MITEL

Inter-Tel® Axxess®

Converged Communications Platform

Installation and Maintenance Manual

Part Number
550.8000

INTER-TEL® AXXESS® CONVERGED COMMUNICATIONS PLATFORM INSTALLATION AND MAINTENANCE MANUAL

Issue 11.0, May 2008

NOTICE

This *Inter-Tel® Axxess® Installation and Maintenance Manual* is released by Inter-Tel (Delaware), Inc. as a guide for certified service personnel. It provides information necessary to properly install, configure, and operate the Inter-Tel Axxess system.

The contents of this document reflect current Inter-Tel standards and are subject to revision or change without notice. Some features or applications mentioned may require a future release and are not available in this release. Future product features and applications are subject to availability and cost. Some features or applications may require additional hardware and/or specific software.

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Portions of this manual and the respective systems and/or methods described therein may be covered under U.S. patent numbers 4,932,022, 4,977,590, 5,109,519, 5,127,003, 5,341,412, 5,351,276, 5,412,715, 5,461,665, 5,465,286, 5,793,861, 5,815,566, 6,205,206, 6,463,051, D479,517, D482,342, D483,749, D483,750, D489,049, 6,754,187, 6,760,427, 6,798,874, 6,829,249, 6,999,431, D499,724, 7,031,443, 7,292,685, 7,318,109; other patents pending. Portions of this system that communicate using the G.729 protocol are covered under patent licenses from third parties.

Product Disposal Instructions

This symbol indicates that the product is classified as electrical or electronic equipment and should not be disposed of with other commercial or household waste at the end of its working life. For appropriate disposal and recycling instructions, contact your local recycling authority or Inter-Tel provider.

The Waste of Electrical and Electronic Equipment (WEEE) Directive (2002/96/EC) was established by the European Union to minimize negative impact on the environment, control hazardous substances, and curtail landfill expansion by using the best available recovery and recycling techniques.

Version 11.0 New Features

For those individuals who are already familiar with the Issue 10.0 Inter-Tel® *Axxess®* *Installation and Maintenance Manual* (part number 550.8000), the following new items are introduced in the version 11.0 release. Throughout the manual, "change bars" like the one next to this paragraph have been placed in the margins to indicate any new or revised information since the Issue 10.0 Manual.

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FEATURES AND ENHANCEMENTS IN V11.0

The following sections summarize new features and enhancements provided with v11.0. For information on the new features, refer to the pages listed below:

- Call Processing Server on Windows Server 2003 (see below)
- Music-On-Hold for Call-Routing Tables (see below)
- Caller ID Forwarding (see below)
- Caller ID Enhancements (see page vii)
- Phantom Devices Enhancement (see page vii)

CALL PROCESSING SERVER ON WINDOWS SERVER 2003

Windows Server® 2003 (Telco version) operating system (OS) with Service Pack 2 (SP2) is now supported for the Call Processing Server (CPS). Field upgrades must be done by certified technicians only. See "Call Processing Server on Windows Server 2003" on page C-3 for details.

MUSIC-ON-HOLD FOR CALL-ROUTING TABLES

Inter-Tel Axxess Call Processing manages the audio connections for incoming Public Switched Telephone Network (PSTN) calls. Currently, when a device places a call on hold, the trunk settings determine how it provides hold audio for the outside party. This determination is based on the settings at the Central Office (CO) trunk group to which the trunk resides. The settings include "Music-On-Hold," "Audio on Transfer to Ring," "Audio on Transfer to Hold," and "Audio on Hold for Transfer Announcement." Each of these fields allows for various options, including Silence, Tick Tone, Ringback, Board, Music-On-Hold, and Use Next Device's Audio Source. See page 6-324 for details.

Call Processing (CP) v11.0 modifies this functionality such that CO trunk calls routed through Call Routing Tables (CRTs) use the matching CRT entry's "Music-On-Hold Profile" to overwrite the settings at the CO Trunk Group. See "Music-On-Hold for Call-Routing Tables" on page 5-98 for details.

CALLER ID FORWARDING

Caller ID forwarding allows Call Processing (CP) to send Caller ID information from any incoming services to the public or private network using Primary Rate Interface (PRI) or Basic Rate Interface (BRI) Integrated Services Digital Network (ISDN) circuits. In the European market Caller ID is known as Calling Line Identification [CLID]. When Caller ID is enabled, the system includes information that identifies the caller to the public or private network. For additional information, see the following locations in the manual:

- Feature Descriptions (see page 5-77)
- Troubleshooting (see page 7-36)

OAI Caller ID Control

System OAI Level 2 Toolkit v9.3 has been enhanced to support Caller ID Forwarding by adding the <*Caller_ID_Number*> and <*Caller_ID_Name*> fields to the syntax of the command string. For details, refer to the latest version of the *System OAI Toolkit Specifications Manual,* document part no. 835.2615.

CALLER ID ENHANCEMENTS

Because some Central Offices are now requiring that calls have Caller ID set, this release adds the "Calling Party Number" and "Calling Party Name" fields to the following settings in Database (DB) Programming:

- Voice Mail (see page 6-369)
- Trunk Group (see page 6-207)

This release also adds the following fields:

- Emergency Calling Party Number (see page 6-209)
- Force Trunk Group Calling Party Name and Number (see page 6-241)

For node networking, if the connected nodes are running Axxess system software earlier than v11.0, the older version nodes ignores all the new fields and function as it does currently.

PHANTOM DEVICES ENHANCEMENT

Phantom device functionality has been enhanced in v11.0. The new phantoms are fully functional virtual devices on the system. For additional information, see the following locations:

- Feature Descriptions (see page 5-120)
- Programming (see page 6-196)
- Troubleshooting (see page 7-50)

ADDITIONAL SYSTEM CHANGES

The following changes are not related to the v11.0 new features and enhancements:

- Windows Vista Support (see below)
- Axxess v9.1 End of Sale (see page vii)
- New Enhanced IP Resource Cards (see page viii)
- Models 8622 and 8662 Multi-Protocol Endpoint PoE Class 2 Support (see page viii)
- New Session Initiated Protocol (SIP) Gateways (see page viii)
- Enterprise Messaging Enhancements (see page ix)
- Unified Messaging Open Standards Edition (see page ix)
- Unified Messaging End of Sale (see page ix)

WINDOWS VISTA SUPPORT

Inter-Tel Axxess is compatible with Microsoft® Windows® Vista®. The supported versions are the latest version of v9.1 and later. These versions' DB Programming and Utilities are also compatible with Windows Vista. For compatibility information, go to Inter-Tel Compatibility Matrix (http://www.inter-tel.com/portal/page/portal/us_edge/technical/compatibilitymatrix) on the Inter-Tel *edGe*® Web site.

AXXESS V9.1 END OF SALE

Axxess v9.1 has reached its end of sale and is no longer supported as of April 30, 2008. Inter-Tel recommends that current v9.1 or earlier installations upgrade to Axxess v10.0 or later.

NEW ENHANCED IP RESOURCE CARDS

The Inter-Tel Axxess Internet Protocol Resource Card (IPRC) "enhanced" daughter card offers new technology for echo cancellation on all calls routed through the IPRC, as well as enhanced diagnostics for troubleshooting echo problems. When the enhanced daughter card is installed and programmed, the new technology on the daughter card replaces the echo cancellers on the IPRC, resulting in improved echo cancellation. There are two types of enhanced daughter cards available:

- **IPRC-DE** (Daughter Enhanced), part no. 550.2277: Provides new echo cancellation technology.
- **IPRC16-DE**, part no. 550.2276: Provides new echo cancellation technology with the DSP circuitry for up to 16 additional Voice over Internet Protocol (VoIP) channels.

For more information, refer to the *IPRC Enhanced Daughter Card Installation Instructions* (document part no. 835.3195) and the *IPC/IPRC and IP Endpoint Firmware Notice* (document part no. 835.2720-9).

MODELS 8622 AND 8662 MULTI-PROTOCOL ENDPOINT POE CLASS 2 SUPPORT

New Models 8622 and 8662 endpoints are now available. They are Class 2 Power over Ethernet (PoE) devices. The new endpoints replace the existing endpoints. Table 1 shows the existing and new part numbers.

Table 1. *Models 8622 and 8662 Part Numbers*

Model	Existing Part Number	New Part Number
Model 8622 – 2-Line Display Multi-Protocol Endpoint	550.8622	550.8622P
Model 8662 – 6-Line Display Multi-Protocol Endpoint	550.8662E	550.8662P

NEW SESSION INITIATED PROTOCOL (SIP) GATEWAYS

The following new SIP gateways are available through the Inter-Tel® CommSource® division:

- Quintum® AFT 400 gateway
- AudioCodes™ MP-114 SIP gateway

These gateways replace the AudioCodes MP-104 SIP Gateway that has gone end-of-life in October 2006. Both gateways support analog Foreign eXchange Office (FXO) ports. The setup procedure is different for each device; the MP-114 gateway uses a Web interface called the AudioCodes Web administrator, while the AFT 400 gateway uses a Configuration Manager program and a Command Line Interface (CLI). Existing Axxess systems must upgrade to v10.0 or later to support the gateways. For more information, refer to the *Inter-Tel Axxess/5000 Quintum SIP Gateway Installation Manual* (document part no. 835.3123) or the *Inter-Tel Axxess/5000 AudioCodes MP-114 SIP Gateway Installation Manual* (document part no. 835.3202).

ENTERPRISE MESSAGING ENHANCEMENTS

The following describes current and future enhancements for Enterprise® Messaging (EM).

EM v1.1

EM v1.1, Service Pack (SP) 2, was released in February 2008. For a list of enhancements, refer to the *Enterprise Messaging Installation and Maintenance Manual* Issue 1.1, Revised February 2008 (document part no. 780.8006). This release does not support Unified Messaging Open Standards Edition (OSE) v2.0 (see below).

EM v1.2

EM v1.2 is scheduled to release in May 2008. This release supports Unified Messaging OSE v2.0 (see below). For information about EM v1.2, refer to the *Addendum to the Issue 1.1 Inter-Tel Enterprise Messaging Installation and Maintenance Manual For Version 1.2 Software* (part number 780.8007).

EM v2.1

The following new features are to be introduced in EM v2.1. These new features require a v11.0 or later system:

- Option Swap: 7 and 9 Keys (see page 4-70)
- VPIM Networking (see page 4-71)

UNIFIED MESSAGING OPEN STANDARDS EDITION

Axxess v8.23x or later supports Unified Messaging OSE, a client-free Unified Messaging distribution for the Inter-Tel Enterprise Messaging voice messaging server. Unlike previous Unified Messaging products from Inter-Tel, users do not have to install a special Unified Messaging client application on their desktop computer to view, play, and manage voice mail messages. With Unified Messaging OSE, the e-mail client can be any Internet Message Access Protocol (IMAP) v4 compliant client that is configured to connect to the IMAP server located on Enterprise Messaging. To use the latest version of Unified Messaging OSE, v2.0, existing Axxess systems must upgrade to v10.2 or later, and to Enterprise Messaging v1.2 or later.

For Unified Messaging OSE v2.0 programming and feature information, refer to the latest version of the *Unified Messaging Open Standards Edition Administrator's Guide* (document part no. 835.3162). For end-user information and instructions for using features, refer to the *Unified Messaging Open Standards Edition User Guide* (part number 835.3164).

UNIFIED MESSAGING END OF SALE

Unified Messaging versions 2.2 (VPU) and 2.3 (EM) have reached their end of sale and are no longer supported as of April 30, 2008.

Inter-Tel recommends that current UM v2.2 and v2.3 installations upgrade to UM Open Standards Edition (OSE). See Unified Messaging Open Standards Edition above for more information.

LICENSING

Table 2 lists the licenses required for v11.0.

Table 2. *Version 11.0 Licenses*

Hardware	Part No.
Axxess Software v11.0	840.0828
Axxess Software Upgrade v10.x to v11.0	840.0829

INSTALLING

Because there is no new hardware for v11.0, there are no new installation instructions in this release.

UPGRADING

Axxess v11.0 now supports the Call Processing Server on Windows Server 2003. Not all hardware configurations can be upgraded to Windows Server 2003. See Table E-2 on page E-8 for the CPS upgrade method to Windows Server 2003. Axxess v11.0 continues to support Windows 2000 Professional OS for the CPS.

A software license is required for all upgrades. If you are upgrading from 10.x or earlier, a v11.0 upgrade license is required. To upgrade an Axxess system v11.0, follow the procedures outlined in "Appendix E — Upgrading to v11.0" on page E-1.

DB PROGRAMMING CHANGES

This section describes minor changes that are made in v11.0 DB Programming. The changes are not related to the v11.0 new features and enhancements.

System – Cabinets

When the Loop Start Card/Daughter cards (LSC-Ds) and Singe Line 16 Cards (SLC-16s) are programmed in DB Programming, and there are LSCs and SLCs installed in the associated slots and vice versa, the cards are not allowed to come online. They are, instead, logged as "Ignored" when plugged in. This change prevents a potential one-way audio problem where the system fails to allocate the necessary voice channels for a hardware board. The problem only exists for hardware boards that do not match the DB Programming board type.

Voice Processor – Devices – Mailboxes – <*node*> – <*mailbox*>

To improve the programming procedures for Unified Messaging, the Fax/E-Mail Forwarding subfolder has been removed. The fields in this folder are now located in the Unified Messaging subfolder under Voice Processor/Devices/Mailboxes/<*node*>/<*mailbox*>.

TROUBLESHOOTING

The following troubleshooting issues are discussed:

- Caller ID Forwarding Issues (see page 7-36)
- Phantom Device Issues (see page 7-50)

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Safety Instructions

For basic installation guidelines necessary for the proper and safe functioning of this equipment, refer to the *Important Safety Instructions* (part number 835.3209) packaged with the system, included on the product software CD, and posted on the Inter-Tel® *edGe* Online Manuals and Guides (www.inter-tel.com/techpublications) Web site.

NOTE: Read and follow all information contained in the *Important Safety Instructions* document before attempting to install or use this product. Only trained, qualified service personnel shall install or maintain this product.

Regulatory Notices

For regulatory information for U.S., Canada, and Europe, refer to the Regulatory Notices section in the *Important Safety Instructions* (part number 835.3209) packaged with the system, included on the product software CD, and posted on the Inter-Tel *edGe* Online Manuals and Guides (www.inter-tel.com/techpublications) Web site.

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Although no telecommunications system or data network is entirely secure, as long as the appropriate security measures are put in place and properly maintained by both the customer and the installing company, the Axxess® system architecture and its associated server-based applications are substantially secure against unauthorized access to the customer's data network via the telecommunications system. Appropriate security measures include, but are not limited to, the proper implementation of user/administrative accounts, passwords, firewalls, Network Address Translation (NAT), access control lists, virus protection, security updates, etc. and the proper maintenance of access points/programs and their respective accounts/passwords.

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Secure Socket Layer

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1. INTRODUCTION

1.1 The Axxess Converged Communications Platform Installation and Maintenance Manual provides the information needed to plan, install, program, implement, maintain an Inter-Tel® Axxess® Converged Communications Platform.

2. AXXESS CONVERGED COMMUNICATIONS PLATFORM

2.1 The Inter-Tel Axxess Converged Communications Platform is a state-of-the-art, digital, voice/data, hybrid communications system. As a hybrid system, it incorporates many of the user-friendly features of key systems with many of the expanded features and flexibility of private branch exchange (PBX) systems.

2.2 The system is designed to meet the needs of growing businesses. In fact, the system's unique digital signal processor (DSP) structure allows it to be easily adapted and expanded as business communication needs change, especially with the Caller ID, Automatic Number Identification (ANI), Dialed Number Identification Service (DNIS), and Integrated Services Digital Network (ISDN) features. The modular design makes the system easy to install and service. And, the programmable features provide an abundance of user-friendly applications to meet each customer's needs. Highlights of the system's design include:

- Advanced microprocessor technology
- Modular, easily replaceable hardware with add-on capabilities for optional features
- Flexible programming to customize many system and station features

2.3 **Windows Vista Support**: Inter-Tel Axxess is compatible with Microsoft® Windows® Vista®. The supported versions are the latest version of v9.1 and later. These versions' Database (DB) Programming and Utilities are also compatible with Windows Vista. For compatibility information, go to Inter-Tel Compatibility Matrix (http://www.inter-tel.com/portal/page/portal/us_edge/technical/compatibilitymatrix) on the Inter-Tel *edGe*® Web site.

3. INTENDED AUDIENCE

3.1 This manual has been developed for technician-users who:

- Possess the knowledge and skill required to set up, maintain, and troubleshoot a programmable circuit-switched telephone system.
- Use standardized telecommunication practices, procedures, and troubleshooting methods when performing their work.
- Possess the ability to effectively apply the principles of:
  - — Internet Protocol (IP) telephony—Voice over IP (VoIP), including Quality of Service (QoS)
  - — Data networking
  - — Technology convergence

4. "ENDPOINTS" AND "PHONES"

4.1 As digital, wireless, and computerized communication methods have evolved, the language of telecommunications has had to change to keep pace. Not long ago, voice and modem-encoded text were communicated from one telephone instrument to another across a global network of copper wire, optical fiber, microwave antennas, and satellites. More recently, however, the development of the Internet and of World Wide Web (www) sites have introduced a multimedia dimension to information and made it all more readily accessible through computer networking. As this evolution has occurred, new words and meanings have been introduced to describe the new methods and equipment used for communicating.

4.2 Telephone systems that once provided only audio phone service to consumers have become "communication platforms" that support many devices that people can use for personal and business communication. Phones now share communication space with personal digital assistants (PDA), with wireless phones capable of text messaging, taking photos, and recording video, with fax machines that transmit and receive messages across Internet Protocol (IP), and with "soft" phones that are displayed and used from computer screens.

A. COMPUTER SCIENCE ROOTS

4.3 When referring to communication protocols in computer science, the term "endpoint" describes an entity on one end of a Transport Layer connection. The Transport Layer is the Transmission Control Protocol/Internet Protocol (TCP/IP) layer in the Open Systems Interconnect (OSI) network model. Utilizing the TCP/IP layer standard, Internet servers send and receive packets of data across the World Wide Web.

B. USAGE IN THIS MANUAL

4.4 This manual uses the term "endpoint" to describe the entire category of devices that the Inter-Tel Axxess family of products supports. In this context, digital and IP phones are endpoints, as are data modules, fax machines, computer telephony (CT) gateways, Single-Line Adapters (SLA), IP Single-Line Adapters (IP-SLA), and answering machines.

4.5 And, to promote clear communication and understanding, commonly accepted names of endpoints are used as appropriate. In particular, desktop IP endpoints are also referred to as IP phones. Digital endpoints are also referred to as digital phones. Wireless endpoints are also referred to as wireless phones. And, analog single line endpoints are also referred to as single line phones or single line sets.

NOTE: In the manual and in DB Programming, a phone may also be called a "station" when reference to the phone and its environment are intended. Or, it may be called "device" in the most general sense.

5. MULTICULTURAL ENGLISH REFERENCES

5.1 This manual supports the requirements of technicians in the United States of America (U.S.) and in the Inter-Tel Europe market, which includes the United Kingdom (U.K.). Because of this dual support, dual references are made to industry features, standards, and jargon, as appropriate throughout the manual. Terms used in the Inter-Tel Europe market appear in brackets [<*European equivalent*>]. For the purposes of this addendum, British English terms are assumed to apply to other English-speaking European cultures, as well.

5.2 For example, the U.S. telecom industry refers to an audio communication line between a public switching system and a private switching system as a "central office (CO) trunk." In the U.K., this same type of line is called a "local exchange trunk." Applying the dual-reference guideline would result in the following statements:

- During system installation, the technician connects CO trunks [local exchange trunks] to the Loop Start cards.
- To turn the Diagnostics Mode on or off at an administrator endpoint, the system administrator enters 9900 [9100 in Europe].

5.3 However, this manual does not make a similar distinction between American English and British English spellings of common words. Only American English spellings appear in this manual. For example, the word "analog" is not also spelled "analogue," because as the meaning is clear.

6. ATM

6.1 Versions 6.0 and later support the standard system configurations as well as a configuration using an Asynchronous Transfer Mode (ATM) Switch. With the ATM technology, an ATM system can more than double the current capacity of the CPU512 embedded system and Windows NT®-based CPU. ATM, which transmits packets of fixed-length cells, uses logical connections to allow various traffic, such as data, voice, and video, to be carried over the same local or wide area network. Using a UTP5 or OC-3 connection, the ATM Switch can simultaneously transmit this traffic, providing a seamless connection for single- or dual-chassis systems.

6.2 Because the ATM system supports such a high capacity, a Call Processing (CP) Server running Windows 2000 is required. See Appendix C — CP Server for more information on the CP Server.

7. SYSTEM CAPACITIES

7.1 The basic chassis has eight circuit card slots, seven of which can be used for station, trunk, and optional resource cards. The fully-expanded, quad-chassis system (actually four basic chassis connected together) has 32 circuit card slots, 28 of which can be used for station, trunk, and optional resource cards. A fully-loaded ATM system (four, dual-chassis systems connected to an ATM Switch) has 64 circuit card slots, 56 of which can be used for device and/or resource cards.

A. SYSTEM CAPACITIES

7.2 The table on the following page lists the maximum number of cards that may be installed in a single-, quad-, or eight-chassis (ATM) system.

Table 1-1. *System Capacities*

Cards	Single[a] (Up to 8)	Dual[b] (Up to 16)	Quad or Non-ATM[c] (Up to 32)	ATM[d] (Up to 64)
Central Processing Unit (CPU) Cards—required	1	1	2	4
IP Resource Card (IPRC) Licensed for Networking	1-2	1-2	1-8[e]	1-28
Digital Keyset Card (DKSC, DKSC-16, or DKSC-16+)—at least one keyset-type card is required	1-7	1-14	1-28	1-56
Internet Protocol Card (IPC) or Internet Protocol Resource Card (IPRC) Programmed for Devices	1-7	1-14	1-28	1-56
For U.S. Systems: Analog Keyset Card (AKSC) or Executone Keyset Card (EKSC) *For European Systems:* AKSC	1-7	1-14	1-28	1-56
Single-Line Card (SLC, SLC-16, SLC-16+)—at least one keyset-type card is required	1-6	1-13[f]	1-28	1-48
For U.S. Systems: Loop/Ground Start Card (LGC) with optional attached Daughter Card (LGC-D) or Loop Start Card (LSC) with optional attached LSC Daughter Card (LSC-D) *For European Systems:* Loop Start Card (LSC) with optional attached LSC Daughter Card (LSC-D)	1-7	1-14	1-28	1-56
For U.S. Systems: T1 Card (T1C) with optional Primary Rate Interface (PRI) Daughter Card or T1/E1/PRI Card *For European Systems:* E1 Card (E1C) with optional Primary Rate Interface (PRI) Daughter Card or T1/E1/PRI Card (see page 1-9 for PRI capacities)	1-7	1-14	1-28	1-56
For U.S. Systems: Basic Rate Interface Card (BRU or BRS) *For European Systems:* Basic Rate Interface Card (BRS)	1-7	1-14	1-28	1-24
Options Card (OPC)	1-7	1-14	1-28	1-56
Network Clock Card (NCC)—only needed with CPU64 or CPU128	1	1	--	--
Embedded Voice Mail Card (EVMC)	1	1	1[g]	--

a. A single cabinet system has 8 slots. 1 slot is fixed for the CPU and the 7 additional slots can be used for any type of peripheral board.
b. A Dual cabinet system has 16 slots. 1 slot is fixed for the CPU, 1 slot is unused, and the 14 additional slots can be used for any type of peripheral board.
c. A Quad or Non ATM system 32 slots. 2 slots are used for CPUs, 2 slots are unused and the 28 additional slots can be used for any type of peripheral board.
d. An ATM system 64 slots. 4 slots are used for CPUs, 4 slots are unused, and the 56 additional slots can be used for any type of peripheral board.
e. Up to 28 IPRCs programmed for networking are supported if using a Call Processing (CP) Server.
f. If the system is configured with 7 SLC-16 cards per cabinet, the system does not have available time slots for providing music and paging to the various devices and functions.
g. Although the EVMC was designed (and is recommended) for smaller, single-chassis CPU64/CPC64 installations, it can also be used in multi-chassis installations that require only a low-performance, non-demanding feature type of voice mail functionality. However, as feature demand or performance needs increase, it may be necessary to move to a higher-performance voice processing product like Enterprise Messaging.

NOTE: *The actual number of stations and trunks available depends on the number and type of devices installed.* For example, if all 56 cards in an eight-chassis ATM system are DKSC-16s, there can be a maximum of 896 phones installed. Or, if only one of the cards is a DKSC-16 and the other 55 are LSC-Ds, the system maximum is 16 phones and 440 trunks. The T1/PRI or E1/PRI card and T1/E1/PRI card provide even greater trunk capacity (limited only by the number of available voice channels in the system). In most circumstances, the theoretical maximums listed in this manual are never reached due to the variety of cards and station instruments installed.

7.3 Voice mail service can be provided using either the Voice Processing Unit, an Enterprise® Messaging system, or the Embedded Voice Mail Card (EVMC). The EVMC is recommended only for single-chassis systems. For more details about the EVMC see page 2-69. For information about the CPU64 configuration, which uses the EVMC, see Appendix D — CPC64/CPU64 Configuration. For information about Enterprise Messaging system, see the *Enterprise Messaging Manual* (document part no. 780.8006).

7.4 In addition to certain resources provided by the system's main processing cards, the 1792 ports can be used to equip a variety of trunks and station instruments, including:

- **Central Office —CO Trunks [Local Exchange Trunks in Europe]**: Loop start and/or ground start trunks, installed in increments of up to four or eight, depending on the number of trunk circuit cards installed in the equipment chassis.
- **T1 (E1) Trunks**: Loop start, ground start, DID [DDI in Europe], E&M (U.S. only), and/or OPX trunks installed in increments of up to 24, depending on the number of T1, E1/PRI, or T1/E1/PRI cards installed in the equipment chassis.
- **Primary Rate Interface (PRI) Trunks**: ISDN PRI trunks installed in increments of up to 24 (23 bearer channels and one data channel) [or 30 in Europe], depending on the number of PRI-equipped T1 or E1 cards in the chassis.
- **Basic Rate Interface (BRI) Trunks**: ISDN BRI trunks installed in up to 2 bearer channels on BRS or BRU (U.S. only) cards in the chassis.
- **Digital Phones**: Axxess Basic, Eclipse™ Basic, Standard Display, Associate Display, Professional Display, Executive Display, Model 8500, Model 8520, and Model 8560 Phones in increments of up to eight or 16, depending on the number of Digital Keyset Cards (DKSCs) and 16-Port Digital Keyset Cards (DKSC-16s or DKSC-16+s) installed in the chassis.

  NOTE: The Professional Display, Associate Display, and Eclipse Basic Digital Phones are supported by U.S. software only.

- **IP Devices**: A combination of up to eight IP PhonePlus, Model 8660 Phone, or IP SLA ports to provide voice-over-IP functionality on Inter-Tel phone systems. The v1.x firmware device Internet Protocol Resource Card (IPRC) supports a combination of up to 16 IP PhonePlus, Model 8660 Phone, or IP SLA ports. This IPRC also supports up to 16 IP trunks via media gateway control protocol (MGCP). The v8.x.x and 9.0.0 firmware IPRC supports a combination of up to 32 IP Phone Plus, Model 8660 Phone, multi-protocol endpoint, Model 8602 IP softphone, IP SLA ports or up to 32 MGCP gateways/endpoints. (The v8.x and 9.0.0 firmware IPRC does not support IP SoftPhones.) The IPC uses a 10Base-T Ethernet interface to the stations and an RS-232 interface for diagnostics and programming. The IPRC uses a 100/10Base-T Ethernet interface to the stations and an RS232 interface for diagnostics and programming. For details, see the latest *IP Devices Manual* (document part no. 835.2195).

  NOTE: If licensed for networking, you cannot use the IPRC for devices. Make sure you have the appropriate software license loaded before you program the IPRC.

- **Multi-protocol Endpoints**: Models 8600, 8620/8622, 8662, and 8690 support either ITP (an Inter-Tel protocol) or Session Initiation Protocol (SIP). These endpoints require a software license and must terminate at a version 8.1 or later firmware IPRC. They cannot terminate at an IPC or a v1.x firmware device IPRC.

- **Model 8602 IP Softphone**: Model 8602 is a new IP softphone that operates like a Model 8662 multi-protocol endpoint and supports Inter-Tel Protocol (ITP) mode on a PC. It requires system software v9.1 or later and IPRC firmware v9.0.0 or later. For details, see the latest *IP Devices Manual* (document part no. 835.2195).
- **Pocket PC SoftPhone**: Model 8601 is a software-based SIP softphone that runs on selected Pocket PC 2002 (or later) Personal Digital Assistant (PDA) platforms. For information about installation and programming, refer to the *Model 8601 Installation Manual*.
- **Wireless Endpoints**: Models 8664, 8665, and 8668 are wireless, compact telephones that use ITP. For details, refer to the *SVP Server and Model 8664/8665 Installation Manual* (part no. 935.4521).
- **Analog Phones**: IMX, Inter-Tel/DVK, GMX, and ESP analog phones in increments of up to eight or 16, depending on the number of Analog Keyset Cards (AKSCs) installed in the chassis.

NOTE: At least one phone circuit should be equipped with a display phone to allow an attendant/administrator to program selected system data, act as the attendant for unsupervised CO/local exchange recalls, receive system alarm messages, etc.

- **Single-Line Sets**: Various types of single-line sets in increments of up to eight or sixteen, depending on the number of Single-Line Cards (SLCs) installed in the chassis. If desired, single-line sets can also be attached to digital phone circuits that are equipped with optional Single-Line Adapters (SLAs).
- **Digital Direct Station Selection/Busy Lamp Field (DSS/BLF) Units**: Up to 5 phones may be equipped with up to 4 DSS/BLF Units each or up to 20 phones may be equipped with single DSS/BLF Units, or any combination of these up to 20 total units. Because each unit is powered by an AC transformer, the units do not require a DKSC or DKSC-16 circuit that is separate from the phone.
- **DSS Consoles**:

NOTE: The following DSS consoles are supported by U.S. software only.

— *With Executone IDS Phones:* IDS phones can be equipped with up to three 48-key DSS Consoles. There can be a total of 80 IDS 48-key DSS Consoles on the system. Each 48-key DSS Console requires an EKSC circuit.

— *With Professional and Associate Display Phones:* PCDPM-equipped Professional and Associate Display Phones may be connected to up to four 60-key DSS/BLF Consoles. There can a total of 20 of these 60-key DSS/BLF Consoles on the system. Because each unit is powered by an AC transformer and is connected to the phone's PCDPM, the 60-key DSS/BLF Consoles do not require a DKSC or DKSC-16/DKSC-16+ circuit that is separate from the phone.

B. SYSTEM FEATURE CAPACITIES

7.5 Some features have maximum capacities that are dependent on the availability of system channels and/or circuits. Features with such capacities are listed in the following table. The numbers listed under the heading labeled "dual" are for two-chassis (16-slot) systems; the numbers listed under the heading "CPS512" are for four-chassis (32-slot) systems; the numbers listed under "CPS1024" are for systems equipped with a Call Processing (CP) Server and that are licensed for 1024 devices; and the numbers under "ATM" are for an eight-chassis, ATM-based system. (See "Appendix C — CP Server" on page C-1 for details about the CP Server.)

NOTE: As of version 8.0, 512 and higher systems require a CP Server. The number of devices supported on these systems, however, is determined by the software license. For example, if using a 512CPS software license with a CP Server, the number of devices available is limited to 512. To support the full capacity of a CP Server, a 1024CPS license is required. Versions prior to 8.0 do not use a software license.

Table 1-2. *System Feature Capacities*

Feature	Capacity 256	Capacity CPS512[a]	Capacity CPS1024	Capacity ATM[b]
Devices	256	512	1024	1792
Non-Blocking Voice Channels	224	448	448	896[c]
Voice Processing PC				
- via CPU-type Card	8	8	8	8
- via Options Card	32	32	32	32
Station instruments	256[d]	512[d]	1024[d]	1792[d]
Phones	224	448	448	896[c]
Single-Line Sets	256	512	1024	1792
DSS/BLF Units				
DSS/BLF Unit-equipped stations	20	20	20	20
DSS/BLF Units per phone	4	4	4	4
Desktop virtual DSS/BLF devices	20	20	150	150
Trunks	336[d]	672[d]	672[d]	1344[d]
Trunk groups	104	208	208	208
Trunks per group	336	672	672	1344
PRI-equipped T1 or E1 cards	8	8	28	28
System RS232 ports	11	11	11	32
Error-free protocol (EFP) sessions	3	6	6	6
Paging				
Ports per system (CPU-type Card)	1	2	2	4
Zones per system	10	10	50	50
Phones in a paging zone	60	60	120	120
Conference circuits available via the CPU(s)	12[e]	12[e]	24[e]	48[e]
Parties per conference	4	4	4	4
Simultaneous four-party conferences	3[e]	6[e]	6[e]	12[e]
System speed dialing				
Numbers per system	1000	1000	5000	5000
Digits per number	48	48	48	48
Station speed dialing				
Numbers per stn. (CO &/or IC)	10	10	10	10
Digits per entry (CO &/or IC)	16	16	16	16
Redial numbers per station	1	1	1	1
Digits per number	32	32	32	32
Callback (queue) requests per station	1	1	1	1
Camp on by a station	1	1	1	1
Stations camped on to a station	256[f]	512[f]	1024[f]	1792[f]
Stations camped on to a trunk group	256[f]	512[f]	1024[f]	1792[f]
Inter-station messages per system	256	512	1024	1792

Table 1-2. *System Feature Capacities (Continued)*

Feature	Capacity 256	Capacity CPS512[a]	Capacity CPS1024	Capacity ATM[b]
Programmable reminder messages Message requests per system	20 256	20 512	20 1024	20 1792
Do-not-disturb messages Messages per system	20 256	20 512	20 1024	20 1792
System forwarding paths Paths per station Points per path	200 3 4	200 3 4	200 3 4	200 3 4
ARS Route groups Facility Groups per route group Trunk groups per facility group Dial rules (16 digits each) - total - programmable Callback requests per system Camp on requests per system	 50 100 104 32 26 256[f] 256[f]	 50 100 208 32 26 512[f] 512[f]	 100 100 208 32 26 1024[f] 1024[f]	 100 100 208 32 26 1792[f] 1792[f]
Toll restriction user groups *(U.S. only)* Area code tables per user group Area codes per table Ext. area codes per user group Office code tables per ext. area code Office codes per table	8 1 800 6 1 800	8 1 800 6 1 800	8 1 800 6 1 800	8 1 800 6 1 800
Station class of service designations	16	16	16	16
Absorbed digit patterns per system Digits per pattern	50 48	50 48	50 48	50 48
Account codes Forced Standard Digits per code	 256 256 3-12	 512 512 3-12	 5000 5000 3-12	 5000 5000 3-12
Hunt groups Per system Stations per hunt group Announce stations per hunt group Overflow stations per hunt group Recall destinations per hunt group Agent IDs per ACD hunt group Stations in an all-ring hunt group	 75 2500[g] 1 1 1 1000 30	 75 5000[g] 1 1 1 1000 60	 300 5000[g] 1 1 1 3000 60	 300 5000[g] 1 1 1 3000 60
Keymap group assignments Keyset DSS/BLF User-programmable buttons per phone	 25 20 35	 25 20 35	 250 20 35	 250 20 35
Extension lists per system Station or trunk numbers per list Stations per list used for hunt groups Stations per list used for paging	256[d] 224 30 60	512[d] 448 30 60	1024[d] 20,000 60 120	1792[d] 20,000 60 120
Busy Hour Call Completion (BHCC)[h]	23,000[i]	23,000[i]	33,500[j]	33,500[j]

a. The 512CPS capacities also apply to quad-chassis systems running system version 7.0 or earlier.
b. ForeRunner LE®155 Asynchronous Transfer Mode (ATM) switch was recently replaced by the Ericsson® (Marconi) ASX™ 200BX ATM switch. The new switch provides the same functionality as the ForeRunner LE155 switch and includes an Ethernet port for remote access. For details, see the ATM Switch Installation Instructions.
c. If Music-On-Hold (MOH) is used in a cabinet, one slot (per cabinet) is subtracted from available time slots. This one slot is dedicated for MOH and held up until the system is reset. MOH is altered each time the system is reset.
d. The maximum number of stations and trunks that can be installed is limited by the number of circuit cards installed and voice channels and/or system memory resources available. For systems using more than 112 ports, use the faster CPC or CP Server.

e. Additional conferencing circuits can be added by installing one or more Options Cards. Each Options Card can have up to four DSPs, and each DSP can be programmed to provide up to 12 or 16 additional conference circuits. Each PCMA card contains 12 conferencing circuits. Conferencing is not amplified.
f. Each station can have only one camp on and one queue request registered at a time.
g. Limited only by the combined total of the extension number lists, which are used to assign to hunt groups, UCD groups, page zones, outgoing access, allowed answer, ring-in, and trunk groups. The combined maximum of all these lists is 2500 or 5000 entries.
h. This is the optimized number of calls that a system can answer during a busy hour per industry accepted practices. Actual capacity varies depending on the version of software, call durations, system configuration and peripheral applications such as System OAI Applications.
i. If not using a CPC, the BHCC value for dual- and quad-chassis systems is 15,500.
j. This number is limited by the call generation capability of Test Lab equipment. Actual BHCC may be higher.

NOTE: A dual-chassis system is limited to 80 concurrent calls across networked IPRCs when the IPRCs are installed in the first chassis. If all IPRCs are installed in the second chassis, this limit is 112.

8. SYSTEM NETWORK CAPACITIES

8.1 With phone system software versions 5.0 and later, you can connect two or more Inter-Tel systems, including ATM systems, to form a network that provides an interface between the systems. To the user, the network appears as though it is one integrated system. With few exceptions, the user can perform all of the functions across the network that he can within a single phone system.

NOTE: If using v6.0 or later in a networked system, all other nodes must be running version 5.2F9 (or later) or 5.349 (or later).

8.2 In this manual, all references to a "phone network" mean two or more phone systems connected by a network span. Each phone system in a network is called a "node."

8.3 The maximum capacities of the network are listed in the following table.

Table 1-3. *System Network Capacities*

Feature/Devices	Capacity
Phone system nodes per network	99
Local devices per node	1792*
Off-node devices per node	25,000*
Devices per network	40,000

* The maximum number of stations and trunks that can be installed is limited by the number of voice channels and/or system memory resources available.

A. 99 NODES SUPPORT

8.4 V10.0 or later software for the U.S., Europe, and Mexico allows the phone network to expand to 99 nodes. Nodes can transparently inter-connect to 98 other systems via T1/E1 PRI networking and IP networking.

8.5 The increased network capacity does not affect endpoint user features. Endpoint users can still dial an extension of a remote user, and the system transparently routes the call throughout the network. From an administrative perspective, you must consider the following as the node capacity increases:

- **Database Management**: As it is done today, you must log in to individual systems to configure the systems. You must plan for supporting a 99-node network by devising a numbering plan for each node. You should be aware that the creation of new devices automatically causes the system to export extensions to the network. The use of export/import may take longer if you attempt to export/import with all 98 remote nodes.
- **Off-Node Device Limit**: As more endpoints are added to nodes within the network, each system broadcasts new endpoint extensions throughout the network. For very large networks, this may create a problem for the Call Processing Card (CPC) systems because there is a limited amount of dynamic memory on the CPCs. You must be aware of the amount of available dynamic heap on the CPCs as the network grows. Even if specific off-node extensions are removed from a node, a DB Programming export performed by a remote node administrator may cause the off-node extensions to reappear. Therefore, it is important to apply the same changes to all the nodes for changes to properly take effect. If you want to delete an off-node device, you need to delete it on the remote node as well.

- **Numbering Plan**: The system supports extensions with a maximum of five digits. It is important to design a network numbering plan that can support all devices in the network.
- **DB Programming Export/Import**: Creation of new devices automatically causes the system to export extensions to the network, therefore, it is extremely critical that the system administrator devise a numbering scheme for ALL 99 nodes.
- **Networking Resources (IP and PRI)**: You must be aware of the necessary Primary Rate Interface (PRI) and voice resources needed to maintain a 99-node network. For voice resources, all nodes must have the appropriate IP bandwidth to support the maximum IP networking calls.
  - — *T1/E1 PRI Networking*—Networking via T1/E1 PRI remains the same as in previous version, except for the increased node capacity. In DB Programming, you can number network nodes from 1 to 99.
  - — *IP Networking*—IP Networking remains the same as in previous versions, except for the increased node capacity. To set up IP Networking, you must assign a P6XXX extension for each Internet Protocol Resource Card (IPRC). If the IPRC does not have a Networking voice resource, you must use another IPRC and enable at least one Networking voice resource. The existence of the Networking voice resource allows the IPRC to communicate with external nodes from IP private networking.
- **OAI Applications**: A new version of the CT Gateway Server is necessary to support more than 63 nodes. The CT Gateway and individual nodes may take a performance degradation based on how OAI applications are added to the network. You must be aware of the number of OAI applications that connect to the CT Gateway. If many OAI applications are connected directly to an individual node, this could cause performance degradation on those nodes.

NOTE: The CT Gateway Server must meet Inter-Tel's performance requirement when it is attached to a 99-node network. When configuring a 99-node network, this could increase the OAI data by 50%. For example, the Attendant Console application requests a large amount of data at power-up. Multiple occurrences of the Attendant Console application in the network, along with fully populated nodes, could cause a 50% increase of data during the Attendant Console power-up sequence.

8.6 For information about programming 99 nodes, see page 6-48.

B. COMPONENT COMPATIBILITY MATRIX FOR 99 NODES SUPPORT

8.7 Table 1-4 shows the version numbers of the components that have been validated to support 99 nodes. Axxess v11.0 is also compatible with additional older versions that are not identified here. It may also be compatible with versions planned for near release. Inter-Tel recommends that the versions listed here are used with Axxess v11.0 if the site has more than 63 nodes.

Table 1-4. *Component Compatibility Matrix*

Component	version
Axxess Component	
ATM Operating System (OS)	8.1.10
Call Processing Card (CPC) Boot Code	2.1.15
CPC/CP Server (CPS)	11.00x US/UK
CT Gateway	4.303
Database Programming	11.00x
Ethernet Adapter Card (EAC)	8.1.5
Enterprise Messaging (EM)	1.2 or later
Internet Protocol Resource Card (IPRC)	10.0.2
Linux-based Embedded Voice Mail Card (EVMC)	11.00x
OS/2-based EVMC	11.00x
System Manager	2.0.6.x
Voice Prompts (WIN2K/NT, OS/2, Linux)	11.0
Windows 2000-based Voice Processing Unit	11.0
Auxiliary Component	
Attendant Console	3.102
Contact Center Suite	4.11 (TBD 4.2 to GA 5/08)
IP PhonePlus	8.100
IP SLA	8.100
Model 8660	8.2.2
Model 8600/8620/8662	2.2.04
Model 8664/8665/8668	1.0.0.4
Model 8690	1.1.8
Network Qualifier	2.006
Session Initiation Protocol (SIP) Server	1.1.1.33
Unified Communicator	3.0.0.13
Unified Messaging Open Standards Edition (OSE)	2.0
Unified Messaging v2.2 (VPU)	Not supported[a]
Unified Messaging v2.3 (EM)	Not supported[a]

a. Unified Messaging versions 2.2 and 2.3 have reached their end of sale. See "Unified Messaging" on page 2-89 for details.

9. VOICE PROCESSING UNIT CAPACITIES

NOTE: As of version 8.0, the OS/2 Voice Processing Unit is no longer supported.

9.1 Listed below are some of the Voice Processing features with maximum capacities that are dependent on system resources.

Table 1-5. *Voice Processing Unit Capacities*

Feature	VP Unit Capacity	EM Unit Capacity
System Audio Interface Ports	40	64
Applications per system[a]	1000	1000
Mailboxes and/or Extension IDs per Voice Processing node Local: Off-Node	10000[b] CPC: 10000[b] 512 CPS: 10000[b] 1024/ATMCPS: 25000[b]	10000 CPC: 10000 512 CPS: 10000 1024/ATMCPS: 25000
Voice Mail Group Lists Members per group list	1000 1500	500 1500
Remote Message Notification Numbers per mailbox	18	1
Audiotex Recordings per system	500	500[c]
Message Notification/Retrieval Applications per system	1	1
System Passwords	4	10
Time Slot Groups	10	10

a. Applications include Call Routing Announcements, STAR applications, Record-A-Call, etc.
b. This is the maximum number of mailboxes allowed by the software.
c. The number of programmed custom recordings is limited only by the size of the local hard disk.

CAUTION

Although version 7.0 or later DB Programming allows you to configure 64 voice mail ports, the Voice Processing unit (not the EM system) currently supports a maximum of only 40 ports. Attempting to configure 64 voice mail ports may cause serious performance issues.

9.2 For Voice Processing Unit capacities when using the Embedded Voice Mail Card (EVMC) see "Appendix D — CPC64/CPU64 Configuration" on page D-1.

10. VOICE PROCESSING UNIT NETWORK CAPACITIES

NOTE: As of version 8.0, the OS/2 Voice Processing Unit is no longer supported.

10.1 A Voice Processing Unit or Enterprise Messaging (EM) unit can be installed on any or all nodes in the phone network. These Voice Processing systems can also be networked together to allow a caller to leave a message on the local Voice Processing system for a mailbox located on another Voice Processing system in the network.

10.2 The maximum capacities of the network are listed in the following table.

Table 1-6. *Voice Processing Unit Network Capacities*

Feature/Devices	VP Unit Capacity	EM Unit Capacity
Voice Processing Units per network	100[a]	200[a]
Mailboxes and/or Extension IDs per Voice Processing node Local: Off-Node:	10000 CPC: 10000 512 CPS: 10000 1024/ATMCPS: 25000	10000 CPC: 10000 512 CPS: 10000 1024/ATMCPS: 25000
System Audio Interface Ports per node	40	24 (for Base I) 32 (for Base II)[b]
Applications[c] per node	1000	1000
Group Lists per node Members per group list	1000 1500	500 1000
Remote Message Notification Numbers per mailbox	18	18
Audiotex Recordings per node	500	500[dd]
Message Notification/Retrieval Applications per node	1	1
System Passwords per node	4	10
Time Slot Groups per node	10	10

a. This is the maximum number of nodes supported by the software. System traffic may limit the actual number of nodes that can be supported without affecting system performance.
b. Available ports are determined by a software license.
c. Applications include Call Routing Announcements, STAR applications, Record-A-Call, etc.
d. The number of programmed custom recordings is limited only by the size of the local hard disk.

10.3 Each Voice Processing system has its own internal message queue (similar to a mailbox's message queues) and stores messages destined for other Voice Processing systems until they are delivered.

NOTE: The Voice Processor AMIS Networking or Inter-Tel Messaging IP Networking software license is required to use Voice Processing Nodes.

10.4 The Voice Processing Node programming area is used to identify and define information about each node (unit) in the Voice Processing network. Before the local Voice Processing Unit can deliver messages to another Voice Processing node in the network, the remote node must be included in the local table.

10.5 Mailboxes programmed on the local node, but associated with mailboxes on remote Voice Processing nodes are referred to as "network mailboxes." They are used by the local Voice Processing unit to identify and locate the mailboxes located on the other Voice Processing nodes. They are not actual mailboxes; they are just "place holders" that tell the local node where to send messages received for that mailbox number. If the mailbox subscriber logs on to a network mailbox, the only options available are the directory name, the password, and possibly the transfer method programming prompt, if enabled.

10.6 There are several methods available for Voice Processing networks. The networking method determines how the local node communicates with the remote node. The networking methods include the following:

- **AMIS**: AMIS is an analog networking protocol that uses the public telephone network.
- **TCPIP**: TCPIP is a direct network connection between the local Voice Processing system and the remote Voice Processing system.

11. SOFTWARE CONFIGURATIONS

A. SOFTWARE PART NUMBERS

11.1 The Inter-Tel system is available in a variety of software configurations with and without networking capabilities. The part number for the software is as follows:

Software	Part No.
Version 11.0 System Software CD-ROM (contains all software)	840.0828
Axxess Software Upgrade v10.x to v11.0	840.0829

B. SOFTWARE LICENSES

11.2 Versions 8.0 and later software controls the device capacity and feature availability of the system through a software license.

11.3 **Call Processing Software Licenses**: All Call Processing Cards (CPCs) and v8.0 (and later) installations require a software license, which differs based on the CPU type, as described below.

- A CPC-Single (64) or CPC-Dual (128 or 256) requires a single software license.
- A 512-CPU, 1024-CPU (CPU/PCM-F), or CPU-ATM (PCM-A) system requires a Call Processing (CP) Server that has a special security key (part no. 827.9775) that supports the software license functionality. The software license is stored on the CP Server PC and works only with the individual special security key.

NOTE: Version 8.0 and later does not require feature PALs. In addition, because version 8.0 no longer detects PALs, you do not need to upgrade the ATM or PCM-F PALs.

11.4 To run version 8.0 or later on a 64, 128, or 256 system, you must use the new CPC. The CPU64, CPU128, and CPU256 cards do not support the new Call Processing license feature and, therefore, do not support version 8.0 or later software.

12. LICENSES

12.1 Software licensing controls device capacity and feature availability on the Axxess system. Axxess platforms are delivered without software licensing. Licenses are generated and downloaded from the Inter-Tel License Management Center (LMC) Web site and then loaded onto the platforms before installing the system. By default, unlicensed platforms can be programmed with the capabilities of fully provisioned systems. However, if the applicable licenses are not provided and authenticated within four hours, the system invokes a reset.

12.2 If DB Programming connects to a system with a valid software license, you cannot restore a database that would invalidate the license. If you need to change the product type, you must first load a new software license through DB Programming. For details, see the PROGRAMMING chapter.

12.3 DB Programming has a folder called "Software License." This folder displays the licenses currently uploaded to the system. The features common to all licenses are displayed in this folder along with the value for each feature according to the current software license. See page 6-97 for details.

A. IP GATEWAY TRUNK LICENSING

12.4 The system requires "IP Gateway Trunk License" for the use of the following two types of IP gateway trunks. One license is required for each trunk to be used.

- Media Gateway Control Protocol (MGCP)
- Session Initiation Protocol (SIP)

B. PREMIUM FEATURE LICENSING

12.5 Premium features, which customers can purchase to enhance their systems, are controlled by software licensing through the Inter-Tel License Management Center (LMC). This section discusses the process used for licensing premium features and the storage of multiple licenses on a system.

General

12.6 Because features are controlled through licensing, the system database cannot be restored with premium features that have not been licensed by Inter-Tel, Inc. If an attempt is made to restore a database that has more premium features than the software license reflects, a warning message appears before the Restore command is executed. When this message displays, the Restore action must be cancelled. For details, see page 6-67.

12.7 Because the software license determines the premium features that can be enabled, changes cannot be made directly in DB Programming. The process to add premium features involves purchasing the license, building the licence from the LMC, generating the license, downloading it to the field PC, and installing it in the system from the PC through DB Programming. For complete procedural instructions, see "Software License Upload" on page 6-45.

Number of Licenses

12.8 The system stores up to five different licenses, each of which is unique based on the system version. For example, only one v11.0 software license can be stored on the system. If a new license that matches the version number of an existing license is loaded, the new license automatically overwrites the existing license.

12.9 In addition, if five licenses are currently stored and a sixth license is saved, the oldest license, based on the date and time the license was created, is overwritten. If, however, the oldest license is currently in use, the next oldest license is overwritten. In addition, because each license is unique, only one license for each version number can be stored at any time.

C. IP ENDPOINT LICENSING

12.10 This section identifies the types of IP endpoints that the Axxess platform supports and the process used for activation of licensed endpoints.

Types of IP Endpoints

12.11 The following list identifies the types of IP endpoints that operate on the system. For specifications and illustrations of the endpoints, see "Station Instruments" on page 2-72.

12.12 Category A Endpoints License:

- Multi-protocol endpoints in ITP mode:
  - Model 8600
  - Model 8622 (and discontinued Model 8620)
  - Model 8662
  - Model 8690
- Non-Inter-Tel endpoint:
  - OAISYS Net Phone, which emulates an Inter-Tel Model 8662 in ITP mode

12.13 Category B Endpoints License:

- Model 8602 IP softphone

Endpoint License Activation Process

12.14 Only the multi-protocol IP endpoints require a license in order to operate on the Axxess system. Although you can program IP endpoints into the database without having an adequate number of licenses, the system allows only the number of endpoints to come on-line for which licenses exist. The system assigns licenses to endpoints on a first-come, first-served basis. However, once the available licenses have been assigned, the system prevents additional IP endpoints from coming on-line. As additional IP endpoint licenses are loaded into software, the system brings programmed but unactivated endpoints on-line, as appropriate. For detailed information about setting-up IP endpoints for service, see the INSTALLATION and PROGRAMMING chapters.

12.15 When IP endpoints fail to come on-line, the Message Print log identifies the excluded endpoint extensions. At the same time, Alarm #127—Unlicensed IP device failed to come on-line—appears on the unit chassis LCD panel and on the Administrator endpoint display(s).

13. HARDWARE SUMMARY

13.1 The SPECIFICATIONS section of this manual explains environmental requirements of the system, describes the hardware, and gives pre-installation information. The hardware descriptions include chassis, power supply unit, circuit cards, station instruments, and additional equipment needed for the optional features.

14. INSTALLATION, PROGRAMMING, AND MAINTENANCE SUMMARY

14.1 The modular design and self-diagnostic capabilities of the system facilitate installation and repair with minimal down-time. Strict quality control standards for manufacturing and thorough field testing provide the system with the reliability demanded by today's high-technology market.

14.2 The INSTALLATION section contains instructions for assembling the main distribution frame (MDF) and for installing the chassis, power supply, circuit cards, station instruments, and other optional hardware.

14.3 The PROGRAMMING section describes the procedures for programming the system features. After the system is installed, the flexible software allows the database to be customized to meet the customer's needs. An external, customer-provided, personal computer (PC) is used to perform this task. The chapter also provides instructions for backing up the system and voice processing databases, uploading software licenses, and programming the IP endpoints.

14.4 The TROUBLESHOOTING section gives instructions for correcting system problems and replacing defective parts. Part numbers are listed in the REPLACEMENT PARTS section.

14.5 The APPENDICES of this manual provide information about special topics such as Networking, RS232 connections, CP Servers, CPC64/CPU64 configuration, upgrade instructions, and network topology.

15. FEATURES SUMMARY

15.1 System, phone, single-line set, DSS/BLF Unit, and administrator features are listed on the following pages. For complete descriptions and operating instructions, see the SPECIFICATIONS and FEATURES sections of this manual.

NOTE: Those features marked with an asterisk (*) require additional equipment.

A. SYSTEM FEATURES

Hardware and General System Features

- Wide variety of software configurations
- System open architecture interface (OAI) for connecting to an Attendant Console or for communicating with an external computer (e.g., for specialized ACD applications)
- Desktop interface for connecting to Desktop OAI applications
- Flexible station instrument configuration and numbering plan for local and network devices
- RS232 connectors on the Central Processing Unit (CPU) card, Call Processing Card (CPC), and the Options Card (OPC) for connecting optional SMDR/error message output devices, a Voice processing PC, or an external PC for programming

NOTE: The system can have a maximum of 32 RS232 ports used for system services. This includes the RS232 ports on the CPU card, CPC, Options Card(s), and PC Data Port Modules. If a Voice Processing PC is installed, at least one of the 32 RS232 ports must be dedicated to the PC.

- Database battery back-up with voltage test points
- Adjustable bit rates for on-site programming and the optional SMDR/error message output devices (300, 1200, 2400, 4800, 9600, 19200, and 38400 bps, depending on the selected port)
- Variable system timers
- Optional Call Processing (CP) Server (a Windows NT- or 2000-based server that communicates with the phone system via the ATM Switch or a 10Base-T Ethernet connection via a cross-connect Ethernet cable to a CPU020 card with an Ethernet Adapter Card (EAC) with or without redundancy

NOTE: An ATM-based system requires the CP Server with Windows 2000 installed.

- Optional modem kit
- Optional PC wall-mount shelf (accommodates most PC models)
- Optional station message detail recording (SMDR) and/or error message recording output device(s)
- Power failure transfer relay on the CPU card and CPC
- Night transfer relay on the Options Card

- Music port on the CPU card, CPC, and the Options Card(s) for connecting an optional external music source
- Industry-standard message waiting capability on SLCs—requires optional Lamp Option Card (LOC)
- •* Optional off-premises extensions (OPXs) with variable ring cadence
- •* Optional OPX repeaters
- •* Optional facsimile machine
- •* Optional battery back-up
- •* Optional external paging equipment
- Optional playback devices for use with the automated attendant and hunt group features
- Support ISDN Basic Rate Interface—Basic Rate S/T Interface (BRS) and Basic Rate U Interface (BRU) cards

NOTE: The BRU cards are supported by U.S. software only.

- User-equippable system speed-dials, account codes, page zones, keymaps, and SPID/DN/TEI [TEI in Europe]
- ACD/Agent ID hunt groups span nodes
- Language selection for trunks, digit translations, and stations
- Programmable system passwords
- Remote automatic call distribution hunt group premium feature
- Private networking support using the Internet Protocol Resource Card (IPRC)
- •* Emergency Outgoing Access support for IP Phones
- InterPrise® products enhancements
- 99 nodes support

Voice Processing Systems

- Voice Processing PC with Microsoft Windows NT or 2000 operating system or Enterprise Messaging with Windows 2000 Professional or Windows 2000 Server

NOTE: As of version 8.0, the OS/2 Voice Processing Unit is no longer supported.

- Call Routing Announcement application with digit translation and Fax-On-Demand capabilities
- Automated Attendant application
- Voice Mail application including undelete message, return call, Record-A-Call, group lists, and remote messaging features
- SMDR buffering
- Scheduled Timed-Based Application Routing (STAR)
- •* Unified Messaging
- Update message latency period timer
- Voice processor enable flag

- Text-to-speech (TTS) and automatic speech recognition (ASR) functionality (available with the EM system only)
- E-mail Reader (available with the EM system only)
- Confirmation tone played by a Call Routing Announcement (CRA) enhancement
- Display Caller ID name and number enhancement

Embedded Voice Mail Card (EVMC)

- Voice Processing Unit with IBM OS/2 or Linux operating system
- All Voice Processing features except Fax-On-Demand, Unified Messaging, TTS, and ASR capabilities

System Organization and Record Keeping Features

- Ten paging zones including the external paging port(s) on the CPU-type card(s)
- •* Station message detail recording (SMDR)
- •* Error message recording
- •* Programmable reports for toll restriction, automatic route selection (ARS), station data, etc.
- Call cost accounting (estimated)
- Forced, standard, and optional account codes
- Flexible attendant arrangements
- Hunt groups with individually-programmed timers
- •* Uniform call distribution (UCD) hunt groups with optional overflow, announcement, and recall destination stations with playback devices and/or supervisor with call monitoring ability
- •* Automatic call distribution (ACD) hunt groups with optional ACD Agent IDs
- •* Analog voice mail hunt groups
- System alarm display and reporting
- Device baseline extensions
- Local 7/10 digit dialing area code flag (U.S. only)

NOTE: An ATM Switch cannot span area codes unless the area codes are all considered local.

- SMDR display of redirect
- Automatic board detect

Trunk Features

- •* Caller ID [CLIP]
- •* Automatic Number Identification—ANI (U.S. only)
- •* Dialed Number Identification Service—DNIS [DDI in Europe]
- Trunk groups
- Automatic incoming trunk answering

- One-button outgoing call access
- Dual-tone multi-frequency (DTMF) or dial-pulse signaling
- Loop start and/or ground start compatibility
- Hybrid balancing (normal or short) for optimal CO trunk performance
- Outgoing-access, allowed-answer, and ring-in assignments on a station-by-station basis (day and night lists)
- Day and night modes of operation
- Day and night toll restriction on a station-by-station and trunk group-by-trunk group basis (including provisions for three area/office code restriction user groups for the U.S.)
- Absorbed-digit dialing
- Automatic route selection (ARS)
- Direct inward system access (DISA)
- * Automated attendant (through the Voice Processing Unit PC)
- T1 AT&T 4ESS Custom, AT&T 5ESS Custom, National ISDN2, and DMS-100 switch types [or E1 Private Networking or PRI Net 5 S-Type switch types in Europe]
- Programmable gain control on T1/E1/PRI card circuits
- Call routing to the public network
- Basic E&M networking (tie trunks) in the U.S. only
- * Direct inward dialing (DID) trunks in the U.S. only
- Flexible DID [DDI in Europe] ring-in assignments
- Pulse dialing for DID and single-line sets on a Single-Line Adapter (SLA)
- Primary rate interface capability with CLIP and DDI in Europe
- Programmable wink, immediate, or delay start options for DID and E&M trunks in the U.S. only

General Station Features

- Programmable feature codes for easy station operation
- Remote Feature Programming
- Group Listen
- Station-to-station intercom calls
- * Off-hook voice announce (OHVA) calls (the ability to place or receive OHVA calls depends on the type of station instrument used and may be enabled/disabled on a station-by-station basis)
- Inter-station messages and message cancel
- Silent messaging (leaves a message at a station without placing an actual call)
- Ring intercom always
- Automatic camp on to busy stations, trunks, and ARS (ability to camp on to ARS may be disabled on a station-by-station basis)
- Busy trunk/station callback (queue)

- Individual hold
- Hold recall
- Call waiting
- Call transfer to extension numbers and outside telephone numbers
- Transfer to hold
- Transfer recall
- Reverse transfer and group call pick-up
- Conference calls – 12 or 24 conference circuits (unamplified) with up to four parties per conference (if desired, more conference circuits may be enabled using Options Cards)
- Agent Help
- System forwarding (routes calls based on the type of call and the status of the intended station)
- Call forward to extension numbers and hunt group pilot numbers
- Call forward to outside telephone numbers
- System speed dialing (including non-display numbers) – can be programmed to override toll restriction
- Station speed dialing
- Redial (last number dialed or last number saved)
- Emergency extensions on a per-station basis
- Music-On-Hold on a per-device basis
- Do-not-disturb with customized messages (the ability to place a station in do-not-disturb may be enabled/disabled on a station-by-station basis)
- Multilingual capability
- Cancel miscellaneous operations
- Hookflash [recall in Europe]
- Hunt group remove/replace
- House phone
- Manual forwarding to public network
- Peer-to-peer (P2P) audio (IP and SIP endpoints only)
- Call Logging
- Intelligent Directory Search

B. PHONE FEATURES

NOTE: For a complete listing of all the optional equipment that may be installed on each phone, see the information in SPECIFICATIONS beginning on page 2-72.

- Various digital and analog phone models are available (for a complete listing of all the phone models available, see page 2-72)
- Executive Display Phones, Professional Display Phones, Standard Display Phones, Associate Display Phones, Axxess Basic Digital Phones, Eclipse Basic Digital Phones, Model 8500 Phones, Model 8520 Phones, Model 8560 Phones, Executone® Phones, IMX AIM phones, and Inter-Tel/DVK phones all allow the user to individually control voice and tone volume levels for intercom calls, outside calls, background music, etc.

NOTE: The Professional Display, Associate Display, Eclipse Basic Digital, and Executone Phones are supported by U.S. software only.

- •* Liquid crystal display (LCD) for viewing numbers dialed, speed-dial numbers, interstation, do-not-disturb, and reminder messages, date/time, phone identification, station programming, user prompts; etc.
- •* Optional Personal Computer Data Port Module (PCDPM) on Standard Display, Associate Display, Executive Display, Professional Display, Model 8520, or Model 8560 Phones for off-hook voice announce receive capability and for the installation of a DSS/BLF Unit and/or an optional Modem Data Port Module (may also be used to connect an output device or a programming PC)
- •* Optional Modem Data Port Module (MDPM) on Standard Display, Executive Display, Model 8520, or Model 8560 Phones for installing a modem-equipped data device (for simultaneous voice/data communication) or a single-line set – requires a PCDPM
- •* Optional Data Port Module on IMX or ESP 12-/24-line, Inter-Tel/DVK, and GMX 24-line analog phones for installing a modem-equipped data device (for simultaneous voice/data communication), or for installing a loud ringing adapter (LRA) and signaling device(s)
- 12-button dialpad
- "Hot" keypad – allows dialing of feature codes, extension numbers, and trunk access codes without pressing the Special button (∞) or lifting the handset
- Up to 250 keymaps for determining the exact layout and function of the buttons
- Feature buttons for one-button access to feature codes (some of the buttons may be designated as user-programmable)
- Speed-dial buttons with light-emitting diode (LED) indicators
- Call appearance buttons with LED indicators (for incoming call indication/access)
- Secondary extension appearance buttons with LED indicators (show call status indications for the designated "primary" station)
- Individual trunk buttons with LED indicators (for incoming and outgoing call indication/access)
- Trunk group buttons with LED indicators (for outgoing call access and trunk group status)
- Speakerphone (allows handsfree operation on outside calls and intercom calls)

- Speakerphone on/off
- Slide-out directory card
- User-programmable ring tone
- Self-test function
- Hearing aid-compatible (HAC) handset
- Reversible baseplate for wall mounting
- * Optional headset
- * Background music
- Microphone mute
- Handsfree answer enable/disable
- On-hook dialing/monitoring
- System hold
- Station call monitoring (hunt group supervisors only)
- Do-not-disturb override (the ability to override do-not-disturb may be enabled/disabled on a station-by-station basis)
- Page remove/replace
- Intercom (extension number) and outside (system speed-dial number) directories
- Feature directory to locate and/or dial feature codes
- Automatic intercom and/or outside call access
- Redirect call feature code
- LCD contrast feature code (Models 8662, 8622, 8620, 8660, and 8560 only)
- Dedicated headset jack (Model 8520 and 8560 Phones only)
- Programmable Message Lamp (Model 8500, 8520, and 8560 Phones only)
- Network Address Translation (NAT) traversal for IP endpoints

C. SINGLE-LINE FEATURES

- Audio message waiting indication capability
- Programmable hookflash/recall recognition
- Programmable ring cadence and gain levels for OPX applications
- Optional SLA for connecting an off-premises extension to the DKSC
- * Caller ID [CLIP] support
- * Pulse dialing support for single-line sets (version 2.x or later software is required)

D. IP DEVICES

15.2 The Internet Protocol Card (IPC) supports a combination of up to eight IP PhonePlus, IP SoftPhone, Model 8660 Phone, or IP Single-Line Adapter (SLA) ports to provide voice-over-IP functionality on Inter-Tel systems. The v1.x firmware device Internet Protocol Resource Card (IPRC) supports a combination of up to 16 IP Phone Plus, IP SoftPhone, Model 8660 Phone, or IP SLA ports or IP trunks (via MGCP). The v8.1 and 9.0.0 firmware IPRC supports up to 32 IP PhonePlus, Model 8660 Phone, multi-protocol phone, Model 8602 IP softphone, or IP SLA ports or IP trunks. (The daughter card is required for the additional 16 ports.) The IPC uses a 10Base-T Ethernet interface to the stations and an RS-232 interface for diagnostics and programming. The IPRC uses a 10/100Base-T Ethernet interface to the stations and an RS-232 interface for diagnostics and programming. For details, see the *IP Devices Installation Manual* (version 1.3 or later).

NOTE: The v8.1 and 9.0.0 firmware IPRC does not support the IP SoftPhone.

E. DIRECT STATION SELECTION/BUSY LAMP FIELD (DSS/BLF) UNIT FEATURES

- •* Up to 20 total units per system (e.g., five Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phones equipped with four units each)
- Each Executone IDS phone can support up to three 48-key DSS consoles
- Each single unit has either 48, 50, or 60 buttons with LED indicators for one-button access to extension numbers/features (the LEDs show the status of the assigned stations, hunt groups, or other resources)
- Up to 20 keymaps for DSS/BLF Units and 30 keymaps for IDS DSS Consoles for determining the layout and numbers accessed by the buttons

NOTE: The Associated and Professional Display Phones, Executone IDS phone, and IDS DSS Consoles are supported by U.S. software only.

F. MINI-DIRECT STATION SELECTION (DSS) UNIT FEATURES

- Number of units per system depends on the number of Model 8560, 8520, or 8660 Phones and circuits available
- Each single unit has 16 buttons with LED indicators for one-button access to up to 16 extension numbers/features (the LEDs show the status of the assigned stations, hunt groups, or other resources)
- Keymap, which is included as part of the Model 8660/8560/8520 Phone keymap, for determining the layout and numbers accessed by the buttons

NOTE: Version 7.0 and later software is required to use the Mini-DSS Unit.

G. ATTENDANT FEATURES

- Accessed by dialing 0 at the stations they serve
- Central operator for incoming calls
- Message center

- Recall destination for unanswered calls
- Immediate off-hook voice announce (OHVA) calls (requires special programming and a DSS/BLF Unit or an Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone with assigned DSS/BLF buttons)

NOTE: The Associated and Professional Display Phones are supported by U.S. software only.

H. ADMINISTRATOR FEATURES

- Place the local node or other nodes in the network in night mode or day mode
- Set the date and time of the local node
- Set the network date and time and re-synchronize clocks in the network.
- Make database changes (see page 5-276 for a list of programming areas)
- Program system speed-dial numbers on the local node
- Receive and clear displayed system and network alarms
- Use diagnostic mode features:
  - — Freeze and unfreeze database history on local nodes or other nodes in the system using freeze "zones"
  - — Print error logs
  - — Seize specific devices for troubleshooting purposes
- Use Diagnostic - dump node information feature code
- Swap extensions without accessing DB Programming
- Enable/disable the modem on the CPC
- Reset the modem on the CPC

I. ATTENDANT CONSOLE FEATURES

NOTE: Versions 7.0 and later software does not support Attendant Console version 2.0. If running version 7.0 or later, Attendant Console version 3.0 is required.

- Designed for use with Microsoft Windows operating system software
- Quick, simple call answering and transfer
- Visual display of system call activity, system directories, system alarms, date and time, etc.
- Automatic call priority
- Optional headset operation
- Swap extension feature

NOTE: See page 2-83 for additional information.

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1. INTRODUCTION

1.1 This section of the manual describes the following Inter-Tel® system hardware:

— Cabling and Main Distribution Frame (MDF)

— ATM Switch

— Chassis

— Power Supply Unit

— Circuit Cards

— Station Instruments

— Additional System Equipment

1.2 The trunks and stations are controlled by the circuit cards in the chassis. A single-chassis system has seven slots available for station, trunk, and optional resource cards. A dual-chassis system has up to 14 slots, a tri-chassis system has up to 21 slots, and a quad-chassis system has up to 28 slots. An eight-chassis ATM-based system has up to 56 slots.

1.3 *The maximum number of stations and trunks that can be installed depends on the number and type of cards and station instruments used*. In most circumstances, the theoretical maximums listed in this manual are never reached due to the variety of cards and station instruments installed.

1.4 A variety of "digital" station instruments can be used on the phone system, including:

- Axxess® Basic Digital Phones
- Eclipse™ Basic Digital Phones
- Standard Display Phones
- Associate Display Phones
- Executive Display Phones
- Professional Display Phones
- Model 8500, 8520, and 8560 Phones

NOTE: There are certain limitations associated with the Model 8500, 8520, and 8560 Phones when used with versions 3.0-6.x system software. See page 5-135 for details.

ALSO: The Professional Display, Associate Display, and Eclipse Basic Digital Phones are supported by U.S. software only.

- 50- or 60-key Digital Direct Station Selection/Busy Lamp Field (DSS/BLF) Units
- 16-key Mini-DSS Units

NOTE: Version 7.0 or later software is required to support the Mini-DSS Units.

1.5 In addition, a wide variety of "analog" station instruments can be used on the Inter-Tel phone system, including:

- IMX 24-Line AIM Phones (display only)
- IMX 12-Line AIM Phones (display only)
- IMX 8-Line Dual-Circuit AIM Phones (display only)
- IMX 24-Line Phones (standard and display)
- IMX 12-Line Phones (standard and display)
- IMX 8-Line Phones (non-display only)
- Inter-Tel/DVK 24-Line Phones (standard and display)
- Inter-Tel/DVK 12-Line Phones (standard and display)
- Inter-Tel/DVK 8-Line Phones (standard and display)
- GMX 24-Line Phones (standard and display)
- GMX 12-Line Phones (non-display only)
- Executone® Model 160, 64, 32, 18, and 12 Phones
- Executone Agent Terminal
- Executone 6-, 17-, and 29-Key Wave Phones
- 48-Key DSS Consoles (for use on Executone phones only)
- Industry-standard single-line, dual-tone multi-frequency (DTMF) or pulse-tone sets (v2.x or later software is required for pulse-tone sets)

NOTE: The volume adjustment range of non-AIM and non-DVK analog phones may not be suitable in all user environments. In such cases, Inter-Tel recommends using AIM and/or DVK phones, if possible.

ALSO: The Executone Model 160, 64, 32, 18, and 12 Phones, Agent Terminal, 6-, 17-, and 29-Key Wave Phones, and 48-Key DSS Consoles are supported by U.S. software only.

1.6 The phone system also supports the following IP devices:

- Axxess IP PhonePlus
- Eclipse IP PhonePlus

NOTE: The v8.1 and 9.0.0 firmware IPRC does not support IP SoftPhone.

- IP SoftPhone (requires a v1.x firmware device IPRC)
- Models 8600, 8620/8622, 8662, and 8690 multi-protocol endpoints (support both SIP and ITP)
- IP single-line adapters (IP SLAs)

1.7 See the latest *IP Devices Installation Manual* (document part no. 835.2195) for details.

1.8 You can also use the following devices with versions 8.1 and later software:

1.9 Versions 8.1 or later:

- Model 8601 Pocket PC SoftPhone—See the *Model 8601 Installation Manual* (document part no. 835.2736) for details
- Models 8664, 8665, and 8668 wireless endpoints—See the *SVP Server and Model 8664/8665/8668 Installation Manual* (document part no. 935.4521) for details

1.10 Versions 9.1 or later:

Model 8602 IP softphone (requires IPRC firmware v9.0.0 or later and supports ITP)

1.11 A wide variety of optional system equipment can also be installed, including:

- Optional Call Processing (CP) Server (a Windows 2000- or 2003-based server that communicates with the phone system via the ATM Switch or a 10Base-T Ethernet connection via a cross-connect Ethernet cable to a CPU020 card with an Ethernet Adapter Card (EAC). See Appendix C — CP Server.

NOTE: A CP Server running Windows 2000 is required for an ATM-based system.

- Multi-port Voice Processing PC and applications software for integrated voice processing (voice mail, automated attendant, hunt group announcement and overflow, optional Fax-On-Demand, SMDR storage, Record-A-Call, etc.) and Unified Messaging. See page 2-104.
- The Embedded Voice Mail Card (EVMC) is an OS/2- or Linux-based voice mail card that can be used to provide small systems with Voice Processing features, except Fax-On-Demand, Unified Messaging, TTS, and ASR. See Appendix D — CPC64/CPU64 Configuration.
- Inside Track® station message detail accounting (SMDA) software and, if the Voice Processing PC is not used, optional call record buffer such as a PollCat (Pollable Call Accounting Terminal), CRB (Call Record Buffer), or ILB (In-Line Buffer). Refer to the manual included with the Inside Track software.

NOTE: Inside Track is not applicable to European systems.

  - — External paging speaker equipment (see pages 2-19 and 3-239).
  - — External music source (see page 3-237).
  - — Power failure transfer equipment (see page 2-103).
  - — Night transfer equipment (see pages 2-30 and 2-103).
  - — Off-premises extensions (OPXs) and OPX repeaters (see pages 2-86 and 3-69).
- Playback devices—if the Voice Processing PC is not used (see pages 2-87 and 3-208).
- Output device(s) for the station message detail recording (SMDR) and error message recording features (see pages 2-100 and 3-234).
- System battery back-up—uninterruptable power supply or standby power supply (see page 2-104).
- Facsimile machine (see page 2-104).
- PC wall-mount shelf—accommodates most PC models (see page 8-9 in REPLACEMENT PARTS).

2. CABLING AND THE MAIN DISTRIBUTION FRAME (MDF)

2.1 Connections between the trunks, station instruments, external equipment, and the station and trunk circuit cards are made at the Main Distribution Frame (MDF). The MDF is made up of industry-standard, 66M1-50-type terminal blocks [237A Krone blocks in Europe] and, if using T1 or E1/PRI cards, eight-conductor (RJ48C) modular jack assemblies [RJ45s in Europe]. Bridging clips are used on all terminal blocks to complete the connections.

2.2 With a U.S. system, the MDF requires at least a 4 × 6-foot (1.2 × 1.8-meter), ¾-inch (2-cm) plywood backboard. (Depending on the number of stations and trunks installed, *a larger backboard may be necessary.*) This allows room to mount the terminal blocks, modular jack assemblies, Single-Line Adapters (SLAs), AC transformers, and any optional peripheral equipment.

A. STATION CONNECTIONS

2.3 Station cable is connected to the MDF at one end, and a modular connecting block at the other end. The modular line cord of the telephone is then plugged into the connecting block.

- For each Executive Display Phone, Professional Display Phone, Standard Display Phone, Associate Display Phone, Axxess Basic Digital Phone, Eclipse Basic Digital Phone, single-line set, Model 8500, 8520, and 8560 Phones, and playback device, twisted single-pair (24 AWG, level-1, 24 gauge thickness or larger—smaller number gauge = larger thickness) cable is run from the station location to the MDF.

  NOTE: The Professional Display, Associate Display, and Eclipse Basic Digital Phones are supported by U.S. software only.

- For each analog phone (including Executone phones), twisted three-pair (24 AWG, level-1, 24 gauge thickness or larger -- smaller number gauge = larger thickness) cable is run from the station location to the MDF.

NOTE: If desired, two- or three-pair twisted cable and four- or six-conductor modular jacks may be used for all station connections. This allows the various types of station instruments to be easily interchanged, if necessary. (If three-pair cable is used, extra terminal blocks and cross-connect wiring techniques are required.)

2.4 For a complete list of station cabling requirements, see page 3-16.

2.5 Industry-standard, 25-pair telephone cable is used to connect the station blocks to their corresponding station cards. Female 50-pin amphenol-type connectors on the cables attach to the male connectors on the station cards in the chassis.

2.6 To help reduce EMI and RFI, it may be necessary to equip the trunk cables with ferrite beads. (See page 3-26 for more information.) Two ferrite split beads (two half beads) are installed on each 25-pair cable just below the amphenol-type connector. The part number for a single split bead is 808.1195.

B. TRUNK CONNECTIONS

2.7 With a U.S. system, the CO trunks are terminated on telephone company RJ-type blocks, as required by FCC regulations. Cross-connect cable is used to connect the RJ-type blocks to the CO terminal blocks on the MDF. The trunks are then connected to the Loop/ Ground Start Cards (LGCs) and/or Loop Start Cards (LSCs) using industry-standard, 25-pair cable. Female 50-pin amphenol-type connectors on the cables attach to the male connectors on the cards.

2.8 Also, the DID trunks are terminated on telephone company RJ-type modular jacks (or on RJ-type blocks that are then cross-connected to modular jack assemblies), as required by FCC regulations. Two-pair, non-reversing, mod-to-mod line cord is used to connect each DID trunk to the appropriate jack on the corresponding SLA.

2.9 With a European system, CO [local exchange] trunks are terminated on the TJF then jumpered to the MDF. Some CO connections are via the fallback system.

2.10 To help reduce electromagnetic interference (EMI) and radio frequency interference (RFI), it may be necessary to equip the trunk cables with ferrite beads. (See paragraph 2.6 for more information.)

2.11 With a U.S. system, T1 spans and Primary Rate Interface (PRI) spans are terminated on telephone company RJ48C modular jacks (or on RJ-type blocks that are then cross-connected to modular jack assemblies), as required by FCC regulations.

2.12 With a European system, Primary Rate Interface (PRI) spans are terminated on network provider RJ48C line jack units (or on RJ-type blocks that are then cross-connected to line unit assemblies).

2.13 Four-pair, non-reversing, mod-to-mod line cord is used to connect each T1 and/or PRI span directly to the T1 or E1/PRI jack on the corresponding T1, E1/PRI, or T1/E1/PRI card. (Or, to use the DB15 connector instead, see page 3-36.)

2.14 Install gas discharge tubes with silicon avalanche suppressors on all non-T1/E1/PRI trunks for lightning protection. (For T1/E1/PRI spans, the CSU itself provides the lightning protection.) Also, in areas with frequent occurrences of lightning, the cable between the telephone company [network provider in Europe] termination and the gas discharge tubes should be at least 75 feet [22.5 meters] long (the cable may be coiled up if desired).

2.15 The Inter-Tel phone system has the following trunk characteristics:

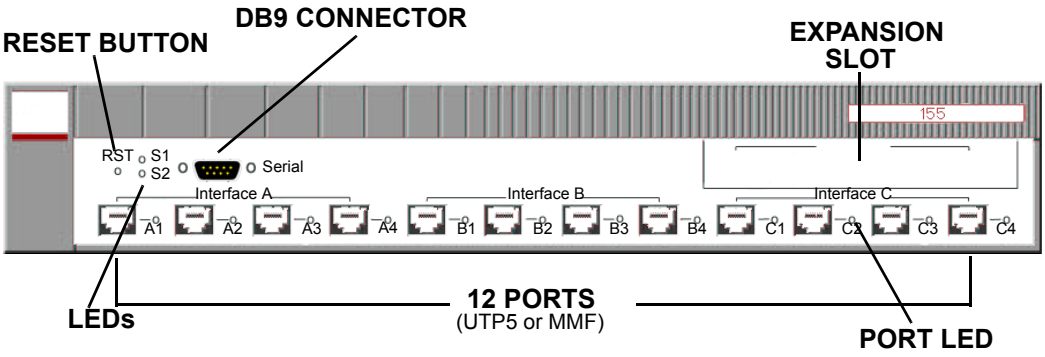
Characteristics	
Loss from:	
CO to phone	0dB (@1 kHz, 0 ft)
CO to single-line set	0dB (@1 kHz, 0 ft)
CO to CO	0dB (@1 kHz, 0 ft)
Ringer equivalence	0.6B
Ringing voltage	40VRMS minimum
Ring frequency	17-63 Hz
Loop current	18mA minimum
Protection	
Tip-to-ring	400V transient

3. ATM SWITCH

3.1 Previously, the Ericsson® (Marconi) *ForeRunnerLE*® 155 ATM Switch was used for expanding the Axxess system to eight chassis. This switch, however, was recently replaced with the Ericsson (Marconi) ASX™ 200BX switch. Each of these switches is described in the following sections.

A. FORERUNNER LE 155

3.2 The Ericsson (Marconi) *ForeRunnerLE* 155 ATM Switch features a 2.5 Gbps, non-blocking switching connection and an integrated CPU to reduce cost and increase reliability. It also provides 12 ports, each with a 155 Mbps UTP Category 5 cable or MMF connector. If necessary, one of two additional modules can be added to the expansion slot. Each module is available with four MMF ports or two MMF and two SMF ports.



3.3 There are four ATM Switch configurations available, based on the connector type(s). These are as follows:

- 12 UTP5 ports with two MMF and two SMF ports in the expansion slot (16 ports total); part no. 550.5270 [IT159 in Europe]
- 12 UTP5 ports with four MMF ports in the expansion slot (16 ports total); part no. 550.5253 [IT130 in Europe]
- 12 UTP5 ports; part no. 550.5251 [IT128 in Europe]
- 12 MMF ports; part no. 550.5252 [IT132 in Europe]

3.4 Each configuration further divides the ports into interfaces. These interfaces, which are labeled A-C (or D for a populated expansion slot), consist of four ports (1-4). Each individual port is then identified by the interface letter and port number. For example, the third port under Interface B is considered Port B3.

3.5 The architecture of the ATM Switch allows each interface to function independently. Using the Element Manager, you can reset one group of ports without affecting the others. For this reason, use a separate interface for each chassis connection when possible. For example, if you have four chassis, connect the first to Port A1; the second to Port B1; the third to Port C1; and the fourth to Port A2 (or D1 if populated). When configured this way, you can reset ATM connections almost entirely on a per-chassis basis. For example, if chassis two has a bad connection but chassis three is running, you can reset Interface B without bringing down the third chassis.

NOTE: The chassis PCMAs must have the same connector types (UTP5, MMF, or SMF) as those provided on the ATM Switch.

B. ASX-200BX

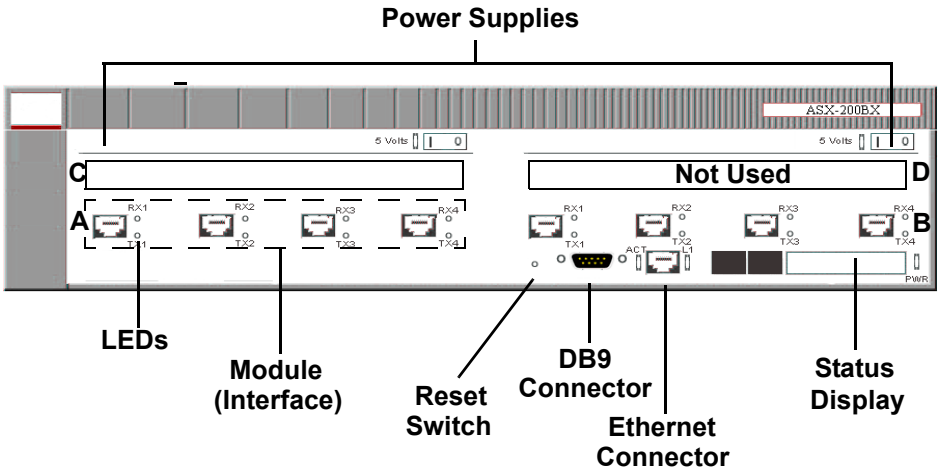
3.6 The ASX-200BX switch provides a 2.5 Gbps non-blocking switching fabric that supports up to three "modules" that are installed into available expansion slots. Because each module provides a different connection type, there are several possible ATM configurations. These are as follows:

- 8-port UTP5 (Copper); part no. 550.5296
- 8-port Multi-Mode Fiber (MMF); part no. 550.5297
- 4-port MMF/4-port UTP5; part no. 550.5298
- 4-port MMF/4-port UTP5/4-port Single-Mode Fiber (SMF); part no. 550.5299

3.7 Each module, or interface, supports up to four ports, for a total of up to 12 ports on one switch. The required modules depend on the type of PCMA cards in the system chassis and the ATM Adapter card in the Call Processing Server (CPS).

3.8 In addition, the ASX-200BX includes a Switch Control Processor (SCP), which continuously monitors the switch. The status is then displayed in the light-emitting diode (LED) Status Display on the front of the switch.

3.9 The following is an illustration of an ASX-200BX with 8 UTP5 connectors.



3.10 Each switch has the following features:

- **DB9 Connector**: Provides a serial interface for connecting to a programming PC. The default settings for the serial port are:
  - *Baud Rate:* 9600
  - *Data Bits:* 8 bits
  - *Parity:* none
  - *Stop Bit:* 1 bit
- **Ethernet Connector**: Provides an IP port for connecting to a programming PC. You must program the IP address for this connector before you can use it (see page 3-166).
- **Redundant Power Supply**: Contains two power supplies, providing backup power capabilities.

- **Status Display**: Indicates the current status of the switch. If the switch is functioning, this displays the switch name.
- **Reset Switch**: Resets the switch and runs power-on diagnostics. You should reset the switch only when instructed to do so by technical support personnel.

3.11 Each module on the ASX-200BX functions as a separate interface, which allows you to reset one module at a time. If configured so that each chassis is connected to a separate module, you can reset the ATM connections on a per-chassis basis. To identify ports, each module is labeled A-C, and each port is numbered 1-4. When referring to a port in programming, use the scheme 1*XY* where *X* is the module letter and *Y* is the port number. For example, the third port in module A is 1A3; the second port in module B is 1B2; etc.

4. CHASSIS

A. CHASSIS DESCRIPTION

4.1 The basic, single-chassis system houses the system power supply, an eight-slot circuit cardfile assembly, and the circuit cards, including:

- Call Processing Card (CPC) or Central Processing Unit (CPU) Card
- Loop/Ground Start Cards (LGCs), Loop Start Cards (LSCs), LGC Daughter Cards (LGC-D), LSC-D
- T1 Card (T1C) with optional Primary Rate Interface (PRI) Daughter Card, E1 Card (E1C) with optional PRI, or T1/E1/PRI Card
- Basic Rate S/T Interface (BRS) and/or Basic Rate U Interface (BRU) Cards
- Digital Keyset Cards (DKSCs), Analog Keyset Cards (AKSCs), Executone Keyset Cards (EKSCs), Single-Line Cards (SLCs)
- Options Cards (OPCs)
- Internet Protocol Cards (IPCs) and/or Internet Protocol Resource Cards (IPRCs)
- Embedded Voice Mail Cards (EVMCs)

4.2 For small, single-chassis systems, the CPU64 configuration, using the CPU64 or CPC64, EVMC, and 4-Amp power supply provides a full-featured system at reduced cost. See Appendix D — CPC64/CPU64 Configuration for more information about this special configuration.

4.3 For increased port capacity, two basic chassis may be connected together to form a dual-chassis system. Each chassis requires the 9-amp power supply and eight-slot circuit cardfile assembly. The second chassis, however, does *not* require its own CPU card or CPC. The two chassis are connected together by a single ribbon cable.

NOTE: The ribbon cable, U.S. part number 813.1636-1B was modified in September 1998 to improve grounding. Before grounding was improved, there were reports of DKSCs and SLCs going off-line periodically. This caused M6029 error messages to appear in the message print output. Systems experiencing this problem should have the newer model ribbon cable (look for the -1B in the part number).

4.4 If required, three basic chassis may be connected together to form a tri-chassis system (see Figure 2-1 on page 2-13 and Figure 2-3 on page 2-14). Like the second chassis, the third chassis has its own power supply unit and eight-slot circuit cardfile assembly. However, unlike the second chassis, the third chassis *does* require its own slave-version CPU020/PCM card. The first and second chassis (master dual-chassis) are connected to the third (slave) chassis through a special fiber-optic cable.

NOTE: CPU020/PCM cards are required for tri- and quad-chassis systems. In addition, versions 8.0 and later require a CP Server for tri- and quad-chassis systems. This means that the first chassis must have an EAC so that it can connect to the CP Server.

4.5 For maximum port capacity without an ATM Switch, four basic chassis may be connected together to form a quad-chassis system (see Figure 2-1 on page 2-13 and Figure 2-3 on page 2-14). Like the other three chassis, the fourth chassis has its own power supply unit and eight-slot circuit cardfile assembly. However, unlike the first and third chassis, the fourth chassis does *not* require its own CPU020/PCM card. The first and second chassis (master dual-chassis) are connected to the third and fourth chassis (slave dual-chassis) through a special fiber-optic cable.

4.6 For maximum port capacity with an ATM Switch, four dual-chassis systems may be connected together to form an eight-chassis node (see page 2-13 for an example of an ATM system). Like the dual chassis, each second chassis has its own power supply unit and eight-slot circuit cardfile assembly. Each chassis pair also has its own CPU020/PCMA card, which is connected to the ATM Switch through a CAT5 or multi-mode fiber (MMF) cable. Because each pair is independently connected to the ATM Switch, there is no master-slave relationship like there is in a quad-chassis system.

Multiple-Chassis Configurations

4.7 With the fiber-optic link, there are a number of valid ways to connect multiple chassis together, as shown in Figures 2-2 and 2-3 on the following pages. Note that for reference purposes, *multiple-chassis configurations are **always** numbered as shown below* – even though the master section may have only one actual chassis installed. (Chassis can be rack mounted instead of the horizontal arrangement shown on page 2-13, but the numbering and connections remain the same.)

NOTE: In tri-/quad-chassis installations, the master T1, E1/PRI, or T1/E1/PRI card (if one exists) should be located in the master dual-chassis. The first (master) and third (slave) chassis in all tri-/quad-chassis installations must be equipped with 9 amp power supplies. The PCM-F card or PCM-F/EAC requires a 9 amp power supply. Also, if the two portions of a tri-/quad-chassis system are installed on the same vertical plane on the same MDF, the first pair of chassis (master dual-chassis) must be separated from the second pair of chassis (slave dual-chassis) by at least one foot to provide proper air circulation. In addition, tri-/quad-chassis systems require a CP Server to run software version 8.0 or later.

Figure 2-1. *Quad-Chassis Configuration*

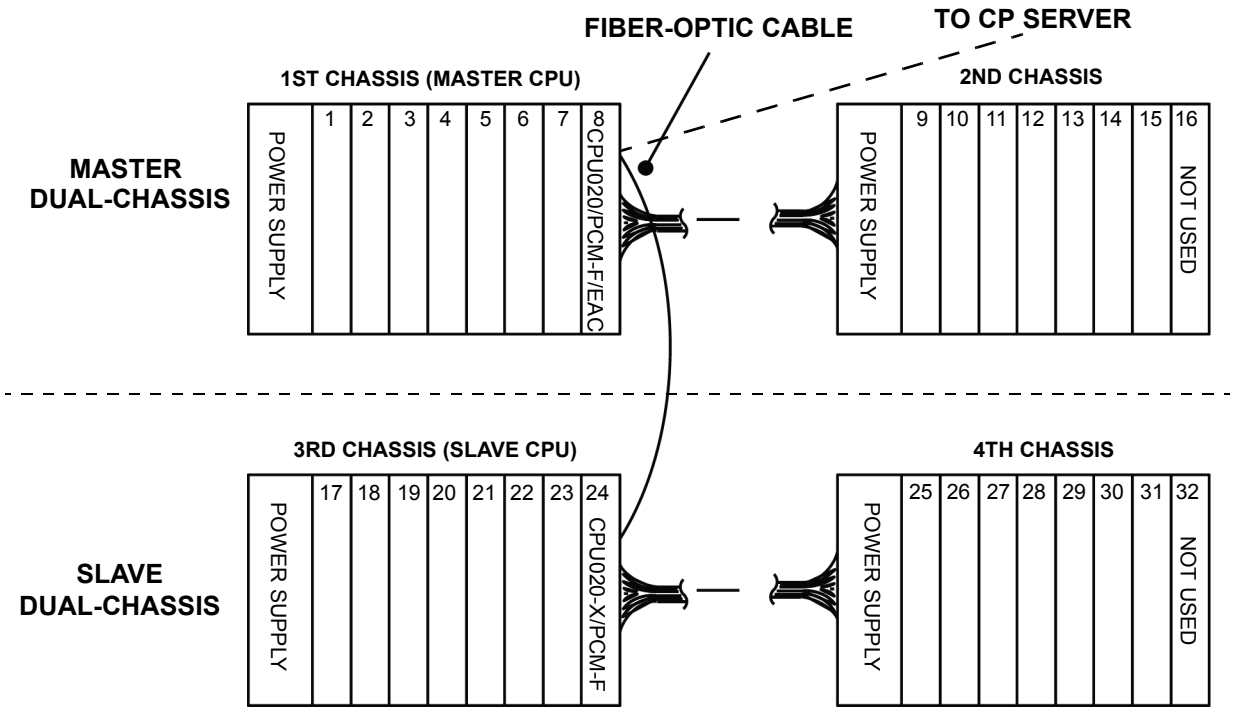


Figure 2-2. *Sample ATM-Based System Configuration*

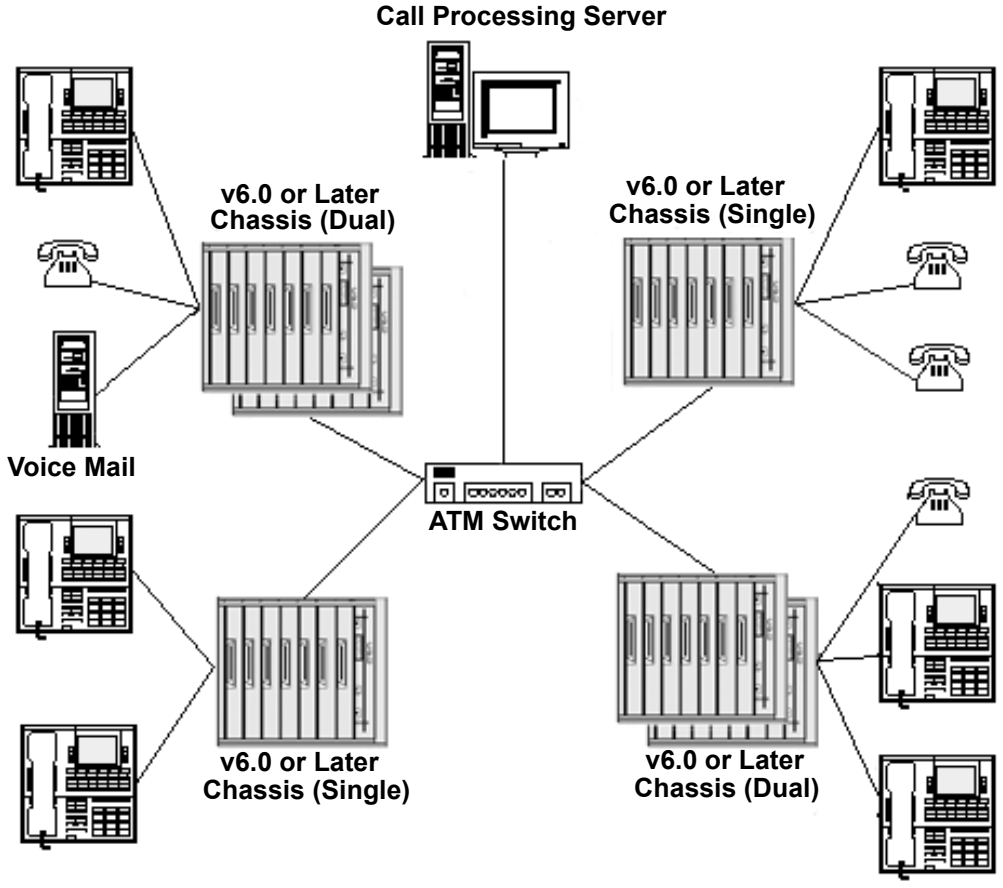
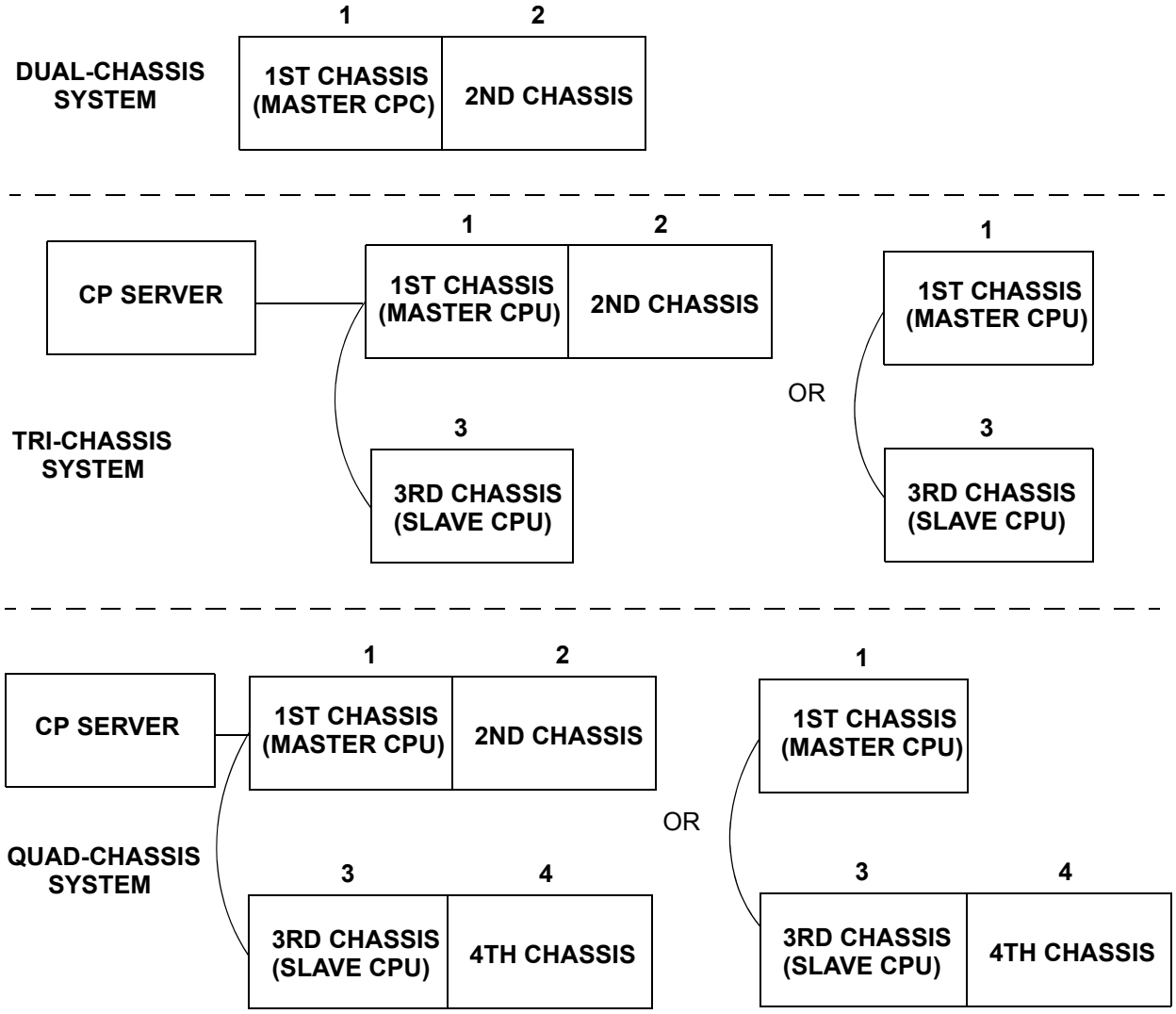


Figure 2-3. *Valid Multiple-Chassis System Configurations without the ATM Switch*



Equipment Chassis Redesigned

4.8 The original equipment chassis, backplane, and power supply unit were redesigned. The new-style equipment chassis has an extra cardfile slot that allows one of the new-style power supply units to be attached directly to the backplane. (With the original chassis, the power supply unit was installed on top and was connected to the backplane through an interface cable.) As a result of the redesign, the new-style chassis is shorter and wider than the original chassis.

NOTE: European systems do not support the original power supply unit.

Original Equipment Chassis

4.9 For an illustration of the original equipment chassis, see Figure 2-8 on page 2-105. For installation instructions, see page 3-74.

4.10 The *approximate* dimensions of a basic equipment chassis are:

Height	15.0 in (38.1 cm)
Width	13.0 in (33.0 cm)
Depth	12.5 in (31.8 cm)
Weight	
- Empty	21.0 lbs (9.45 kg)
- Loaded	30.0 lbs (13.5 kg)

4.11 The *approximate* dimensions of a dual-chassis system are:

Height	15.0 in (38.1 cm)
Width	26.0 in (66.0 cm)
Depth	12.5 in (31.8 cm)
Weight	
- Empty	42.0 lbs (18.9 kg)
- Loaded	58.5 lbs (26.3 kg)

New-Style Equipment Chassis

4.12 For an illustration of the new-style equipment chassis, see Figure 2-9 on page 2-106. For installation instructions, see page 3-83.

4.13 The *approximate* dimensions of a basic equipment chassis are:

Height	10.5 in (26.7 cm)
Width	16.5 in (41.9 cm)
Depth	11.5 in (29.2 cm)
Weight	
- Empty	16.0 lbs (7.3 kg)
- Loaded	30.0 lbs (13.5 kg)

4.14 The *approximate* dimensions of a dual-chassis system are:

Height	10.5 in (26.7 cm)
Width	33.0 in (83.8 cm)
Depth	11.5 in (29.2 cm)
Weight	
- Empty	32.0 lbs (14.5 kg)
- Loaded	58.5 lbs (26.3 kg)

B. FUNCTIONAL DESCRIPTION

4.15 Together, the circuit cards in the chassis perform the switching activities for the system, detect incoming calls, process data-controlled features, control the interaction of station instruments, trunks, intercom calls, and Voice Processing applications, etc.

4.16 The system is a microprocessor-controlled, pulse code modulation (PCM), time-division multiplex (TDM) system.

- The CPU64 and CPU128 cards have 6 megabytes (MB) flash ROM and 4 MB random access memory (RAM).
- The CPU128 with memory daughter card has 6 MB flash ROM and 6 MB RAM.

NOTE: Versions 8.0 and later require the CPC for 64, 128, or 256 systems.

- The CPU020-D card has 4 MB flash ROM and 4 MB RAM, the CPU020-Q card has 4-6 MB flash ROM and 8 MB RAM.
- The CPU020-X has 2 MB flash ROM and 2 MB RAM.

NOTE: Versions 8.0 and later require a CPU020/PCM/EAC with a CP Server for 512 or higher systems.

- The CPC has 16 MB flash ROM, 32 MB SDRAM, and 8 MB battery-backed SRAM.
- Each phone contains a microprocessor that communicates via the Digital Keyset Card with the microprocessor on the CPU card or CPC.
- The CPU020/PCM can have extended memory added. The kit (U.S. part number 828.1405) contains four RAM chips that are installed in sockets on the CPU020/PCM.
- The CPU020/PCM/EAC and CPU (CP Server) are designed to support heavily loaded systems with high call traffic. The call processing software has been moved from the system CPU to a CP Server that communicates with the phone system via a 10Base-T Ethernet connection. For more information about this special CPU, see Appendix C — CP Server.
- The CPU020/PCMA and Windows 2000-based CPU (CP Server) are designed to support an ATM-based system. The call processing software has been moved from the system CPU to a Windows 2000-based server that communicates with the phone system via the ATM Switch. For more information about this special CPU, see Appendix C — CP Server.

Voice Channel Allocations

4.17 The circuitry and software in each single equipment chassis combine to produce an "integrated switching matrix" of 128 time slots (4 PCM highways [buses] × 32 time slots per highway). A quad-chassis system has 512 time slots (16 PCM highways [buses] × 32 time slots per highway). In general, the circuit cards in slots one and two are assigned to highway one, the cards in slots three and four are assigned to highway two, and so on. Therefore, there are at least 16 time slots available for each card installed in the chassis.

NOTE: The ATM connection to the chassis in a tri-/quad-chassis configuration provides 256 time slots for communication between the two halves (8 channels × 32 time slots per channel). Of the 256 times slots, 224 are available as non-blocking voice channels. All of the DSP resources on the CPU020 cards, OPCs, and DKSC-16s can be shared across the fiber-optic connection.

4.18 For the most part, having 16 time slots available per card presents no problem. For example, each Digital Keyset Card (DKSC), whether equipped with any Single-Line Adapters or not, requires 16 time slots. Each Loop/Ground Start Card (LGC) and each Loop Start Card (LSC) requires four time slots. (If equipped with Daughter Cards, LGCs and LSCs require eight time slots.) And, each Single-Line Card (SLC) or 16-Port Single-Line Card (SLC-16) requires eight or sixteen time slots respectively.

4.19 Some cards could require more than 16 time slots. For example, each Options Card (OPC) and each BRI card (BRS or BRU) could possibly need more than 16 time slots, depending on the number of Digital Signal Processors (DSPs) installed on the card. With a U.S. system, each T1 Card (T1C) and each T1/PRI card and each T1/E1/PRI card could require up to 24 time slots. With a European system, each E1/PRI Card and each T1/E1/PRI Card could require up to 30 time slots. And, each 16-Port Digital Keyset Card (DKSC-16) could require up to 32 time slots.

4.20 To reduce the chance of T1C, T1/PRI, E1/PRI, T1/E1/PRI, OPC, BRS, BRU, and DKSC-16 circuit blocking, the time slots for these types of cards are allocated dynamically (e.g., they are allocated as needed). In addition, since T1Cs, T1/PRIs, E1/PRIs, T1/E1/PRIs, OPCs, BRS, BRU, and DKSC-16s can use unused time slots from any highway in its respective equipment chassis, the chance of any optional DSP resource or circuit being blocked is extremely remote and can be prevented (see the following note). Therefore, the phone system is considered to be a virtually non-blocking system.

NOTE: If multiple OPCs, T1Cs, T1/PRIs, E1/PRIs, T1/E1/PRIs, BRSs, BRUs, and/or DKSC-16s are installed in expanded chassis, resource blocking can be prevented by splitting the cards between the cardfiles and/or installing them in the same cardfile as the LGCs or LSCs. In tri-/quad-chassis installations, the master T1C, E1/PRI card, or T1/E1/PRI card (if one exists) should be located in the master dual-chassis.

4.21 The following chart shows the number of voice channels that are used by the various cards. Note that dynamically allocated voice channels are not used until they are needed.

Table 2-1. *Maximum Voice Channels for Various Cards*

Card	Max. Voice Channels Used
E1C	30 – Dynamically Allocated
T1C	24 – Dynamically Allocated
T1/PRI	23 – Dynamically Allocated
E1/PRI	30 – Dynamically Allocated

Table 2-1. *Maximum Voice Channels for Various Cards (Continued)*

Card	Max. Voice Channels Used
BRS	16 – Dynamically Allocated
BRU	16 – Dynamically Allocated
DKSC-16 or DKSC-16+	32 – Dynamically Allocated
DKSC	16 – Permanently Allocated
AKSC	16 – Permanently Allocated
EKSC	24 – Dynamically Allocated
IPC or IPRC	16 – Dynamically Allocated
IPRC with IPRC-D	32 – Dynamically Allocated
SLC	8 – Permanently Allocated
SLC-16 or SLC-16+	16 – Permanently Allocated
LGC	4 – Permanently Allocated
LGC-D	8 – Permanently Allocated
LSC	4 – Permanently Allocated
LSC-D	8 – Permanently Allocated
OPC	Determined by Card Configuration—Dynamically Allocated (see the chart on the following page)

4.22 The number of voice channels the OPC uses depends on how the card is configured. The following chart lists the OPC features and the maximum number of voice channels assigned for each. (All voice channels are dynamically allocated.)

Table 2-2. *Maximum Voice Channels for the OPC*

Feature	Max. Voice Channels
16 Conference Circuits	16
8 Caller ID [CLIP] Receivers	8
6 Caller ID [CLIP] Transmitters	6
6 DTMF Receivers	6
12 Conference and 4 DTMF	16
4 Enhanced Speakerphones	8
8 Standard Speakerphones	16
Voice Mail Connection (v5.3 and earlier)	24
Voice Mail Connection (v6.0 and later)	32

4.23 With the exception of shared speakerphone devices, each DSP resource (e.g., conference circuit, DTMF receiver, etc.) requires one time slot per resource. Each shared speakerphone device (for Axxess Basic Digital, Eclipse Basic Digital, Standard Display, Associate Display, Model 8560, Model 8500, and Model 8520 Phones) requires two time slots.

NOTE: The Eclipse Basic Digital and Associate Display Phones are supported by U.S. software only.

DTMF Receiver Allocations

4.24 It is possible to have more call traffic than the standard system DTMF receivers (two on a PCM card; six on a PCMA card) can process efficiently. If DTMF receiver circuits are often busy (e.g., single-line set users are experiencing a delay in receiving dial tone), it may be necessary to install an Options Card and equip it with optional DSPs programmed as DTMF receivers. (See page 2-67 for more information on optional DSPs.) You can also install a DKSC-16+, which has four available DTMF receivers.

NOTE: DTMF receivers are used when single-line sets are dialing and when features such as E&M (for the U.S. only), DISA, automated attendant, CO call continuation, etc. are used. They are in use only while the dialed number is being processed, not for the duration of the call.

External Paging Ports

4.25 As many external paging ports as desired can be used for connecting external paging speaker equipment. Each external paging port can be connected to an available LGC or LSC circuit or to the speaker wires on a phone. In addition, the CPU card and CPC each has its own dedicated external paging port (see page 2-30 for more information).

NOTE: When connecting external paging equipment to an LGC or LSC circuit, use a paging amplifier that provides talk battery. When connecting external paging equipment to a phone or to the CPU card or CPC, use a paging amplifier that *DOES NOT* provide talk battery.

CAUTION

Using a paging amplifier with talk battery on a phone may damage the phone or the Digital Keyset Card.

Music Source

4.26 If desired, external music sources may be connected to the system for background music and music-on-hold. The music jacks on the CPU card, CPC, and OPC are the inputs for the optional external music sources (radio, tape player, etc.). The music source connected to the primary CPU or CPC can be used for music-on-hold and/or background music. However, the music sources connected to the secondary CPU and Options Cards can only be used for music-on-hold.

4.27 The music-on-hold input circuitry on the CPUs, CPCs, and OPCs are high-impedance inputs (8.7K ohm). This allows a connection to any audio device with an output impedance of 4 - 8.7 ohms. In addition, the music-on-hold inputs are rated at 1 volt RMS. If the input power exceeds this rating, the audio may be distorted. Because of this, Inter-Tel recommends limiting the input power to 1 Watt.

4.28 In an ATM-based system, the CPU board in equipment chassis 1 is the default source for music-on-hold. This CPU is also the only available source for background music.

4.29 The system software supports multiple music-on-hold sources, using the “Music 2” jacks on the Options Cards and the music jacks on the CPUs or CPCs.

NOTE: In v6.0, the resource manager was changed drastically to support ATM systems. As an indirect result, when OPC cards use the MOH voice channel, the OPC circuit busy light stays on until the system resets. If the OPC card is reset, the light remains on.

Card Slot Arrangement

4.30 The circuit card slot on the far right (8th slot) of the basic equipment chassis is reserved for the CPU card or CPC. Other than that, any station, trunk, or optional resource card may be placed in any of the remaining seven slots. If a second chassis is attached, the CPU card or CPC in the first (master) chassis also controls the second chassis (e.g., slot number 16 remains empty).

4.31 For a non-ATM system, if a third chassis is attached, it must be equipped with a "slave-version" CPU020-X/PCM-F card in slot number 24. If a fourth chassis is attached, the CPU020-X/PCM-F card in the third chassis also controls the fourth chassis (e.g., slot number 32 remains empty). As of version 8.0, tri- and quad-chassis systems require the CP Server; so, the CPU in slot number 8 must have an EAC so that it can connect to a CP Server.

NOTE: When using the CP Server, the "slave" CPU is used just as it would be with the other CPU cards.

4.32 ATM-based systems do not have a master-slave relationship, which means that any chassis pair can go offline without affecting the rest of the system. There are several advantages to this:

- Should a CPU or other piece of hardware fail, the entire system does not fail, just that equipment chassis.
- Hardware replacements can be handled on a per-chassis basis without having to power off the entire system.
- The system can be designed to spread critical business operations over several chassis to protect the system should any chassis go offline, whether scheduled or not. After the chassis comes back online, it powers up and begins functioning with the rest of the system.

4.33 Because the ATM system chassis are no longer dependent on each other, each chassis pair is identified by the PCMA number assigned when configuring the chassis (see page 3-14). For example, if a dual-chassis system houses PCMA #1, that system is referred to as PCMA Group 1.

NOTE: Any equipment chassis in an ATM system should be able to go offline and back online without affecting the other chassis in the system. Any calls associated with or using the resources of a chassis that goes offline, however, are dropped.

4.34 Information on the type of card installed in each card slot must be entered through Database (DB) Programming.

C. ENVIRONMENTAL REQUIREMENTS

4.35 The equipment chassis and the station instruments require the following environmental conditions:

Table 2-3. *System Environmental Requirements*

Requirements	In Operation	In Storage
Temperature (Chassis) Temperature (Station Instruments) Relative Humidity (Non-Condensing) Altitude	32° to 104 F (0° to 40° C) 32° to 113 F (0° to 45° C) 5% to 95% Up to 10,000 ft (3,048 m)	-40° to 185° F (-40° to 85° C) -40° to 185° F (-40° to 85° C) 5% to 95% Up to 40,000 ft (12,192 m)

NOTE: Do not exceed the maximum operating temperature (as listed above). Therefore, when installing the chassis and station instruments, allow a sufficient margin for error in case of air conditioning failure, routine maintenance, plant shutdown, etc. As a general guideline, if the conditions are suitable for office personnel, they are also suitable for all equipment and station instrument operation. A properly controlled environment helps to extend the operating life of the equipment.

D. SYSTEM POWER SUPPLY UNIT

4.36 The power supply unit provides power to all of the circuit cards and stations. It *must* have an isolated, dedicated, AC outlet of 100-125 VAC, 50-63 Hz, single-phase commercial power source with the appropriate amperage circuit (see the Power Calculator table on page 2-23). For details, see the first NOTE on page 3-10 in INSTALLATION. If an expanded chassis is installed, all of the power supplies should be plugged into a single power strip that is connected to the isolated, dedicated outlet.

Power Supply – 4 Amp

4.37 The 4-amp system power supply unit – part no. 550.0121 [PS2747 in Europe] is installed in the far left slot of the equipment chassis.

4.38 Located on the front side of the power supply unit are the AC power cable connector, the ON/OFF power switch, three LED voltage indicators, and four voltage test points. During normal system operation, all three LEDs (+5V, -5V, and +24V) should be lit. If any of the LEDs are blinking or are unlit, the power supply is defective and the entire unit must be replaced.

4.39 Each power supply unit has one, 250V, 5A fuse. (This fuse may *not* be replaced in the field. If the fuse is blown, return the entire power supply unit for repair.) Each unit's maximum power outputs are +24VDC, 4A; +5VDC, 6A; and -5VDC, 0.5A.

4.40 Up to two, 16-Port Digital Keyset Cards (DKSC-16s) can be installed in chassis equipped with this 4-amp power supply, depending on the number and variety of other circuit cards to be installed. If necessary, use the following chart to determine how many cards can be installed, while staying within the 4-amp limit.

Table 2-4. *Power AMPs Requirements*

Card Type	Amps Required	Card Type	Amps Required
DKSC-16/DKSC-16+	1.28	BRU	.33
AKSC	1.28	BRS	.19
EKSC	1.28	LGC-D	.30
SLC	.75	LSC-D	.30

Table 2-4. *Power AMPs Requirements (Continued)*

Card Type	Amps Required	Card Type	Amps Required
SLC-16 or SLC-16+	.40	OPC	.30
DKSC	0.64	LGC	.15
T1C	0.40	LSC	.15
T1/PRI or E1/PRI	0.45	IPC or IPRC	.34
T1/E1/PRI	0.20	IPRC-D	.48

4.41 The 4-amp power supply can support one EVMC, which draws power from the 5VDC output.

Power Supply Unit – 6 Amp

NOTE: European systems do not support the 6-amp power supply unit.

4.42 The 6 amp system power supply unit (part no. 550.0104) is installed in the far left slot in the new-style chassis. (See page 2-15 for information on the new chassis design.)

4.43 Located on the front side of the power supply unit are the AC power cable connector, the ON/OFF power switch, three LED voltage indicators, and four voltage test points. During normal system operation, all three LEDs (+5V, -5V, and +24V) should be lit. If any of the LEDs are blinking or are unlit, the power supply is defective and the entire unit must be replaced.

NOTE: With the new-style equipment chassis, the grounding lug is no longer located on the front of the power supply unit. It is located on the left, inside wall of the chassis.

4.44 Each power supply unit has one, 250V, 4 amp fuse. (This fuse may *not* be replaced in the field. If the fuse is blown, return the entire power supply unit for repair.) Each unit's maximum power outputs are +24VDC, 6A; +5VDC, 3A; and -5VDC, 1.5A.

4.45 With some limitation, the 16-Port Digital Keyset Cards (DKSC-16s) can be installed in chassis equipped with this 6 amp power supply. Depending on the number and variety of other circuit cards to be installed, up to two of the cards can be DKSC-16s (or up to four in dual-chassis systems, and so on). If necessary, use the chart above to determine how many DKSC-16s can be installed using the 6 amp power supply.

NOTE: The 6 amp power supply cannot be used with an EVMC because this power supply does not provide enough amperage on the +5 VDC buss to power the card. The EVMC requires 2.5 to 3 Amps from the 5 volt buss in order to run. To use an EVMC voice mail card the system must have either a 4 amp or 9 amp power supply. Note that the 4 amp power supply is only for the Axxess or Eclipse^2 64 systems.

Power Supply Unit – 9 Amp

NOTE: The first (master) and third (slave) chassis in all tri-chassis and quad-chassis installations must be equipped with 9 amp power supplies. The PCM-F card requires a 9 amp power supply.

4.46 The 9-amp system power supply unit – part no. 550.0131 [PS2735 in Europe] is installed in the far left slot in the new-style equipment chassis. (See page 2-15 for information on the new chassis design.)

NOTE: The part number for the 9 amp power supply unit was changed from 550.0110 to 550.0131. There is no functional difference between the old power supply unit and the new unit except the inrush current on the 550.0131 is higher then the 550.0110; however, both power supplies have the same maximum input amperage.

4.47 Located on the front side of the power supply unit are the AC power cable connector, the ON/OFF power switch, three LED voltage indicators, and four voltage test points. During normal system operation, all three LEDs (+5V, -5V, and +24V) should be lit. If any of the LEDs are blinking or are unlit, the power supply is defective and the entire unit must be replaced.

NOTE: With the new-style equipment chassis, the grounding lug is no longer located on the front of the power supply unit. It is located on the left, inside wall of the chassis.

4.48 Each power supply unit has one, 250V, 8 amp fuse. (This fuse may *not* be replaced in the field. If the fuse is blown, return the entire power supply unit for repair.)

Each unit's maximum power outputs are:

+24VDC, 9A +5VDC, 7A
-5VDC, 1.5A

4.49 With the 9 amp power supply unit, there is no power limit as to how many DKSC-16s can be installed in the equipment chassis.

Power Calculator

4.50 The following table can be used to calculate power requirements. This chart is for the input wattages for calculating UPS backup time.

Table 2-5. *Power Calculator*

Power Supply Part Number	550.0121	550.0103	550.0110	550.0131
Input Wattage	172	225	345	345
Inrush Current (Surge Current)	1-1.5Amps	1.5 - 2 Amps	1.5 - 2 Amps	2.5 - 3 Amps
Input Amperage	110VAC/3.8Amp 240VAC/1.8Amp	100-115VAC/5Amp 230-240VAC/2.5Amp	100-115VAC/6Amp 230-/3Amp	100-115VAC/6Amp 230-240VAC/3Amp

Voltage Surge and Spike Protection

4.51 To reduce the effects of AC voltage surges and spikes that may cause system malfunctions, false logic, and/or damage to the electronic components, a surge/spike protector with the following specifications is recommended:

- Clamp voltage transients at 300VDC within five nanoseconds when exposed to waveforms as described in the ANSI/IEEE Standard C62.41-1980 (IEEE 587).
- Reduces RFI/EMI noise by at least 20 dB at frequencies between 5 kHz and 30MHz.
- Ground lug for grounding Single-Line Adapters, trunk gas discharge tubes, etc. to the chassis AC outlet.

4.52 Inter-Tel's CommSource™ division or Inter-Tel Europe sells a surge/spike protector (part number 900.0500) that meets all of these specifications.

Battery Back-Up

4.53 To provide back-up power in the event of a power failure or brownout condition, the power supply can have optional battery back-up using a customer-provided uninterruptable power supply (UPS) unit or a standby power supply (SPS) unit. See page 2-104 for more information.

NOTE: The CP Server PC should be connected to the same type of surge protector and battery backup unit as the chassis. System operation is dependent on both the PC and the chassis being operational.

E. CALL PROCESSING UNIT (CPU) CARD

4.54 The CPU card contains the main controlling microprocessor and associated control logic and memory circuits. It is under the control of a program stored in flash memory, which is activated when the system is powered up. The following information is described in this section:

- CPU128 Card (see below)
- CPU64 Card (see page 2-26)
- CPU020/PCM Card (see page 2-26)
- CPU Design Features (see page 2-27)
- CPU (see page 2-32)
- CP Server (see page 2-35)

NOTE: A CPC is required to run version 8.0 or later on a 64, 128, or 256 system. See page 2-32 for details.

CPU128 Card

4.55 The CPU/MEM card (a.k.a. CPU 112) has been replaced by the CPU128 card – part no. 550.2010 [PB2590 in Europe]. Instead of the pre-programmed ROMs used on the CPU/MEM card, the CPU128 card employs flash memory circuitry. This technology allows you to install and upgrade system software through the programming PC, rather than having to replace the ROMs on the card.

NOTE: CPU/MEM cards are not available for use on the Inter-Tel system for versions 5.0 or later.

4.56 The CPU128 card can be used only in single-chassis or dual-chassis systems.

4.57 If any T1 or T1/E1/PRI Cards are to be installed in an equipment chassis using the CPU128 card configuration, a single T1 or T1/E1/PRI Network Clock Card (NCC) must also be installed. Then, instead of using the clock circuitry on the CPU card, the NCC becomes the clock for the entire system. The NCC provides the smooth, error-free clock transitions required for T1 or T1/E1/PRI applications. (See page 2-55.) Inter-Tel recommends using only one T1/PRI or E1/PRI card with the CPU128.

NOTE: When the CPU128 card is used with a T1 or T1/E1/PRI Card in a high call traffic scenario (e.g., multiple ACD groups), along with OAI in use (e.g., Call Center type of application), it may be necessary to upgrade to a larger CPU256 card. The CPU256 enhances system performance.

4.58 **CPU128 with Memory Daughter Card**: The CPU128 card comes equipped with a memory daughter card (part no. 550.2017) to support increased system features. A daughter card adds extra RAM to the existing CPU128 card. Additional capacity enables all available features on a system to function properly. This card replaces an existing CPU128 (part no. 550.2015).

CPU64 Card

NOTE: Versions 8.0 and later do not support the CPU64 card. To install a CPU64 configuration, a Call Processing Card (CPC) is required.

4.59 The CPU64 card is identical to the CPU128 card, except that it only supports up to 64 devices and it can be used only in a single-chassis system. Like the CPU128, the CPU64 card uses flash memory circuitry.

4.60 The CPU64 can be used with any of the voice mail platforms (EVMC or Windows-based Voice Processing Unit). The CPU64 can be used with the 4-amp, 6-amp, or 9-amp power supplies. (However, the EVMC cannot be used with the 6-amp power supply.)

4.61 If any T1 or T1/E1/PRI cards are to be installed in an equipment chassis using the CPU64 card configuration, a single T1, E1/PRI, or T1/E1/PRI Network Clock Card (NCC) must also be installed. Then, instead of using the clock circuitry on the CPU card, the NCC becomes the clock for the entire system. The NCC provides the smooth, error-free clock transitions required for T1 or E1 applications. You can use only one PRI-equipped T1 or E1 card with the CPU64.

4.62 The CPU64 card can be used in a node in a phone system network. However, the node should not be a hub and its voice mail system should not support any other nodes.

4.63 For CPU64 information, see Appendix D — CPC64/CPU64 Configuration. For installation instructions, see page 3-95.

CPU020/PCM Card

NOTE: Versions 8.0 and later require a CP Server for 512 or higher systems.

4.64 There are six types of CPU020/PCM combinations available:

- CPU020-D/PCM-D Card (a.k.a. CPU 256)
- CPU020-Q/PCM-F Card (a.k.a. CPU 512 Master)
- CPU020-X/PCM-F Card (a.k.a. CPU 512 Slave)
- CPU020-A/PCMA-C Card (a.k.a. ATM Copper)
- CPU020-A/PCMA-F Card (a.k.a. ATM Multi-Mode Fiber)
- CPU020-A/PCMA-FS Card (a.k.a. ATM Single-Mode Fiber)

4.65 “D” is short for dual, “Q” is short for quad, “X” is used for slave, “F” is short for fiber, and “A” is short for ATM. The four CPU020 cards (-D, -Q, -X, and -A), the two PCM cards (-D and -F), and the three PCMA cards (-C, -F, and -FS) listed above are purchased separately. In addition, each CPU020 card must be equipped with the proper boot ROM and memory map PAL software kit. See REPLACEMENT PARTS for the appropriate part numbers.

NOTE: The memory map PAL is not the same as the feature-unit PAL. As of version 8.0, the feature-unit PAL is no longer required.

4.66 For a non-ATM system, the CPU020-D/PCM-D card can be used only in single-chassis or dual-chassis systems. In tri-chassis or quad-chassis systems, the CPU020-Q/PCM-F card must be used in the first (master) chassis and the CPU020-X/PCM-F card must be used in the third (slave) chassis, as shown in the following chart (see also Figure 2-1 on page 2-11):

Table 2-6. *CPU020-D/PCM-D Configuration*

System Configuration	CPU020-	PCM-
Single-/Dual-Chassis	D (or Q)	D (or F)
Tri-/Quad-Chassis – Master	Q	F
Tri-/Quad-Chassis – Slave	X	F

4.67 CPU020/PCM card combination kits are available. Each kit includes a CPU020 card, a PCM card, the proper boot ROM and memory map PAL, and any additional memory components required. The combined cards are shipped pre-assembled, with the current production version system software already loaded.

4.68 Inter-Tel recommends using the CPU020/PCM card under any of the following conditions:

- Systems using more than 128 ports
- Systems with unusually high call volume or extensive use of multiple ring-ins and/or all-rings (e.g., more than 100 ringing stations per minute)
- Systems with more than one T1 or E1/PRI Card

NOTE: Inter-Tel recommends that fully-loaded phone systems, with a large number of trunks and heavy call traffic, have extended memory added on the CPU020/PCM. Without added memory on the CPU020/PCM, fully-loaded systems may periodically reset due to memory shortages. The reset is the result of the CPU020/PCM attempting to recover memory. All new CPUs purchased with versions 5.2 and later software come equipped with the extended memory installed.

4.69 For an ATM-based system, the CPU020-A/PCMA card must contain the appropriate connector type. For example, if using multi-mode fiber (MMF), you must use the CPU020-A/PCMA-F card. You may require more than one type of PCMA card, depending on the ATM Switch configuration (see page 2-9). As with the CPU020/PCMs, card combination kits are available. Each kit includes a CPU020-A card, a PCMA card, the proper boot ROM and memory map PAL, and any additional memory components required.

CPU Design Features

4.70 The CPU cards contain the following system resources (see Figures 3-48 to 3-54 on pages 3-98 to 3-109):

- **DB15 PCM Connector**: (15-pin D-subminiature female connector) Connects the first eight ports to the Audio Interface Card (AIC) in the Voice Processing PC, or connects to the DB15 PCM connector on the Embedded Voice Mail Card (EVMC).

NOTE: In tri-/quad-chassis installations, the DB15 PCM connector on the slave PCM-F card cannot be used to provide additional AIC ports. Use an Options Card instead.

- **RS232 Jacks**: (three 8-pin, 8-conductor modular jacks/line jack units on the CPU020-D/-Q, one on the CPU020-X) Connects to the RS232 port on the Voice Processing PC or EVMC, or connects SMDR/error message output devices, a personal computer for programming the system database, and/or a modem for remote programming/maintenance (see pages 2-99 to 2-104 for requirements).

NOTE: The system can have a maximum of 32 RS232 ports for system services. This includes the RS232 ports on the CPU card, Options Card(s), and PC Data Port Modules. If an Inter-Tel Voice Processing PC or EVMC is installed, at least one of the system RS232 ports must be dedicated to Voice Processing.

- **RJ45 Jack (on PCMA-C only)**: Connects the CAT5 cable to the appropriate connectors on the ATM Switch. The maximum length for this connection is 100 meters (.06 miles).
- **LED Indicators**: Shows main processor operation (CPU ACTIVE - green), database in default state due to error (DATABASE ERROR - red), minor system alarms (MINOR ALARM - yellow), external clock activity (EXTERNAL CLOCK ACTIVE - green), and database back-up battery connection (BATTERY OFF - red). The PCM-F card has two additional LEDs that show local fiber-optic status (LOCAL SYNC - green) and remote fiber-optic status (REMOTE SYNC - green). The PCMA cards include an ATM LED that indicates when there is no active connection to the ATM Switch (ATM LINK BAD - red). See Figure 7-2 for detailed LED information.
- **Fiber-Optic Jacks (on PCM-F, PCMA-F, and PCMA-FS only)**: Installed as one transmit and one receive on each PCM-F card for connecting the CPU020-Q/PCM-F card in the first (master) chassis to the CPU020-X/PCM-F card in the third (slave) chassis. The fiber-optic jacks on the PCMA-F and PCMA-FS cards are used to connect to the OC-3 connectors on the ATM Switch. See the table below for loop limits without repeaters.

Table 2-7. *Fiber-Optic Jack Loop Limits*

Desc.	Connector	Connects To	Medium	Maximum Length
512CPU	ST	Slave	Multimode Fiber (62.5/125 outer/inner core in microns)	1 km (.6 miles)
512CPU/ EAC	ST	CPS	Multimode Fiber (62.5/125 outer/inner core in microns)	1 km (.6 miles)
PCMA-F	SC	ATM Switch	Multimode Fiber (62.5/125 outer/inner core in microns)	2 km (1.2 miles)
PCMA-FS	SC	ATM Switch	1320 nm Single Mode Fiber	> 2 km (1.2 miles) but less than 15 km (9.3 miles)

For more information about the 512 CPUs, see the following chart.

Table 2-8. *512 CPU*

Fiber Diameter	62.5/125 (outer/inner core in microns)
Mode	Multi-Mode
Wavelength	820 nm
Maximum Attenuation	5 db/km
Terminating Connector	ST
Pairs	1
Maximum Length	1 km (.6 miles)
B-L Product	20 MHz-km

NOTE: A ten-foot (3-meter) cable is available using part no. 813.1628 [C118 in Europe]. Or, if necessary, contact Inter-Tel's CommSource™ division or Inter-Tel Europe for information on purchasing longer cable lengths. Custom cables can also be built using industry-standard ST-type connectors.

- **Reset Switch**: (one in single-/dual-chassis systems, two in tri-/quad-chassis systems, four in ATM systems with eight chassis) Manually resets the system during trouble-shooting. This is a software-controlled system reconfiguration. Pushing this switch does the following:
  - — Preserves the battery-backed database information, all calls in progress, outside calls being dialed, pages in progress, camped-on calls, queue requests, system and user programming, inter-station messages, calls on system and individual hold, do-not-disturb messages, and reminder messages.
  - — Restores DTMF receivers, modem access, and speech channels, and resets the station clocks to match the system clock. Updates trunk button, DSS/BLF button, speed-dial button, and feature button lamp status.
  - — Disconnects calls ringing in and all ISDN calls in progress.

NOTE: If necessary, the reset switch can also be used to place the CPU card into "receive software upload" mode, as described on page 3-170.

- **Database Back-Up Battery**: (3.6V, ½AA lithium battery) Supports the database for at least two months of accumulated system down time.
- **Battery Voltage Test Points And Check Button**: Checks the database back-up battery charge. If the battery voltage is not greater than 3.3VDC, replace the battery.
- **Battery Back-up Jumper Strap**: Activates the database back-up battery. The jumper strap should be placed in the ON position (over the bottom two pins) before the CPU card is installed. When the card is taken out of service for repair or storage, the jumper strap should be placed in the OFF position (top two pins) to preserve the battery charge.

NOTE: Placing the BATTERY jumper strap in the OFF position erases the database when the card is unplugged or loses power.

- **Music Jack**: Connects an external music source (such as a radio, tape player, etc.) that can be heard by callers when they are placed on hold or camped on. It may also be used for phone background music. This is an 1/8-inch (3-mm), two-conductor (mono), mini-phone jack. For installation instructions, see page 3-237. In DB Programming, music can be enabled or disabled, and each trunk group (as well as transfer audio) can be assigned music, silence, or tick-tones. See page 6-234 in PROGRAMMING.

NOTE: In tri-/quad-chassis installations, the music jack on the slave PCM-F card can also be used to provide an additional music-on-hold music source (but it cannot be used for background music).

- **Relay Jack**: (6-pin, 6-conductor modular jack or line jack unit) Currently used for power failure transfer equipment only. A power failure transfer jack can be used for connecting one or more single-line sets to predetermined CO [local exchange in Europe] trunks in the event of processor card failure or a power failure. The output is one normally-open (deactivated) double-pole, double-throw (DPDT) relay (actually wired as SPDT) that is held closed (activated) during system operation. Additional equipment needed includes a customer-provided PFT relay card and card mounting case. The relay contact ratings are listed below. (See page 2-103 for additional requirements.)

CPU Card Relay Contact Ratings

0.6A at 125VAC
0.6A at 110VDC
2.0A at 30VDC

- **Real-Time Clock**: Provides more accurate date and time function.
- **External Paging Jack**: (RCA-type phono jack with an output impedance of 600 ohms – one in single-/dual-chassis systems, two in tri-/quad-chassis systems, four in eight-chassis ATM systems) Connects external paging speaker equipment to allow paging to large areas, such as warehouses or loading docks. For installation instructions, see page 3-239.

NOTE: Some older CPU128 cards do not have a paging port.

- **Network Clock Circuitry**: **(CPU020/PCM or CPU020/PCMA Only)** Provides the smooth, error-free clock transitions required for T1 or E1/PRI applications. If any T1 or E1/PRI cards are installed in the chassis, the network clock circuitry provides the clock function for the entire system.

CAUTION

Do not use a Network Clock Card (NCC) with the CPU020/PCM (including EAC CPUs) or CPU020/PCMA cards. Using an NCC with these cards causes erratic problems which may disable the system. The NCC is used for the CPU64 and CPU128 only.

NOTE: The following four system resources are provided by the single Digital Signal Processor (DSP) on the CPU card. For additional and/or other system resources, one or more Options Cards (with up to four DSPs each) may be installed in the chassis. Tri-chassis and quad-chassis systems have double the resources listed below.

- **Conferencing Resources (CPU020/PCM Only)**: Connect up to 12 parties in multiple conferences. There can be up to three simultaneous four-party conferences or up to four simultaneous three-party conferences on a PCM card (non-ATM). Conferencing is not amplified.
- **Conferencing Resources (CPU020/PCMA Only)**: Connect up to 12 parties in multiple conferences. Due to the request/response nature of the ATM signaling, all conferences that use resources on the PCMA card are treated as if they are four-party conferences (e.g., a three-party conference call reserves all four conference circuits, whether used or not). Because there are 12 conference circuits available, a maximum of three simultaneous conferences can be created on a PCMA card. For normal conferences, the conference circuits on the other PCMA boards on the system can be shared.
- **DTMF Receiver Circuits**: Provides circuits (two for non-ATM cards; six for ATM cards) that are used when single-line sets connected to MDPMs are dialing and when features such as DISA and analog voice mail are used. These receivers convert DTMF signals to the digital codes used by the system. They are in use only while the dialed number is being processed, not for the duration of the call. If all DTMF receivers are busy, the user may try again (DISA users are redirected to the primary attendant).
- **DTMF Transmitter Circuits** (two for non-ATM cards; four for ATM cards): Generates DTMF tones whenever needed by the system. For example, when an Axxess Basic Digital, Eclipse Basic Digital, Standard Display, Associate Display, Model 8500, Model 8520, or Model 8560 Phone user manually dials a number while on a call or when a station user uses one of the system auto-dial features (ARS, call forward to the public network, station or system speed dialing, etc.). Executive Display and Professional Display Phones can generate their own DTMF tones.

NOTE: The Eclipse Basic Digital, Associate Display, and Professional Display Phones are supported by U.S. software only.

- **Tone Generating Circuit**: Produces all of the system tones (busy, reorder, do-not-disturb, etc.) heard by phone, single-line set, DISA, and automated attendant users.
- **Caller ID [CLIP] Receivers (CPU020/PCMA Only)**: Provides four resources for receiving Caller ID [CLIP]. These resources are in addition to those available on the OPC (see page 2-67).

Call Processing Card (CPC)

4.71 Version 8.0 and later software support a new CPU card called the Call Processing Card (CPC). This card, which replaces the CPU64, CPU128, and CPU256 cards, uses a software license that allows you to upgrade the capacity of the system without having to replace the card (see page 5-11 for details about the software license feature). Because this software license contains premium feature information, the CPC or CPU020 cards do not require a feature unit PAL. In addition, the CPC does not require an attached PCM card because the CPC houses the typical PCM functionality and hardware. See Figure 3-47 on page 3-94 for an illustration.

NOTE: To use version 8.0 or later software on 512 and higher systems, a Call Processing (CP) Server is required. To use version 8.0 or later software on 64, 128, or 256 systems, the CPC is required.

4.72 Features of the CPC include the following:

- **Memory**: Includes 16 MB Flash, 16 MB SDRAM, and 2 MB battery-backed RAM.
- **IP Port (10/100 Base-T)**: Provides IP connections directly to the system, which is similar to the functionality of the IP port found on the CP Server. In addition, you can use the IP Port to upload Call Processing software, upload the software license, or program the database. See page 6-178 for more information.

CAUTION

Security is the responsibility of the customer's network administrator. Inter-Tel is not responsible for network problems due to security violations involving the IP address exposure of the CPC. This includes, but is not limited to, interrupted network service. Recommendations remain to place the CPC in a private IP network. In addition, Inter-Tel recommends that you install the CPC behind a firewall or Network Address Translation (NAT) box.

- *** Button**: Allows you to bypass the backup database, verify the status of Call Processing, and place the CPC into the download state. See page 7-88 for details.

NOTE: The * button is **not** a reset button.

- **RS232 Jacks**: (two 8-pin, 8-conductor modular jacks/line jack units) Connects to the RS232 port on the Voice Processing PC or EVMC, or connects SMDR/error message output devices, a personal computer for programming the system database, and/or a modem for remote programming/maintenance (see pages 2-99 to 2-104 for requirements).

NOTE: The system can have a maximum of 32 RS232 ports for system services. This includes the RS232 ports on the CPC, Options Card(s), and PC Data Port Modules. If an Inter-Tel Voice Processing PC or EVMC is installed, at least one of the system RS232 ports must be dedicated to Voice Processing.

- **Relay**: (6-pin, 6-conductor modular jack or line jack unit) Currently used for power failure transfer equipment only. A power failure transfer jack can be used for connecting one or more single-line sets to predetermined CO [local exchange] trunks in the event of processor card failure or a power failure. The output is one normally-open (deactivated) double-pole, double-throw (DPDT) relay (actually wired as SPDT) that is held closed (activated) during system operation. Additional equipment needed includes a customer-provided PFT relay card and card mounting case. The relay contact ratings are listed below. (See page 2-103 for additional requirements.)

CPC Relay Contact Ratings

0.6A at 125VAC
0.6A at 110VDC
2.0A at 30VDC

- **Music-On-Hold Port**: Connects an external music source (such as a radio, tape player, etc.) that can be heard by callers when they are placed on hold or camped on. It may also be used for phone background music. This is an 1/8-inch (3-mm), two-conductor (mono), mini-phone jack. For installation instructions, see page 3-237. In DB Programming, music can be enabled or disabled, and each trunk group (as well as transfer audio) can be assigned music, silence, or tick-tones. See page 6-234 in PROGRAMMING. This port supports A-law or mu-law companding, which is configured in DB Programming (System\Flags\Companding Type).
- **External Paging Jack**: (RCA-type phono jack with an output impedance of 600 ohms) Connects external paging speaker equipment to allow paging to large areas, such as warehouses or loading docks. For installation instructions, see page 3-239.
- **Built-in Network Clock Card (NCC)**: Provides the smooth, error-free clock transitions required for T1 or E1/PRI applications. If any T1 or E1/PRI Cards are installed in the equipment chassis, the network clock circuitry provides the clock function for the entire system.

CAUTION

Do not use a Network Clock Card (NCC) with the CPC. Using an NCC with this card causes erratic problems which may disable the system. If upgrading an existing CPU64 or CPU128 to a CPC, you must remove the NCC before installing the CPC. If the CPC detects an NCC, the system continually resets.

- **Software Security and Feature Program**: Supports a software license that enables functionality on the card. This feature serves two purposes: 1) the PAL is no longer required; and 2) feature and/or device upgrades are possible without replacing the card. For details about the software security and feature program, see page 7-95.
- **Modem**: Provides a built-in modem that is capable of transferring data at speeds up to 33600 bps. This modem is always equipped and is basically a system serial port that you must "call." Because the connection is established via a call, each CPC modem in a chassis is assigned an extension (1500 by default). For details about the modem, see page 6-180.
- **Rechargeable Battery**: Replaces the lithium battery that is available on CPU cards and is rechargeable. If the power to the CPC is lost, this battery stores the system database for up to a month (30 days). After power is restored, the battery recharges itself. This battery can be recharged more than 400 times before it needs to be replaced.

- **Flash Memory**: Stores a backup copy of the system database. The system automatically backs up this database once a day or whenever scheduled in DB Programming. The backup database is then restored if the same version of Call Processing software is reloaded or if the system detects a corrupt database on power up. See page 6-69 for details.

NOTE: The following four system resources are provided by the single Digital Signal Processor (DSP) on the CPC. For additional and/or other system resources, one or more Options Cards (with up to four DSPs each) may be installed in the equipment chassis.

- **Conferencing Resources**: Connect up to 12 parties in multiple conferences. There can be up to three simultaneous four-party conferences or up to four simultaneous three-party conferences on a PCM card (non-ATM). Conferencing is not amplified.
- **DTMF Receiver Circuits**: Provides circuits (two for non-ATM cards; six for ATM cards) that are used when single-line sets connected to MDPMs are dialing and when features such as DISA and analog voice mail are used. These receivers convert DTMF signals to the digital codes used by the system. They are in use only while the dialed number is being processed, not for the duration of the call. If all DTMF receivers are busy, the user may try again (DISA users are redirected to the primary attendant).
- **DTMF Transmitter Circuits**: Provides four circuits for generating DTMF tones whenever needed by the system. For example, when an Axxess Basic Digital, Eclispe Basic Digital, Standard Display, Associate Display, Model 8500, Model 8520, or Model 8560 Phone user manually dials a number while on a call or when a station user uses one of the system auto-dial features (ARS, call forward to the public network, station or system speed dialing, etc.). Executive Display and Professional Display Phones can generate their own DTMF tones.

NOTE: The Eclipse Basic Digital and Associate Display Phones are supported by U.S. software only.

- **Tone Generating Circuit**: Produces all of the system tones (busy, reorder, do-not-disturb, etc.) heard by phone, single-line set, DISA, and automated attendant users.
- **Caller ID [CLIP] Receivers**: Provides four resources for receiving Caller ID [CLIP]. These resources are in addition to those available on the OPC (see page 2-67).

4.73 The CPC supports Call Processing versions 8.0 and later only. In addition, if the CPC is installed on a networked system (i.e., one with multiple nodes), each node on the system must be running version 5.2F9 or later. The CPC also supports the U.S. and European system versions of software.

Call Processing (CP) Server

NOTE: As of version 8.0, a CP Server is required for 512 and higher systems.

4.74 The Call Processing (CP) Server is a PC-based CPU for the phone system. This PC runs one of the following operating systems:

- Windows NT
- Windows 2000
- Windows 2003 (see "Call Processing Server on Windows Server 2003: Windows Server® 2003 (Telco version) operating system (OS) with Service Pack 2 (SP2) is now supported for the Call Processing Server (CPS). Field upgrades must be done by certified technicians only." on page C-3 for details.)

4.75 For an ATM-based system, Windows 2000 or later is required. The call processing software has been moved from the chassis CPU to this CP Server which communicates with the system via the ATM Switch or via a 10Base-T Ethernet connection.

4.76 For a non-ATM system, the PC and equipment chassis are connected using a cross-connect Ethernet cable and use the UDP (User Datagram Protocol) to transfer data across a direct connection or a LAN connection. For an ATM-based system, the PC connects directly to the ATM Switch via a multi-mode fiber (MMF) or CAT5 cable.

4.77 A CP Server is recommended for the following types of installations:

- Any node using the ATM Switch (Windows 2000 is required)
- Any node with more than 355 display phones
- Any networked node with a Voice Processing Unit with more than 16 ports ***and*** more than 128 Executive Display, Professional Display, or Model 8560 Phones
- Any networked node with a Voice Processing Unit with Automated Attendant and/or Call Routing Applications that handle more than 1000 calls in an hour
- A node with centralized voice mail that supports more than four nodes ***or*** a total of 1500 off-node devices
- A node with all-ring or multiple-ring hunt groups with ten or more stations that handle more than 500 calls in an hour
- A node with all-ring hunt groups that contain off-node devices
- A node with hunt groups containing more than 100 members
- A node with a page zone containing more than 60 stations that receives more than 30 pages in an hour
- A node used as an outgoing call center with more than 100 agents
- A heavily-loaded node that uses any System OAI applications
- A node with more than five ISDN Primary Rate trunks for private networking or to the central office [local exchange in Europe]

4.78 The non-ATM CPU020 has an 80-pin connector that attaches to the Ethernet Adapter Card (EAC). In addition, the PCM faceplate has an extra cutout for the RJ45 connector on the EAC. The three components, CPU020, PCM, and EAC, are available in two configurations:

- CPU020/PCM-D with an EAC (U.S. part no. 550.9026)
- CPU020/PCM-F with an EAC (U.S. part no. 550.9025)

4.79 The EAC supports only 10Base-T and is connected to the PC via an industry-standard category 5 crossed patch cable (U.S. part no. 813.1724). The cable is 14 feet [4.2 meters] in length allowing some freedom of placement of the server. (See next page for a pinout of the cable.) Per IEEE 802.3 10Base-T specification, this cable can be up to 328 feet [100 meters] long.

4.80 A cable is provided with the CP Server. For CP Servers with an OC-3 ATM adapter card, an MMF cable is provided. For standard CP Servers or for those with a UTP5 ATM Adapter card, a 14-foot [4.2-meter] 10base-T CAT5 Cross-Connect cable is supplied with the CP Server PC.

4.81 The 10Base-T CAT5 Cross-Connect cable has an 8-pin/8-conductor modular connector on each end. However, if you need a longer cable, make sure the pinout is as follows:

Table 2-9. *Pin Assignments*

Pin	Pin
1	3
2	6
3	1
4	4
5	5
6	2
7	7
8	8

4.82 The memory map PALs for both the dual and quad configurations have been combined into a single memory map PAL (in socket U41). The U.S. part number for this PAL is 827.8967.

4.83 For more information about the CP Server, see Appendix C — CP Server.

F. STATION CIRCUIT CARDS

4.84 Station instruments are connected to station circuit cards in the equipment chassis. Digital phones are connected directly to Digital Keyset Cards (DKSCs) and/or 16-Port Digital Keyset Cards (DKSC-16s or DKSC-16+s). Analog phones are connected directly to Analog Keyset Cards (AKSCs). Executone IDS and Wave phones are connected directory to Executone Keyset Cards (EKSCs). Industry-standard single-line DTMF or pulse-tone sets and playback devices (AC-ringing instruments only) are connected directly to Single-Line Cards (SLCs) and/or 16-Port Single-Line Cards (SLC-16s or SLC-16+s) or indirectly to DKSCs/DKSC-16s through Single-Line Adapters (SLAs). Version 2.x or later software is required for pulse-tone sets.

NOTE: At least one keyset card should be installed to provide a display phone to an administrator who can receive system alarm messages, program system data, etc.

CAUTION

Do not install a Professional, Associate, or Eclipse Basic Phone on an Executone Keyset Card (EKSC) or Executone IDS devices on a Digital Keyset Card (DKSC). Installing a phone on the wrong type of station card can damage both the phone and the card.

4.85 The following information is described in this section:

- Station Circuit and Extension Numbers (see below)
- Digital Keyset Cards—DKSC/DKSC-16/DKSC-16+ (see page 2-38)
- Single-Line Adapter—SLA (see page 2-39)
- Analog Keyset Card—AKSC (see page 2-40)
- Executone Keyset Card—EKSC (see page 2-40)
- Single-Line Card—SLC (see page 2-41)
- 16-Port Single-Line Card—SLC-16/SLC-16+ (see page 2-42)

Station Circuit and Extension Numbers

4.86 Each station card circuit is assigned an eight-digit hardware address number (WW.XX.YY.ZZ) according to the node (WW), the card's slot in the equipment chassis (XX = 01-07 or 09-15), the location of the circuit on the card (YY = 01-08), and whether or not the circuit has more than one purpose (ZZ = 01-03). For example, circuit 01.01.02.01 is the second circuit on the first card installed in the chassis (Node 1). The first two numbers of the hardware address depict the node number.

4.87 In some circumstances, a station circuit can have more than one purpose. For example, two single-line devices can be installed on a single DKSC or DKSC-16 circuit. The first device is assigned circuit number WW.XX.YY.01, and the second device is assigned circuit number WW.XX.YY.02. Similarly, a phone's PC Data Port Module and Modem Data Port Module can be assigned their own circuit numbers. In this case, the phone is assigned circuit number WW.XX.YY.01, the first data port is assigned number WW.XX.YY.02, and the second data port is assigned number WW.XX.YY.03.

NOTE: In system software v9.0 and later, IP Devices (see page 6-147) display only the node and the board number (WW:XX) in the Address field. Earlier software versions displayed an address for IP endpoints that was similar to other station devices using the WW:XX:YY:ZZ convention explained above.

4.88 When the system is first powered up, only the CPC or CPU and one DKSC-16 (in card slot 1) are pre-configured. Changes to the default configuration can be made through DB Programming.

4.89 The default system assigns extension number 1000 to station circuit 01.01.01.01, extension number 1001 to station circuit 01.01.02.01, and so on to match the initial configuration. Circuit 01.01.01.01 is equipped as an administrator and as the primary attendant's phone when the system is in the default configuration. If desired, the circuits can be assigned 1-5 digit extension numbers through DB Programming.

NOTE: When 8-line IMX AIM phones are assigned to dual circuits, the "B" circuits do not have extension numbers until they are assigned in DB Programming.

Digital Keyset Cards (DKSC, DKSC-16, and DKSC-16+)

4.90 **DKSC**: Each DKSC provides up to eight circuits for connecting Executive Display Phones, Professional Display Phones, Standard Display Phones, Associate Display Phones, Axxess Basic Digital Phones, Eclipse Basic Digital Phones, Model 8500, 8520, or 8560 Phones, or Single-Line Adapters (SLAs) to the system. DKSCs have eight self-correcting thermistors (one for each circuit) to protect the system from damage due to a short circuit in the station cabling. If a short circuit causes a thermistor to open, it automatically closes when the short is removed.

4.91 **16-Port DKSCs (DKSC-16 and DKSC-16+)**: Each DKSC-16 and DKSC-16+ provides up to 16 circuits for connecting Executive Display Phones, Professional Display Phones, Standard Display Phones, Associate Display Phones, Axxess Basic Digital Phones, Eclipse Basic Digital Phones, Model 8500, 8520, or 8560 Phones, or Single-Line Adapters (SLAs) to the system. The boards have 16 self-correcting thermistors (one for each circuit) to protect the system from damage due to a short circuit in the station cabling. If a short circuit causes a thermistor to open, it automatically closes when the short is removed. Each DKSC-16 also has a single Digital Signal Processor (DSP) and each DKSC-16+ has two DSPs to provide the following card and/or system resources:

- **Shared speakerphones** (10 on a DKSC-16 and 16 on a DSKC-16+) for Axxess Basic, Eclipse Basic, Standard Display, Associate Display, Model 8500, Model 8520, and Model 8560 Phones. For information on shared speakerphones, see page 5-125.
- **Tone generating circuit** for providing system tones (busy, reorder, do-not-disturb, etc.) to the phones connected to the card, if necessary. It is available in case there are no voice channels available in the system to issue tones from the CPC or CPU when needed.
- **Switching matrix** to allow access to any available time slot on the card's PCM highway (bus). This is provided since the card may need up to 32 simultaneous voice channel connections.
- The DKSC-16+ also provides system resources for four **Caller ID [CLIP] receivers**, eight **Caller ID [CLIP] transmitters** (four if running v8.1 or 9.0.0 firmware), and four **DTMF receivers**. For information on Caller ID [CLIP] features, see page 5-72.

NOTE: The additional DKSC-16+ resources are available only on system versions 5.2 and later. On older version systems, the DKSC-16+ operates the same as a DKSC-16.

4.92 Up to 20 phones on the DKSCs (including the DKSC-16 and DKSC-16+) may be equipped with single DSS/BLF Units or up to five Executive Display Phones, Professional Display Phones, Standard Display Phones, Associate Display Phones, Model 8560 Phones, or Model 8520 Phones may be equipped with four units each (or any combination of the two up to 20 total units). Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones connect to a 60-key or 50-key DSS/BLF Unit. Each DSS/BLF Unit is connected to the optional PC Data Port Module (PCDPM) on the back of the corresponding phone. Unlike many other systems, the DSS/BLF Units do *not* require equipment chassis cabling separate from the phone's cabling; instead, they are powered by their own AC transformer.

NOTE: The Mini-DSS does not affect the number of DSS/BLF Units that you can attach to a phone. For example, if you have a Model 8560 with a Mini-DSS, you can still connect up to four DSS/BLF Units to the phone. The Mini-DSS can connect to a Model 8520 or Model 8560 Phone only.

4.93 Each card has a green LED (CARD ON-LINE) to indicate the card's functional readiness and a yellow LED (CIRCUIT BUSY) to indicate if any circuit on the card is busy (see Figure 3-55 on page 3-112). Also, each card has a metal EMI shield attached to the front edge of the card to help prevent electromagnetic interference.

NOTE: As of version 6.0, the CIRCUIT BUSY LED lights whenever a station residing on the DKSC-16 or DKSC-16+ is listening to background music.

4.94 **DKSC-16 or DKSC-16+ Extended Loop Length Modification**: For longer station loop lengths (up to 154 ohms/3000 ft), an optional customer-supplied, external power supply can be connected to the DKSC-16 termination block. You can use the Inter-Tel 30V power supply for 108 ohms/2100 feet. Or you can purchase the +36VDC External Power Supply Kit (kit part no. 928.3705).

4.95 This option is available only on DKSC-16s that are at revision level "2.0" or later or that have been modified and labeled with FSM100995. The external power supply is connected to pins 50 (+36V, V/SL) and 25 (GND, SL/V) on the termination block. In addition, the strap on jumper JP1 must be placed over pins 1 and 2 to enable the external power supply.

Single-Line Adapter (SLA)

4.96 An SLA, which can be installed on an available DKSC or DKSC-16 circuit, is used for connecting up to two AC-ringing single-line DTMF or pulse-tone sets, playback devices, and/or off-premises extensions. As many SLAs as desired can be installed on the system (up to the maximum number of available system ports – 256 for dual-chassis systems, 512 for quad-chassis systems). Version 2.x or later is required for pulse-tone sets.

4.97 *With a U.S. system*, an SLA can also be used for connecting up to two direct inward dialing (DID) trunks. (Or one single-line set and one DID trunk.) The SLAs support pulse dialing for both DID and single-line sets. See FEATURES, page 5-70, for more information on using DID trunks. SLAs are shipped with both ports set for single-line device operation. Before connecting a DID trunk, be sure to open the unit and set the corresponding port for DID operation. Also, if necessary, adjust the hybrid balance setting to better accommodate longer loop lengths. See page 3-22 for details.

4.98 AC-ringing off-premises stations can be connected directly to the SLA circuit (*external* HVRA Units, power supplies, and ring generators are not needed).

NOTE: SLAs do not support *visual* message waiting applications. If single-line sets with message waiting lamps are installed on SLAs, the lamps do not function. However, single-line sets can be configured in DB Programming to receive *audible* message waiting tones.

4.99 Each SLA, whether used for single-line sets or DID trunks or both, is powered by its own AC transformer. Also, SLAs have two self-correcting thermistors. If a short circuit causes a thermistor to open, it automatically closes when the short is removed.

NOTE: For optimal system performance, the Voice Processing PC and applications software should be used for voice mail and automated attendant functions, rather than using SLA circuits.

Analog Keyset Card (AKSC)

4.100 Each AKSC provides up to eight circuits for connecting analog phones to the system. (See page 2-4 for a list of analog phone models that can be installed on the Inter-Tel system.) If desired, the IMX 8-Line Dual-Circuit AIM Phones can be installed *two to a circuit* on AKSCs.

4.101 AKSCs have eight self-correcting thermistors (one for each circuit) to protect the system from damage due to a short circuit in the station cabling. If a short circuit causes a thermistor to open, it automatically closes when the short is removed.

NOTE: Analog DSS/BLF units and analog Attendant Computer Consoles cannot be used on the Inter-Tel phone system.

4.102 Each AKSC has a green LED (CARD ON-LINE) to indicate the card's functional readiness and a yellow LED (CIRCUIT BUSY) to indicate if any circuit on the card is busy (see Figure 3-56 on page 3-114). Also, each card has a metal EMI shield attached to the front edge of the card to help prevent electromagnetic interference.

Executone Keyset Card (EKSC)

NOTE: EKSCs are not supported in Europe.

4.103 Each EKSC provides up to 12 circuits for connecting Executone IDS equipment to the system. IDS equipment includes Model 160, 64, 32, 18, and 12 Phones, Agent Terminals, and 48-key DSS Consoles.

4.104 EKSCs have self-correcting thermistors on each circuit to protect the system from damage due to a short circuit in the station cabling. If a short circuit causes a thermistor to open, it automatically closes when the short is removed.

4.105 The EKSC also provides a secondary audio path supported on all 12 circuits.

4.106 The 48-key DSS Consoles supported by the EKSC require a separate circuit and their cabling is separate from the associated phone's cabling.

4.107 The EKSC is equipped with a single Digital Signal Processor (DSP) to provide system resources for allocating shared speakerphones and DTMF receivers.

NOTE: For information on Caller ID [CLIP] features, see page 5-72. For information on shared speakerphones for Eclipse Basic Digital and Associate Display Phones, see page 5-124.

4.108 The EKSC has a tone generating circuit for providing system tones (busy, reorder, do-not-disturb, etc.) to the phones connected to the card, if necessary. It is available in case there are no voice channels available in the system to issue tones from the CPC or CPU when needed.

4.109 Also, it has a switching matrix to allow access to any available time slot on the card's PCM highway (bus). This is provided since the card may need up to 24 simultaneous voice channel connections.

4.110 The green LED (CARD ON-LINE) on the EKSC indicates the card's functional readiness and a yellow LED (CIRCUIT BUSY) indicates whether any circuit on the card is busy (see Figure 3-56 on page 3-114). Also, each card has a metal EMI shield attached to the front edge of the card to help prevent electromagnetic interference.

4.111 EKSC Extended Loop Length Modification: For longer station loop lengths up to 103 ohms/2000 ft. (609 m.), if necessary, an optional external power supply with the following characteristics can be connected to the power supply connector on the EKSC:

- 36VDC
- 120mA per phone

Single-Line Card (SLC)

4.112 Each SLC provides up to eight circuits for connecting on-premises, AC-ringing, single-line DTMF or pulse-tone sets and playback devices. (Version 2.x or later is required for pulse-tone sets.)

4.113 The card has one or two onboard, square-wave, AC ring generators (depending on the card version). Each ring generator has a total ringer equivalence number (REN) of 8.0 (although no circuit on the card can have a REN above 3.0). The REN provided by the onboard ring generator(s) should be adequate for most installations. However, if a fax machine, modem, or single-line set does not respond to the ringing provided by the on-board ring generator(s) or, if the REN capability of the card is exceeded, an external (customer-provided) 20 Hz sine-wave ring generator should be used instead.

NOTE: Off-premises stations cannot be legally connected to SLCs; use SLAs instead. Also, for optimal system performance, the Voice Processing PC and applications software should be used for voice mail and automated attendant functions, rather than using SLC or SLA circuits.

4.114 Each SLC can be equipped with an optional Lamp Option Card (LOC) to provide visual message waiting capabilities for single-line sets equipped with message lamps. Single-line sets are then configured in DB Programming to receive *visual* and/or *audible* message waiting indication tones.

4.115 SLCs have self-correcting thermistors to protect the -48V output. If a short circuit causes a thermistor to open, it automatically closes when the short is removed.

4.116 Each SLC has a green LED (CARD ON-LINE) to indicate the card's functional readiness and a yellow LED (CIRCUIT BUSY) to indicate if any circuit on the card is busy (see Figure 3-57 on page 3-117). Also, each card has a metal EMI shield attached to the front edge of the card to help prevent electromagnetic interference.

16-Port Single-Line Card (SLC-16 or SLC-16+)

4.117 *With a U.S. system*, the SLC-16 can support up to 16 on-premises AC-ringing single-line devices. Like the 8-circuit SLC, it does not support off-premises stations and does not have DID capability. The SLC-16+ is identical to the SLC-16 and provides no additional functionality.

4.118 *With a European system*, the SLC-16 can support up to 16 on-premises AC-ringing single-line devices. Like the 8-circuit SLC, it does not support off-premises stations. The SLC-16+ is required for supporting CLIP to single-line sets.

4.119 An SLC-16 cannot be installed in a card slot that is programmed for an SLC. The ports must be specifically programmed for an SLC-16. However, if port programming is changed from SLC to SLC-16, the existing eight circuits retain their individual programming and the new circuits stay in the default state. If the port is changed from an SLC-16 to an SLC, the first eight circuits retain their programming, but the last eight are lost.

4.120 The SLC-16 uses 72VDC to generate trapezoidal ringing waveforms, which have the same effect as sine ringing waveforms. The card is capable of delivering a total ringer equivalence number (REN) of 16.0 at a frequency of 20 Hz (e.g., 1.0 REN per circuit). No single circuit should exceed a REN of 5.0. If the card requires more than 16.0 REN total, use a larger DC power supply (available from Inter-Tel). If a fax machine, modem, or single-line set does not respond to the ringing provided by the SLC-16, consult the manual for that device and/or contact Inter-Tel Technical Support.

4.121 A power supply is required for each SLC-16 because the cards are not designed to pull talk battery, ring, or message lamp power from the system power supply. Power supplies are available that support one card or up to four cards.

4.122 A 16-port Lamp Options Card (LOC-16) is also available to provide visual message waiting capabilities for single-line sets equipped with message lamps. The single-line sets are then configured in DB Programming to receive *visual* and/or *audible* message waiting indication tones.

NOTE: The SLC-16 and LOC-16 can be used only with version 4.3 or later system software.

G. INTERNET PROTOCOL CARD (IPC) AND INTERNET PROTOCOL RESOURCE CARDS (IPRC)

4.123 There are five different types of Internet Protocol (IP) cards available for use with the Inter-Tel system:

- **New Enhanced IP Resource Cards**: The Inter-Tel Axxess Internet Protocol Resource Card (IPRC) "enhanced" daughter card offers new technology for echo cancellation on all calls routed through the IPRC, as well as enhanced diagnostics for troubleshooting echo problems. When the enhanced daughter card is installed and programmed, the new technology on the daughter card replaces the echo cancellers on the IPRC, resulting in improved echo cancellation. There are two types of enhanced daughter cards available:
  - — **IPRC-DE** (Daughter Enhanced), part no. 550.2277: Provides new echo cancellation technology.
  - — **IPRC16-DE**, part no. 550.2276: Provides new echo cancellation technology with the DSP circuitry for up to 16 additional VoIP channels.

  For more information, refer to the *IPRC Enhanced Daughter Card Installation Instructions* (document part no. 835.3195) and the *IPC/IPRC and IP Endpoint Firmware Notice* (document part no. 835.2720-9).
- **Version 9.0.0 Firmware IPRC**: Supports both IP devices and private networking. When version 8.1.x or later IPRC firmware is used with a v9.0 or later Axxess system, the IPRC allows an Axxess system to share IP endpoint and IP networking capabilities on the same IPRC. This functionality allows for a cost reduction where one IPRC provides sufficient resources for both the IP endpoints and IP networking. Each IPRC supports up to 32 IP devices (16 IP devices without an IPRC daughter card), MGCP trunks, SIP trunks, multi-protocol devices, Model 8602 IP softphone, or IP networking ports. The new IPRC configuration replaces both the Private-Networking IPRC and Devices IPRC in previous releases. When upgraded to v9.0, the system prevents IPRCs running with firmware older than version 8.1 to come online.
- **Version 8.1 Firmware IPRC**: Supports up to 32 IP devices, MGCP trunks, multi-protocol devices, and IP networking ports (it does not support the IP SoftPhone). This card is also required if using peer-to-peer (P2P) audio. This card uses a software license and does not require a PAL. Unlike prior versions of IPRC firmware, the v8.1 IPRCs cannot be placed behind firewalls (from the endpoint perspective) or Network Address Translation (NAT) boxes. The new firmware also does not provide a BOOTP server for IP address assignment. The card requires a static IP address or the use of either a third-party BOOTP sever or a DHCP server. Inter-Tel, however, strongly recommends that you use a static IP address for the IPRC.

  NOTE: For devices, the software license regulates the number of multi-protocol endpoints only, such as the Model 8600, Model 8662, Model 8620, and Model 8690. It does not regulate the number of endpoints that use IP only, such as the Model 8660 and IP PhonePlus, or the number of MGCP gateways/endpoints.

- **Version 1.x Firmware Networking IPRC**: Supports 16 private networking ports using IP. This card requires a networking PAL. If desired, you can upgrade this card to a v8.1 firmware IPRC.

- **Version 1.x Firmware Device IP Resource Card (IPRC)**: Supports the same ports as the IPC in addition to 32 IP trunks via media gateway control protocol (MGCP). This card requires a PAL to determine the number of available ports. If desired, you can upgrade this card to a v8.1 firmware IPRC.
- **IP Card (IPC)**: Supports IP PhonePlus, IP SoftPhone, Model 8660, and IP Single-Line Adapter (SLA) ports, providing the functionality required to make phone calls through the IP network to the telephone system.

4.124 Unless specified otherwise, the term "device IPRC" indicates the v1.x firmware IPRCs that are programmed for devices. The term "v9.0.0 IPRC" refers to the IPRC firmware and configuration that supports both private networking and IP devices on one IPRC.

4.125 The IPRC is equipped with the following resources:

- **LED Indicators**: Show the status of the card and circuits. See the TROUBLESHOOTING section of this manual for additional LED information.
- **Serial Port Jacks**: (RS232, 8-pin, 8-conductor modular jack or line jack unit) Connect diagnostics monitors.
- **IP Port and LEDs**: Provides an RJ-45 port for connecting to the Local Area Network (LAN). The LEDs indicate when the network connection is valid and when data is being transmitted. See TROUBLESHOOTING for more information.

IP Devices Support

4.126 To support the IP phones and SLA, the slot in the equipment chassis that is used by the IPC or the v1.x firmware IPRC must be programmed as a DKSC-16 Card. (The "Amps Required" field shows that the card requires the 0.64 amps needed by a DKSC. However, an IPC or IPRC only requires 0.34 amps.) If using a v8.1 or later firmware IPRC, the slot is programmed as an IPRC (32 Device), even if the daughter card is not installed.

4.127 If using an IPC, the first eight circuits must be programmed according to the type of equipment you are connecting. The remaining eight circuits are unused, but can be programmed as "phantom" extensions in the phone system. If using a v1.x firmware IPRC, you can program eight or 16 circuits, depending on the PAL. Program circuits as "Keyset" if an IP PhonePlus, Model 8660 Phone, or IP SoftPhone is connected without loop-start access. Program circuits as "Single-Port LSA" if an MGCP gateway is connected. Program the circuits as "Dual Single-Line" for an IP SLA (even though the IP SLA supports only one single-line set). If a daughter card is installed, you can program up to 32 circuits, depending on the PAL.

4.128 If using a v8.x firmware IPRC, you can program up to 16 (32 with a daughter card) IP devices, including IP phones, IP single-line adapters (SLAs), and Media Gateway Control Protocol (MGCP) gateways and endpoints. The v8.x firmware IPRC also supports the multi-protocol endpoints, such as the Model 8600, which can use either Session Initiation Protocol (SIP) or Inter-Tel Protocol (ITP). To terminate a multi-protocol endpoint configured to use SIP, the latest version of SIP Server is required.

4.129 If using a v9.0.0 firmware IPRC, you can program up to 32 IP devices (16 IP devices without an IPRC daughter card), MGCP trunks, SIP trunks, multi-protocol devices, Model 8602 IP softphone, or IP networking ports. The v9.0.0 firmware supports all of the Inter-Tel hard and soft IP endpoints, including the multi-protocol endpoints running in ITP mode or SIP mode.

4.130 To support more than 16 devices, the IPRC requires the IPRC daughter card and v8.1 firmware. Also, you must first change enable the voice resource for that port. By default, all ports are set to "Out of Service." After you have enabled the voice resources, you can program the devices. For all IP devices except the IP SLA, program the circuit as "IP Phone;" for IP SLAs, program the circuits as "IP SLA;" for MGCP gateways and endpoints, program the circuits as "MGCP Gateway and Endpoint" or "MGCP Endpoint."

NOTICE

The IP SoftPhone requires v1.x firmware and is not supported by IPRCs that have v8.x or later firmware. Other IP endpoints, such as the Model 8660 Phone, however, require the 8.x or later IPRCs to support the features. For these reasons, Inter-Tel recommends that you do not connect IP SoftPhones and other IP endpoints to the same IPRC. If you do, you may lose feature or phone functionality.

4.131 Each IP Card (IPC) can support up to eight IP devices; each v1.x firmware device IPRC can support up to 16 IP devices; and each v8.1 or later firmware IPRC can support up to 32 devices (with the daughter card).

IP Networking Support

4.132 Previously, the Internet Protocol (IP) Networking Module (3200N) was the primary method to connect Inter-Tel systems to IP networks (for details on the IP Networking Module, see the *IP Networking Module Manual,* document part no. 730.8001). As of version 7.0, however, a v1.x firmware networking Internet Protocol Resource Card (IPRC), a v8.1 firmware IPRC, or a v9.0.0 firmware IPRC can be programmed to provide private networking using IP.

NOTE: The IPRC does not support modem connections and faxing over the IP network.

4.133 To use the pre-9.0.0 IPRC for private networking, the slot in the equipment chassis that is used by the IPRC must be programmed as IPRC (Private Networking). You must then program the voice resources and appropriate IP connection information (see page 6-12 for details). To use the v9.0.0 firmware IPRC for private networking, you must assign a networking VoIP resource to the IPRC in DB Studio (see page 6-145).

4.134 If programmed for IP networking, the IPRC supports 16 channels. To use an additional 16 channels (32 total), you must attach the daughter card to the IPRC (IPRC-D).

4.135 For more information about the IPC, IPRC, and IP devices, refer to the latest *IP Devices Installation Manual* (document part no. 835.2195).

H. LOOP START AND GROUND START TRUNK CIRCUIT CARDS

NOTE: LGCs and LGC-Ds are not supported in Europe.

4.136 Each LGC and LSC has circuitry for up to four analog trunks. For increased capacity, a daughter card (LGC-D or LSC-D) can be attached to each LGC or LSC to provide four additional circuits.

4.137 LSCs and LSC-Ds can have only loop start trunks connected to them. LGCs and LGC-Ds can have either loop start or ground start trunks. Individual trunks may be designated as either loop start or ground start in DB Programming.

NOTE: LGC-Ds cannot be installed on LSCs, and LSC-Ds cannot be installed on LGCs.

4.138 For optimal CO trunk transmission performance, each analog trunk circuit should be "hybrid balanced" in DB Programming. Hybrid balancing refers to selecting the balance network – normal or short – that best matches the trunk's impedance.

4.139 Each LGC, LGC-D, LSC, and LSC-D has four specially-designed fuses soldered to the card to protect it from large voltage surges due to lightning. If a fuse is open (blown), it may mean that other parts of the card have also been damaged. For this reason, open fuses must not be replaced in the field. Instead, the entire card *must* be returned for testing and possible repair. Failure to follow this policy voids the warranty on the card.

4.140 Each card has a green LED (CARD ON-LINE) to indicate the card's functional readiness and a yellow LED (CIRCUIT BUSY) to indicate if any circuit on the card is busy (see Figures 3-59 and 3-60 on pages 3-122 and 3-123). Also, each card has a metal EMI shield attached to the front edge to help prevent electromagnetic interference.

4.141 When the system is in the default configuration, all trunks are configured for DTMF signaling. If necessary, some or all of the trunks can be reprogrammed for dial-pulse signaling through DB Programming.

NOTE: Single-line sets generate their own DTMF tones when dialing. If used on dial-pulse trunks, both the DTMF tones generated by the phone and the dial-pulse signals generated by the system are sent to the central office [local exchange], which may be a problem if the central office [local exchange] recognizes both. To avoid this problem, single-line sets should be restricted to ARS-only when using dial-pulse trunks. Then, only the ARS-generated dial-pulse signals are sent to the central office [local exchange].

4.142 The CO circuits are assigned hardware address numbers according to the card's slot in the equipment chassis and the location of the circuit on the card. For example, when using the initial configuration, CO circuit 07.04.01 is the fourth circuit on the LSC installed in slot 7. Each trunk card circuit is assigned a six-digit circuit number XX.YY.ZZ, where XX = the card slot location (01-07 or 09-15), YY = the circuit number on the card (01-08), and ZZ = whether the circuit has more than one purpose (currently 01 only).

4.143 When the system is in the default configuration, the first four buttons on each phone are assigned as CALL buttons for incoming call indication/access. (The ANSWER button may also be used to access incoming calls.) No individual trunk buttons or trunk group buttons are assigned. Button assignments can be designated in DB Programming. See PROGRAMMING, page 6-306, for more information.

4.144 LGC and LGC-D Modification: Under certain conditions, previous versions of the LGC (part no. 550.2309) and the LGC-D (part no. 550.2310) were not able to always properly detect the ground start signal provided by the central office (CO). A modification to the LGC and LGC-D has been implemented to correct this problem.

4.145 More specifically, the previous versions of the LGC and LGC-D did not function properly in ground-start mode when terminated on a CO trunk that had a constant current battery feed of less than approximately 25mA. The new cards have a built-in 50VDC power supply and therefore no longer rely on the current from the CO to detect the "ground start" signal.

4.146 A customer site may have this problem if a phone user's display shows "LINE IS UNPLUGGED" when attempting to access a ground start trunk on an LGC or LGC-D. To determine if the card really does have a problem detecting the ground start signal from the CO, follow these steps:

1. Ensure that the appropriate trunk circuit on the card is programmed for ground start mode. (Loop start mode is not subject to this problem.)
2. Verify that the card is properly installed and connected.
3. Verify that tip and ring for the trunk circuit are properly connected. Here is a simple method that works for most cases:
   a. At the associated CO block, remove the bridging clips for the trunk.
   b. On the telco side of the block, attach a test set (butt set) in monitor mode (on-hook) to tip and ring.
   c. Apply ground to the ring terminal.
   d. *If you hear dial tone (only while ground is applied),* tip and ring are properly connected. Replace the bridging clips and make sure the card can still receive dial tone.

      If you do not hear dial tone, apply ground to the tip terminal. If you now hear dial tone, tip and ring are reversed.

> **NOTE:** Be sure the test set is in monitor mode. If it is in off-hook mode, sometimes the line can be seized regardless of the polarity, and the test is invalid.

4.147 Identifying the Modified Cards: Remove the LGC and/or LGC-D and look at the part number stenciled on the card. Immediately after the part number, there should be a dash (-) and a single letter contained within a box. This denotes the revision level of the card. The following chart shows the revision level of the current, modified cards:

Table 2-10. *LGC/LGC-D Revision Levels*

Card	Part No.	ECN No.	Revision
LGC	550.2309	4813	826.5223-4D (or higher) 826.5223-5D (or higher) 826.5223-6 (any letter)
LGC-D	550.2310	4814	826.5224-3B (or higher) 826.5224-4B (or higher) 826.5224-5 (any letter)

4.148 Improperly paired cards do not work together and the trunks attached to these cards do not function.

4.149 LGC -6 cards only work with LGC-D -5 cards (and vice versa). They are not backwards compatible. LGC -4 and -5 cards only work with either LGC-D -3 and -4 cards.

4.150 If a system is experiencing this problem ***and*** if it is equipped with older, unmodified LGC and/or LGC-D cards, the card(s) may be replaced by contacting Inter-Tel Technical Support.

I. T1 CARDS (T1CS), E1/PRI CARDS (E1/PRIS), AND T1/E1/PRI CARDS

4.151 The following information is described in this section:

- T1 Card—T1C (see below)
- E1/PRI Card (see page 2-53)
- T1/E1/PRI Card (see page 2-54)
- Special T1 or E1/PRI Cabling Requirements (see page 2-54)
- T1C, E1/PRI, or T1/E1/PRI Cards on a Network (see page 2-55)
- T1 or E1/PRI Network Clock Card—NCC (see page 2-55)
- T1C, E1/PRI, or T1/E1/PRI Cards Features (see page 2-56)

T1 Cards (T1Cs)

NOTE: T1Cs are not applicable to European systems.

4.152 The acronym for the T1 Card (T1C) should not be confused with the special T1C digital carrier system (which operates at twice the data rate of the standard T1 carrier system). The T1 Card does not support T1C.

4.153 The term "T1" refers to a specific digital method of transmitting voice and data; it is the basic 24-channel time-division multiplex (TDM), 1.544 Mb/s pulse code modulation (PCM) system as used in the United States.

4.154 The T1 standard, which was developed as a method of reducing cabling and increasing transmission quality between distant central offices, is actually a subset of a multi-level Digital Signal (DS) scheme. In popular usage, T1 refers to the DS1 rate as transmitted via any type of media (e.g., copper wire, fiber optics, or microwave).

4.155 Because each T1 span actually consists of 24 individual circuits (or channels) multiplexed together, it is often less expensive to purchase a single T1 span than it is to purchase multiple individual trunks. Each of the 24 T1 circuits can be programmed as a loop start, ground start, DID, E&M, or OPX trunk (or as unequipped). Any combination is possible.

4.156 "Fractional" T1, often called FT1, is a special T1 service where only a portion of the 24 channels are used and paid for. To implement fractional T1 (if available), equip only those circuits that are provided.

NOTE: Currently, the T1C and T1/E1/PRI cards are used to transmit voice signals only, not data. In the future, data transmissions over T1 circuits may also be supported.

4.157 To interface with the public network, each T1 Card is equipped with an on-board-channel Service Unit (CSU). Because the CSU is on-board, the programmer must select the line build-out (LBO) attenuation, which is available from the T1 provider.

NOTE: To use an external CSU instead, the card can be set to DSX mode. However, the card and the external CSU must be located in the same building, and the cabling between them must be no longer than 133 feet.

4.158 When a T1 span is used to connect the phone system to the public network or to another telephone system, it is important that the two endpoints communicate at the same frequency. Because it is practically impossible for both endpoints to have clocks running at identical frequencies, one endpoint is given "master" clock status and the other endpoint is given "slave" clock status. The slave clock can then extract and lock onto the master clock frequency from the incoming T1 information. This master/slave synchronization is programmed in the database. When connected to the public network, the T1 Cards should be designated as public network slaves.

4.159 When information (currently, voice only) is transmitted over a T1 span, a "framing" bit is added so that the receiving end can recognize where the information begins and ends (as long as the clock circuitry of the two endpoints is synchronized). Currently, the phone system supports the two major types of framing schemes in use today: D4 Superframe, which is the most common, and Extended Superframe (ESF), which is required to support future ISDN and DMI features. For voice transmissions, D4 Superframe is quite adequate. However, for data transmissions (not currently supported), ESF is much more reliable and efficient.

4.160 Digital information is transmitted over the T1 span as a series of 1s and 0s (e.g., 1000 0000 0000 0001...). If there are too many consecutive zeroes, the T1 clock extraction circuitry does not operate correctly. To prevent this, T1 carrier equipment limits the number of consecutive zeroes. The Inter-Tel system supports the two major types of zero code suppression schemes in use today: Alternate Mark Inversion (AMI), also called bit-7 stuffing, and Bipolar Eight Zero Substitution (B8ZS). Again, for voice transmissions, AMI is quite adequate. However, for 64 kbps clear channel ***data*** transmissions (not currently supported), B8ZS is required.

4.161 Like the line build-out attenuation, the framing scheme type and zero code suppression scheme type are also available from the T1 provider. In fact, when ordering a T1 span, always request the following information from the T1 provider:

- **Framing Scheme**: D4 Superframe or ESF
- **Zero Code Suppression Scheme**: AMI or B8ZS
- **Recommended Line Build-Out Attenuation**: 0dB, 7.5dB, 15dB, or 22.5dB

4.162 Each card has circuitry for up to 24 trunk channels. The 24 channels (or circuits) can be any combination of loop start trunks, ground start trunks, DID trunks, E&M trunks, or OPX trunks.

4.163 T1 Card Firmware

NOTE: The T1C is a discontinued card and is replaced by the T1/E1C. The T1C requires firmware EPROMs while the T1/E1C uses flash memory to load the firmware.

4.164 For T1Cs to work properly with version 4.0 and later system software, the T1Cs must be at revision level 2.G or later. To use PRI or the Networking feature, a PRI daughter card (part no. 550.2733) is required. In addition, the T1Cs must be equipped with specific firmware on U4 and U5. To upgrade the T1C for version 5.0, use kit number 828.1459. To upgrade the T1C for versions 5.1 and 5.2, use kit number 828.1506, which contains part no.s 827.8980 (U4) and 827.8981 (U5). To upgrade the T1C for versions 5.3, 5.4, 6.0, 7.0, 8.0, and 9.0 use kit number 828.1552, which contains part no.s 827.9831 (U4) and 827.9832 (U5). To upgrade the E1C for versions 5.3, 5.4, 6.0, 7.0, 8.0, and 9.0 use kit number 828.1553, which contains part no.s 827.9833 (U4) and 827.9834 (U5). Revision level 2.F or earlier T1Cs cannot be upgraded; they must be replaced with new T1Cs. This information is summarized in the following chart:

Table 2-11. *T1C Revision Levels*

T1C Rev. Level	System Software Level
up through 2.F	v2 or v3 T1 only
2.G to 3.J	v2 (or later) T1, and v4.1 (or later) PRI (if equipped with 828.1367 kit and PRI Daughter Card)
3.K or later	v2 (or later) T1, and v4.1 (or later) PRI (if equipped with PRI Daughter Card)

NOTE: For standard T1 support, the current version of T1 firmware and/or flash is backwards compatible with all system versions of software. Inter-Tel recommends using the latest firmware for the T1C or the latest flash for the T1/E1 when using the standard T1 feature.

4.165 The T1 OPX Disconnect feature is available in all versions of T1/E1 flash and T1C firmware (part no. 827.8664 or greater for U4 and part no. 827.8665 or greater for U5). To remind you, a warning appears in DB Programming whenever the flag is changed. The warning states that the flag has no effect unless the correct T1 firmware is installed on the card.

4.166 The adjustable gain control is available in all versions of T1/E1 flash and T1C firmware starting with 827.8787 (U4) and 827.8788 (U5). This is supported in system versions 4.4 and 5.1 and later.

NOTE: For firmware information on PRI or standard T1 support for versions older than 5.1, see Tech Tip 550.0228.

4.167 T1 Card with Primary Rate Interface (PRI) Daughter Card

4.168 A PRI Daughter card can be attached to up to eight T1 Cards to allow the system to interface with an ISDN Primary Rate Interface trunk. Each PRI trunk provides a 23B+D interface that delivers 23 bearer channels (B-channels), each operating at a rate of 64,000 bits per second (bps), one data signaling channel (D-channel) also operating at 64,000 bps, and framing information at 8,000 bps – for a total bandwidth of 1.544 megabits per second.

4.169 The PRI trunk is physically connected to the T1 Card using the RJ48C jack or the DB15 connector normally used to connect a T1 trunk.

4.170 Each PRI trunk must be configured for the proper switch type of service (dedicated service, on-demand B-channel selection, or a combination of the two). See page 5-60 for more information. The Inter-Tel system is designed to support only AT&T 4ESS Custom, AT&T 5ESS Custom, National ISDN, National IDSN2, and DMS-100 switch types. No other switch types are currently supported by the phone system.

4.171 The Inter-Tel system currently supports the following ISDN Primary Rate Interface services:

- Automatic Number Identification (ANI)
- On-Demand B-Channel Selection
- Dialed Number Identification Service (DNIS)
- Calling Party Number
- Call-By-Call Service Selection

4.172 T1 Gain Control (TGC) Daughter Card

4.173 If experiencing excessive E&M gain levels on a T1 Card that is not equipped with a PRI Daughter card, two special software components and a TGC Daughter card (kit part no. 828.1397) can be attached to the card to provide 6dB of attenuation.

4.174 To use the TGC Daughter card, the T1 Card must be at revision level 2.G or later.

E1/PRI Cards (E1/PRIs)

NOTE: E1/PRI cards are applicable to European systems only.

4.175 The E1C Card requires a PRI Daughter Card to interface with an ISDN Primary Rate Interface (PRI) trunk. Also, to work properly with system version 4.0 and higher system software, E1C/PRIs must have a hardware revision level 2.G or later.

4.176 The firmware for the E1C/PRI Card resides in EPROMs (located at U4 and U5) and needs to be compatible with the version of system software in which it is being used. For version 5.2, use part numbers 827.9067 (U4) and 827.9068 (U5), which can also be obtained in the SW206 upgrade kit.

4.177 The E1C/PRI Card supports programmable gain control (as described on page 6-257) for the new versions of system software that support programmable gain control.

4.178 E1/PRI Cards Specifications

4.179 Each E1/PRI trunk provides the system with a digital interface that delivers 30 bearer channels (B-Channels), one signaling channel, and one synchronization channel. Each channel operates at a rate of 64,000 bits per second (bps) for a total bandwidth of 2.048 megabits per second.

4.180 Currently, the E1/PRI Cards support Euro-ISDN Net5. This specification is also referred to as ITU-T Recommendation I.421, although it is actually ETS 300-156. The layer 3 signaling is normally called ITU-T Recommendation Q.931 but is actually ETS 300-102. The ITU (International Telecommunication Union) is an international organization that coordinates global telecom networks and services and publishes telecommunication technology, regulatory and standards information. Recommendations I.421 and Q.931 are published by the ITU-T (Telecommunication Standardization Sector of the ITU). The ETS specifications are published by the European Telecom Standards Institute.

4.181 The system utilizes CLIP and DDI as presented by these services and allows routing to be based on either or a combination of both.

4.182 Currently ISDN voice services are the only ones supported. In the future, ISDN data and video services may be supported.

4.183 The information that is programmed specific to each circuit (e.g., each of the 30 channels) on the E1/PRI Card includes the Circuit Type, which can be programmed as B-Channel or None. See PROGRAMMING, page 6-126.

4.184 The E1 circuits are assigned hardware address numbers according to the card's slot in the equipment cabinet and the location of the circuit on the card. For example, E1 circuit 6.1.21 is the 21st circuit on the 6th card, an E1/PRI, installed in the equipment cabinet.

4.185 Each PRI trunk must be configured for the proper switch type of service. See page 5-55 for more information. The system is designed to support only PRI Net5 and Private Networking switch types. No other switch types are currently supported by the system.

4.186 The system can support up to eight E1/PRI cards for system version 5.1 or higher. Older versions support up to four E1/PRI cards.

T1/E1/PRI Cards

4.187 The T1 or E1/PRI card is compatible with any version of system software that supports T1 or E1/PRI (domestic or international). The T1 or E1/PRI board is converted to a T1/E1/PRI by changing a feature PAL on the board. (There is no daughter board for PRI.) The system can support up to eight T1/E1/PRI cards.

4.188 The T1/E1/PRI card supports programmable gain control, as described on page 6-257. However, it also has DIP switches to program gain control for older versions of software. (If the phone system is upgraded to a version of software that supports programmable gain control, the programmed setting in the database overrides the settings programmed by the DIP switch.)

4.189 The firmware for the T1/E1/PRI card resides in flash memory and needs to be compatible with the version of system software in which it's being used. For 5.1 or newer versions, the system software downloads this firmware to the T1/E1/PRI card automatically whenever there is a compatibility mismatch. For older systems, the corresponding T1/E1/PRI firmware is provided on a disk and can be uploaded into the flash memory via a programming PC. See page 3-126 in INSTALLATION for details.

Special T1 or E1/PRI Cabling Requirements

4.190 Inter-Tel recommends that special high-speed data cable with the following characteristics be used for all T1 or E1/PRI installations:

- 22 AWG (22 gauge thickness or larger -- smaller number gauge = larger thickness)
- Individually shielded two-pair (or with the transmit and receive pairs run in separate cables or binder groups)
- ABAM or PIC (a level-3 cable designed specifically for the T-carrier transmission rate)
- Maximum capacitance of 15pF/ft
- Maximum attenuation of 5dB/1000 ft (300 m) at 772 kHz and 70° F (21° C)

> **NOTE:** The programming prompts for LBO attenuation (and DSX mode, if used) assume that proper T1 cable (with the above characteristics) is being used. When transmitting over shorter distances, it may be possible to use standard two-pair voice frequency cable instead; however, the programmer may have to compensate by selecting a smaller LBO setting. Also, no matter which type of cable is used, the transmit and receive pairs must be shielded from each other (or run in separate cables or binder groups).

4.191 If the DB15 connector on the T1 Card is used in place of the standard RJ48C jack, a flat, 26 AWG (26 gauge thickness or larger -- smaller number gauge = larger thickness), twisted-pair ribbon cable with a maximum capacitance of 16pF/ft may be used for distances of up to 50 feet [15 meters]. See page 3-36 for more information. For distances longer than 50 feet [15 meters], use the special high-speed data cable described in paragraph 4.190.

4.192 The PRI trunk is physically connected to the T1, E1/PRI, or T1/E1/PRI card using the RJ45 jack. The DB15 connector is normally used to connect test/monitoring equipment.

T1C, E1/PRI, or T1/E1/PRI Cards on a Network

4.193 Each network span requires two T1C, E1/PRI, or T1/E1/PRI cards, one for each end of the span. The cards require the most recent firmware (use upgrade kit part number 828.1459).

4.194 For each network span, order a clear channel, point-to-point T1/E1/PRI span from the service provider (not PRI service, even though you are using T1C, E1/PRI, or T1/E1/PRI cards).

NOTE: The system requires T1 channel 24 in any network span.

ALSO: For fractional E1 (not all 32 channels in use) in any network span, the system requires E1 channel 0 for synchronization and E1 channel 16 for signaling information.

4.195 When a T1/PRI, E1/PRI, or T1/E1/PRI card is programmed for networking (Private Network switch type), the system automatically creates a node trunk group. (All B-Channels residing on the same T1/PRI card are in the same node trunk group.)

4.196 The system allows up to eight node trunk groups per node (one for each T1/PRI, E1/PRI, or T1/E1/PRI card).

T1 or E1/PRI Network Clock Card (NCC)

NOTICE

System versions 8.0 and later do not require a Network Clock Card (NCC) for 64 or 128 systems. If upgrading to 8.0, you must remove the NCC before installing the CPC. If the NCC is not removed, the system continually resets.

4.197 If any T1 or E1/PRI cards are to be installed in a chassis using the CPU64 or CPU128 card, a single Network Clock Card must also be installed. Then, instead of using the clock circuitry on the CPU card, the NCC becomes the clock for the entire system. The NCC provides the smooth, error-free clock transitions required for T1 or E1/PRI applications.

NOTE: Do not install an NCC in chassis equipped with the CPU020/PCM card; it could cause T1 or E1 timing problems. If one is accidently installed, the programming PC displays a warning.

4.198 The NCC is installed in the expanded chassis connector slot (J10) on the far right side of the equipment chassis backplane. When two basic chassis are attached together to form a dual-chassis system, the NCC is installed only in the second (right-hand) chassis.

NOTE: Be sure the system power is turned OFF before inserting or removing the Network Clock Card.

T1C, E1/PRI, or T1/E1/PRI Cards Features

4.199 The T1C, E1/PRI, or T1/E1/PRI cards are equipped with the following resources:

- **LED Indicators**: Show the status of the card and circuits and to show T1, E1/PRI, or T1/E1/PRI trunk alarms and board-level error occurrences. See the TROUBLESHOOTING section of this manual for additional LED information.
- **Serial Port Jack**: (RS232, 8-pin, 8-conductor modular jack or line jack unit) Connects a diagnostics monitor.
- **T1 or E1/PRI Span Jack**: (RJ48C, 8-pin, 8-conductor modular jack [or RJ45 line jack unit in Europe] Connects a single, 24-channel T1 span to the T1 Card or 30-channel E1 span to the E1/PRI Card. (If this jack is used, do not use the DB15 connector also provided on the card.)

NOTE: All T1 or E1/PRI spans should be disconnected from their cards before turning the system power on or off.

- **Dual Bantam Jack**: (with transmit and receive audio ports) Connects T1 or E1 test equipment to the T1C or E1/PRI card and monitoring the transmit and receive signals.

CAUTION

When a live T1 or E1/PRI span is installed on the T1C or E1/PRI card, test equipment should be used for monitoring purposes only. Do not attempt to send signals over either the transmit or receive audio ports on the T1C or E1/PRI card if a T1 or E1/PRI span is installed. Also, be sure the test equipment is set in the “bridge” mode.

- **Alternate T1 or E1/PRI Span DB15 Connector**: (15-pin D-subminiature female connector) Connects a single, 24-channel T1 or 30-channel E1/PRI span to the T1 Card.

NOTE: In certain situations, it may be more convenient to use the DB15 connector instead of the standard RJ48C jack also provided on the card.

- **Make-Busy Switch and LED**: Removes all circuits on the card from service and places them back in again. Pressing the switch lights the yellow make-busy LED and begin the process of “busying out” each circuit not in use. The circuits that are in use are then busied out as soon as the users hang up. When all circuits are busied out, the make-busy LED flashes slowly, the circuit-busy LED is unlit, and the card can be removed from its slot in the equipment cabinet. If the card is not busied out and removed within a certain amount of time, a programmable timer expires and the card returns to normal operation; however, the make-busy LED then flashes rapidly until the make-busy switch is released.
- **DTMF Transmitters**: *(T1/E1/PRI card only)* Generates DTMF tones for traffic on the T1/E1/PRI card. To use these transmitters, the Enable T1/E1/PRI DTMF Dialing flag must be enabled (see page 6-260). If this flag is disabled, the DTMF transmitters on the CPC or CPU card are used instead.

NOTE: The T1/E1/PRI card DTMF transmitters are not shared resources. Only the T1/E1/PRI card can use these transmitters.

4.200 Compared to the other trunk cards in the system, the T1, E1/PRI, or T1/E1/PRI card is very sophisticated and offers some unique programming options. The information that is programmed *specific to each T1, E1/PRI, or T1/E1/PRI card* installed in the system includes:

- **Framing Type**: The type of framing scheme used by the T1 trunks connected to the card can be D4 Superframe or Extended Superframe (ESF). The type of framing scheme used by the E1/PRI trunks connected to the card can be Common Channel Signaling or Channel Associated Signaling.
- **Zero Code Suppression Scheme**: *With a T1 Card*, the programmer must select the type of zero suppression scheme, AMI (bit-7 stuffing) or Bipolar Eight Zero Substitution (B8ZS), used by the T1 span connected to each card. *With an E1/PRI card*, the system allows this field to be programmed for either HDB3 or None. Note that this parameter needs to match the configuration of the remote end of the E1 span to work properly; therefore, verify that the remote end is configured in the same manner. Digital information is transmitted over the E1 span as a series of 1s and 0s (e.g., 1000 0000 0000 0001...). If there are too many consecutive zeroes, the E1 clock extraction circuitry does not operate correctly. To prevent this, E1 carrier equipment can limit the number of consecutive zeroes by using a zero-code suppression scheme. The system supports the HDB3 type of zero code suppression, which should be selected for PRI to function correctly.
- **Line Build-Out (LBO)** *(T1 and T1/PRI Only)*: The programmer must designate the LBO attenuation of the T1 span connected to the card. This value can be determined automatically by the system, or it can be set according to the T1 provider's recommendation.
- **CRC Processing** *(E1/PRI Only)*: The system allows CRC Processing to enabled or disabled. Note that this parameter needs to match the configuration of the remote end of the E1 span to work properly; therefore, verify that the remote end is configured in the same manner. An optional cyclic redundancy check (CRC) has been incorporated within the E1 bit stream to provide additional protection against simulation of the frame alignment signal, and /or where there is a need for an enhanced error monitoring capability. CRC information is transmitted within the synchronization channel and also requires a multi-frame synchronization process.
- **Reference Clock Programming**: *With a U.S. system*, it is crucial that all T1 Cards and the CPC or CPU card share the same clock signal so that they can remain synchronized with each other and the T1 span. This is accomplished by allowing one of the T1 Cards to act as the system reference clock. The programmer must designate whether the T1 Card is a private network master, private network slave, or a public network slave (the public network always acts as the master clock). Cards that are used for clock recovery from the network should always be located in the master part of a quad configuration. *With a European system*, the system has several clocking options. However, whenever the system is connected digitally to the network, it should always be a slave to the network. Currently, E1/PRI cards are the only cards than can recover clock from the network and provide it to the rest of the cards in a system, synchronizing all the system cards with each other and making the system a slave to the network.

4.201 If more than one T1, E1/PRI, or T1/E1/PRI card is installed in the system, the software uses an ordered list to determine a "back up" to the system reference clock. Then, if the T1, E1/PRI, or T1/E1/PRI card acting as the system reference clock fails or is taken out of service, the next T1, E1/PRI, or T1/E1/PRI card on the back-up priority list takes over.

- **Error Diagnostics**: If desired, the programmer can generate the current T1, E1/PRI, or T1/E1/PRI error counts (by hour and by day) for the selected card. The error counts are displayed in tabular format.
- **Error Thresholds**: The programmer can set numerical thresholds (number per hour and number per day) for certain T1, E1/PRI, or T1/E1/PRI card errors. If a threshold is exceeded, an error is printed on the SMDR report.

4.202 The information that is programmed *specific to each circuit* (e.g., each of the 24 channels) on the T1 Card includes:

- **Circuit Type**: Each individual circuit can be programmed as a loop start, ground start, DID, E&M, or OPX trunk. See PROGRAMMING, page 6-126.
- **DTMF or Dial-Pulse Signaling**: When the system is in the default configuration, all trunk circuits are configured for DTMF signaling. If necessary, some or all of the circuits can be reprogrammed for dial-pulse signaling.

4.203 The T1, E1/PRI, and T1/E1/PRI circuits are assigned hardware address numbers according to the card's slot in the equipment chassis and the location of the circuit on the card. For example, T1 circuit 6.1.21 is the 21st circuit on the 6th card, a T1C, installed in the equipment chassis.

J. BASIC RATE INTERFACE (BRI) CARDS (BRSS/BRUS)

NOTE: BRU cards are not supported in Europe.

4.204 The Inter-Tel system includes Basic Rate S/T Interface (BRS) and Basic Rate U Interface (BRU) cards to support ISDN Basic Rate Interface. These cards enable the system to connect to video, voice, and data devices such as video conference units, ISDN modems, LAN bridges, and ISDN telephones. The cards provide a cost-effective alternative to PRI for small systems requiring ISDN services. However, BRI cannot be used for networking two different systems.

NOTE: The Inter-tel Private Networking feature is not an option when using the BRI card. To use the Private Networking feature with the Inter-Tel systems, the PRI card must be used.

ALSO: The U.S. software does not automatically upload the BRI software to the BRS and BRU cards as it does with T1 or E1/PRI Cards. To ensure that the most current version of BRI software is loaded, the BRS/BRU software should be manually uploaded at the time of installation. The U.K. version software automatically uploads the BRI software to the BRS card.

4.205 BRS and BRU cards support:

- BRI stations that connect to data/video devices behind the PBX via BRS card with an S interface and a BRU card with a U interface.
- BRI trunks that connect to the network via BRS cards with an NT1 device.
- Data/video switching from behind the PBX out any U interface trunk (e.g., PRI trunk and BRI trunk) to the public network.

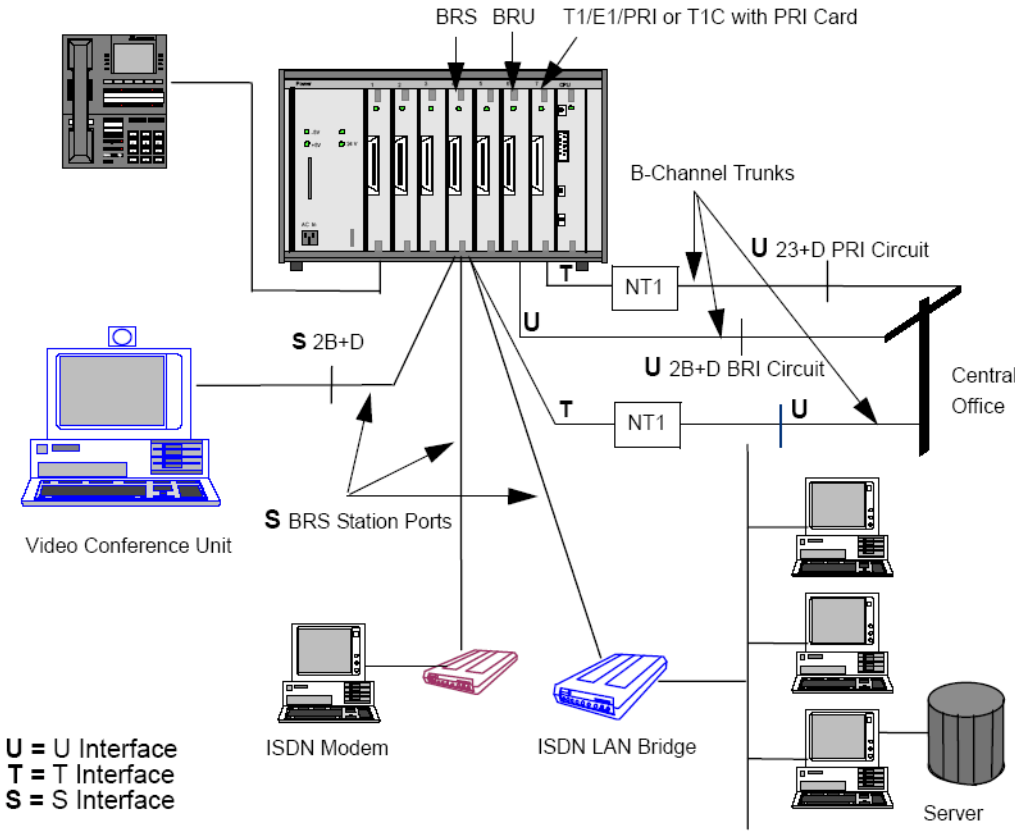
4.206 BRI stations operate like other stations with the following exceptions:

- BRI station ports do not support passive bus configuration (one ISDN device per BRS port).
- A BRI station cannot perform features while connected on a call that require a hookswitch or SPCL button, including Individual Hold, System Hold, Transfer, Record-A-Call, and Conference.
- The BRI station cannot have a mailbox, but can have an extension ID.
- A message cannot be left for a BRI station.
- Centrex options are not supported behind the BRS or BRU card.
- Caller ID [CLIP] information does not include the caller's name (i.e., only the caller's number is available).

4.207 A BRI station cannot place a call on both B-channel stations with a single setup message. The bonding must be performed outside the D-channel signaling. In other words, a video conference unit or data device must place two separate calls and perform the bonding at the external device level. The routing user information provided on the D-channel at this time is call setup and call termination messages only.

4.208 A system with 64 or 128 devices is limited to two BRI cards (BRS, BRU, or a combination), while systems with CPU020-D or CPU020-Q cards may have eight cards installed in any combination of BRS and BRU. The additional BRI cards do not increase the T1/PRI or E1/PRI card capacities.

Figure 2-4. *Example of a System Using BRI*



4.209 The drawing above is an example of a system with BRI. It includes the following:

- A video conference unit connected to port 1 of a BRS card in slot 4.
- An ISDN modem connected to port 2 on the same BRS card.
- An ISDN LAN bridge connected to port 3 of the BRS card in slot 4.
- Trunks connected to a BRU card in slot 6 and a T1/E1/PRI or T1C with PRI card in slot 7.
- A Phone in port 1 of the DKSC in slot 1.

4.210 The phone can use the 23 B-channels on the PRI card, 2 B-channels on the BRU card [30-channels on the European PRI], and the 2 B-channels on the BRS card going to the NT1 to place outgoing voice calls via ARS, trunk group, or individual trunk access. The phone can receive incoming voice calls on the same B-channels because all the B-channel trunks are in the same trunk group that routes calls by trunk number via a call routing table.

4.211 The video conference unit can place an outgoing call on one or both of its B-channels out any one of the B-channel trunks (either BRI or PRI). The video conference unit can receive an incoming call on one or both of its B-channels via a call routing table used by the trunk group that contains the B-channels. Note that the video conference unit could place an IC call to another video conference unit in the same system by dialing the extension or extensions of the B-channel station of the other video conference unit.

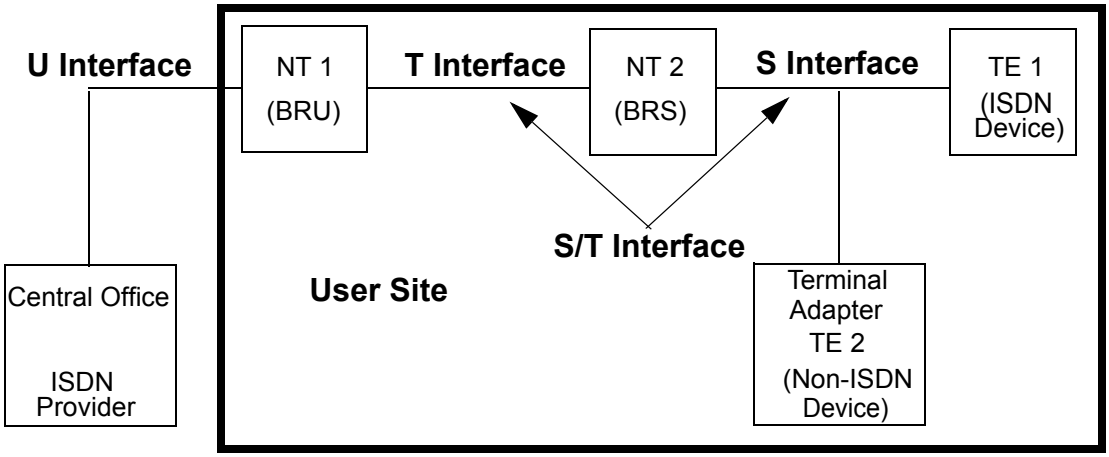
4.212 The ISDN modem and ISDN LAN Bridge function the same way as the video conference unit for incoming and outgoing calls.

4.213 Basic Rate Interface terminology and the two different types of BRI cards are described in the following pages.

Basic Rate Interface Terminology

4.214 The following terms are defined to clarify the functional aspects of the two different types of BRI cards available for the system (see the diagram following the definitions):

BRI ISDN Reference Points



- **Terminal Equipment (TE)**: A general term for a device at the user end of a telephone connection. BRS cards are connected to either a Network Terminator (NT) or a user device. The BRS port is the TE when it is connected to a Network Terminator (NT). If a user device is connected to the BRS port, then that device is the TE.
- **Terminal Equipment Type 1 (TE1)**: An ISDN device that terminates the user end of a telephone connection, such as an ISDN telephone, an ISDN modem, an ISDN LAN bridge, or a video conference unit.
- **Terminal Equipment Type 2 (TE2)**: A non-ISDN device that terminates the user end of a telephone connection, such as an analog telephone, fax machine, or analog modem.
- **Terminal Adaptor (TA)**: A device that terminates the ISDN connection and interfaces to an analog user device, such as an analog telephone, fax machine, or modem.
- **Network Terminator (NT)**: A general term for a device on the customer end of a connection to the Central Office [Local Exchange in Europe] that resides between the user's terminating ISDN device (TE or TA) and the Central Office [local exchange]. For systems with BRS cards, the NT is the BRS port. In the U.S., the NT device is provided by the customer. In other countries this device is often supplied by the ISDN provider.
- **Network Terminator 1 (NT1)**: The primary terminating device that resides between the Central Office and the customer site. The Central Office [local exchange] side connection to this device is called the U interface. The customer side connection is called the T interface. A BRU port has a built-in NT1 on each port. With BRI (U.S. only), the NT1 is normally provided by the local service company or consumer. With PRI, NT1 is not normally provided by the local service company or consumer, instead the ISDN provider or Central Office [local exchange] provides the NT1 for the customer location.
- **Network Terminator 2 (NT2)**: This terminating device resides between the primary terminating device (NT1) and the user's ISDN device (TE or TA). The side of the connection that links to the NT1 is called the T interface (see above), and the side that connects to a user device is called the S interface. A BRS port is an NT2 device and so is the T1/E1, T1C, and E1C.

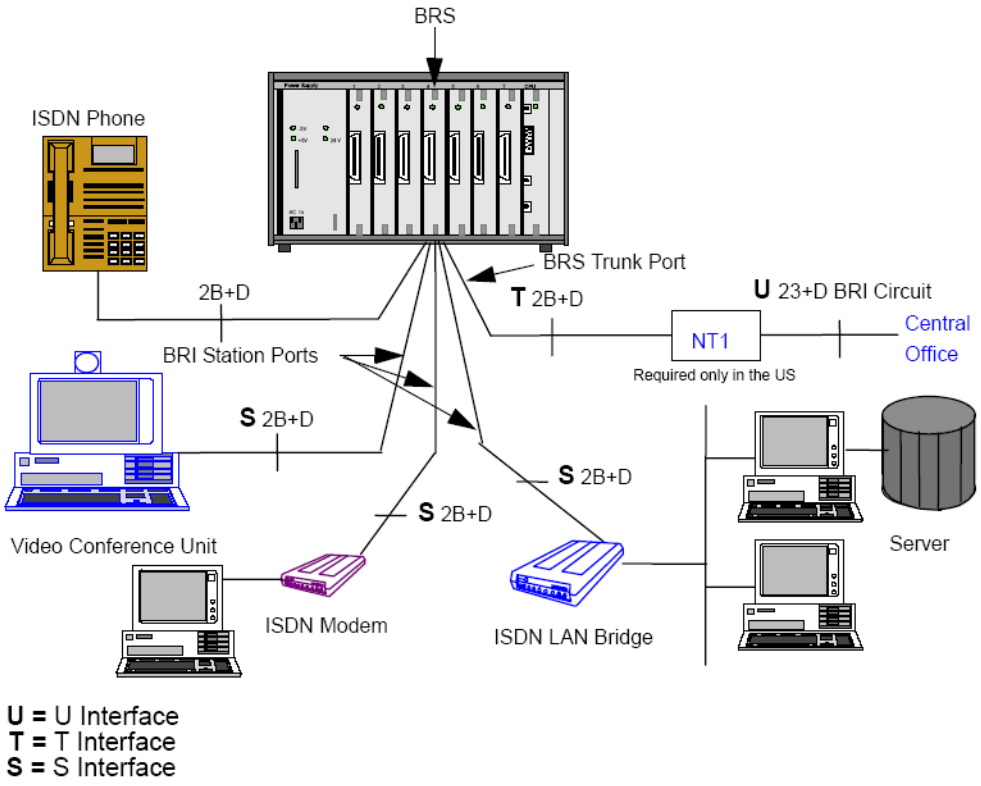
- **U Interface (User Interface)**: A connection coming into the system from an external transmission system (ISDN provider), which is typically a Central Office [local exchange]. (BRI U interface is a two-wire connection and PRI U interface is a four-wire connection.) All BRI ISDN devices with a U interface normally have an NT1 built-in to interface directly to the Network (ISDN provider or Central Office/Local Exchange).
- **T Interface (Trunk Interface)**: A four-wire connection between the NT1 that terminates the transmission coming from the Central Office [local exchange] and the NT2 that connects to the user's TE. This interface is located at the customer site.
- **S Interface (Station Interface)**: A four-wire connection between the NT2 and the user's TE. This interface is located at the customer site.
- **S/T Interface (Station/Trunk Interface)**: Because the BRI Station card (BRS) can be programmed for either a station port (S) or a trunk port (T) the interface, LIT is referred to as the S/T Interface.

Basic Rate S/T Interface (BRS) Card

4.215 The Basic Rate S/T Interface card (BRS) allows the system to connect to North American or European ISDN basic rate trunks via S/T interface. The card supports eight ports. Each port consists of two bearer channels (B-channels), each operating at a rate of 64,000 bits per second, and one data channel (D-channel) operating at a rate of 16,000 bits per second.

4.216 BRS station ports can be connected to a data device (e.g., ISDN Modem or ISDN LAN Bridge), a video device (e.g., video conference unit), or an ISDN phone via S/T interface. BRS trunk ports may be connected to a Central Office [local exchange] ISDN BRI circuit via an S/T interface and a Network Terminator Type 1 (NT1). In the U.S., the Central Office terminates BRI circuit with a U interface. Therefore, an NT1 device is required to convert the U interface to an S/T interface.

Figure 2-5. *Sample BRS Installation*



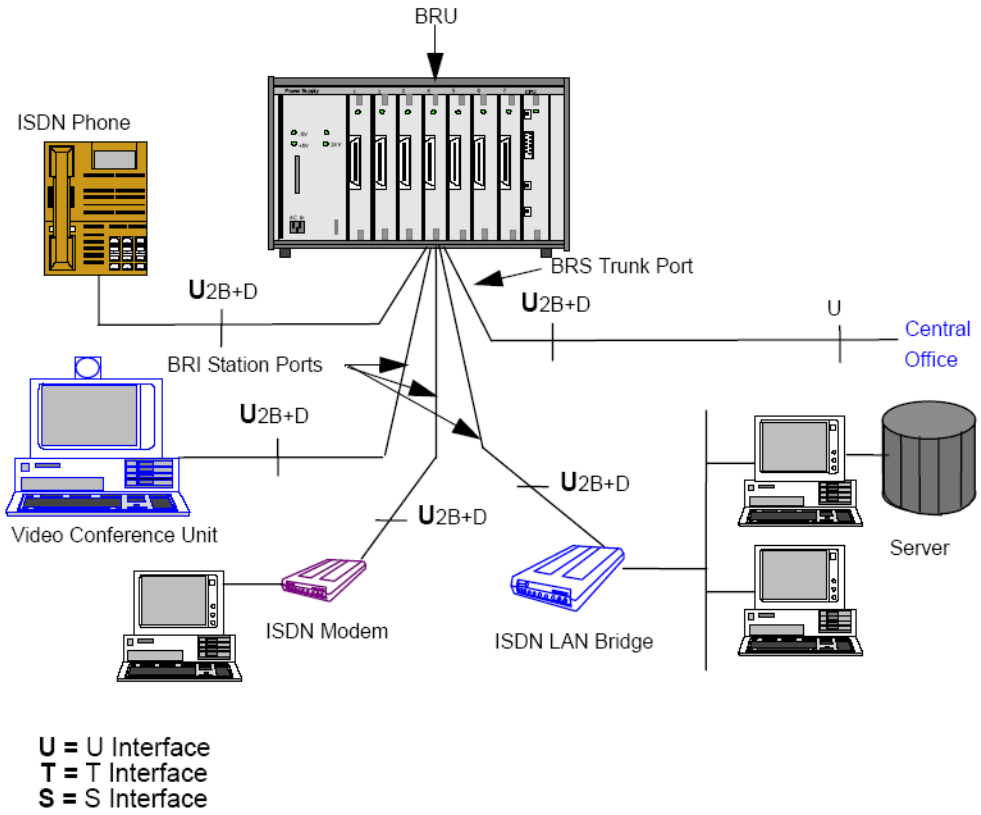
Basic Rate U Interface (BRU) Card

NOTE: BRU cards are not supported in Europe.

4.217 The Basic Rate U Interface card (BRU) allows the system to connect to North American ISDN basic rate trunks via U interface. The card supports eight ports. Each port consists of two bearer channels (B-channels), each operating at a rate of 64,000 bits per second, and one data channel (D-channel) operating at a rate of 16,000 bits per second.

4.218 BRU station ports can be connected to a data device (e.g., ISDN Modem or ISDN LAN Bridge), a video device (e.g., video conference unit), or an ISDN phone via U interface. BRU trunk ports can be connected to the central office via U interface. Note that each port has a built-in Network Terminator Type 1 (NT1) device allowing the port to be connected directly to the CO line. Most BRU station devices, such as video conference units, come with an S interface. However, some BRU station devices, such as ISDN modems, may have a built-in NT1 device and must be connected directly with a BRI station port (U interface).

Figure 2-6. *Sample BRU Installation*

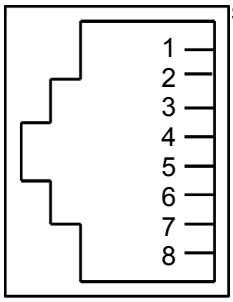


NOTE: All the BRI station ports, shown above, are U interface with a built-in NT1.

BRS and BRU Physical Characteristics

4.219 Both the BRS and BRU cards have the following characteristics, with the exception of the external power connector and jumpers, only present on the BRS card:

- **Amphenol-type Connector**: Each card has a 50-pin Ampheno1-type connector that allows the installer to connect all eight BRI ports.
- **LED Indicators**: Each card has five LEDs:
  - — **Card On-Line**: Indicates that the card is on line.
  - — **Circuit Busy**: Indicates that at least one of the 16 possible B-channel devices is busy.
  - — **Provide Reference**: Indicates that one of the BRI trunk ports on the card is providing the reference clock to the system.
  - — **Make-Busy Indicator**: Shows the current state of the Make-Busy switch. This LED has the same function as the Make-Busy LED on the T1/E1/PRI card.
  - — **Halted**: Indicates that the card is not functioning due to a hardware or software error.
- **Make-Busy Switch**: This switch allows the installer to busy out all BRI ports on the card. Pressing the switch lights the yellow Make-Busy LED and busy out all circuits that are not in use. The circuits that are in use are busied out as soon as the users hang up. When the circuits are all busied out, the LED flashes slowly, the Circuit-Busy LED is unlit. Now the card can be removed from the slot. If the card is not fully busied out and removed within a certain time (the default setting is one minute), the card returns to normal status and the Make-Busy LED flashes rapidly until the switch is released.
- **Serial Port Jack**: Each card has an RS232 port to enable the installer to perform diagnostics on the card and to upload new firmware. The pinout of the RS232 port is as follows:



	SIGNAL TYPE	SIGNAL DIRECTION
1	Not Used	
2	Not Used	
3	Ground	
4	TXD	Serial data input to the card
5	DTR	Handshake input to the card
6	RXD	Serial data output from the card
7	DSR	Handshake output from the card (always enabled)
8	DSR	Handshake output from the card

- **BRS External Power Connector**: This connector is present only on the BRS card. It allows an auxiliary power supply to be connected to the card. This allows the card to support devices which require line power via the BRS port. In order for the BRS card to provide line power, an Inter-Tel adapter cable (part number 813.1740, included with the BRS card) and an Inter-Tel -48VDC power supply – part number 550.0114 [PS 2743 in Europe] are connected to the card. See page 3-37 for details.

NOTE: With a European system, the BRS power supply is not connected to the BRS External Power Connector. Therefore, this is not recommended in Europe.

- **BRS Power Supply Jumpers**: Each of the eight BRS ports can be configured to receive power from the BRS card or from an external source by setting the port's jumper. If the device connected to a BRS port uses an external power supply, the jumper for that port must be set over the bottom four pins. In the default position, the jumper is set over the top four pins, which sets the port to use power from the BRS. (See page 3-37 for details on the BRS jumper settings.)

BRU and BRS Cards Cabling Requirements

4.220 Inter-Tel recommends that special high-speed data cable with the following characteristics be used for all BRI cards installations:

- 22 AWG (22 gauge thickness or larger -- smaller number gauge = larger thickness)
- Individually shielded two-pair (or with the transmit and receive pairs run in separate cables or binder groups)
- ABAM or PIC (a level-3 cable designed specifically for the T-carrier transmission rate)
- Maximum capacitance of 15pF/ft
- Maximum attenuation of 5dB/1000 ft (300 m) at 772 kHz and 70° F (21° C)

K. OPTIONS CARDS (OPCS)

4.221 Each Options Card (OPC) is equipped with the two ROMs, from kit part numbers 828.1561 [DI241 in Europe], that allow the card to work with version 2 or later system software. The part numbers of the two ROMs in the kit are 827.9949 (U3) and 841.0202 (U29).

4.222 Each OPC contains the following system resources (see Figure 3-81 on page 3-155):

- **DSPs**: (one standard, up to three more optional) Allocates Caller ID [CLIP] receivers (up to eight per DSP), Caller ID [CLIP] transmitters (up to six per DSP), shared speakerphones for Axxess Basic Digital, Eclipse Basic Digital, Associate Display, Model 8500, Model 8520, Model 8560, and Standard Display Phones (up to eight per DSP), additional conferencing circuits (up to 16 more per DSP), and additional DTMF receivers (up to six more per DSP).

  NOTE: For information on Caller ID [CLIP] features, see page 5-72. For information on shared speakerphones for Axxess Basic Digital, Eclipse Basic Digital, Associate Display, Model 8500, Model 8520, Model 8560, and Standard Display Phones, see page 5-125.

  With European systems, previously, only the DSP on the DKSC-16+ provided the Caller ID [CLIP] transmitters required for CLIP for single-line sets. As of version 8.224 (U.K. v8.225), each DSP on the OPC provides an additional six Caller ID [CLIP] transmitters. If necessary, you can upgrade the firmware on an existing OPC with the A-Law OPC Upgrade Kit (part no. 828.1671). See the *A-Law Options Card Firmware Replacement Instructions* for details.

  NOTE: This functionality was originally implemented for the U.S. (Mu-law) software in v8.1.

- **LED Indicators**: Indicates the card's functional readiness (CARD ON-LINE - green) and whether or not any resource on the card is busy (CIRCUIT BUSY - yellow). See TROUBLESHOOTING for detailed LED information.
- **DB15 PCM Connector**: (15-pin D-subminiature female connector) Connects an additional 4-32 ports (port numbers 9-40) to the Audio Interface Card (AIC) in the optional Voice Processing PC.

  NOTE: In tri-/quad-chassis installations where the third (slave) chassis is located away from the first (master) chassis, be sure to install the OPC in the master dual-chassis if it is used to provide additional Voice Processing channels.

- **RS232 Jack**: (8-pin, 8-conductor modular jack or line jack unit) Connects an SMDR/error message output device, a PC for programming the system database, or a modem for remote programming/maintenance (see page 2-102 for requirements). Do *not* use this port to connect to the Voice Processing PC.

- **Relay Jack**: (6-pin, 6-conductor modular jack or line jack unit) Currently used for night transfer only (e.g., to turn on or off lights, an alarm system, or other devices). The relay, which is activated when an attendant places the system in night mode, causes the swing connection to move from the "normally open" to the "normally closed" position. The OPC relay contact ratings are listed below. (See page 2-103 for additional requirements.)

Table 2-12. *OPC Relay Contact Ratings*

OPC Relay Contact Ratings
0.6A at 125VAC 0.6A at 110VDC 2.0A at 30VDC

- **Music Jack**: Connects a second external music-on-hold music source (cannot be used for phone background music).
- **Logic Inputs Connector**: Currently reserved for future use.

4.223 Each OPC must be equipped with two special software components (kit no. 828.1279) that allow the card to work with version 2 or later system software. If an earlier version of system software is ever re-installed, the two new components must be removed and the old components must be re-installed.

L. OS/2-BASED EMBEDDED VOICE MAIL CARD (EVMC)

4.224 The Embedded Voice Mail Card (EVMC) was recently modified. Except where noted, the old EVMC and the newer EVMC are identical.

4.225 The EVMC comes equipped with a single DSPM card to provide a four-port OS/2- or Linux-based voice mail unit on a single circuit card. It is typically installed in slot 7 of the equipment chassis and it derives power from the chassis backplane.

4.226 The EVMC can have an additional DSPM card to support an additional four voice mail ports. The 2.0 Gigabyte hard drive on the EVMC supports up to 180 hours of voice messages.

4.227 The PAL (U23) on the EVMC determines how many mailboxes it supports (32, 64, or 200 mailboxes). Standard mailboxes and networking mailboxes (i.e., AMIS mailboxes) are both included when determining the total number supported. The 32- and 64-mailbox PALs can be used only with the CPC64 or CPU64. The 200-mailbox PAL can be used with any CPC or CPU, if the system is running software version 5.270 or later. If you exceed the supported mailbox limit, users receive the "mailbox full" message when they try to leave a message in any mailbox. Currently, the 4-port EVMC is shipped with the 32-mailbox PAL.

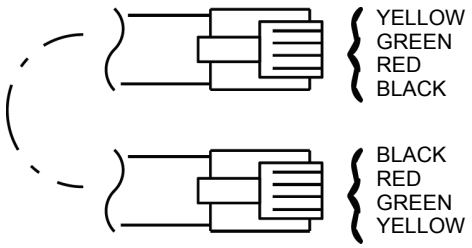
4.228 Timeslots for the EVMC are supplied through the AIC ribbon cable. For example, the Options Card (OPC) could possibly need more than 16 time slots, depending on the number of Digital Signal Processors (DSPs) installed on the card. The OPC timeslots do not decrease the number of timeslots or ports provided by the CPU64 configuration.

4.229 The EVMC supports AMIS networking and database saves to a zip or floppy drive. It does not support Fax-On-Demand, Unified Messaging, TTS, or ASR.

4.230 The parallel port on the EVMC can be used, like the Voice Processing system parallel port, for performing save and restore with a Zip drive. You can use an optional Zip Drive Backup Kit (828.1460) that contains an Iomega™ Zip Drive, Zip disks, a Save Boot floppy disk, and a Restore Boot floppy disk. To use this kit you also need an external floppy drive, which is available in the Floppy Drive Kit (828.1486).

4.231 There are connectors on the EVMC for a VGA monitor and a keyboard (there is a connector for a mouse, but it is not supported in this EVMC version).

4.232 The old EVMC has two RS232 serial ports, however, only the first RS232 port is operational at this time. The RS232 interface consists of a standard, two-pair, reversing, mod-to-mod line cord. A reversing line cord can be identified by examining the modular plug connectors side-by-side as shown below.



TWO-PAIR REVERSING LINE CORD

4.233 The newer EVMC has one RS232 serial port and one DB9 serial port. The RS232 interface functions the same as it does on the old EVMC. The DB9 serial port provides a connection for Inside Track (not applicable to European systems).

NOTE: If using the DB9 port, an SMDR buffer box is not required.

4.234 In addition, the newer EVMC no longer has the Reset button, which was for development purposes only and should not be used.

4.235 The environmental requirements of the EVMC are:

Table 2-13. *EVMC Environmental Requirements*

Requirements	In Operation	In Storage
Temperature Relative Humidity (Non-Condensing)	41° to 104° F (5° to 40° C) 8% to 90%	-4° to 140° F (-20° to 60° C) 8% to 90%

4.236 For more information about the CPU64 configuration and the EVMC, see Appendix D — CPC64/CPU64 Configuration.

M. LINUX-BASED EMBEDDED VOICE MAIL CARD (EVMC)

4.237 Version 8.224 [Europe v8.225] or later supports the EVMC with the Linux operating system instead of the IBM® OS/2® Warp system. The EVMC hardware has not changed; only the operating system on the card is different. Many of the operations that you performed on the older EVMCs are handled the same with the Linux version. For detailed information on the Linux-based EVMC, see the *Linux-Based EVMC Installation and Maintenance Instructions* (document part no. 835.2839).

NOTE: Although the hardware has a port labeled “Optional LAN” the EVMC currently does not have LAN support.

4.238 There are, however, a few differences with the Linux-based EVMC, as described below.

- Because Linux manages disk space in a way different from OS/2, you can use more of the drive to store your data and log files. You can store over 800 hours of voice mail data and have room for backups.

  Approximate disk space requirements for voice mail storage are as follows (note that the requirements are the same for OS/2 or Linux):

  — 1 second = 3 K
  — 1 minute = 180 K
  — 1 hour = 10.8 MB
  — A default mailbox is 30 minutes maximum = 5.4 MB (180 K x 30 minutes)
  — 64 mailboxes = 346 MB (5.4 MB per mailbox x 64 mailboxes)
  — 200 mailboxes = 1.1 GB (5.4 MB per mailbox x 200 mailboxes)

- Because drivers are not available for Linux, you cannot use a Zip drive for back ups. However, you can avoid using multiple floppy disks by using some of the additional space on the drive.

- In the event that the AVDAP services crashes, Linux automatically generates files with the extension “.bt.” These files can be used by Inter-Tel Technical Support to troubleshoot the cause(s) of the crash. Information about the crash is contained in files named **`alarms.bt`** and **`exception.bt`**.

- The PALs are implemented differently with the Linux-based EVMC, as follows:
  - — With an Axxess 64 and no PAL or a 64 mailbox PAL, the EVMC defaults to 64 mailboxes and no limit on messages.
  - — With an Axxess 128 or 256 and no PAL or a 64 mailbox PAL, the EVMC defaults to 64 mailboxes with a limit of one message of one minute length.
  - — With a 200 mailbox PAL, the EVMC has 200 mailboxes and no limit on messages.

NOTE: The 32 mailbox PAL is not supported by the Linux-based EVMC. If you have a CPC-64 and remove the 32 mailbox PAL, you still have 64 mailboxes.

ALSO: The behavior of the OS/2-based EVMC does not change with respect to the PALs.

5. STATION INSTRUMENTS

5.1 A variety of station instruments can be used on the Inter-Tel system. The combination of station instruments depends on the number and type of station cards installed.

5.2 At least one display phone should be installed to provide an administrator the capability to receive system alarm messages, program system data, and receive unsupervised CO recalls. When the system is in the default configuration, circuit 01.01.01 is designated as a phone, an administrator, and the primary attendant.

NOTE: Because most systems is installed with a variety of station and trunk cards and a variety of station instruments, the maximum station and trunk capacities listed in this manual is rarely reached.

A. DIGITAL PHONES

5.3 The following digital phone models can be installed on the Inter-Tel system:

- Axxess Basic Digital Phones
- Eclipse Basic Digital Phones
- Standard Display Phones
- Associate Display Phones
- Executive Display Phones
- Professional Display Phones
- Models 8500, 8520, and 8560

NOTE: Eclipse Basic Digital, Associate Display, and Professional Display Phones are not supported in Europe.

NOTICE

These are proprietary phones and are to be used **only** with Inter-Tel systems. Also, version 8.0 or later software is required to support the Eclipse Basic Digital, Professional Display, and Associate Display Phones.

5.4 For illustrations of these phones, see the drawings beginning on page 2-107. The feature buttons, speed-dial buttons, call buttons, and station programmable buttons shown in the drawings are as they appear when the system is in the default configuration.

5.5 See page 3-178 for phone installation instructions.

5.6 The *approximate* dimensions of the Axxess Basic, Standard Display, and Executive Display Phones are:

Height	3.0 in (7.6 cm)
Width	7.5 in (19.1 cm)
Length	9.0 in (22.9 cm)
Weight	2.3 lb (1.0 kg)

5.7 The *approximate* dimensions of the Eclipse Basic, Associate Display, and Professional Display Phones are:

Height	3.25 in (8.6 cm)
Width	9.25 in (23.5 cm)
Length	7.75 in (19.7 cm)
Weight	2.09 lb (.95 kg)

5.8 The *approximate* dimensions of the Model 8662, 8622, 8620, 8660, 8560, and 8520 are:

Height	2.5 in (6.4 cm)
Width	8.8 in (22.3 cm)
Length	8.3 in (21.1 cm)
Weight	2.7 lb (1.2 kg)

5.9 The *approximate* dimensions of the Model 8500 and 8600 are:

Height	2.5 in (6.4 cm)
Width	6.4 in (16.3 cm)
Length	8.3 in (21.1 cm)
Weight	2.1 lb (0.9 kg)

5.10 Digital phones have the following design features in their default configuration:

- 12-button dialpad
- A variety of feature buttons (some of which may be designated as user-programmable)
- Four call buttons with LED indicators for incoming call access and indication
- Integrated speakerphone capability (optional on Axxess Basic Digital, Eclipse Basic Digital, Standard Display, Associate Display, Model 8500, Model 8520, and Model 8560 Phones)
- Ring and voice volume controls (using the VOLUME button)
- Slide-out directory card (Executive Display, Professional Display, Standard Display, Associate Display, Axxess Basic Digital, and Eclipse Basic Digital Phones only)
- Self-test feature (for testing phone functions)
- Hearing aid-compatible (HAC) electret handset
- Reversible baseplate for wall mounting (Executive Display, Professional Display, Standard Display, Associate Display, Axxess Basic Digital, and Eclipse Basic Digital Phones only)
- Adjustable baseplate that allows you to change the tilt angle of the phone (Model 8500, 8520, and 8560 Phones only)
- Dedicated headset jack (Model 8520 and 8560 Phones only)
- LCD contrast adjustment control (Model 8520 and 8560 Phones only)
- Programmable message lamp (Model 8500, 8520, and 8560 Phones only)

Integrated Speakerphone

5.11 All Executive Display and Professional Display Phones have a built-in, integrated, speakerphone, which can be used in either standard or enhanced mode. (See page 5-125 for more information.)

5.12 All Axxess Basic, Eclipse Basic, Standard Display, Associate Display, Model 8560, Model 8520, and Model 8500 Phones have a speaker and a microphone for handsfree intercom reply, but they do not have a true integrated speakerphone. However, the DSPs on the Options Card can be configured to provide shared speakerphone circuitry (in either standard or enhanced mode) for Axxess Basic, Eclipse Basic, Standard Display, Associate Display, Model 8520, and Model 8500 Phones. In addition, each DKSC-16/DKSC-16+ has a DSP for providing up to 10 speakerphones for Axxess Basic, Eclipse Basic, Standard Display, Associate Display, Model 8560, Model 8520, and Model 8500 Phones. (See pages 5-122 and 5-125 for more information.)

NOTE: If desired, speakerphone capability can be disabled on a station-by-station basis through DB Programming. (See page 6-211.)

5.13 The integrated speakerphone allows users to place and receive outside calls and intercom calls without lifting the handset. After a call is connected, the phone user may speak handsfree over the speakerphone or lift the handset to speak privately. When using the handset, the user may switch to the speakerphone by pressing the Speaker button and replacing the handset. If enabled, pages and/or background music may be broadcast over the speaker when the phone is not being used.

Liquid Crystal Display (LCD)

5.14 All Executive Display, Professional Display, Standard Display, Associate Display, Model 8560, and Model 8520 Phones and most Executone phones have an LCD with two 16-character lines at the top. The display helps the user to process calls more efficiently and reliably. There are display messages for date and time, numbers dialed, call cost data, reminders, do-not-disturb, forwarding, voice mail processing, incoming calls, station and system status, system alarms, user prompting, and programming. These display messages are indicated throughout the FEATURES section of the manual.

NOTE: The LCD contrast is adjustable for the Models 8662, 8622, 8620, 8660, 8560, and 8520 endpoints (see page 5-130 for details).

5.15 Executone Model 160, Model 64, and 29-Key Wave Phones and Executive Display, Professional Display, and Model 8560 Phones have additional display lines that are associated with the menu selection buttons that border them. The menu display changes to reflect the functions of the menu buttons. Pressing the button next to the displayed option selects that option. (When there is only one option on a line, pressing the button on either end of the line selects that option.)

NOTE: Version 7.0 or later system software is required to support the menu display and buttons on the Model 8560.

5.16 Axxess Basic Digital, Eclipse Basic Digital, and Model 8500 Phones cannot be equipped with LCDs.

Message Lamp

5.17 Model 8500, 8520, and 8560 Phones are equipped with a Message Lamp, located in the upper-right corner of the phones. By default, this lamp is programmed to indicate when a message is waiting. You can, however, program this lamp for a different feature using the keymap in DB Programming (see page 6-308 for details). If you change the feature associated with this lamp, the lamp flashes when the feature is active. For example, if you program the lamp for Hold, the lamp flashes whenever a call is held at the station.

NOTE: Version 7.0 or later system software is required to support the Message Lamp feature for the Model 8500, 8520, and 8560 Phones.

Electret Circuitry

5.18 Inter-Tel phones and handsets have been designed using "electret" microphone circuitry, rather than the "dynamic" microphone circuitry used on many previous Inter-Tel phone models. Electret phones and handsets are not electrically compatible with dynamic phones and handsets.

5.19 If a dynamic handset is attached to an electret phone, the handset transmit levels are noticeably lower. If an electret handset is attached to a dynamic phone, the handset does not transmit. (In both cases, neither the phone nor the handset is damaged.)

5.20 Electret phones can be identified by the manufacturer's label on the bottom of the phone. Electret handsets can be identified by the "E1" embossed in the recess for the wall-mount hook (near the receiver).

Optional Headsets

5.21 For the Executive Display, Professional Display, Standard Display, Associate Display, Axxess Basic Digital, Eclipse Basic Digital, and Model 8500 Phones, a headset may be attached to any phone by unplugging the handset from the handset jack on the base of the phone, plugging the headset into the handset jack, and entering a feature code to enable the headset. The Speaker button, which is used to turn the headset on and off, is lit when placing and receiving calls and unlit when the headset is not in use. The phones are compatible with industry-standard, four-conductor, modular headsets that have electret microphones. See page 3-188 for installation instructions.

5.22 The Model 8520 and 8560 Phones have a dedicated headset jack on the back of the phone. This allows the phone to have a headset and handset plugged in simultaneously. The headset, however, must still be enabled using the feature code. Like the Executive Display, Professional Display, Standard Display, Associate Display, Axxess Basic Digital, and Eclipse Basic Digital Phones, the Speaker button on the Model 8520 and 8560 Phones is used to turn the headset on and off and to indicate when the headset is not in use. In addition, the hookswitch is enabled even when the headset is turned on. This allows the user to answer a call with the handset without having to disable the headset.

NOTE: Version 7.0 or later software is required to use the dedicated headset jack on the Model 8520 and 8560 Phones. If using any of these phones on a system that is running software versions 3.0-6.x, you must plug the headset into the handset jack.

Optional Personal Computer Data Port Module (PCDPM)

5.23 For off-hook voice announce receive capability, and to allow an optional DSS/BLF Unit and/or an optional Modem Data Port Module to be installed, any Standard Display, Associate Display, Executive Display, Professional Display, Model 8520, or Model 8560 Phone may be equipped with an optional PC Data Port Module. For the Executive Display and Standard Display Phones, PCDPM – part no. 550.3018 or 550.3014 [PB2585 or PB2549-A in Europe] or is required. For the Model 8520 and 8560 Phones, the PCDPM required is part no. 550.8400. For the Professional Display and Associate Display Phones, the PCDM requires is part no. 560.3018 is required.

5.24 The original model PCDPM (part no. 550.3014) for the Executive Display and Standard Display Phones was replaced with a new model (part no. 550.3018). The new PCDPM (part no. 550.3018), the Model 8520/8560 PCDPM (part no. 550.8400), and the Professional Display and Associate Display PCDM (part no. 560.3018) do not require an external AC transformer (see Figure 2-7 on page 2-78 for a diagram). Refer to the revised installation instructions included with the new unit. If a DSS/BLF Unit is not installed on the module's RS232 port, the port can be used to connect an SMDR/error message output device or a personal computer. See page 3-188 for instructions on installing the optional PC Data Port Module.

NOTE: Older Axxess PCDPMs – part no. 550.3014 [PB2549-A in Europe] require AC transformers when they are connected to DSS/BLF Units only or PC/output devices only. PCDPMs do *not* require AC transformers when they are used for off-hook voice announce only or have attached MDPMs.

5.25 If the PCDPM uses an RS232 connection (e.g., when using system serial port or desktop OAI applications), see Appendix B — RS232 to ensure a successful RS232 connection.

Optional Modem Data Port Module (MDPM)

5.26 Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phones equipped with PC Data Port Modules may also be equipped with optional Modem Data Port Modules – part no. 550.3015 [PB2549-A in Europe]. The MDPM contains a jack that can be used to connect a data device (such as a personal computer equipped with a modem) or a single-line set to the Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone. Each MDPM is powered by its own AC transformer. The MDPM – part no. 550.3015 [PB2549-A in Europe] includes an AC transformer; it no longer needs to be ordered separately.

NOTE: MDPMs always require their own separate AC transformers. (For a diagram of AC transformer requirements, see Figure 2-7 on page 2-78.)

5.27 The data device attached to the MDPM must have a direct-connection modem. The modem must be externally powered (or capable of operating on 20mA of loop current) and have an RJ11 CO interface.

NOTE: With a European system, the modem must be externally powered (or capable of operating on 20mA of loop current) and have a BS 6312 plug. The MDPM has a socket, but is supplied with an RJ11 to BS 6312 adaptor. The wiring of the RJ11s can vary; therefore, Inter-Tel does not support installers who do not use this adaptor.

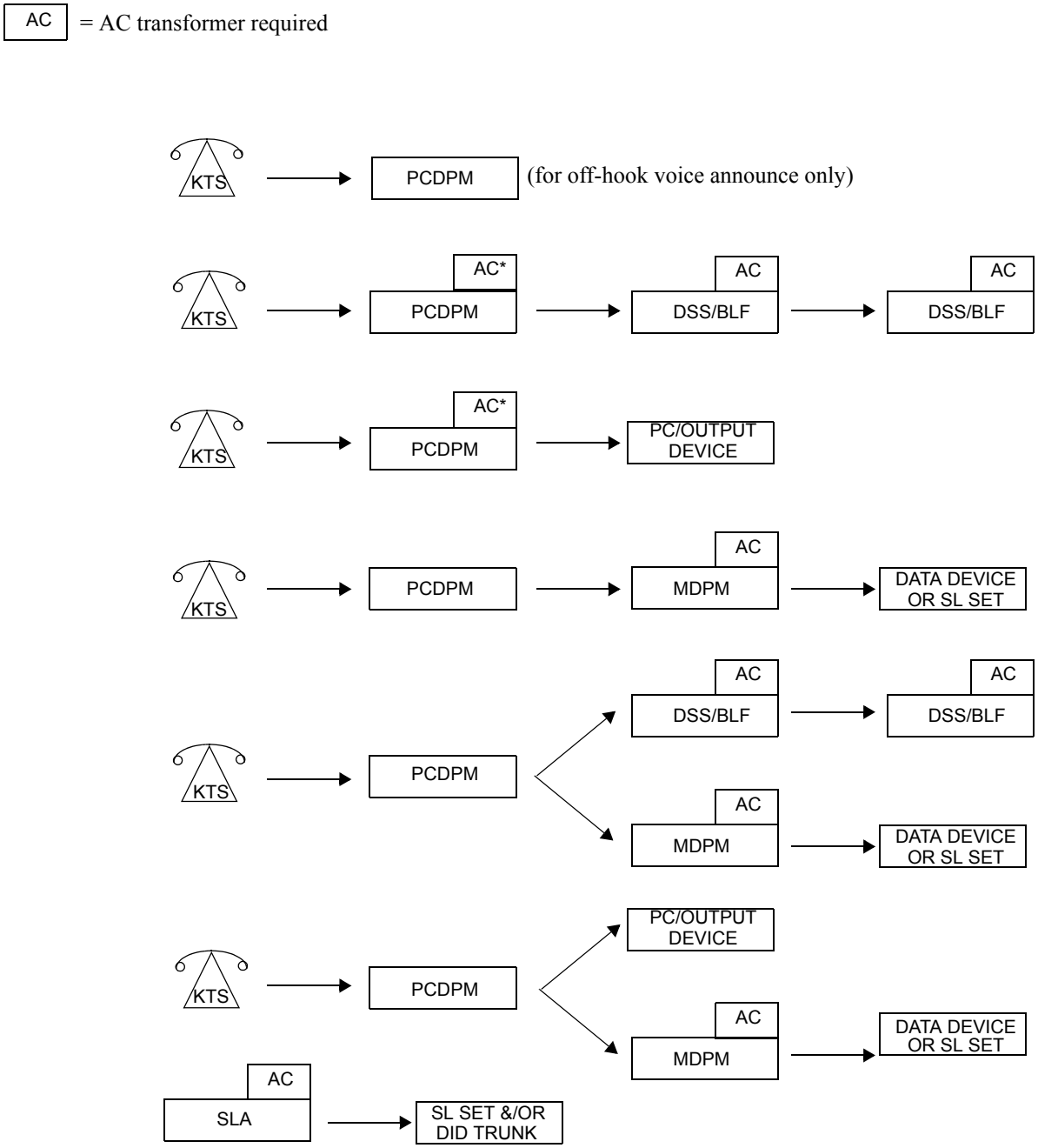
5.28 The data device can be used by the associated Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone(s) to communicate with remote data equipment over voice channels being used for CO [local exchange] or intercom calls. See pages 3-201 and 5-269 for installation and operation instructions.

NOTE: Each “equipped” MDPM installed counts as one single-line device.

5.29 The MDPM ring signal waveform has the following characteristics:

- 20 Hz
- 200 volts peak-to-peak
- 48% duty cycle
- No DC offset (superimposed on circuit ground)

Figure 2-7. *AC Transformer Requirements*



* The AC transformer is only required if using PCDPM part no. 550.3014. See Note below.

NOTE: DSS/BLF Units, MDPMs, and SLAs always require their own separate AC transformers. Older Axxess PCDPMs (part no. 550.3014) require AC transformers when they are connected to DSS/BLF Units or PC/output devices only. PCDPMs do not require AC transformers when they are used for off-hook voice announce only or have attached MDPMs.

B. IMX ANALOG PHONES

5.30 Refer to a version 7.0 or earlier *Axxess Installation and Maintenance Manual* for IMX analog phone installation instructions. The following IMX analog models are available:

- 24-Line AIM Phones (display only)
- 12-Line AIM Phones (display only)
- 8-Line Dual-Circuit AIM Phones (display only)
- 24-Line Phones (standard and display)
- 12-Line Phones (standard and display)
- 8-Line Phones (non-display only)

5.31 For illustrations of the phones, dimensions, features, etc. refer to a version 7.0 or earlier *Axxess Installation and Maintenance Manual*.

C. GMX ANALOG PHONES

NOTE: GMX analog phones are not supported in Europe.

5.32 Refer to a version 7.0 or earlier *Axxess Installation and Maintenance Manual* for GMX analog phone installation instructions. The following Inter-Tel GMX phones may be used on the Inter-Tel system:

- 24-Line Standard GMX Phones
- 24-Line Display GMX Phones
- 12-Line GMX Phones (non-display only)

5.33 Refer to a version 7.0 or earlier *Axxess Installation and Maintenance Manual* for illustrations, dimensions, features, etc.

D. EXECUTONE IDS PHONES

NOTE: Executone IDS phones are not supported in Europe.

5.34 The following Executone IDS keyset models can be installed on the Inter-Tel system:

- Model 64, 32, 18, and 12 Keysets
- Model 160 and Agent Terminal
- 48-Key DSS Console (up to three per keyset, total of 80 per system)
- 6-, 17-, and 29-Key Wave Phone

NOTE: The 28-Key Wave Phone is not supported.

5.35 Refer to a version 7.0 or earlier *Eclipse Installation and Maintenance Manual* for illustrations, dimensions, features, etc.

E. INTER-TEL/DVK ANALOG PHONES

NOTE: Inter-Tel/DVK analog phones are not supported in Europe.

5.36 Refer to a version 7.0 or earlier *Installation and Maintenance Manual* for Inter-Tel/DVK analog phone installation instructions. The following Inter-Tel/DVK analog models are available:

- Inter-Tel/DVK 24-Line Phones (standard and display)
- Inter-Tel/DVK 12-Line Phones (standard and display)
- Inter-Tel/DVK 8-Line Phones (standard and display)

5.37 For illustrations of the phones, dimensions, features, etc. refer to a version 7.0 or earlier *Installation and Maintenance Manual*.

F. DIGITAL DIRECT STATION SELECTION/BUSY LAMP FIELD (DSS/BLF) UNITS

5.38 The Inter-Tel system supports three types of DSS/BLF Consoles/Units. The unit used depends on the phone.

- **With Executone IDS Phones**: IDS phones can be equipped with up to three 48-key DSS Consoles. There can be a total of 80 IDS 48-key DSS Consoles on the system. Each 48-key DSS Console requires an EKSC circuit.

  NOTE: For details about the 48-key DSS/BLF Unit, see a version 7.0 or earlier *Eclipse Installation and Maintenance Manual.*

- **With Executive Display, Standard Display, Professional Display, and Associate Display Phones**: PCDPM-equipped Executive Display, Professional Display, Standard Display, and Associate Display Phones may be connected to up to four 60-key DSS/BLF Units. There can a total of 20 of these 60-key DSS/BLF Consoles on the system. Because each unit is powered by an AC transformer and is connected to the phone's PCDPM, the 60-key DSS/BLF Consoles do not require a DKSC or DKSC-16/DKSC-16+ circuit that is separate from the phone. There is a 6-row model that is designed for Executive Display and Standard Display Phones and a 12-row model that is designed for Professional Display and Associate Display Phones
- **With Model 8520 and Model 8560 Phones**: PCDPM-equipped Model 8520 and Model 8560 Phones may be connected to up to four 50-key DSS/BLF Units. There can a total of 20 of these 50-key DSS/BLF Units on the system. Because each unit is powered by an AC transformer and is connected to the phone's PCDPM, the 50-key DSS/BLF Consoles do not require a DKSC or DKSC-16/DKSC-16+ circuit that is separate from the phone

5.39 There are two types of DSS/BLF Units available. One is a 60-key model and the other is a 50-key model. Although the 60-key, 6-row model is designed for Executive Display and Standard Display Phones and the 50-key model is designed for Model 8520 or Model 8560 Phones, you can use either DSS/BLF Unit on any of these phones. Each single unit provides one-button access to 50 or 60 extension numbers and/or other features/resources, while four units provide access to 200 or 240 extension numbers and/or other features/resources. (See page 3-202 for DSS/BLF Unit installation instructions.)

NOTE: The presence of a Mini-DSS does not affect the number of DSS/BLF Units that you can attach to a phone.

5.40 DSS/BLF Units are physically attached to the RS232 ports on Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phones equipped with PC Data Port Modules (PCDPMs). Although the PCDPMs for the Model 8520 and 8560 Phones are different from the PCDPMs for the Executive Display, Professional Display, Associate Display, and Standard Display Phones, the DSS/BLF Units work with either PCDPM. For example, you can attach a 60-key DSS/BLF Unit to a Model 8520 or 8560 Phone.

5.41 The DSS/BLF Units are powered by their own AC transformers and do not require a DKSC or DKSC-16 circuit that is separate from the phone's circuit. If desired, all 20 units can be connected to Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phones installed on the same Digital Keyset Card.

NOTE: DSS/BLF Units always require their own separate AC transformers. For a diagram of AC transformer requirements, see Figure 2-7 on page 2-78.

5.42 For an illustration of the DSS/BLF Unit (and other station instruments), see the drawings beginning on page 2-107. The DSS/BLF Unit dimensions are listed in the following table.

Table 2-14. *DSS/BLF Unit Dimensions*

	60-KEY DSS/BLF	50-KEY DSS/BLF
Height	3.0 in (7.6 cm)	2.5 in (6.4 cm)
Width	7.5 in (19.1 cm)	6.4 in (16.3 cm)
Length	9.0 in (22.9 cm)	8.3 in (21.2 cm)
Weight	1.7 lb (0.8 kg)	1.2 lb (0.6 kg)

5.43 As stated earlier, each single DSS/BLF Unit provides one-button access to up to 50 or 60 extension numbers, trunks, features, or other resources. The numbers programmed in the buttons can be station extension numbers, hunt group numbers, trunk access codes, or feature codes. The layout of the buttons and the numbers they access are assigned using one of up to 20 keymaps in DB Programming. The intercom number/button assignments of each keymap are programmed in four lists, pages 1-4. For example, single DSS/BLF Units are assigned page 1 of the desired keymap, tandem units are assigned two pages of the desired keymap (page 1 for the first unit and page 2 for the second unit), and so on.

5.44 Together, the lamps in the buttons create a busy lamp field that indicates the status of each station, hunt group, or other feature/resource assigned to the buttons. If assigned to a station, the LED indicator in the button is solidly lit when the associated station is busy or unplugged, flashes slowly when the station is in do-not-disturb, flashes rapidly when the station has a call ringing in, or flutters continuously if the station is causing a STATION OFF-HOOK system alarm. If assigned to a hunt group, the LED indicator is solidly lit when all stations in the hunt group are unavailable (busy, in do-not-disturb, or removed from the hunt group), and it rapidly flashes when a call is camped on to a hunt group. If assigned to a feature or resource, the LED indicator is lit to show busy status and unlit to show idle status.

NOTE: Group Call Pick-up does not affect DSS lamp status any differently than standard hunt group calls or individual calls.

G. MINI-DIRECT STATION SELECTION (DSS) UNITS

5.45 One Mini-DSS Unit can be installed on every Model 8520 and/or Model 8560 Phone in the system. Because the Mini-DSS does not utilize any of the resources required by a DSS/BLF, a phone can have one Mini-DSS and up to four DSS/BLF Units.

NOTE: Version 7.0 or later software is required to use the Mini-DSS.

5.46 Each single unit provides one-button access to 16 extension numbers and/or other features/resources. (See page 3-206 for Mini-DSS Unit installation instructions.)

5.47 Mini-DSS Units are physically attached to a special cable pin connector on Model 8520 or Model 8560 Phones. Because the Mini-DSS Units draw power from the phone, external AC transformers are not required. In addition, separate circuits on the DKSC-16 or DKSC-16+ are not required.

5.48 For an illustration of the Mini-DSS Unit (and other station instruments), see the drawings beginning on page 2-107. Mini-DSS Unit dimensions are:

Table 2-15. *Mini-DSS Unit Dimensions*

Height	1.5 in (3.8 cm)
Width	2.6 in (6.5 cm)
Length	7.7 in (21.1 cm)
Weight	2.1 lb (0.13 kg)

5.49 As stated earlier, each single DSS/BLF Unit provides one-button access to up to 16 extension numbers, trunks, features, or other resources. The numbers programmed in the buttons can be station extension numbers, modem extensions, hunt group numbers, trunk access codes, or feature codes. The layout of the buttons and the numbers they access are assigned using the Mini-DSS keymap in DB Programming. The intercom number/button assignments of each keymap are programmed in the Model 8662/8660/8620/8560/8520 keymap.

5.50 The lamps in the buttons on the Mini-DSS function the same as they do on a DSS/BLF Unit.

NOTE: Group Call Pick-up does not affect DSS lamp status any differently than standard hunt group calls or individual calls.

H. DIGITAL ATTENDANT CONSOLES

NOTE: Analog Attendant Computer Consoles cannot be installed on the system.

5.51 The Attendant Console is a special software program and hardware security key that allows attendants to use their personal computers (PCs) to answer and manage incoming call traffic. It is primarily designed for use by attendants of larger phone systems – those systems with so many stations that using DSS/BLF Units is not practical. However, due to its enhanced functionality, including electronic text messaging and directory dialing, the Attendant Console is an attractive option for smaller systems as well.

5.52 The Attendant Console PC, which requires a System Level 2 OAI interface, is connected to the system through a TCP/IP connection. for more details refer to the *Attendant Console User Guide* supplied with the console software.

NOTE: Version 7.0 or later software does not support Attendant Console version 2.0. If running version 7.0 or later, Attendant Console version 3.0 is required.

I. AGENTSET

5.53 The AgentSet is used in place of a phone with an attached PCDPM. It is connected to a Digital Keyset Card (DKSC or DKSC-16/DKSC-16+) port just as a phone would be. It has a jack for plugging in an Inter-Tel handset or headset. It also has a serial connector (DB9) for attaching it to a PC.

5.54 To operate properly, the AgentSet ***must*** be programmed for headset mode. This can be done in one of the following ways:

- Enable Headset Mode via Station Programming in the system database.
- At the AgentSet, unplug the headset or handset (to take the AgentSet off hook) and enter the Headset On/Off feature code (317) to enable headset mode. Then plug in the headset or handset.

NOTE: There is nothing to prevent the user from disabling headset mode using the Headset On/Off feature code at the station.

5.55 Because the AgentSet does not have feature buttons, a dialing pad, or a speaker, it must be connected to a PC that is equipped with an OAI product that emulates a phone. The OAI product is used for dialing and feature access.

5.56 The AgentSet also requires an Inter-Tel headset or handset. Only Inter-Tel equipment should be used. Other manufacturer's equipment may not be compatible with the AgentSet.

5.57 If the user wants to receive audible signals at the headset, there is a station flag that sends a single tone to the headset when a call is ringing.

J. IP DEVICES

5.58 The Internet Protocol (IP) PhonePlus, Model 8660, IP SoftPhone, multi-protocol endpoints, Model 8602 IP softphone, and IP Single-Line Adapter (SLA) provide the functionality that allows you to make phone calls through the IP network to the IP Card (IPC) or IP Resource Card (IPRC) on the telephone system.

5.59 Most IP endpoints resemble existing endpoints that use the TDM highway. For example, the Axxess IP PhonePlus looks like an Executive Display Phone, Models 8660 and 8662 look like the Model 8560, the Model 8620 looks like the Model 8520, etc. The Model 8690, however, is a special display phone that uses Windows CE as the interface. The IP devices operate the same as their counterparts installed directly on the system, except for the following:

- They cannot use a PCDPM or MDPM.
- They do not support Desktop OAI applications.
- They do not support the "enhanced speakerphone mode" feature.

5.60 The IP SLA operates like a standard Inter-Tel single-line set.

CAUTION

IMPORTANT NOTE FOR ADMINISTRATORS FOR EMERGENCY (911/999) CALLS: Unless programmed for emergency outgoing access using a local loop (see the latest version of the *IP Devices Installation Manual*, document part no. 835.2195, for details), when an IP endpoint such as the model 8602 IP softphone, dials the emergency number, this includes 911 (default for U.S. systems), 999 (default for European systems), or the appropriate emergency dial sequence for the locality, from a remote location, the call is placed from where the phone system is physically located. Because these emergency services use Caller ID [CLIP] to help locate the caller, emergency service could be misdirected or delayed. Also, the emergency service contacted may be local to the phone system, but not to the IP device. All IP device users should be alerted to this situation and instructed to use a local telephone line for placing emergency calls. Also note that the IP device may not function in the event of a power or network failure at either the local site or the phone system location unless operational backup power provisions have been implemented by the installing company.

5.61 For more information about the IPC, IPRC, and IP devices, refer to the *IP Devices Installation Manual* (version 1.3 or later).

K. MULTI-PROTOCOL ENDPOINTS

5.62 System software versions 8.1 and later support multi-protocol endpoints. These endpoints support either Session Initiation Protocol (SIP) or Inter-Tel Protocol (ITP). Endpoints that can operate in either mode include the following:

- **Models 8600, 8620/8622, and 8662**: Look and operate like Models 8500, 8520, and 8560, respectively. New Models 8622 and 8662 endpoints are now available. They are Class 2 Power over Ethernet (PoE) devices. The new endpoints replace the existing endpoints. Table 2-16 shows the existing and new part numbers.

Table 2-16. *Models 8622 and 8662 Part Numbers*

Model	Existing Part Number	New Part Number
Model 8622	550.8622	550.8622P
Model 8662	550.8662E	550.8662P

- **Model 8690**: Has a touch screen that displays feature, dialpad, and menu buttons.

5.63 These endpoints operate differently if using SIP versus IPT mode. For details about the available features and how to program these phones to use SIP, see the *SIP Server Installation and Configuration Manual* (document part no. 835.2605).

NOTE: To use IP endpoints in SIP mode, SIP Server v1.1 or later is required.

5.64 In addition, the software license regulates the number of multi-protocol endpoints that are available on the system.

L. MISCELLANEOUS ENDPOINTS

5.65 Versions 8.1 and later also support the following endpoints:

- **Model 8601 SoftPhone for Pocket PC**: Is a software-based SIP softphone that runs on selected Pocket PC 2002 (or later) Personal Digital Assistant (PDA) platforms. For information about installation and programming, refer to the *Model 8601 Installation Manual* (document part no. 835.2736).
- **Models 8664, 8665, 8668**: Are wireless, compact telephones that use ITP. For details, refer to the *SVP Server and Model 8664/8665/8668 Installation Manual* (document part no. 935.4521).

NOTICE
This manual does not provide the instructions necessary to install and program these phones. If installing these phones, please see the appropriate manual previously specified.

M. SINGLE-LINE SETS

5.66 Up to 512 industry-standard, AC-ringing, single-line DTMF or pulse-tone sets may be installed on the Inter-Tel system. (Version 2.x or later software is required for pulse-tone sets.) The single-line sets are connected to Single-Line Cards (up to eight per card), 16-Port Single-Line Cards (up to 16 per card), or to Single-Line Adapters (up to two per adapter) that are connected to available DKSC or DKSC-16/DKSC-16+ circuits. A single-line set may also be connected to any MDPM-equipped Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone.

NOTE: DC-ringing single-line devices may not be connected to SLCs or SLAs.

5.67 Single-line set users access some station features simply by lifting the handset and entering a feature code. Other features are accessed using a combination of a hookflash [recall in Europe] and a feature code. See the FEATURES section of this manual for details.

5.68 Each SLC can be equipped with an optional Lamp Option Card (LOC) to provide visual message waiting capabilities for single-line sets equipped with message lamps. SLAs do not support *visual* message waiting applications. Single-line sets, whether connected to SLCs or SLAs, are then configured in DB Programming to receive *visual* and/or *audible* message waiting indication tones.

5.69 **Optional Handset Amplifiers**: The typical handset amplifier is an external device that plugs into the single-line set (where the handset is normally connected); the handset is then plugged into the amplifier. Receiver voice volume is controlled by turning a thumbwheel (or similar control) located on the amplifier. Such amplifiers are generally equipped with a transformer that requires an AC outlet. See page 3-209 for installation instructions.

N. OFF-PREMISES EXTENSIONS (OPX)

5.70 An OPX is an AC ringer-equipped single-line DTMF set that is placed in a remote location and connected to the system through a telephone company [network provider in Europe] OPX circuit or a customer-provided circuit.

5.71 The circuits are connected using Single-Line Adapter (SLA) circuits, which are connected to available Digital Keyset Card (DKSC) and/or 16-Port Digital Keyset Card (DKSC-16) circuits. OPX circuits can also be connected using T1/E1 circuits, which are connected to available T1, E1, T1/PRI or E1/PRI card circuits.

NOTE: OPXs do not support *visual* message waiting applications. If single-line sets with message waiting lamps are installed as off-premises extensions, the lamps do not function. However, single-line sets can be configured in DB Programming to receive *audible* message waiting indication tones.

5.72 With a U.S. system, the OPX circuits provided by the telephone company are identified with Facility Interface Codes (FIC): Class A, OL13A; Class B, OL13B; or Class C, OL13C. The service order code is 9.0F (fully protected private circuit). If the impedance does not exceed a loop measurement of 800 ohms (loop of 15,600 feet (52 km) using 24AWG (0.6 mm) wire), a customer-provided circuit between the remote location and the system may be used.

5.73 With a Europe system, the OPX circuits are either provided by the network provider, or, if the impedance does not exceed a loop measurement of 800 ohms (loop of 15,600 feet/4754 m. using 0.5 mm. wire), a customer-provided circuit between the remote location and the system may be used.

5.74 The 800-ohm loop limitation is usually suitable for accessing on-premises stations from off-premises stations. However, for applications where one off-premises station is calling another through the station card, the impedance between the two off-premises stations (both circuits added together) must be less than 800 ohms (loop measurement). With a U.S. system, a Class A or B FIC circuit is preferable.

5.75 When an SLA circuit is designated for OPX use in DB Programming (see page 6-118), its voice volume levels are increased approximately 3dB over normal single-line set voice volume levels. This is an attempt to compensate for any loss in the external OPX circuit. However, in certain off-premises applications, the voice volume levels may still be unacceptable. This degradation in voice volume is due to the natural voice frequency range limitations of the telephone company [network provider in Europe] or customer-provided circuit. To increase the frequency range, installation of a two-wire, negative impedance repeater unit is recommended. Such units typically provide from 0-15 decibels of voice volume gain and allow regulation of the gain in each direction when simultaneous voice transmission occurs. The switching sensitivity on the unit should also be adjustable.

5.76 The repeater unit is installed at the MDF between the station block and the cabling to the off-premises station. (See page 3-69 for OPX installation instructions.) When ordering a repeater unit, consult with the supplier for ordering the proper mounting shelf and power supply for the unit. Additional information on operating and adjusting the repeater unit is generally included with the product.

NOTE: Each OPX circuit can have extended ring cadence enabled or disabled (see page 6-211). The extended setting lengthens the duration of the ring signal to meet the requirements of many OPX and/or repeater applications. The default setting is *disabled*.

O. PLAYBACK DEVICES

NOTE: For automated attendant features and for optimal playback device performance and functionality, use the Voice Processing PC and applications software instead of SLA circuits.

5.77 AC-ringing playback devices may be connected to SLCs (up to eight per card) and/or SLC-16s (up to 16 per card) or to SLAs (up to two per adapter). (See page 3-208 for playback device installation instructions.) A playback device is an answering machine that answers the call, plays a message, and then disconnects from the call. Playback devices are installed like single-line sets, using one-pair twisted cable and four-conductor modular jacks (line jack units). The device must be capable of the following functions: detect ringing, provide ring trip, and automatically disconnect when the announcement is completed.

5.78 Playback devices may be used with hunt groups to speed call processing. Hunt groups can have three types of special stations: announcement stations, overflow stations, and recall destination stations. These types of stations may be equipped with a playback device that answers the call, then disconnects to transfer it back to the hunt group. (See FEATURES, page 5-37, for more information.)

5.79 The playback device is programmed as a regular single-line station and is assigned an extension number. If the device is sensitive to camp-on tones and disconnects calls, the programmer can disable the camp-on indications for that station circuit (see PROGRAMMING, page 6-210).

6. VOICE PROCESSORS

6.1 The Voice Processing Unit runs either Windows NT or Windows 2000. The Enterprise® Messaging (EM) system runs either Windows 2000 Professional or Windows 2000 Server. Both processors are installed on an IBM-compatible PC that is connected to the Inter-Tel system to provide integrated voice processing (voice mail, automated attendant, hunt group announcement and overflow, optional Fax-On-Demand, Record-A-Call, etc.). For installation instructions, see page 3-210.

NOTE: As of version 8.0, the OS/2 Voice Processing Unit is no longer supported.

6.2 The Voice Processing PC is equipped with the following items:

- For NT-based units, Microsoft Windows NT Workstation version 4.0 operating system software (equipped with Service Pack 5)
- For 2000-based units, Microsoft Windows 2000 (equipped with Service Pack 4)
- For NT-based units, Pentium® 133 MHz or AMD K6-2 or higher microprocessor (depending on the PC version)
- For 2000-based units, 300 MHz or higher microprocessor
- 32 MB or 64 MB RAM (depending on the PC version)
- 2.1 GB or larger hard disk drive
- 3.5-inch (1.44 MB) double-sided/high-density floppy disk drive
- CD-ROM
- Network Interface Card (NIC) (This is a specially-designed network communications processor card that can transfer data packets between the Voice Processing PC and a Local Area Network. The RJ45 connector on the Network Interface Card connects to the LAN via a customer-provided standard network interface cable.)
- Standard VGA monitor, keyboard, and mouse

NOTE: A Voice Processing PC is required to achieve the maximum capacities of a v6.0 or later system (see page 1-15).

6.3 For details about the EM system, see the *Enterprise Messaging System Installation Manual* (document part no. 780.8006).

6.4 Optional equipment that can be used with the Voice Processor includes:

- Fax card (for the optional Fax-On-Demand feature)
- SMDR retrieval cable kit, part number 828.1282 [not supported in Europe]
- Parallel printer for producing various reports
- Zip Drive Kit (part number 828.1460) for database save and restore

A. ENVIRONMENTAL REQUIREMENTS

6.5 The Voice Processor and any peripheral PC equipment require the following environmental conditions:

Table 2-17. *Voice Processor/Peripheral PC Equipment Environmental Requirements*

Requirements	In Operation	In Storage
Temperature (PCs) Relative Humidity (Non-Condensing) Altitude Shock	41° to 131 F (5° to 55° C) 5% to 95% Up to 10,000 ft (3,048 m) 20 G, 2 ms	-40° to 140° F (-40° to 60° C) 5% to 95% Up to 40,000 ft (12,192 m) 200 G, 2 ms

NOTE: Do ***not*** exceed the maximum operating temperature (as listed above). Therefore, when installing the Voice Processing PC and other peripheral PC equipment, allow a sufficient margin for error in case of air conditioning failure, routine maintenance, plant shutdown, etc. As a general guideline, if the conditions are suitable for office personnel, they are also suitable for all equipment and station instrument operation. A properly controlled environment helps to extend the operating life of the equipment.

B. VOICE PROCESSING APPLICATIONS SOFTWARE

6.6 Each new Voice Processor is shipped with the Voice Processing applications software already installed on the hard disk drive. For more information about voice processing application software, see the *Windows 2000 Voice Processor Notice* (part number 835.2761).

C. UNIFIED MESSAGING

6.7 Unified Messaging versions 2.2 (VPU) and 2.3 (EM) have reached their end of sale and are no longer supported as of April 30, 2008.

6.8 Inter-Tel recommends that current UM v2.2 and v2.3 installations upgrade to UM Open Standards Edition (OSE). See below for more information.

D. UNIFIED MESSAGING OPEN STANDARDS EDITION (OSE)

6.9 Axxess v8.23x or later supports Unified Messaging OSE, a client-free Unified Messaging distribution for the Inter-Tel Enterprise Messaging voice messaging server. Unlike previous Unified Messaging products from Inter-Tel, users do not have to install a special Unified Messaging client application on their desktop computer to view, play, and manage voice mail messages. With Unified Messaging OSE, the e-mail client can be any Internet Message Access Protocol (IMAP) v4 compliant client that is configured to connect to the IMAP server located on Enterprise Messaging. To use the latest version of Unified Messaging OSE, v2.0, existing Axxess systems must upgrade to v11.0 or later, and to Enterprise Messaging v1.2 or later.

6.10 For Unified Messaging OSE v2.0 programming and feature information, refer to the latest version of the *Unified Messaging Open Standards Edition Administrator's Guide* (document part no. 835.3162). For end-user information and instructions for using features, refer to the *Unified Messaging Open Standards Edition User Guide* (part number 835.3164).

E. DIAGNOSTICS

6.11 For advanced troubleshooting purposes, the Voice Processing applications software includes special diagnostics software. If a Voice Processing applications problem cannot be resolved, Customer Support may instruct the technician to connect the monitor and keyboard and perform certain diagnostic procedures. For example, see page 7-107 for details on retrieving system log files.

F. AUDIO INTERFACE CARD

6.12 The Voice Processing PC's Audio Interface Card (AIC) is a specially-designed Input/Output (I/O) circuit card that connects to the DB15 PCM connector on the CPU card, CPC, or Options Card. The initial connection to the CPU or CPC provides up to eight audio interface ports (e.g., dedicates eight system voice channels) for communication between the telephone system and the Voice Processing PC. For an additional 4-32 ports, the AIC may be connected to the DB15 PCM connector on the Options Card.

G. VOICE PROCESSING CARDS

6.13 To control the audio interface ports provided by the AIC, as many as three Voice Processing Cards (VPCs) can be installed in the Voice Processing PC (at least one is required). The VPCs are connected to the AIC using a special interface cable.

6.14 There are several different models of Voice Processing Cards (VPCs) available: 4-Port, 4-Port "Mini", 8-Port, 8-Port "Mini", 12-Port, 16-Port, and 24-Port. Other than their smaller size, these mini-size cards install and function the same as the full-size cards. As many as three VPCs (in any combination) can be installed up to the system maximum of 40 audio ports.

H. VIDEO CARD

6.15 The Voice Processing PC is equipped with a Video card (currently VGA) so that the monitor and keyboard (IBM AT-compatible) can be connected for performing diagnostics or for loading software and/or voice prompts.

I. OPTIONAL FAX CARD

6.16 The Voice Processing PC can be equipped with a special Fax card to provide Fax-On-Demand services to callers. See page 4-29 for more information.

J. MESSAGE STORAGE

6.17 The amount of voice mail message storage space available depends on the size of the hard disk drive installed in the Voice Processing PC. Approximately 10 MB of disk space is required for each hour of message storage. A 405 MB hard disk drive, for example, can store approximately 30 hours of messages in addition to the operating system software. A 540 MB hard disk drive can store approximately 45 hours of messages.

NOTE: The 10 MB/hour capacity does not include SMDR or Fax-On-Demand storage allocations. If these features are used, there is a corresponding reduction in available message storage space.

K. SMDR RETRIEVAL CONNECTION REQUIREMENTS

NOTE: The SMDR Retrieval cable kit is not supported in Europe.

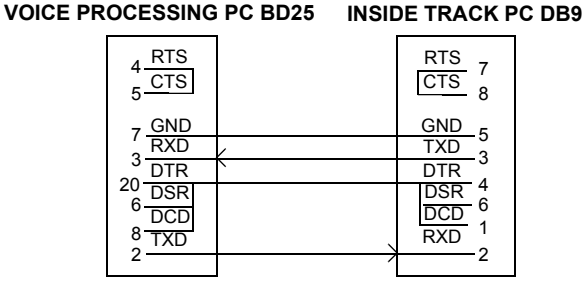
6.18 If necessary, the DB25 (COM2) serial port on the back of the Voice Processing PC can be used to connect another PC for SMDR data retrieval using Inter-Tel's Inside Track call accounting software (not applicable to European systems).

6.19 To do this, use the modular adapters and line cord included in the optional SMDR retrieval cable kit (part no. 828.1282), as listed below and described in the *Inside Track User Guide*.

- DB25 Modular Adapter (female, reversing, four-wire) – part number 804.2425 (MOD-TAP part number 06-986-G99-33)
- DB9 Modular Adapter (female, non-reversing, four-wire) – part number 804.2420 (MOD-TAP part number 08-988-610-00)
- Line Cord (two-pair, reversing) – part number 813.1592

6.20 If the SMDR retrieval cable kit is not available, it is possible to use a standard straight-through cable along with a null-modem adapter and, if necessary, a gender changer. These items are readily available at most electronic/computer equipment stores.

6.21 Another option is to use the following wiring diagram to build the necessary cable/connectors.



DB25 internal jumpers: 4-5 and 20-6-8
DB9 internal jumpers: 7-8 and 4-6-1

NOTE: When building cable/connectors from this diagram, note that *the connections between the Voice Processing Unit DB25 port and the other device include the modular adapters and a line cord*. The connection must be made as indicated in the diagram (correct pins connected) regardless of whether a reversing or non-reversing line cord is used.

L. PARALLEL PRINTER CONNECTION REQUIREMENTS

6.22 If desired, the DB25 parallel port on the back of the Voice Processing PC can be used to connect a parallel printer for producing various reports. The pin functions of the parallel port are listed in the following chart.

Table 2-18. *Parallel Printer Connection Requirements*

Signal Name	Function	Pin
-STROBE	OUT	1
DATA BIT 0	OUT	2
DATA BIT 1	OUT	3
DATA BIT 2	OUT	4
DATA BIT 3	OUT	5
DATA BIT 4	OUT	6
DATA BIT 5	OUT	7
DATA BIT 6	OUT	8
DATA BIT 7	OUT	9
-ACK	IN	10
BUSY	IN	11
PE	IN	12
SLCT	IN	13
-AUTO FEED XT	OUT	14
-ERROR	IN	15
-INIT	IN	16
-SLCT IN	OUT	17
GND	-	18-25

M. SECURITY KEYS FOR UNIFIED MESSAGING FEATURES

6.23 The Unified Messaging feature requires special security keys. These keys are attached to the parallel port of the Voice Processing PC. They control the number of users and level of features allowed. The available security keys are listed in the REPLACEMENT PARTS section of this manual.

N. VOICE PROCESSOR NETWORK CAPABILITIES

6.24 A Voice Processor can be installed on any or all nodes in the phone system network. These Voice Processors can also be networked together to allow a caller to leave a message on the local Voice Processor for a mailbox located on another Voice Processor in the network. There can be up to 100 nodes in the Voice Processor network.

6.25 Each Voice Processor has its own internal message queue (similar to a mailbox's message queues) and stores messages destined for other Voice Processing Units until they are delivered.

NOTE: The Inter-Tel Messaging IP Networking or Voice Processor AMIS Networking software license is required to use Voice Processing Nodes. See page 6-97.

6.26 The Voice Processing Node programming area is used to identify and define information about each node (unit) in the Voice Processing network. Before the local Voice Processing Unit can deliver messages to another Voice Processing node in the network, the remote node must be included in the local table.

6.27 Mailboxes programmed on the local node, but associated with mailboxes on remote Voice Processing nodes are referred to as "network mailboxes." They are used the by the local Voice Processor to identify and locate the mailboxes located on the other Voice Processing nodes. They are not actual mailboxes; they are just "place holders" that tell the local node where to send messages received for that mailbox number. If the mailbox subscriber logs on to a network mailbox, the only options available are the directory name, the password, and possibly the transfer method programming prompt, if enabled.

6.28 There are several methods available for Voice Processing networks. The networking method determines how the local node communicates with the remote node. The networking methods include the following:

- **AMIS**: AMIS is an analog networking protocol that uses the public telephone network.
- **TCPIP**: TCPIP is a direct network connection between the local Voice Processor and the remote Voice Processor.

7. IP NETWORK SPECIFICATIONS

CAUTION

Inter-Tel strongly recommends using the IP devices on a managed private network. If using a private network, the following recommended guidelines are relatively easy to attain. However, if connected to the public Internet, IP devices do function, but the quality may suffer due to the dynamic bandwidth availability. The possible problems could be voice quality degradation, garbled speech, dropped calls, equipment resets, etc. Also, the VoIP suitability of any Internet connection can change at any time, with no advance notice. **Inter-Tel cannot guarantee any voice quality when connected to the public Internet. Therefore, Inter-Tel is not responsible for network quality issues that are caused by using the public Internet.**

7.1 Because the IPRC uses Internet Protocol (IP) to connect Inter-Tel systems, your data network must meet certain specifications to ensure a quality phone network. This section describes these network requirements.

NOTE: Currently the IPRC does not support faxing over the IP network.

7.2 Inter-Tel recommends that you run the Network Qualifier (for at least 24 hours) to assess your network before purchasing or installing the hardware required to set up voice over IP (VoIP). The Network Qualifier currently supports up to eight simulated VoIP connections. While in the Network Qualifier, set the Audio Frames/IP Packet and Minimum Playback Buffer Time fields to the appropriate values (see page 2-98). If the assessment reveals that your network is operating outside the specifications listed below, contact your network administrator. The Network Qualifier application is available on the Inter-Tel Web site (www.inter-tel.com) and on the phone system software CD.

A. PERFORMANCE

7.3 The quality of the data network is one of the most important factors in achieving a high-quality Internet Protocol (IP) telephony call. The variables involved include the speed and capacity of the network configuration, type of connection, routing scheme, and amount of traffic being passed through the network.

7.4 Inter-Tel recommends the following network performance characteristics for voice:

- **Packet Loss**: (The percentage of unusable packets) No more than 5% random packet loss
- **Successful Packet Rate**: (The opposite of packet loss) 95% or more
- **Latency**: (The time it takes for information to pass from one point to another) Less than 120ms of one-way delay and less than 250ms of total end-to-end delay

  NOTE: End-to-end delay is the sum of the one-way network, buffering, and processing delays. For example, at default, the total buffer and processing delays of the IPRC is 130ms, leaving 120ms (250ms - 130ms) of allowable network delay.

- **Jitter**: (Changes in delay) 40ms or less (indicated by the results from the Network Qualifier)

NOTE: If the Network Qualifier returns a value greater than 40ms for jitter, voice packets received over the network may be unusable. If this occurs, the unused packets contribute to (and increase) the packet loss.

- **Hops**: (The number of router points a data packet must pass through during transmission) 15 or fewer to minimize the number of potential problem points
- **Committed Information Rate (CIR) and Burst Rate (BR)**: 512 Kbps each (This means that 512 Kbps is always available, and the link may burst up to 1024 Kbps for short periods if the bandwidth is available from the WAN provider.) There is usually no guarantee of BR availability.

NOTE: Keep the voice traffic below the CIR value. When voice traffic exceeds the CIR value, packets may begin to queue within the frame relay network. When this happens, frame packets may become subject to delay and could eventually be tagged as Discard Eligible (DE). (Packets marked as DE could be discarded before reaching their destination. This potential queuing delay and packet loss can adversely affect voice quality.

7.5 Whenever possible, Inter-Tel recommends using Ethernet switching mechanisms rather than Ethernet hubs. The switches offer more inherent Ethernet advantages over hub (shared) technology, such as segmentation, full-duplex operations, layer 2 prioritization, etc.

NOTE: Call Processing does not support firewalls or Network Address Translation (NAT) tables. Because Call Processing controls the IPRC, the IPRC cannot be installed behind a firewall or use a Network Address Translation (NAT) table.

7.6 If using a private network, these recommended guidelines are relatively easy to attain. If connected to the public Internet, the IPRC does function, but the quality may suffer due to the dynamic bandwidth availability. Therefore, Inter-Tel recommends using the IPRC(s) on a managed private network.

7.7 For any router that has IPRC traffic passing through it, program the router to prioritize the packets sent by the IPRC. You can usually do this by programming the router to prioritize the packets sent by a specific IP address (i.e., the IPRC's IP address), by prioritizing UDP packets (because all audio packets sent by the IPRC are UDP packets), or by prioritizing the UDP port numbers that the IPRC uses. You can also use Quality of Service (QoS) features, like Type of Service (ToS) or Differentiated Services (Diffserv). Under the IP Connection folder in DB Programming, set the Audio RTP Type of Service field to 184 if your routers use ToS or Diffserv. (Although 184 is the recommended setting for the Audio RTP Type of Service, you can set this field to any value for a Diffserv code point in your network.) In addition, program the external router to fragment large data packets into smaller packets. This reduces the possibility that voice packets have to wait for a large data packet at the router.

B. BANDWIDTH UTILIZATION

7.8 The IPRC uses all of the channels as talk paths, and signaling is transmitted via the Ethernet interface along with the voice traffic.

7.9 To ensure enough bandwidth (i.e., the IP, Ethernet, and/or Frame Relay overhead) is available, use the formulas listed below to calculate the amount required to make inter-node IPRC calls. Although bandwidth varies depending upon call control messages, tone detection, voice activity detection (VAD), and the RTP profile, these formulas provide a good estimate for planning the network.

NOTE: The amount of bandwidth used on any network segment (i.e., WAN or LAN) should never exceed 75% of the available bandwidth. There should always be a 25% margin factored in when planning the network capacity for voice, data, etc.

7.10 The bandwidth is calculated based on the following programmable settings:

- **Vocoder (voice codec)**: Determines the codec bandwidth and the frame size (audio sample size).
- **Audio Frames per IP Packet**: Determines the number of voice frames placed into each packet.
- **RTP Profile**: Determines the Real-time Transport Protocol (RTP) header size.

7.11 The specific variables in the formula are defined in the following table.

Table 2-19. *Variables*

Variable	Value
RTPHeadSize	4 for Inter-Tel RTP; 12 for Standard RTP
IP/UDPHeadSize	28 bytes
EthHeadSize	18 bytes
FrameHeadSize	6 bytes (worst case)
AudioFramesPerIPPacket	Number of voice frames used per packet
VoiceFrameSize	Size, in milliseconds, for one voice frame, which is dependent on the voice codec (vocoder)
CodeBW	Codec bandwidth (in kbps)
BurstDataPerCall	Data bursts in the call control socket for each call setup
BurstDataDB	200 kbps when importing/exporting in DB Programming
#Calls	Number of calls

7.12 While the maximum burst data per call (BurstDataPerCall) varies from system to system, you can estimate the bandwidth required based on the values in the following table.

Table 2-20. *Required Bandwidth*

Event	KBPS
IC Call	10
All-Ring Hunt Group Call	10 per member

7.13 The CodecBW and VoiceFrameSize values for each of the supported vocoders are listed in the following table.

Table 2-21. *Codes Bandwidth and Voice Frame Size*

Voice Codec	Codec Bandwidth	Voice Frame Size
G.729	8 kbps	10 ms
G.711	64 kbps	10 ms

Voice Over IP

$$\text{Bandwidth in each direction} = \left[\frac{(\text{EthHeadSize} + \text{IP/UDPHeadSize} + \text{RTPHeadSize}) \times 8 \text{ bits/byte}}{\text{AudioFramesPerIPPacket} \times \text{VoiceFrameSize (10ms for G.729)}} + (\text{CodecBW} + \text{BurstDataPerCall})\right] \times \#\text{Calls} + \text{BurstDataDB}$$

Voice Over IP Across Frame Relay

$$\text{Bandwidth in each direction} = \left[\frac{(\text{FrameHeadSize} + \text{IP/UDPHeadSize} + \text{RTPHeadSize}) \times 8 \text{ bits/byte}}{\text{AudioFramesPerIPPacket} \times \text{VoiceFrameSize (10ms for G.729)}} + (\text{CodecBW} + \text{BurstDataPerCall})\right] \times \#\text{Calls} + \text{BurstDataDB}$$

7.14 For example, using the default settings (G.729, three audio frames per packet, and Inter-Tel RTP) and including Ethernet overhead, the bandwidth for 32 concurrent voice over IP calls would result in the following:

$$\left[\frac{(18 + 28 + 4) \text{ bytes} \times 8 \text{ bits/byte}}{3 \times 10 \text{ ms}} + (8 \text{ kbps} + 10 \text{ kbps})\right] \times 32 \text{ calls} = 1002.7 \text{ kbps}$$

C. PROCESSING LATENCY

7.15 The amount of latency introduced by a IPRC depends on the specific vocoder being used. When calculating latency, you must consider two vocoder-specific variables and one programmable parameter, as described below.

- **Frame Size (FS)**: Is a fixed number and is inherent to the specific vocoder being used. The following table distinguishes the Frame Size for each vocoder in milliseconds.

Table 2-22. *Frame Size*

Vocoder	Frame Size
G.729	10 ms
G.711	10 ms

- **Audio Frames per IP Packet (AF)**: Is a programmable parameter (Audio Frames Per IP Packet; see page 6-185) for each Node IP Connection Group that indicates how many vocoder frames are packed into a single packet. For minimal latency (but more IP or Frame overhead), configure the system with one audio frame per IP packet. For lower overhead, use larger values for this field. By default, a value of three is used for the Audio Frames Per IP Packet field because it provides a good balance between low latency and low overhead. You can configure this value while in the Network Qualifier.
- **Minimum Playback Buffer Time (MPBT)**: Defines the average time, in milliseconds, that packets wait in the receive buffer before the system plays the audio. The higher this number, the more latency in the signal; however, it is less likely that network problems like jitter causes lost or late audio packets. The lower the minimum playback time, the less latency there is in the signal; however, there is a greater chance of jitter affecting the call. The default is 80 ms, and the range is 1 to 320 ms. You can configure this value while in the Network Qualifier.

7.16 The basic formula for calculating end-to-end latency is as follows:

Latency = ([AF + 2] * FS) + MPBT + estimated one-way network delay

Where the estimated one-way network delay is based on dividing the round-trip ping time by two.

NOTE: The actual end-to-end latency on a live network may be higher due to normal fluctuation in network delay.

7.17 The following is an example of how the total end-to-end latency value is calculated. This example uses the default settings (G.729, three audio frames per IP packet, and 80 ms buffer time) and assumes a 20 ms one-way network delay.

Latency = ([3 + 2] * 10) + 80 + 20 = 150 ms

8. ADDITIONAL SYSTEM EQUIPMENT

8.1 The following additional system equipment is available:

- Personal Computer for On-Site or Stand-Alone/remote Programming (see below)
- Modem Connection (see page 2-100)
- MSDR/Error Message Output Device (see page 2-100)
- PC Data Port Module Connection (see page 2-100)
- CT Gateway and Transceivers (see page 2-101)
- IP and Frame Relay Devices (see page 2-102)

8.2 The following requirements are also described in this section:

- Communication Parameters (see page 2-102)
- Ferrite Beads (see page 2-103)
- Power Failure Transfer Connection (see page 2-103)
- Night Transfer Connection Requirements (see page 2-103)
- System Battery Back-Up Requirements (see page 2-104)
- Additional Information (see page 2-104)

A. PERSONAL COMPUTER FOR ON-SITE OR STAND-ALONE/REMOTE PROGRAMMING

8.3 **PC Requirements**: The device used for on-site or stand-alone/remote programming must be an IBM-compatible PC running Windows 98, Windows NT 4.0, Windows 2000, Windows ME, Windows XP, or Windows Vista with the following specifications:

Table 2-23. *PC Requirements*

Minimum Requirements	Recommended Specifications
Pentium II 266 MHz or higher microprocessor	Pentium III 500 MHz or higher
32 MB RAM	128 MB RAM or more
100 MB free disk space to install 80-100 MB to run	100 MB free disk space to install 80-100 MB to run
VGA at 640x480 or greater	256-color display at 800x600 or greater
CD-ROM	CD-ROM
Mouse	Mouse

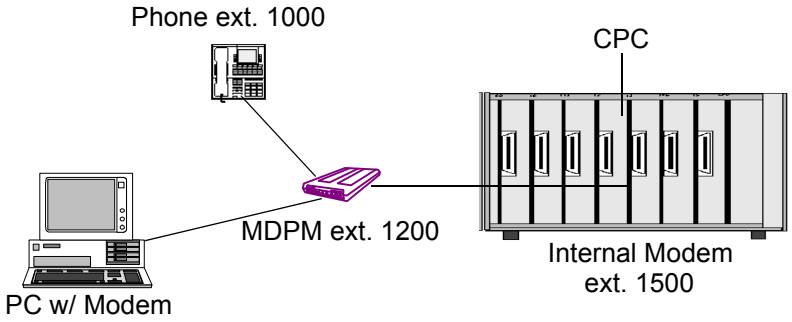
8.4 For environmental requirements, see page 2-89.

8.5 For remote maintenance, a modem is also needed, which is available on the CPC. To connect the PC to the Inter-Tel system for on-site programming, see the following information.

B. MODEM CONNECTION

8.6 If desired, a modem for remote programming/maintenance can be connected to one of the RS232 ports on the CPU card or Options Card(s). Although PCDPM RS232 ports can also be used for remote programming if necessary, it is not recommended. See Appendix B — RS232 to ensure a successful RS232 connection.

8.7 If using a CPC, you can use the built-in modem. To connect to the modem, you need a phone that has a Modem Data Port Module (MDPM) for connecting to a PC. You can then use that PC to connect to the CPC modem over a system intercom call, as shown in the following illustration. Because modem connections are slower than IP connections, use the modem connection only when necessary.



NOTICE
You cannot change the modem's extension if you are currently using the modem to connect to DB Programming.

C. SMDR/ERROR MESSAGE OUTPUT DEVICE

8.8 The optional output device(s) for the SMDR and Error Message Recording features must be equipped with RS232 communications ports. An output device may be a printer, a terminal, or a PC equipped with terminal emulation software. See Appendix B — RS232 to ensure a successful RS232 connection.

8.9 If using System Manager, you can also have the phone system send SMDR information to the System Manager server. This allows Web client users to access SMDR data remotely. For details, see the *System Manager Installation Manual*.

D. PC DATA PORT MODULE OAI CONNECTION

8.10 If the PCDPM is not connected to a DSS/BLF Unit, its RS232 port can be used to connect an SMDR/error message output device or a personal computer.

8.11 If the PCDPM uses an RS232 connection for a system serial port or desktop OAI application, see Appendix B — RS232 to ensure a successful RS232 connection.

E. CT GATEWAY AND TRANSCEIVERS

8.12 System OAI Level 2 applications can be used in conjunction with the CT Gateway and Transceivers.

- **CT Gateway**: Provides a TCP/IP System OAI interface that supports a multi-node network. Each node in a network functions individually, as far as the System OAI streams are concerned. In other words, you cannot program a System OAI serial port on Node 1 that controls devices or receive events for other nodes. If you need to control devices on multiple nodes, you must install a CT Gateway. The CT Gateway receives the OAI streams from multiple nodes and consolidates them into a single stream for a System OAI application. The CT Gateway sends and receives all of its data via TCP/IP connections.

  NOTE: Upgrade the CT Gateway to version 4.0. Version 8.0 or later requires CT Gateway version 3.0 or later.

- **Transceivers**: Provide an interface between RS232 and TCP/IP. Because the CT Gateway only supports TCP/IP, and CPC or CPU serial ports and older System OAI applications communicate with RS232, a Transceiver is needed to translate the data protocol. Because Transceivers can be located on separate PCs, they allow the CT Gateway to communicate with multiple nodes, regardless of the number of serial ports available on the CT Gateway PC and regardless of the distance between the nodes and the CT Gateway. There are two different Transceivers:

  — The Switch Transceiver provides the interface between the serial ports and TCP/IP.

  — The Application Transceiver provides the interface between System OAI application RS232 serial ports and TCP/IP. (System OAI applications that use TCP/IP do not require an Application Transceiver.)

8.13 The Switch Transceiver is not required if the connection to the telephone system uses a CP Server or CPC TCP/IP socket connection. The Application Transceiver is not required if the connection from the CT Gateway to the application uses TCP/IP.

8.14 See the *CT Gateway Installation Manual* for details. (For environmental requirements, see page 2-89.)

F. IP AND FRAME RELAY DEVICES

8.15 You can use IP or frame relay devices, such as the IP Networking Modules (previously called InterPrise 3200N), to provide networking support between system nodes. No special programming is required when using IP devices or frame-relay devices in place of T1, E1/PRI, or T1/E1/PRI connections. Each remote node still has programmed node trunk groups and off-node devices.

8.16 When planning the system, note that each node trunk group can have up to 23 trunks [30 trunks in Europe], thus limiting the number of incoming and outgoing calls. Make sure you have enough T1/PRI, E1/PRI, or T1/E1/PRI cards connected to the IP or frame-relay device(s) to support the customer's call load.

8.17 Refer to the manuals provided with the IP or frame relay device for hardware information and programming procedures.

8.18 **U.S. and U.K. Networking**: The U.K. versions software allow the capability to network U.S. and U.K. systems together using the Inter-Tel IP Networking Module Products. Any 5.x version of U.S. software can network with a U.K. system. This feature is enabled by a flag called Network with United States, which is available in the U.K. version only. This flag is located under System/Flags.

8.19 Although this is not a choice for the U.S. systems, it is mentioned in this document so that U.S. users are aware of the option to have U.S. systems networked to U.K. systems.

NOTE: The U.S./U.K. network must utilize the IP Networking Module products at version 1.3 or later.

G. COMMUNICATION PARAMETERS

8.20 The on-site programming PC and SMDR/error message output devices must communicate at 300, 1200, 2400, 4800, 9600, 19200, or 38400 bps, depending on the selected port. System bit rates are set independently in DB Programming. Other specifications include:

- The data format must be 8 bit standard ASCII.
- Parity is OFF (ignored).
- Communication is full duplex.
- There is one stop bit.
- Hardware flow control is enabled.

NOTE: For software uploads only, the CPU020/PCM card is also capable of using 115,000 bps. See page 3-170 for more information.

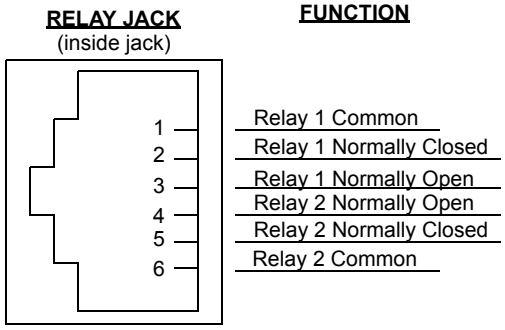
8.21 For more information on configuring a serial port, see Appendix B — RS232.

H. FERRITE BEADS

8.22 To help reduce electromagnetic interference (EMI) and radio frequency interference (RFI), it may be necessary to equip the RS232 cables with ferrite beads. (See INSTALLATION, page 3-26.) Two ferrite split beads (two half beads) are installed on each cable. The part number for a single split bead is 808.1195.

I. POWER FAILURE TRANSFER CONNECTION REQUIREMENTS

8.23 To connect power failure transfer (PFT) relay equipment to the relay jack (six-conductor modular jack) on the CPU card or CPC (see page 2-30), one end of the cable between the two must have a 6-pin modular plug connector. The other end of the cable must be compatible with the PFT equipment. Consult the owner's guide for the equipment to verify that the pin functions are the same as those shown in the chart at the right. If necessary, rewire the connector on the device to match the requirements.



8.24 During normal system operation, both relays are energized. When power is removed, connection is made between the common and normally closed contacts. On the CPU020/PCM card, R1C is wired in parallel to R2C, R1NC is wired in parallel to R2NC, and R1NO is wired in parallel to R2NO.

J. NIGHT TRANSFER CONNECTION REQUIREMENTS

8.25 To connect night transfer relay equipment to the relay jack (six-conductor modular jack or line jack unit) on the Options Card (see page 2-67), one end of the cable between the two must have a 6-pin modular plug connector. The other end of the cable must be compatible with the night transfer equipment. Consult the owner's guide for the equipment to verify that the pin functions are the same as those shown in the chart above. If necessary, rewire the connector on the device to match the requirements.

8.26 The output devices, whether used for SMDR or error messages, must be able to print reports with a width of 80 characters. The devices must not have cables longer than 50 feet (15 meters).

K. SYSTEM BATTERY BACK-UP REQUIREMENTS

8.27 The CPC is equipped with a rechargeable battery that backs up the database. In the event of a power failure or brownout condition, this battery supports the card for up to 30 days. When power is restored to the card, the battery is automatically recharged.

8.28 To provide back-up power for the system power supply and the Voice Processing PC power supply, you can install optional battery back-ups using a customer-provided uninterruptable power supply (UPS) unit or standby power supply (SPS) unit. Contact Inter-Tel's CommSource division or Inter-Tel Europe for a list of available models.

8.29 Inter-Tel recommends that a power supply unit with the following characteristics be installed:

- Sine wave output
- Transfer time of less than 20 milliseconds
- Low AC line transfer voltage (or cutoff circuit) of minimum 100VAC [200VAC in Europe]

NOTE: The system back-up time is determined by the battery ampere-hour ratings. The unit's batteries can be either external or internal. Also, even if the power supply unit has the specifications listed above, it cannot be guaranteed that it works properly with the Inter-Tel system. Contact Inter-Tel's CommSource division or Inter-Tel Europe and/or Technical Support for a listing of approved UPS/SPS units and installation instructions.

8.30 The following chart shows the input wattage requirements, under full load for the system power supplies.

Table 2-24. *Input Wattage Requirements*

Power Supply (U.S. Part No.)	Input Wattage
4-Amp (550.0121)	172 watts
6-Amp (550.0104)	225 watts
9-Amp (550.0110)	263 watts

L. ADDITIONAL INFORMATION

8.31 If necessary, contact Inter-Tel's Customer Support department and/or CommSource division or Inter-Tel Europe for additional information on the optional equipment listed below:

- Standby Power (system battery back-up)
- Lightning and AC Surge Protection
- Headsets
- Paging Equipment
- Single-Line Telephones
- Cordless Telephones
- Music-On-Hold
- OPX Repeaters
- Inside Track SMDA Software and Buffer Box (not applicable to European systems)
- Facsimile Machine
- Playback Devices
- Power Failure Transfer

Figure 2-8. *Original Basic Equipment Chassis*

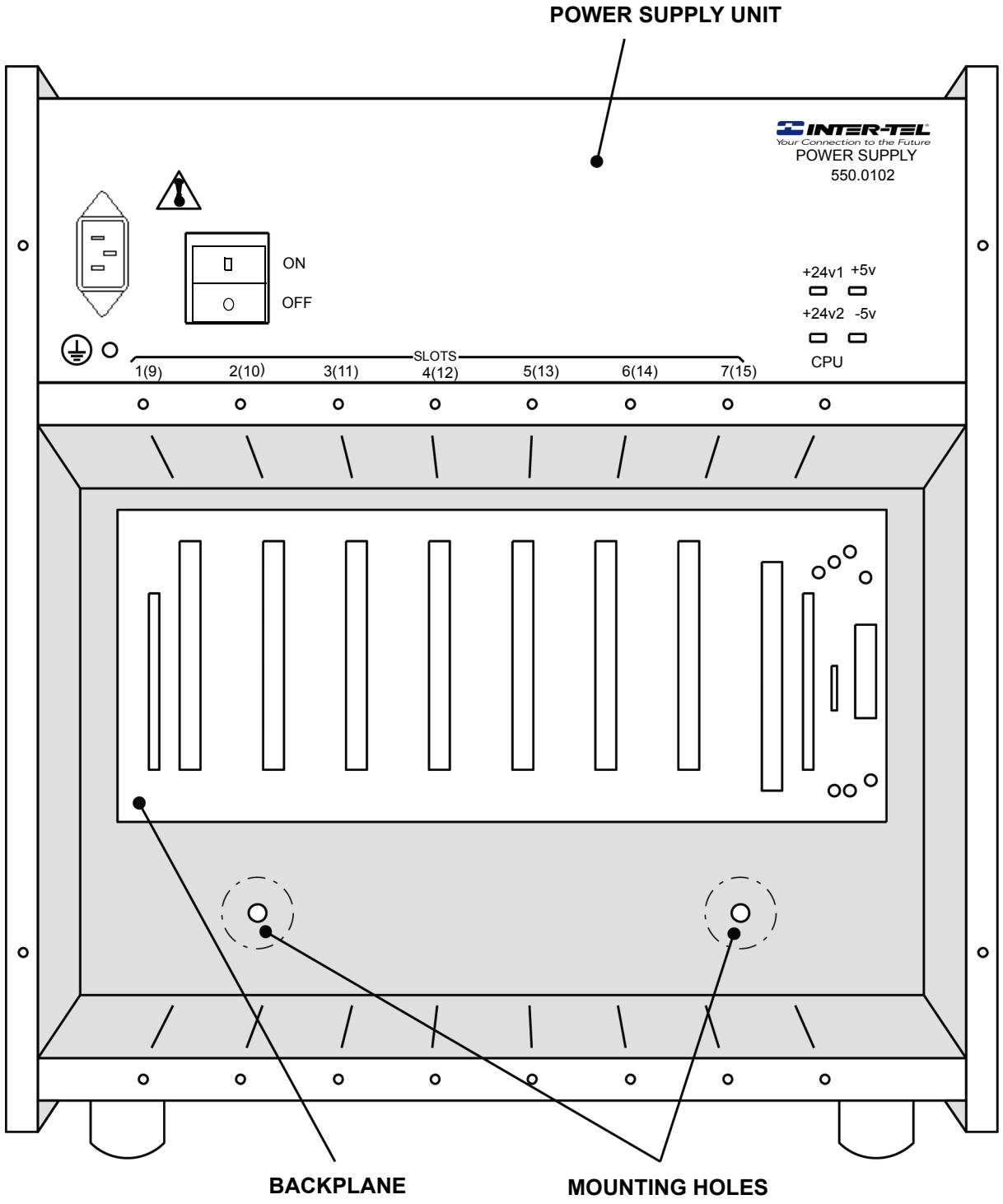


Figure 2-9. *New-Style Basic Equipment Chassis*

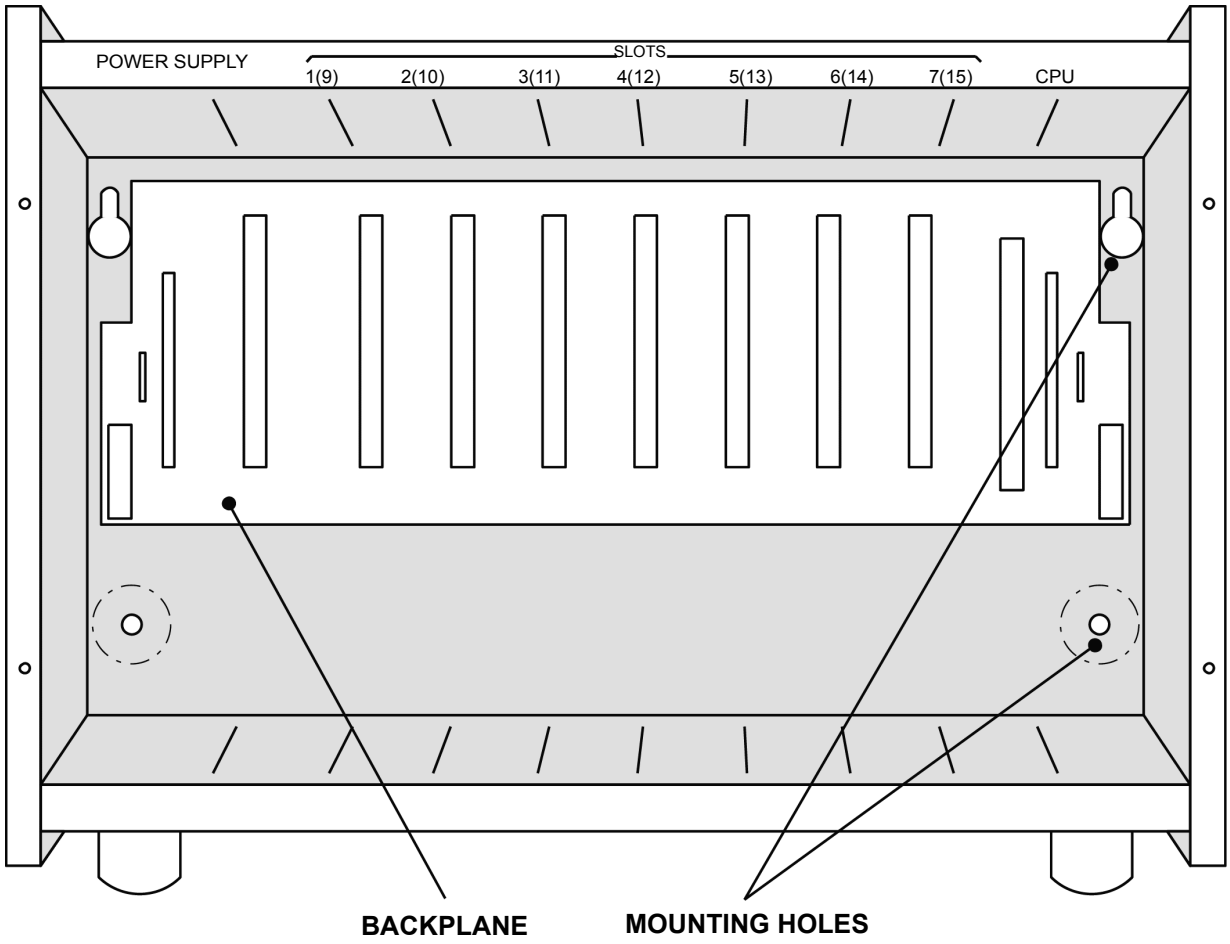
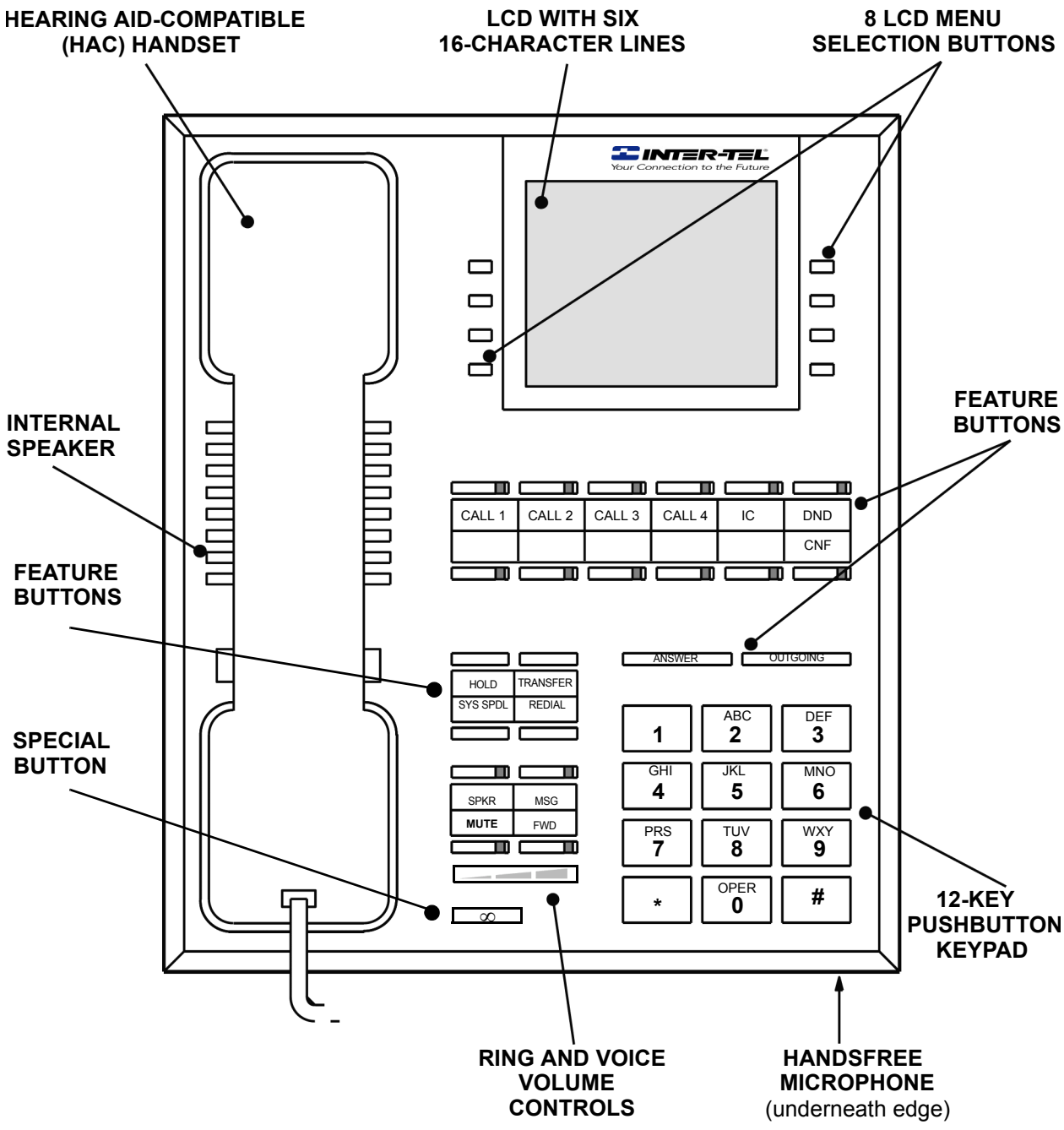


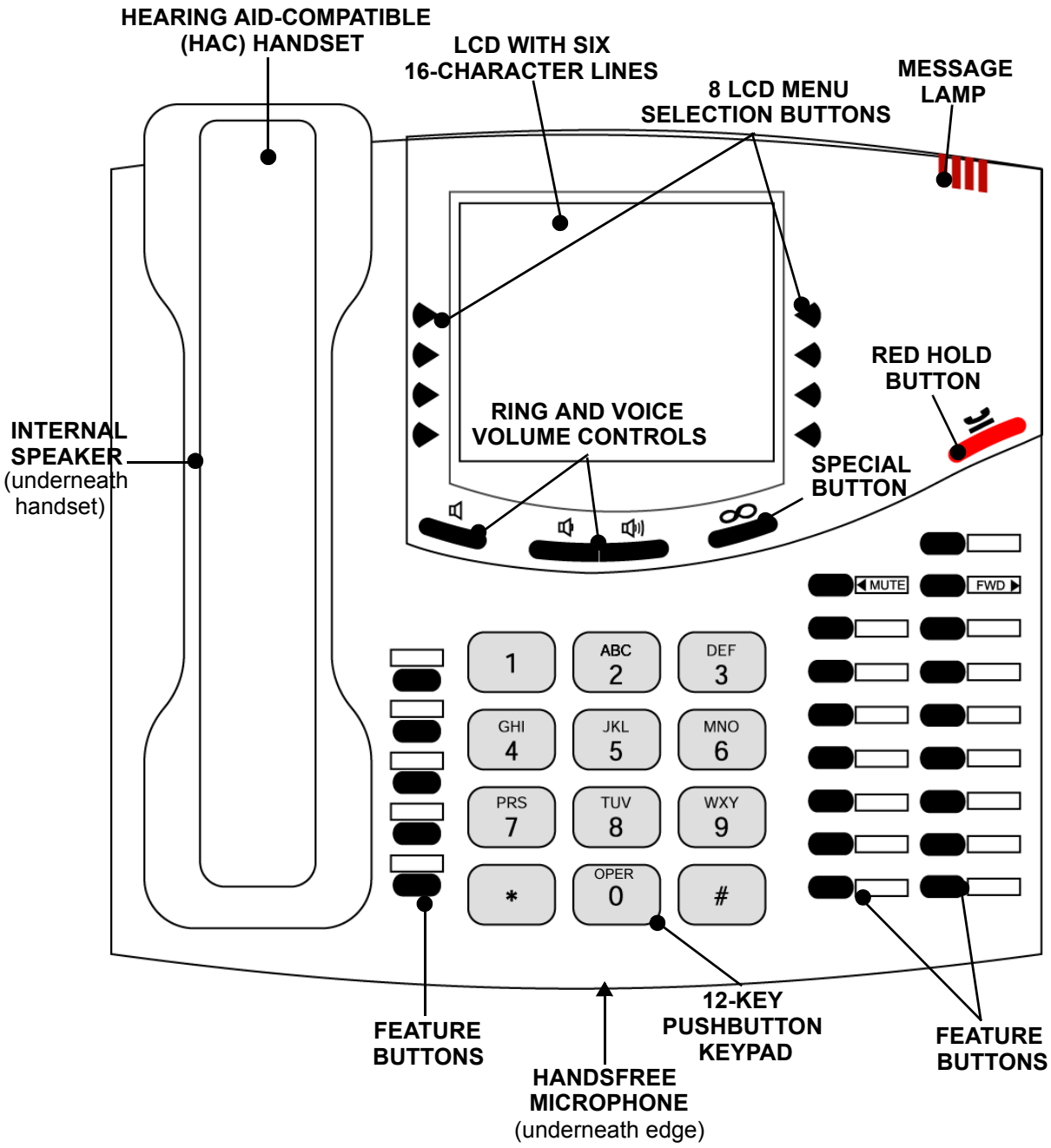
Figure 2-10. *Executive Display Phone*



NOTE: The buttons are shown as they appear in the default keymap configuration.

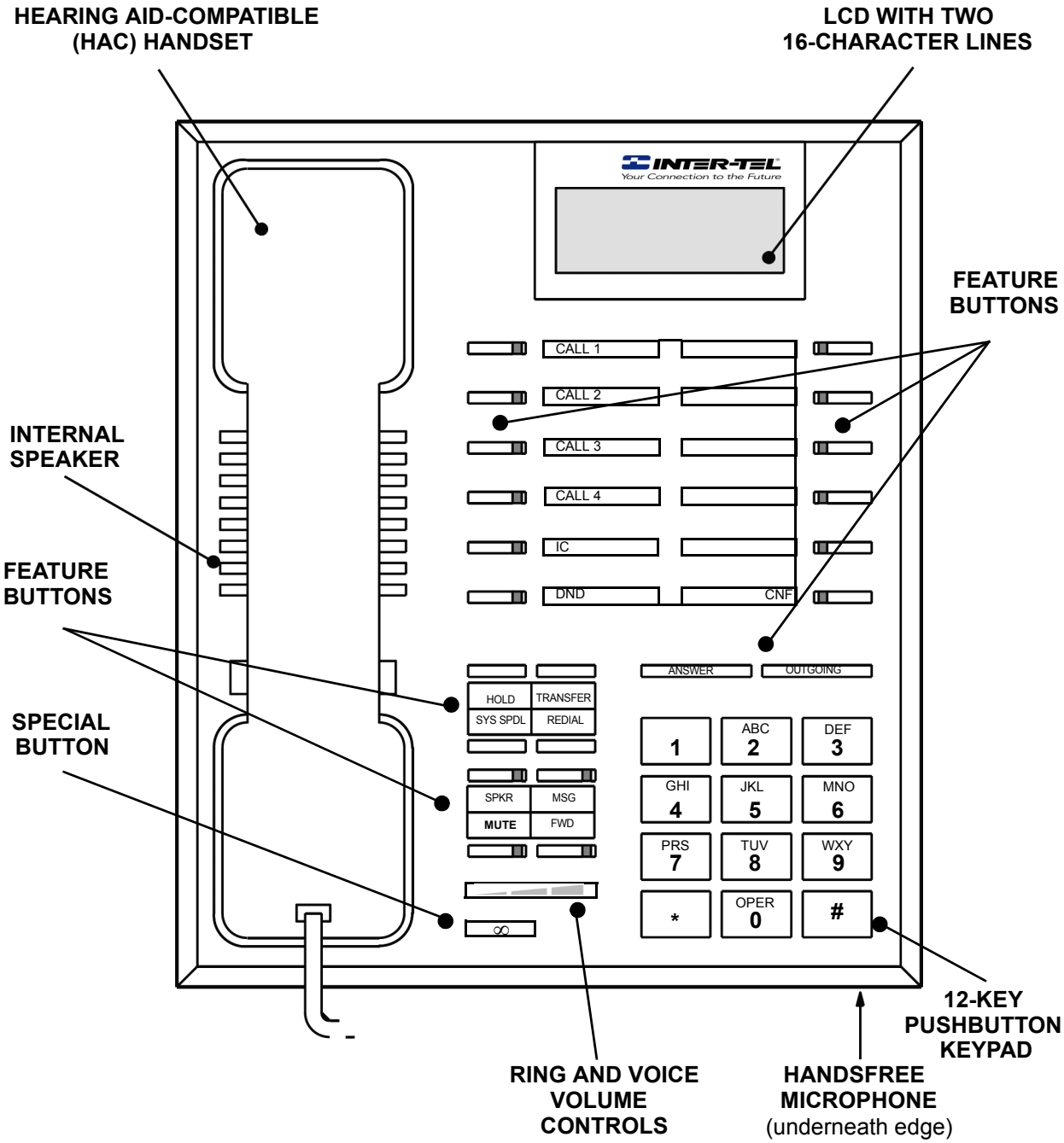
Figure 2-11. *Professional Display Phone*

Figure 2-12. *Model 8560*



NOTE: The buttons are shown as they appear in the default keymap configuration.

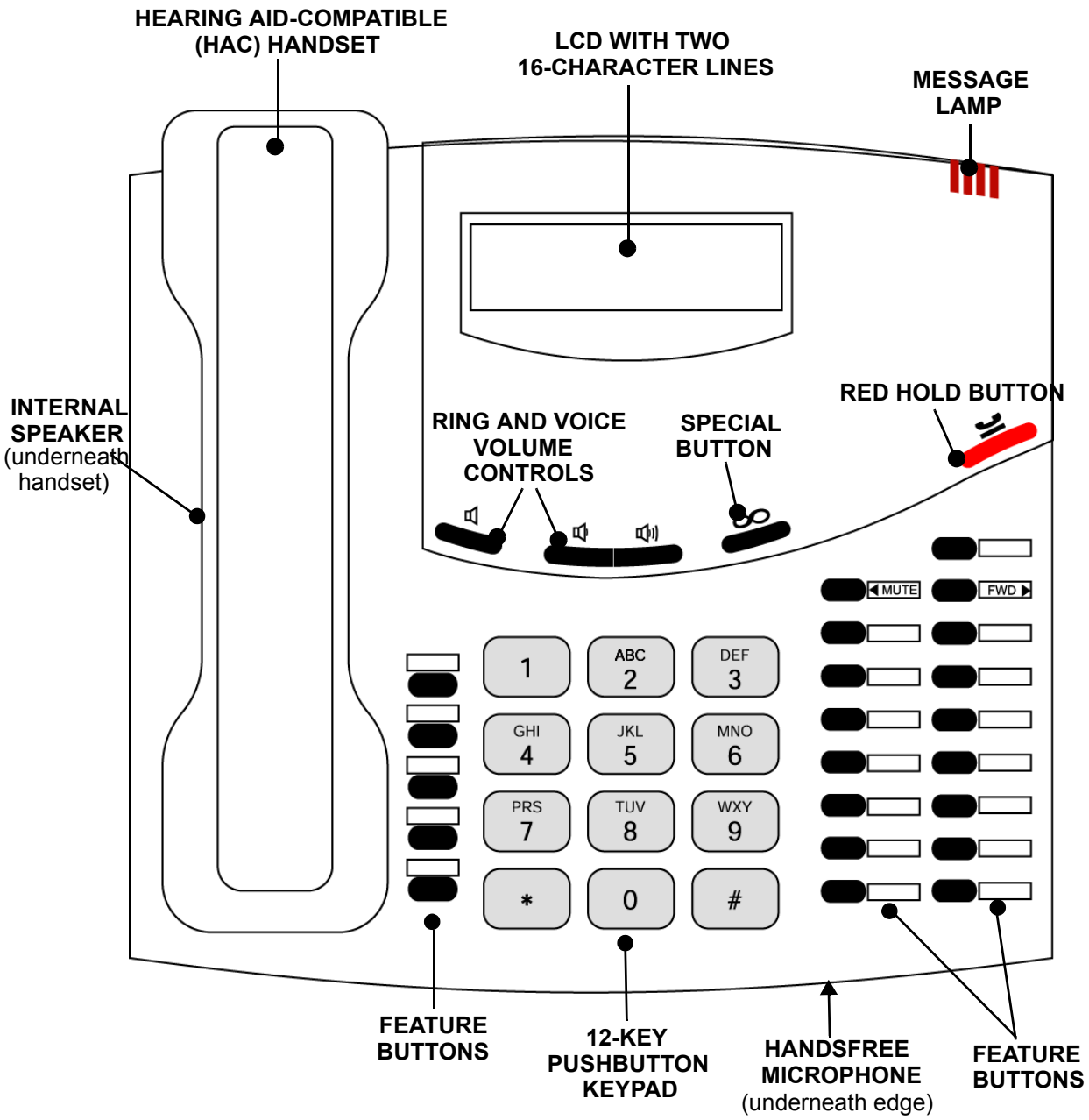
Figure 2-13. *Standard Display Phone*



NOTE: The buttons are shown as they appear in the default keymap configuration.

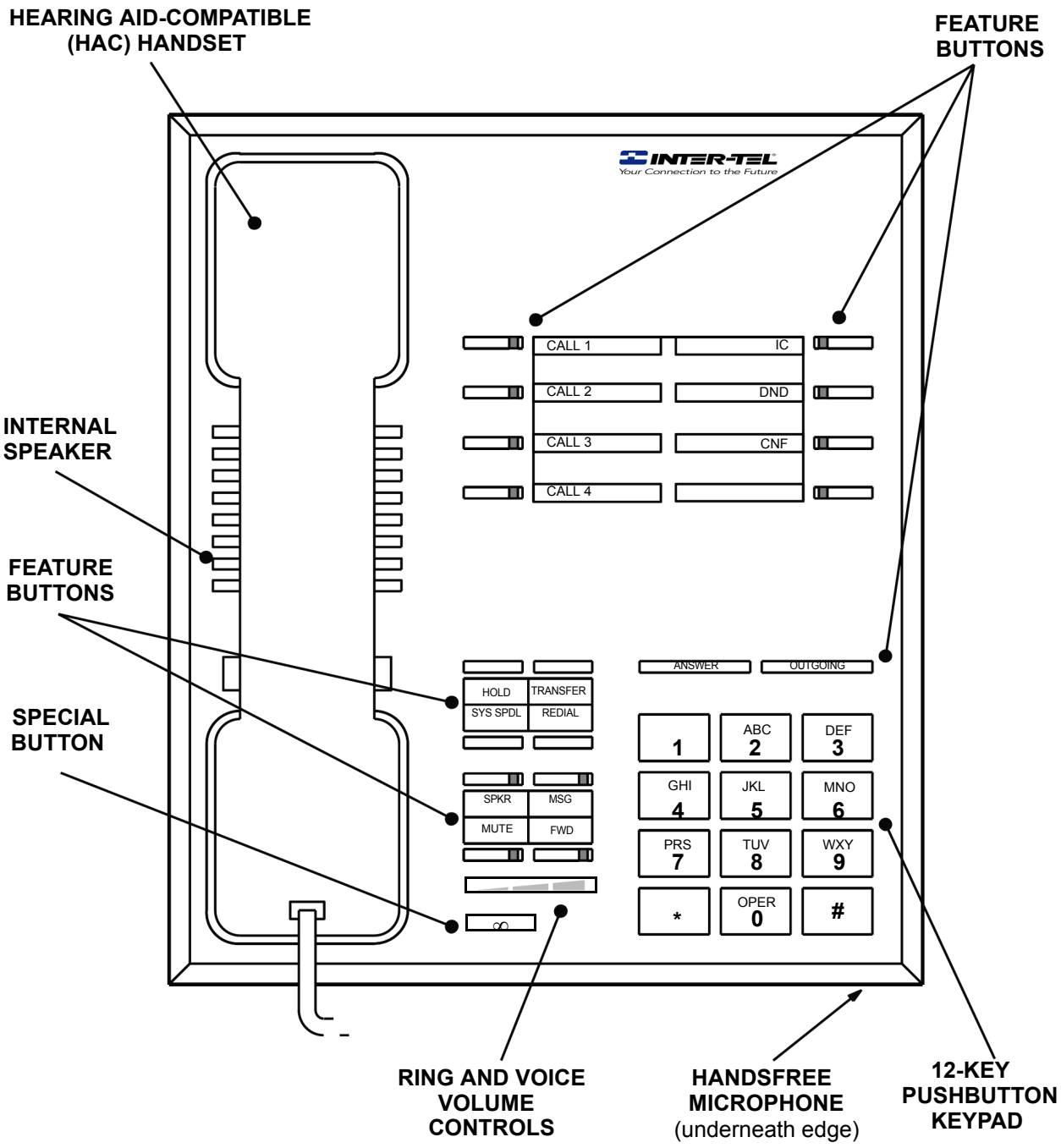
Figure 2-14. *Associate Display Phone*

Figure 2-15. *Model 8520*



NOTE: The buttons are shown as they appear in the default keymap configuration.

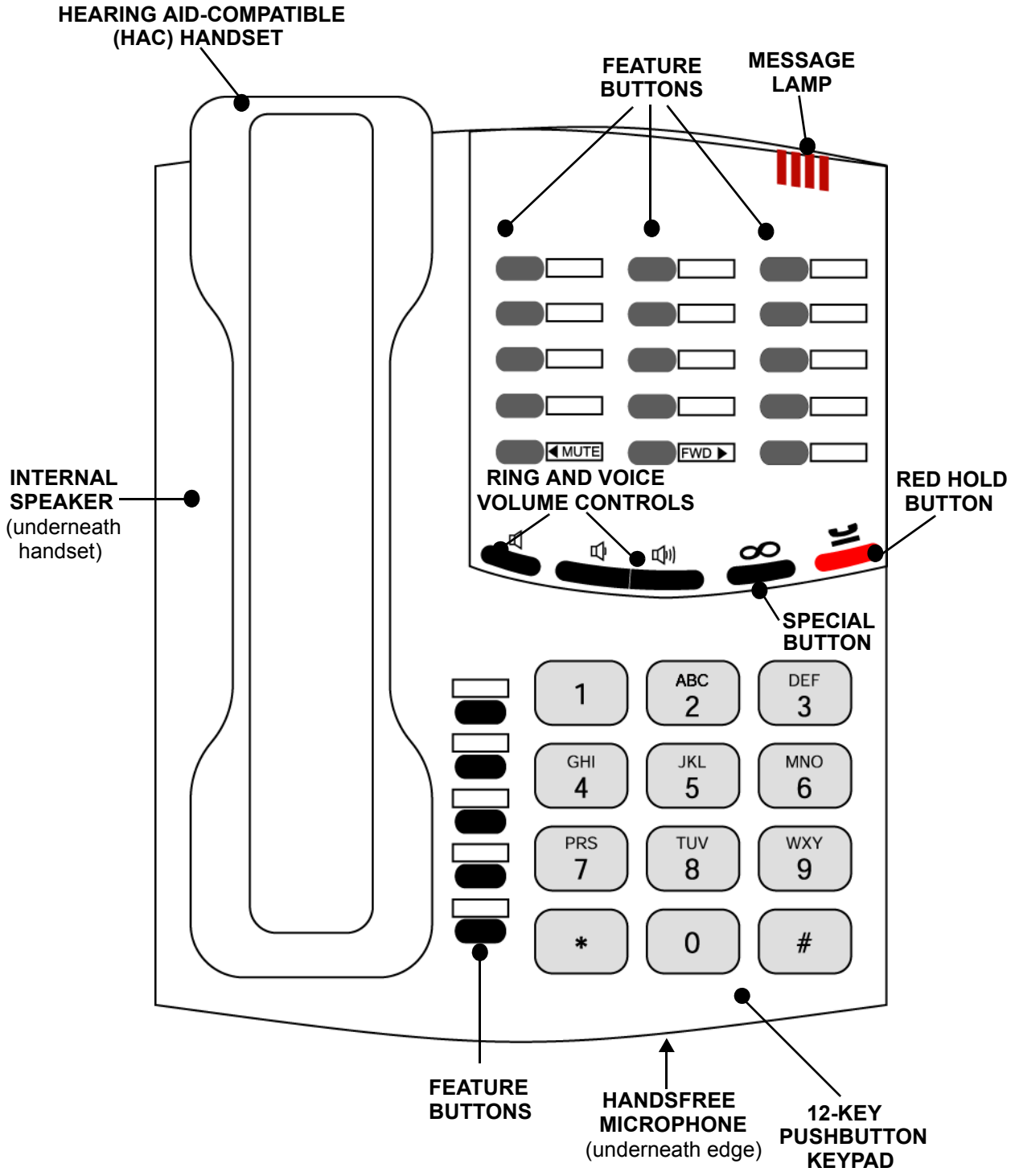
Figure 2-16. *Axxess Basic Digital Phone*



NOTE: The buttons are shown as they appear in the default keymap configuration.

Figure 2-17. *Eclipse Basic Digital Phone*

Figure 2-18. *Model 8500*



NOTE: The buttons are shown as they appear in the default keymap configuration.

Figure 2-19. *60-Key DSS/BLF Unit*

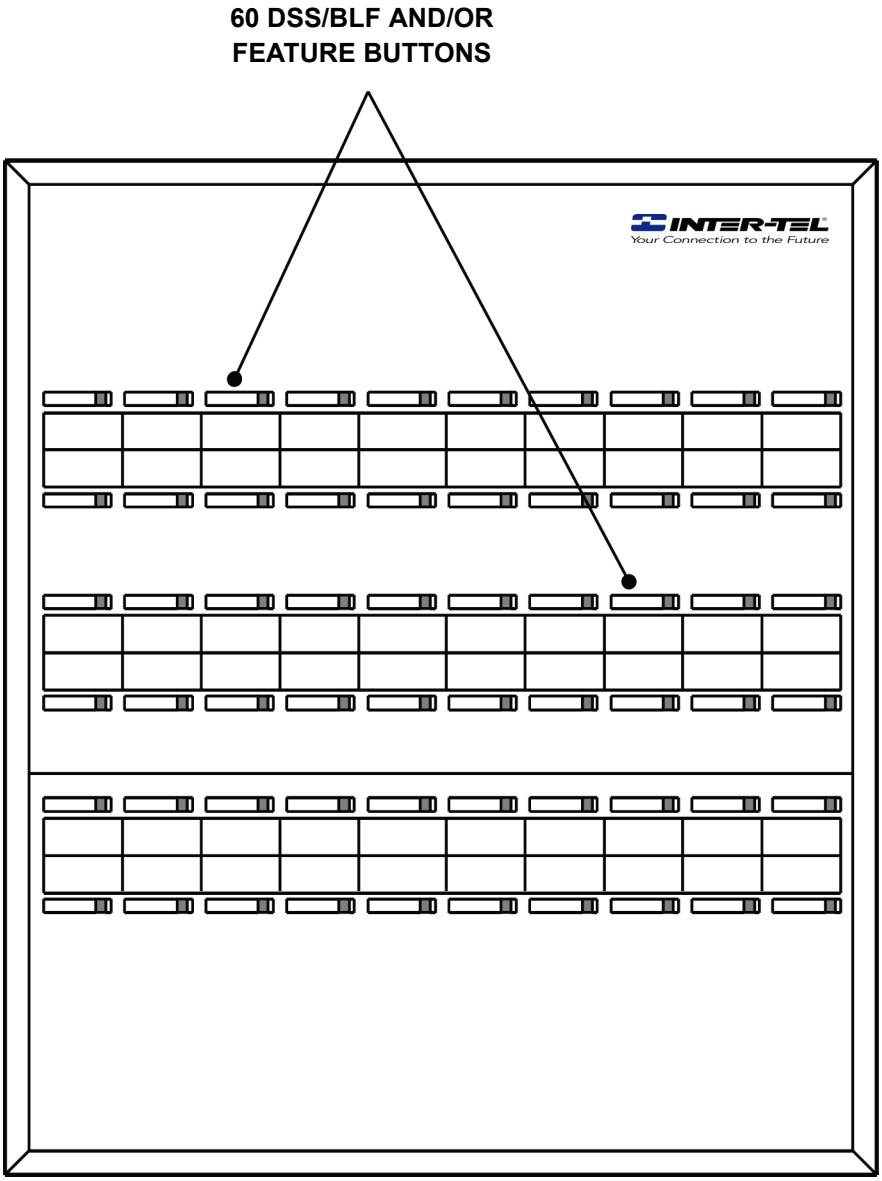


Figure 2-20. *50-Key DSS/BLF Unit*

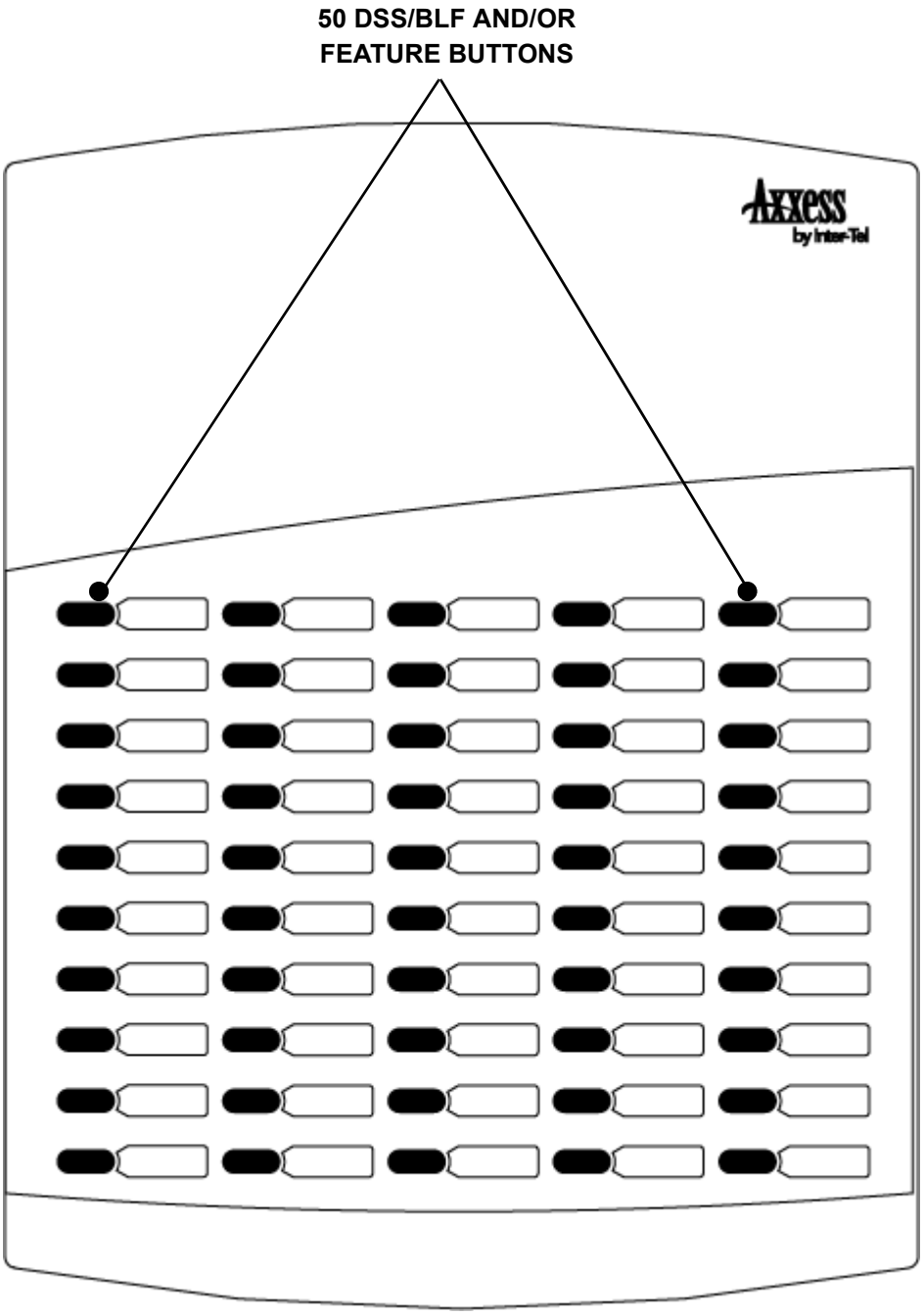
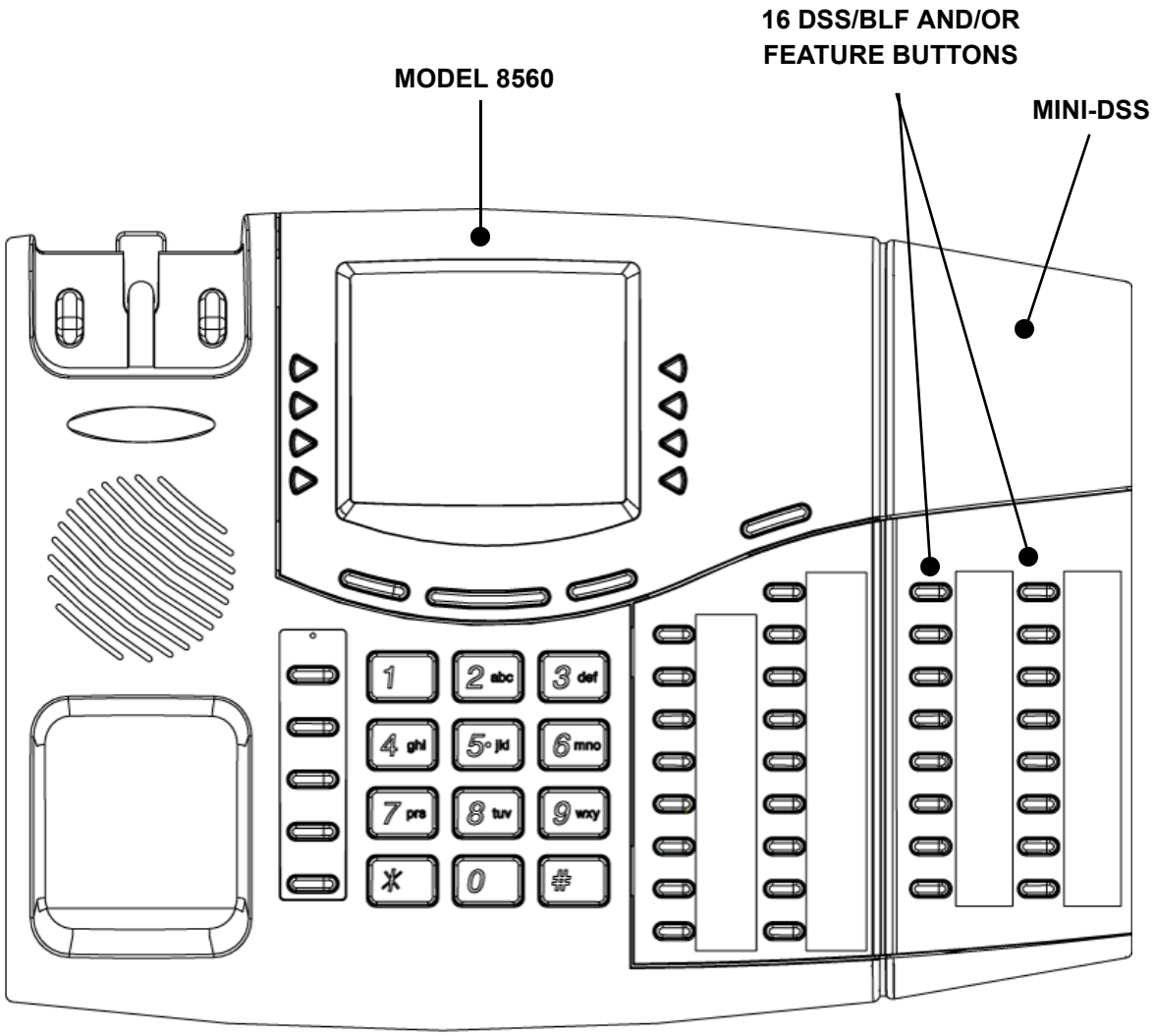


Figure 2-21. *Mini-DSS and Model 8560 Phone*



Installation

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1. INTRODUCTION

1.1 This section describes the recommended procedures for installing the Inter-Tel® phone system and Voice Processing PC. See the SPECIFICATIONS section of the manual for hardware descriptions.

CAUTION

This *Inter-Tel Installation and Maintenance Manual* instructs certified field technicians on the proper installation practices for the Inter-Tel phone system. This manual does not provide step-by-step instructions for premises wiring practices as dictated by the National Electrical Code [BS 6701 in Europe], which includes, but is not limited to, cable layouts, cable installation, AC power installation, proper AC grounding, eliminating or preventing external interferences (including, but not limited to, RFI, EMI, lightning, AC power disturbances, static discharge), and other telephony practices standard within the industry. Cable installers, electricians, and field technicians are expected to be properly trained and, if applicable, licensed in their trade practices. *Official Inter-Tel system certification is required for technical assistance.*

2. NEW SYSTEM INSTALLATION OUTLINE

2.1 System installation is performed in the following order. Detailed instructions and figures for each step are located throughout the INSTALLATION section.

1. Plan the installation, including the equipment chassis (with the ATM Switch, if applicable) and main distribution frame (MDF) location, station locations, cable runs, and optional equipment.
2. Run cables to the phones, single-line sets, and playback devices. Run wiring to any optional equipment, such as the Voice Processing PC, power failure transfer equipment, external paging equipment, general signaling devices, music source, etc.

   NOTE: DSS/BLF Units do not require a chassis cable run that is separate from the phone's cabling. DSS/BLF Units are connected directly to phones equipped with PC Data Port Modules, and they are powered by their own separate AC transformers.

3. Terminate the station cables on modular jack [line jack units in Europe] assemblies at the station locations.
4. Mount the MDF backboard and assemble the terminal blocks on the backboard. If any T1 or E1 and/or Primary Rate Interface (PRI) trunks are to be installed, terminate them on eight-conductor modular jack (line jack unit) assemblies at the MDF. If installing any single-line sets or non-T1 DID trunks (U.S. only), you may want to attach the necessary Single-Line Adapters (SLAs) and their corresponding modular jack (line jack unit) assemblies on the backboard.

   NOTE: The T1/E1 or PRI trunk is typically connected to the standard RJ48C modular jack [RJ45 line jack unit] provided on the T1, E1/PRI, or T1/E1/PRI card. However, in certain situations, it may be more convenient to use the card's DB15 connector instead. See page 3-37 for more information.

5. Connect the trunks, station cables, and circuit card cables to the corresponding terminal blocks [or, for T1 or E1 and PRI, modular jack assemblies, or RJ45 sockets in Europe].
6. Perform the station loop resistance test for each station cable (as described on page 3-69).

7. Position the equipment chassis near the MDF backboard or mount the chassis on the backboard.
8. Install the circuit cards in the chassis and connect them to the cables that run from the MDF blocks. For an ATM-based system, connect the cables from the PCMA cards and CP Server to the ATM Switch.
9. Install the station instruments and any optional station equipment, such as headsets, handset amplifiers, digital PC Data Port Modules, analog Data Port Modules, etc.
10. Install any optional system equipment, such as the Voice Processor, output device(s) for station message detail recording (SMDR) and error message recording, external paging network, external music source, etc.
11. Ensure that all equipment is working properly.
12. See the PROGRAMMING section of this manual to configure and program the system.

3. NETWORK INSTALLATION CONSIDERATIONS

3.1 To install a network, you need the following:

- **Software License**: As of version 8.0, premium features, including networking, are controlled by a software license. The license uploaded to the system must have the appropriate networking feature.

  NOTE: If installing a version 7.0 or earlier system, every node in the network must have the special networking PAL installed. Versions 7.0 and earlier have PALs in several unit values that support networking.

- **T1 or E1 Service**: For each network span, order a clear channel, point-to-point T1 or E1/PRI span from the service provider (not PRI service, even though you are using T1, E1/PRI, or T1/E1/PRI cards).
- **T1/PRI, E1/PRI, or T1/E1/PRI Cards**: Each network span requires two T1/PRI, E1/PRI, or T1/E1/PRI cards, one for each end of the span. The cards require the most recent firmware. Use upgrade kit part number 828.1521 (T1C) or 828.1522 (E1C) that contains required software components.
- **CPCs or CPU Cards**: If you use the newer CPU cards, you do not need to use upgrade kits. The new CPUs include the EAC connector and extended memory. Call Processing Cards (CPCs) do not require an upgrade as well. (See REPLACEMENT PARTS for part numbers.)

3.2 If desired, you can create a private network using the networking Internet Protocol Resource Card (IPRC). This card allows private networking over an IP network and is a lower-cost solution than the IP Networking Module (3200N). If using the IPRC, the T1 or E1 service and cards are not required.

3.3 Installation recommendations include the following:

- Each node ***must*** have at least one central office [local exchange in Europe] trunk. Emergency calls must be programmed to use the local trunk.
- All nodes should have access to the node where the Voice Processing Unit is located, with a minimum number of hops through other nodes. This has a significant impact on the network layout.

NOTE: If using version 6.0 or later in a networked system, all other nodes must be running version 5.2F9 (or later) or 5.349 (or later).

3.4 The network does not allow one node to use another node's resources (DSP resources, serial ports, etc.) with the exception of external page ports, SMDR output ports, and message print output ports.

4. INSTALLATION GUIDELINES FOR ADDING TO OR UPGRADING A SYSTEM

4.1 When adding to or upgrading an Inter-Tel system, you should follow the guidelines in the following sections. Detailed instructions and figures for each step are located throughout the INSTALLATION section.

4.2 The following steps outline the procedures you should perform when upgrading or adding to any Inter-Tel system. For details about version-specific information, see Appendix E — Upgrading to v11.0.

1. If the system needs to be turned OFF, advise the customer and plan to do so when it has the least impact on their operation.
2. If the changes affect the CPC, CPU card, or the customer's database, perform a complete save of the customer's database. If the Voice Processing Unit is affected, also save the Voice Processing database and prompts.

   NOTE: If performing an upgrade, follow the instructions provided with the software for saving, converting, and restoring software.

3. If adding an equipment chassis, power supply, ATM Switch, or Voice Processor, see the corresponding sections in this chapter for installation information.
4. Make sure the software license has the appropriate features required for the installation or upgrade. If the license does not match Database (DB) Programming, an error occurs (see page 6-45 for more information).
5. If new cabling is required, run cables to the stations and run wiring to any optional equipment. Then, assemble the new terminal blocks on the backboard. If any T1or E1 and/or Primary Rate Interface (PRI) trunks are to be installed, terminate them on eight-conductor modular jack (line jack unit) assemblies at the MDF. If installing any single-line sets or non-T1 DID trunks, it may be desirable to attach the necessary Single-Line Adapters (SLAs) and their corresponding modular jack (line jack unit) assemblies on the backboard.

   NOTE: With a U.S. system, the T1 or E1/PRI trunk is typically connected to the standard RJ48C modular jack provided on the T1, E1/PRI, or T1/E1/PRI card. However, in certain situations, it may be more convenient to use the card's DB15 connector instead. See page 3-37 for more information. With a European system, the PRI trunk is typically connected to the standard RJ45 line jack unit provided on the E1/PRI or T1/E1/PRI Card.

6. Connect any new trunks, station cables, and circuit card cables to the corresponding terminal blocks (or, for T1/E1 and/or PRI, modular jack assemblies or RJ45 sockets).
7. Perform the station loop resistance test for each station cable (as described on page 3-69).
8. Install any new circuit cards in the chassis and connect them to the cables that run from the MDF blocks.
9. If necessary, upload the system software into the flash memory on the CPC or the CPU. Also load new Voice Processing software, if required.
10. Install any new station instruments and any optional station or system equipment.
11. Ensure all equipment is working properly.

12. If adding Voice Processing (VPC) or Fax cards to the Voice Processing Unit, install them according to the instructions in this chapter. Verify the switch settings, cable connections, and card options.

13. See the PROGRAMMING section of this manual to restore the database and configure and program the additions to the system. If you added an Options Card (OPC) for additional voice channels, make sure the voice channels are programmed correctly in the database. If you upgraded to an ATM-based system, use the Database Merge Utility to update the database (see page 6-23).

14. Train the end users on any new features or options that have been installed.

5. PRE-INSTALLATION CHECKLIST

5.1 To make installation easier, use the following checklist when preparing to install the system. (Hardware specifications are included in the SPECIFICATIONS section.)

A. ESTABLISH SUITABLE ENVIRONMENTAL CONDITIONS FOR THE SYSTEM

- Locate the equipment chassis within 5 feet (1.5 meters) of an isolated, dedicated, 105-125VAC, 57-63 Hz, 15A, single-phase commercial power source.

NOTE: This *must* be an isolated, dedicated AC circuit for proper operation. All three wires (power, neutral, and ground) must be run separately from the outlet to the breaker panel without being bonded to any other wire or circuit. DO NOT plug any other equipment into this outlet. To maintain the protection provided by the isolated, dedicated circuit, the length of the AC power cord limits the distance between the power supply and the outlet; DO NOT use an extension cord. DO NOT defeat (or leave floating) the ground wire in the AC power cord. Also, to protect the system from AC voltage surges, a surge/spike protector is recommended (see page 2-24 for specifications).

- Select the chassis location to minimize cable run length. Station instruments connected to the system must not exceed the limits (using 24AWG, 24 gauge thickness or larger -- smaller number gauge = larger thickness) listed in the table on page 3-69. The resistance values (in ohms) are loop measurements; the distance values (in feet/meters) are the maximum one-way measurements from the chassis.
- Do not expose the equipment chassis to direct sunlight, high humidity, heat, dust, or strong magnetic fields (such as those generated by heavy motors and large copy machines).
- With a U.S. system, the MDF requires at least a 4 × 6-foot (1.2 × 1.8-meter), ¾-inch (2 cm) plywood backboard. This should provide sufficient room for all blocks and peripheral equipment.

NOTE: If installing a number of AC transformers and SLAs (along with their corresponding modular jack assemblies) at the MDF, a larger backboard may be necessary.

- For cooling purposes, provide ample air space around the chassis assembly (minimum one foot on all sides). Do not place or store items on top of the equipment chassis.
- The SMDR/error message output device(s) must not have cables longer than 50 feet (15 meters).
- Locate the equipment in a climate-controlled room with the following environmental conditions.

Table 3-1. *System Environmental Requirements*

Requirements	In Operation	In Storage
Temperature (Equipment Chassis)	32° to 104 F (0° to 40° C)	40° to 185° F (-40° to 85° C)
Temperature (Station Instruments)	32° to 113 F (0° to 45° C)	-40° to 185° F (-40° to 85° C)
Relative Humidity (Non-Condensing)	5% to 95%	5% to 95%
Altitude	Up to 10,000 ft (3,048 m)	Up to 40,000 ft (12,192 m)

NOTE: Locate the equipment in a climate-controlled room. Do ***not*** exceed the maximum operating temperature (as stated above). Therefore, when installing the chassis and station instruments, allow a sufficient margin for error in case of air conditioning failure, routine maintenance, plant shutdown, etc. As a general rule, if conditions are suitable for office personnel, they are also suitable for all equipment and station instrument operation. A properly controlled environment helps to extend the operating life of the equipment.

From UL 1459, a product safety specification governing telephone equipment:

- Never install telephone wiring during a lightning storm.
- Never install telephone jacks in wet locations unless the jack is specifically designed for wet locations.
- Never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
- Use caution when installing or modifying telephone lines.

B. ASSEMBLE THE NECESSARY TOOLS AND SUPPLIES

- **Digital and Executone® Phones**: Industry-standard, one-pair (two-conductor) twisted cable to run from the MDF to all station instruments and four-conductor modular jack (line jack unit) assemblies for all station instruments, including phones, single-line DTMF or pulse-tone sets, and playback devices (version 2.x or later is required for pulse-tone sets). (For exceptions, see the NOTE following paragraph 6.5 on page 3-16.)

NOTE: Executone Phones are not supported in Europe.

- **Multi-protocol endpoints operating in SIP mode**: A v1.1 or later SIP Server is required. See the *SIP Server Installation and Configuration Manual* (document part no. 835.2605) for details.
- **Models 8664, 8665, 8668 wireless endpoints**: A NetLink SVP (SpectraLink® Voice check box Priority) server is required. See the *NetLink SVP Server and Model 8664/8665/8668 Installation Manual* (document part no. 935.4521) for details.
- **Analog Phones**: Industry-standard, three-pair (six-conductor) twisted cable to run from the MDF to all analog phones and six-conductor modular jack (line jack unit) assemblies for all station instruments.
- A high-impedance digital multimeter to check the power supply and ensure correct wiring of the modular jack (line jack unit) assemblies. Multimeter accuracy must be ±0.5% or better.
- With a U.S. system, standard terminal blocks (66M1-50 type) and bridging clips for constructing the MDF. With a European system, Krone blocks for the MDF jumper field.
- With a U.S. system, industry-standard, 25-pair cable for connecting the MDF terminal blocks to the circuit cards in the equipment chassis.
- Industry-standard, cross-connect wire [industry-standard jumper in Europe] may be needed for certain applications.

- With a U.S. system, 50-pin female amphenol-type connectors (with recommended non-conducting, plastic covers) and a connecting machine. With a European system, 50-pin female amphenol-type connectors (with recommended non-conducting, plastic covers) pre-wired to Krone blocks or without Krone blocks for Patch Panel cabling
- Ferrite beads for all station, trunk, and RS232 cables, if necessary. (See SPECIFICATIONS, page 2-7.)
- Gas discharge tubes with silicon avalanche suppressors for lightning protection on the CO [local exchange], DID (U.S. only), and OPX trunks.
- Grounding terminal and 10AWG (10 gauge thickness or larger -- smaller number gauge = larger thickness) wire for grounding the chassis, Single-Line Adapters, etc.
- Standard telephone hand tools and the mounting hardware for the MDF backboard, trunk and station blocks, modular jack (line jack unit) assemblies, etc.
- For on-site or stand-alone/remote system programming, an IBM-compatible personal computer (see page 2-99 for specifications).

5.2 If installing any T1 or E1 and/or Primary Rate Interface (PRI) trunks: Depending on the type of T1/PRI or E1/PRI termination used by the telephone company [network provider] and how close it is to the MDF, the following tools and supplies are also needed.

NOTE: With a U.S. system, the T1 or PRI trunk is typically connected to the standard RJ48C modular jack provided on the T1 or T1/E1/PRI card. However, in certain situations, it may be more convenient to use the card's DB15 connector instead. See page 3-37 for more information. With a European system, the E1 or PRI trunk is typically connected to the standard RJ45 line jack unit provided on the E1 or T1/E1/PRI Card.

5.3 For T1/PRI trunks terminated on RJ48C jacks near the MDF or for E1/PRI trunks terminated on RJ45 jacks near the MDF:

- Four-pair, non-reversing, mod-to-mod line cords (one for each T1/PRI trunk installed)
- For T1/PRI or E1/PRI trunks terminated on RJ48C [RJ45] jacks away from the MDF (see Figure 3-13 on page 3-32)
- Eight-conductor modular jack (line jack unit) assemblies – two for each T1/PRI or E1/PRI trunk installed
- Four-pair, non-reversing, mod-to-mod line cords (two for each T1/PRI or E1/PRI trunk installed)
- T1 or E1 cable – enough to extend from each telco RJ48C [RJ45] jack to the MDF (see page 2-54 for cable specifications)

5.4 For T1/PRI or E1/PRI trunks terminated on an RJ-type block:

- Eight-conductor modular jack (line jack unit) assemblies – one for each T1/PRI or E1/PRI trunk installed. (See Figure 3-15 on page 3-34)
- Four-pair, non-reversing, mod-to-mod line cords (one for each T1/PRI trunk installed)
- T1 or E1 cable – enough to extend from each telco [network provider] termination to the MDF (see page 2-54 for cable specifications)

C. PLAN STATION LOCATIONS AND TYPES OF STATION INSTRUMENTS

- Prepare a cable identification plan using station *circuit* numbers; do not use extension numbers (see page 3-16 for details).
- **Phones**: The expanded, quad-chassis, Inter-Tel system has the capacity for up to 448 digital and/or analog/Executone phones. A fully-equipped ATM system can support up to 896 digital and/or analog/Executone phones. (See page 2-72 for a listing of the available phone models and for more information on maximum station capacities.)

NOTE: Executone phones are not supported in Europe.

- **DSS/BLF Units**: Up to 20 digital DSS/BLF Units can be installed on the system. A single phone can be equipped with up to four units. In addition, the Executone IDS phones support a 48-key analog DSS/BLF Console.

NOTE: Analog DSS/BLF Units cannot be installed on the European system. Also, digital DSS/BLF Units do not require a cabinet cable run that is separate from the phone's cabling.

- **Mini-DSS Units**: Model 8520 and 8560 Phones can connect directly to a Mini-DSS. The number of Mini-DSS units is not limited by system resources.
- **Single-Line Sets**: Up to 512 industry-standard, single-line DTMF or pulse-tone sets can be installed on a quad-chassis system. (Version 2.x or later software is required for pulse-tone sets.) For an ATM-based system, up to 1792 single-line sets can be installed. Single-line sets are connected to Single-Line Cards (SLCs), or they can be connected to Single-Line Adapters (SLAs), which are in turn connected to Digital Keyset Card circuits.
- **Playback Devices**: Playback devices can be used in place of single-line stations. If the Voice Processing PC is not used, playback devices are especially useful as hunt group overflow/announcement stations. (See page 2-87 for specifications.)
- **Attendant Consoles**: If desired, up to 30 digital Attendant Consoles can be installed on the system. The consoles are connected to the system through TCP/IP connection via a System OAI Server (CT Gateway, CP Server, or System OAI Switch Transceiver)

NOTE: Analog Attendant Computer Consoles cannot be installed on the system. Also, version 7.0 or later software does not support Attendant Console version 2.0. If running version 7.0 or later, Attendant Console version 3.0 is required. For additional information on the Attendant Console, refer to the user guide included with the console package.

D. PLAN THE ATM CONFIGURATION

5.5 Before installing an ATM system, you must know how to map the chassis to the appropriate boards in the system. The following table defines which PCMA board should connect to which chassis for proper operation.

Table 3-2. *PCMA Configuration*

PCMA/CPU #	Chassis #	Boards
1	1 and 2	1-16
2	3 and 4	17-32
3	5 and 6	33-48
4	7 and 8	49-64

5.6 When physically placing the system components, make sure all the PCMA boards and CP Server PCs have easy access to the ATM Switch. The voice mail unit should also be placed near the first chassis so that the voice mail PCM cables reach PCMA #1.

5.7 The CPU board in chassis 1 (slot 8) acts as the default CPU for the Music-On-Hold source and the first eight voice mail ports. This CPU is also the only available source for background music.

5.8 In addition, the ATM-based system requires one common Time Division Multiplexing (TDM) clock source for all backplanes. Because of this, the CP Server configures the system so that all backplanes are synchronized to a single public clock source, such as a T1 or E1 clock.

E. ASSEMBLE THE OPTIONAL EQUIPMENT

- Voice Processing PC and applications software for integrated Voice Processing (voice mail, automated attendant, hunt group announcement and overflow, optional Fax-On-Demand, SMDR storage, etc.).
- CT Gateway/Transceiver for using System OAI applications on a network.

NOTE: A CT Gateway is recommended for high-capacity systems, especially an ATM system. CT Gateway version 3.0 or later is required for system version 8.0 or later.

- Headsets (electret compatible) for phone stations and handset amplifiers for single-line set stations.
- Digital phone Personal Computer Data Port Modules (PCDPMs) for using Off-Hook Voice Announce (OHVA) features and for installing a digital DSS/BLF Unit and/or an optional Modem Data Port Module. If a DSS/BLF Unit is not installed, the module's RS232 port (4800-bps) can be used to connect an SMDR/error message output device or a personal computer for programming the system database.
- Digital phone Modem Data Port Modules (MDPMs) for connecting modem-equipped data devices or single-line sets.
- Analog phone Data Port Modules for hooking up either modem-equipped data terminals or loud ringing adapters and external signaling devices.
- AgentSets used in place of a phone with an attached PCDPM. It has a jack for plugging in an Inter-Tel handset or headset. It also has a serial connector (DB9) for attaching it to a PC. Because the AgentSet does not have feature buttons, a dialing pad, or a speaker, it must be connected to a PC that is equipped with an OAI product that emulates a phone.

- SMDR/error message output device(s) must not have cables longer than 50 feet (15 meters). (See page 2-100 for specifications.)
- System battery back-up – uninterruptable power supply or standby power supply. (See page 2-104 for specifications.)
- OPX repeater(s) for amplifying voice volume levels, if necessary. (See page 2-86 for specifications.)
- External music source.
- External paging speakers and amplifier(s). (See the CAUTION on page 3-242.)
- Power failure transfer equipment (customer-provided PFT relay card, card mounting case, and external power source).
- Electronic equipment (alarms, lights, etc.) to connect to the night transfer relay.
- Voice mail equipment. (See page 2-104.)
- Facsimile machine. (See page 2-104.)
- PC wall-mount shelf (accommodates most PC models); see page 8-9.

6. STATION CABLING

6.1 Floor plans should be developed to aid in proper station cabling in a star (home run) configuration from the equipment chassis. The cables are run from the station locations to the station blocks at the MDF. See page 2-7 for cabling requirements.

6.2 Both ends of each cable should be labeled with the station's circuit number (hardware address). The circuit number (XX.YY.ZZ) designates the station card's position in the equipment chassis (XX = 01-07, 09-15, 17-23, or 25-31), the circuit position on the card (YY = 01-16), and whether or not the circuit is a multi-purpose circuit (ZZ = 01-03; see next paragraph). For example, circuit number 04.06.01 identifies the sixth circuit on the card installed in slot number four.

NOTE: All circuit numbers are preceded with a two-digit node number. For example, 02.02.16.01 is a port on node 2, card 2, circuit 16.

6.3 In some circumstances, a station circuit can have more than one purpose. For example, two single-line devices can be installed on a single DKSC circuit. The first device is assigned circuit number XX.YY.01, and the second device is assigned circuit number XX.YY.02. Similarly, a phone's PC Data Port Module and Modem Data Port Module can be assigned their own circuit numbers. In this case, the phone is assigned circuit number XX.YY.01, the first data port is assigned number XX.YY.02, and the second data port is assigned number XX.YY.03. For more information, see page 6-118.

6.4 When the system is first powered up, the default configuration is used to assign extension numbers to the station circuits. For example, in dual-chassis systems, card slots 1-5 and 9-13 are configured as DKSCs, while slots 6-7 and 14-15 are configured as LSCs with attached Daughter cards (LSC-Ds). In tri-chassis and quad-chassis systems, card slots 17-23 and 25-31 are unequipped. Changes to the default configuration can be made through DB Programming. See PROGRAMMING for more information.

6.5 The default system assigns extension number 1000 to station circuit 01.01.01, extension number 1001 to station circuit 01.02.01, and so on to match the initial configuration. Circuit 01.01.01 is equipped as an administrator and as the primary attendant's phone when the system is in the default configuration. Through DB Programming, the circuits can be assigned 1-5 digit extension numbers.

NOTE: DSS/BLF Units do *not* require a DKSC circuit and equipment chassis cabling that is separate from the associated phone's circuit and cabling. Instead, each unit is connected directly to the phone (equipped with a PC Data Port Module) and is powered by a separate AC transformer.

A. RUNNING CABLE

6.6 The station cable is connected to the MDF at one end, and a modular connecting block at the other end. The modular line cord of the telephone is then plugged into the connecting block.

- **For devices connected to an EKSC**: From the MDF location, run industry standard, two-pair (four-conductor) twisted (24AWG, level-1 or better/24 gauge thickness or larger -- smaller number gauge = larger thickness) cable to Executone IDS and Wave phones connected to each EKSC. Both pair of wires are required for the telephone to function. One pair transmits digitized voice information and control signals from the system to the telephone. The other pair receives digitized voice information and control signals from the telephone.

NOTE: EKSCs are not supported in Europe.

- For devices connected to an AKSC: From the MDF, run industry standard, three-pair (six-conductor) twisted (24AWG, level-1 or better/24 gauge thickness or larger -- smaller number gauge = larger thickness) cable to *analog* phones.
- **For devices connected to an DKSC or SLC**: From the MDF location, run industry standard, one-pair (two-conductor) twisted (24AWG, level-1 or better/24 gauge thickness or larger -- smaller number gauge = larger thickness) cable to the digital phones, single-line phones, and playback devices connected to each DKSC, DKSC-16, or SLC.

NOTE: If desired, two- or three-pair twisted cable and four- or six-conductor modular jacks (line jack units) may be used for *all* station connections. This allows the various types of station instruments to be easily interchanged, if necessary. (If three-pair cable is used with a U.S. system, extra terminal blocks and the use of cross-connect wiring techniques are required.)

6.7 Follow these guidelines:

- Install proper type cable for the application according to the National Electrical Code and local building codes [BS 6701 in Europe].
- Avoid cable runs parallel to fluorescent light fixtures or AC lines not in conduit. If these obstacles are unavoidable, run the cables across them at right angles.
- Do not run station cables inside electrical conduit already occupied by AC power cable. (To do so is a violation of the National Electrical Code or BS 6701.)
- Do not run cables near equipment with electric motors or through strong magnetic fields, such as those generated by large copy machines, arc welding equipment, heavy motors, etc.
- Do not place station cables where they can be stepped on or where they can be rolled over by office furniture.
- If using multi-pair (e.g., 25-pair) cable runs to multiple station locations, do not include AC-ringing single-line sets, AC-ringing auxiliary equipment, or trunks in a cable being used for analog or Executone phones. Stations using analog phone circuits should be routed in separate multi-pair cable runs.
- Do not exceed the loop limit measurements (using 24AWG, 24 gauge thickness or larger -- smaller number gauge = larger thickness) for the station cable lengths as outlined in the table on page 3-69. The resistance values (in ohms) are loop measurements; the distance values (in feet/meters) are the maximum one-way measurements from the equipment chassis.

6.8 Because the phone system utilizes digital voice and data transmission between the digital phones and the chassis, there are some additional digital phone-only cabling requirements:

- Do not use shielded cable, and do not use cable smaller than 24AWG (24 gauge thickness or larger -- smaller number gauge = larger thickness).
- Do not parallel cables (e.g., do not "double up on the copper").
- Do not "bridge" or "Y" off the cable to another location.
- Unlike analog phones, the digital or Executone phones do not need to be routed in separate multi-pair cable runs. For example, if using 25-pair cable runs to multiple station locations, it is acceptable to include single-line sets, auxiliary equipment, and trunks in a cable being used for phones.

B. TERMINATING THE CABLES AT STATION LOCATIONS

Digital Phones, Single-Line Sets, and Executone Phones

NOTE: Executone phones are not supported in Europe.

6.9 Terminate the digital phone, single-line set, Executone phone, and playback device station cables on four-conductor modular jack [six-conductor line jack unit in Europe] assemblies at the station locations. (For exceptions to this, see the NOTE following paragraph 6.6 on the previous page.)

6.10 Do not mount the assemblies on the wall at this time; they are wall mounted later when the station instruments are installed. See Figure 3-1 through Figure 3-5 on the following pages for a wiring diagram.

CAUTION

If tip and ring are reversed when installing a phone, the phone does not function.

Analog Phones

6.11 Terminate the analog phone station cables on six-conductor modular jack (line jack unit) assemblies at the station locations.

6.12 Do not mount the assemblies on the wall at this time; they are wall mounted later when the station instruments are installed. See Figures 3-3 and 3-4 for a wiring diagram.

CAUTION

If the power pair (W/BL, BL/W) is reversed, installing a phone opens the associated thermistor on the Analog Keyset Card.

Figure 3-1. *Digital Phone/Single-Line Set Modular Jack Assembly Wiring (U.S.)*

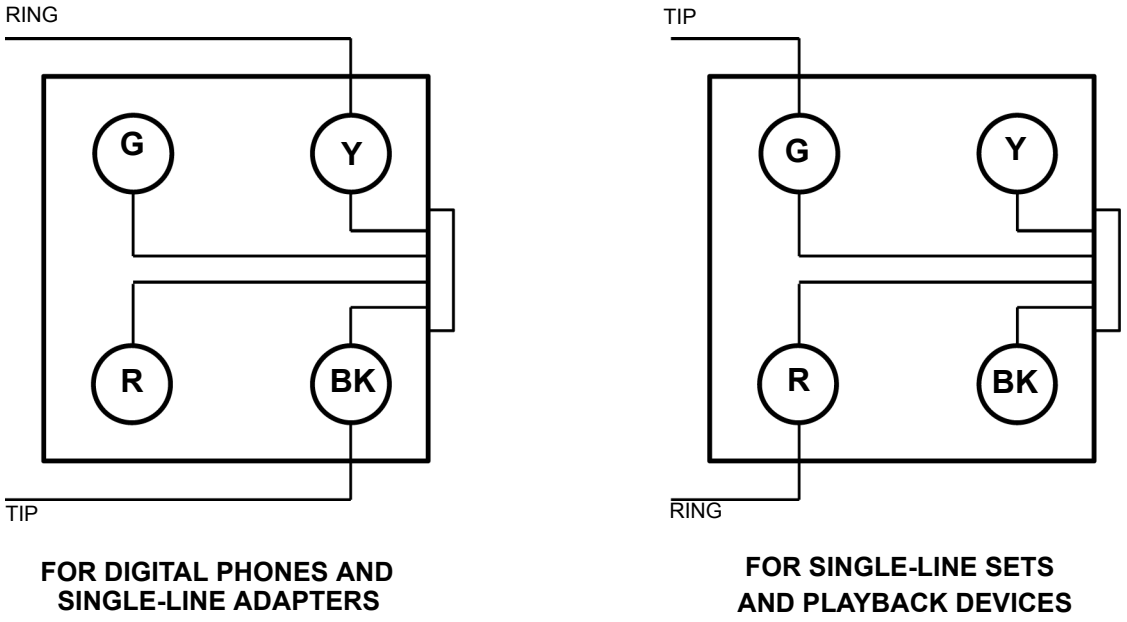


Figure 3-2. *Digital Phone/Single-Line Set Line Jack Unit Assembly Wiring (Europe)*

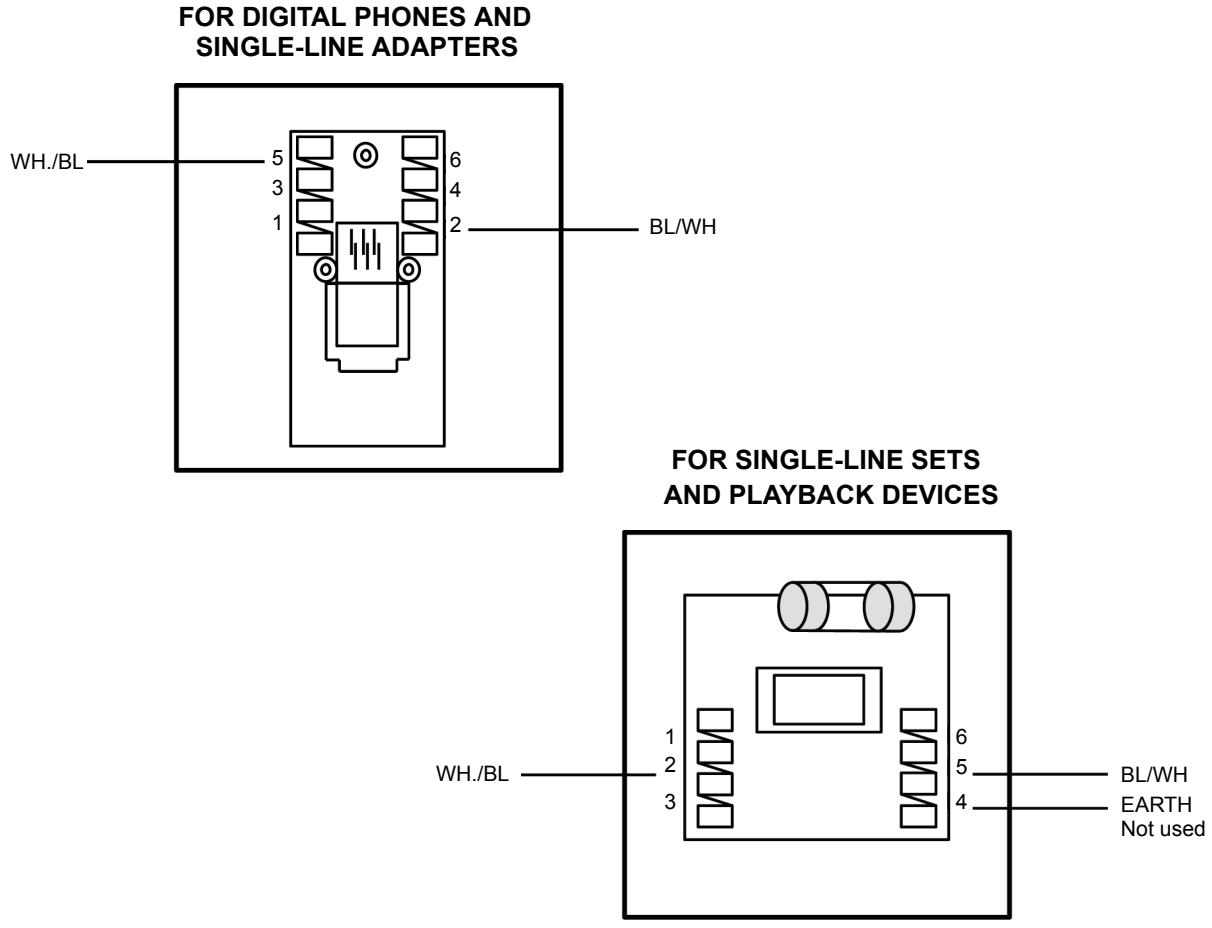


Figure 3-3. *AKSC Analog Phone Modular Jack Assembly Wiring (U.S.)*

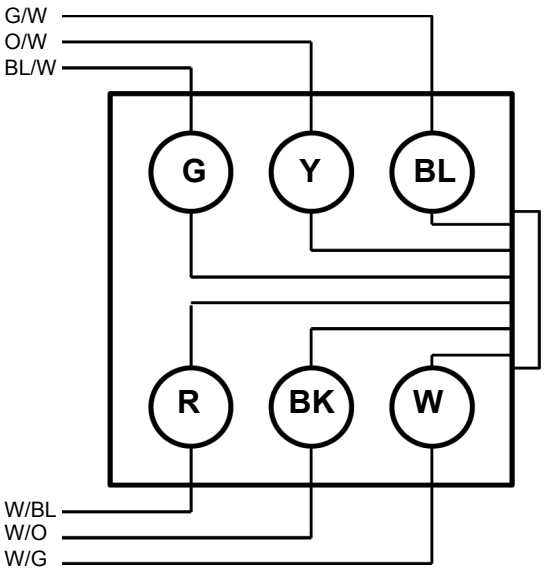
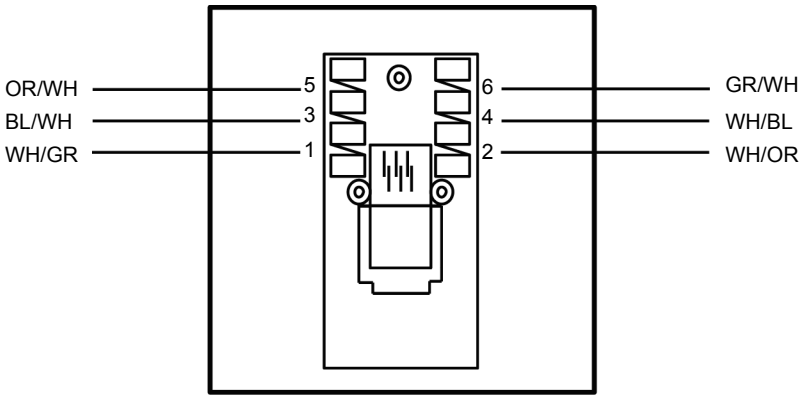
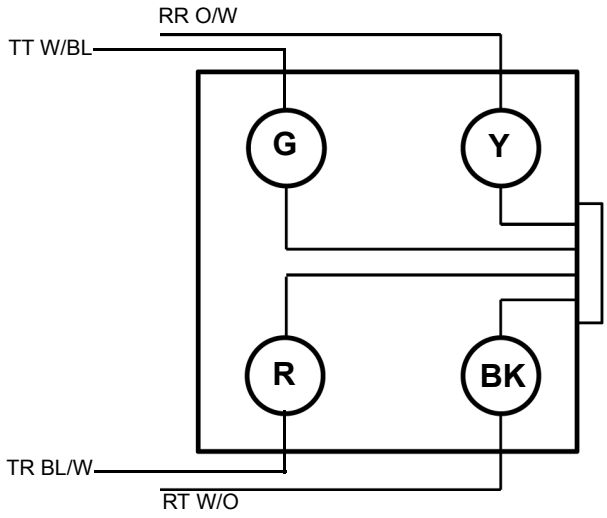


Figure 3-4. *Analog Phone Line Jack Unit Assembly Wiring (Europe)*



ANALOG PHONES

Figure 3-5. *EKSC Executone Phone Module Jack Assembly Wiring (U.S. Only)*



7. ASSEMBLING THE MAIN DISTRIBUTION FRAME (MDF) BACKBOARD

7.1 The MDF is the point at which the station instruments, trunks, and auxiliary system equipment are connected to terminal blocks [Krone blocks in Europe] (or, in the case of T1/E1 or PRI trunks, modular jack/line jack unit assemblies are used instead). These terminal blocks [Krone blocks] are then connected to the circuit cards in the Inter-Tel equipment chassis. It is extremely important that the connections be made carefully and accurately.

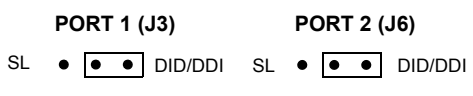
7.2 With a European system each card is normally supplied with 2 meter cable (4 meter cables are available by special order). These cables are supplied with the appropriate number of Krone blocks for the card type.

NOTE: To simplify the MDF installation instructions, this manual outlines basic one block-to-one card installation using bridging clips. However, it is quite acceptable, and in some situations more functional, to use additional blocks and then cross connect between them.

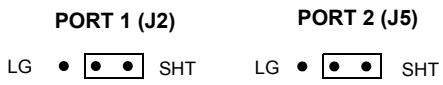
7.3 Assemble the MDF as follows:

1. If desired, mount at least a 4 × 6-foot (1.2 × 1.8-meter), ¾-inch (2 cm) plywood backboard at the MDF location. (Depending on the number of stations and trunks to be installed, a larger backboard may be necessary.)
2. With a U.S. system, attach the 66M1-50-type terminal blocks to the plywood backboard. With a European system, Attach the Krone blocks to the plywood backboard. See Figure 3-6 on page 3-23 for a sample MDF block layout and cable assignments with a U.S. system. Blocks include:
   - Up to 28 blocks for connecting CO trunks and the 25-pair cables for the Loop/Ground Start Cards (LGCs) and/or Loop Start Cards (LSCs).
   - Up to 28 blocks for connecting station cables and the 25-pair cables for the station cards – Digital Keyset Cards (DKSCs) and/or 16-Port Digital Keyset Cards (DKSC-16s) for digital phones and SLAs, Analog Keyset Cards (AKSCs) for analog phones, Executone Keyset Cards (EKSCs) for Executone IDS and Wave phones, and Single-Line Cards (SLCs) and/or 16-Port Single-Line Cards (SLC-16s) for single-line sets and playback devices.

     NOTE: If three-pair cable is used, extra terminal blocks and the use of cross-connect wiring techniques are required.
3. *If installing any T1/E1 and/or PRI trunks,* attach eight-conductor modular jack (line jack unit) assemblies to the MDF as required. (For complete information on installing T1/PRI or E1/PRI trunks, see page 3-31.)
4. *If installing any BRI trunks (BRSs or BRUs),* attach 50-pin Amp connectors to the MDF.
5. *If installing any single-line devices or DID [DDI in Europe] trunks*, attach the necessary SLAs and corresponding modular jack (line jack unit) assemblies to the plywood backboard. See Figure 3-12 on page 3-30 and Figure 3-27 on page 3-53 for complete diagrams of the SLA setup.
   a. **Changing the port type**: SLAs are shipped with both ports set for single-line device operation. If connecting a DID [DDI] trunk:
      1.) Remove the cover and set the corresponding port for DID [DDI] operation.



2.) If necessary, adjust the hybrid balance setting from short (<.6 miles/1km) to long (>.6 miles/1km).



3.) Replace the cover.

b. **Wall-mounting the unit**: The SLA can be placed horizontally on a table, or it can be wall-mounted using the two L-shaped brackets that accompany the unit, as follows:

1.) Remove the two screws on one side of the unit and use them to attach the two L-shaped wall mounting brackets (see Figure 3-7).

2.) Position the SLA on the MDF backboard and mark the location of the mounting screw holes on the wall. Set the SLA aside.

3.) Drive a screw into the center of each mounting hole marking, allowing the heads of the screws to protrude slightly.

NOTE: Use screws of sufficient strength to support the SLA.

4.) Hang the SLA on its screws. If necessary, adjust the screws to ensure the SLA is held firmly in place.

Figure 3-6. *Sample MDF Block Layout and Cable Assignments (U.S. Only)*

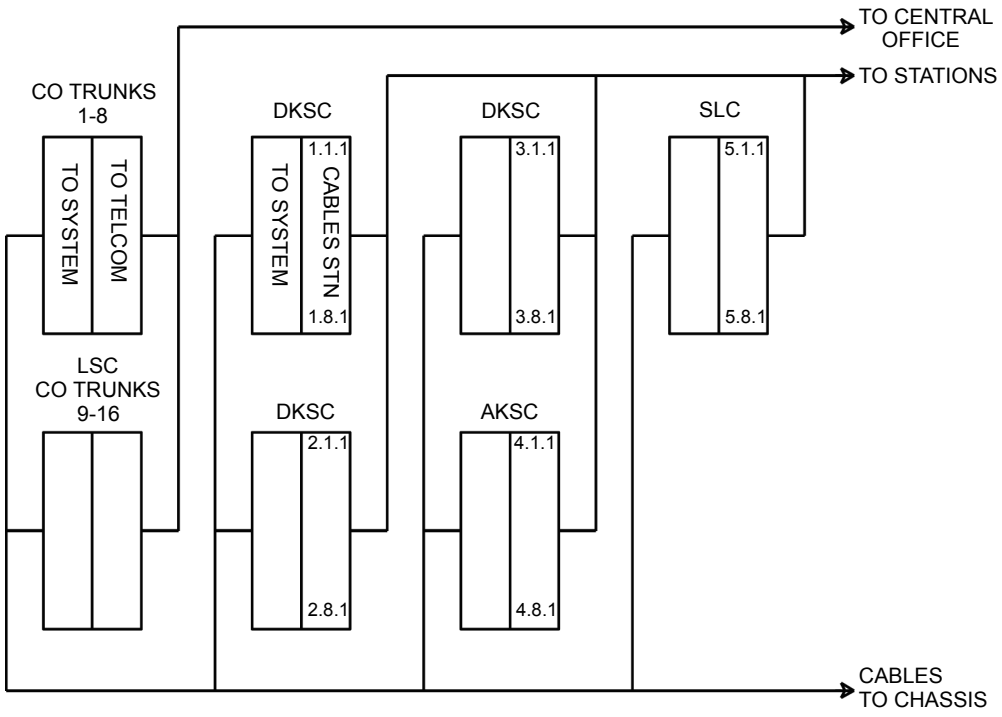
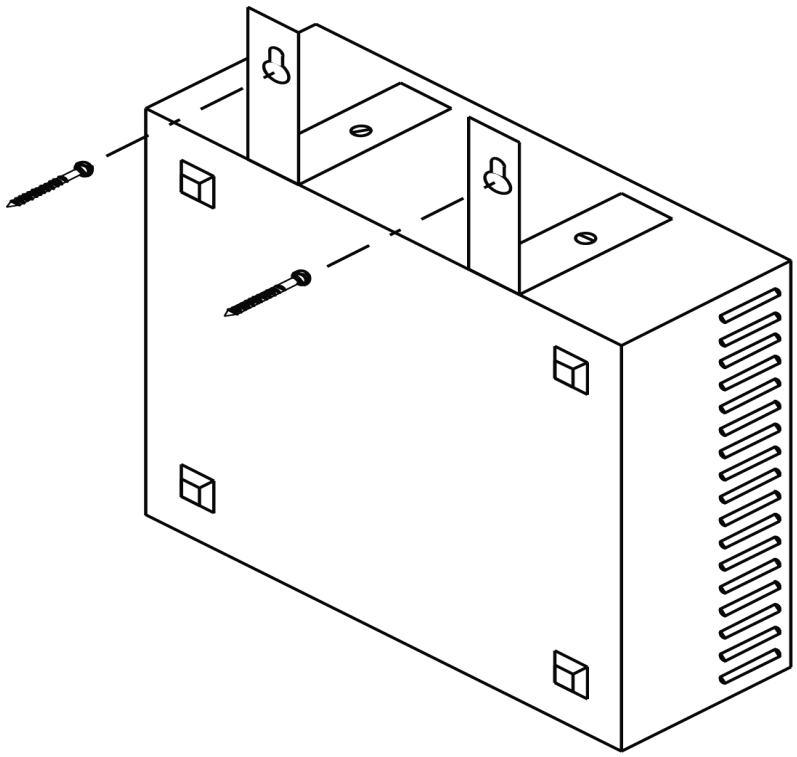


Figure 3-7. *Wall Mounting the SLA*



A. CONNECTING THE TRUNK CABLES TO THE MDF

CO Trunks in a U.S. system

7.4 Connect the CO trunks and Loop/Ground Start Card (LGC) and/or Loop Start Card (LSC) cables as outlined below. Note that step 2 mentions two installation methods. Although the first method is more flexible, the second method can be used to minimize the number of terminal blocks that are used. Refer also to Figures 3-9 and 3-10 on pages 3-27 and 3-28.

1. Cross connect the CO trunks from the telephone company RJ-type blocks to the *right* side of the LGC/LSC blocks.

   NOTE: For additional lightning protection, see paragraph 7.9.

2. *Method A:* Using enough 25-pair cable to run from the LGC/LSC blocks to the cards, make the termination cables. Attach a 50-pin female amphenol-type connector to one end of each LGC/LSC cable. Label each connector and cable end with the appropriate slot number (e.g., SLOT-6). These connectors are attached to the cards after they are installed in the equipment chassis.

   Method B: Using enough 25-pair cable to run from the LGC/LSC blocks to the cards, make the special termination cables. Attach up to three, 50-pin female amphenol-type connectors to one end of each LGC/LSC cable as shown in Figure 3-10. Connect CO trunks 1-4 (or 1-8 if a LGC or LSC Daughter Card is installed) to one connector, trunks 5-8 (or 9-16) to another connector, trunks 9-12 (or 17-24) to a third connector, and so on for each cable. Label each connector with the appropriate slot number (e.g., SLOT-6). These connectors are attached to the cards after they are installed in the equipment chassis.

3. *If necessary,* using standard electrical tape, tape two ferrite split beads (part no. 808.1195) around each 25-pair cable just below the amphenol-type connector (as shown in Figure 3-8 on page 3-26).

NOTE: Ferrite beads may be necessary to help reduce EMI and RFI.

4. Terminate the other end of each 25-pair cable on the *left* side of the corresponding LGC/LSC block.
5. Install bridging clips to complete the connections.

CO Trunks in a European system

7.5 Some trunks are connected via the fallback system. See page 3-73 and Figures 3-36 and 3-37.

7.6 The trunk connections are made to the designated Krone blocks. See Figure 3-11 on page 3-29 and Figure 3-8 on page 3-62.

NOTE: For additional lightning protection, see page 3-26.

7.7 Attach the two ferrite split beads supplied around each cable just below the amphenol type connector (as shown in Figure 3-8).

DID Trunks (U.S. Only)

7.8 Connect the Direct Inward Dialing (DID) trunks to the Single-Line Adapters as outlined below. See Figure 3-12 on page 3-30 for a diagram.

NOTE: DID trunks are also supported through T1 circuits. In addition to DID trunks, SLAs can also be used to install single-line sets, playback devices, and off-premises stations. Any combination using the two ports on the SLA is possible. See pages 3-44 to 3-69 for station installation instructions.

1. Mount one four-conductor modular jack assembly next to each telephone company RJ jack.
2. For each modular jack assembly mounted in step 1, mount a corresponding four-conductor modular jack assembly near the SLA location.
3. To connect the modular jack assemblies mounted in steps 1 and 2:
   a. Run one-pair cable between the corresponding modular jack assemblies.
   b. Wire each end of the one-pair cables onto their respective modular jack assemblies. See Figure 3-12 on page 3-30 for a diagram of the wiring.
4. Plug one end of a two-pair, mod-to-mod line cord into each RJ jack and plug the other end into the corresponding modular jack assembly beside it.
5. At the MDF backboard, plug one end of a two-pair mod-to-mod line cord into each modular jack assembly. The other end is *later* plugged into the corresponding DID jack on the SLA.

Lightning Protection

7.9 For additional lightning protection, install gas discharge tubes with silicon avalanche suppressors to ground (grounding rod or copper, cold water pipe) on each CO, DID, and OPX trunk. This must be done external to the system. Each gas discharge tube is installed directly between the telephone company [network provider] RJ-type block and the system's CO/DID/OPX input. This protection should provide energy absorption and filter low-level surge potentials.

7.10 Also, in areas with frequent occurrences of lightning, the cable between the telephone company [network provider] RJ-type blocks and the gas discharge tubes must be at least 75 feet (22.5 meters) long (the cable may be coiled up if desired).

Figure 3-8. *Example of Ferrite Bead Installation*

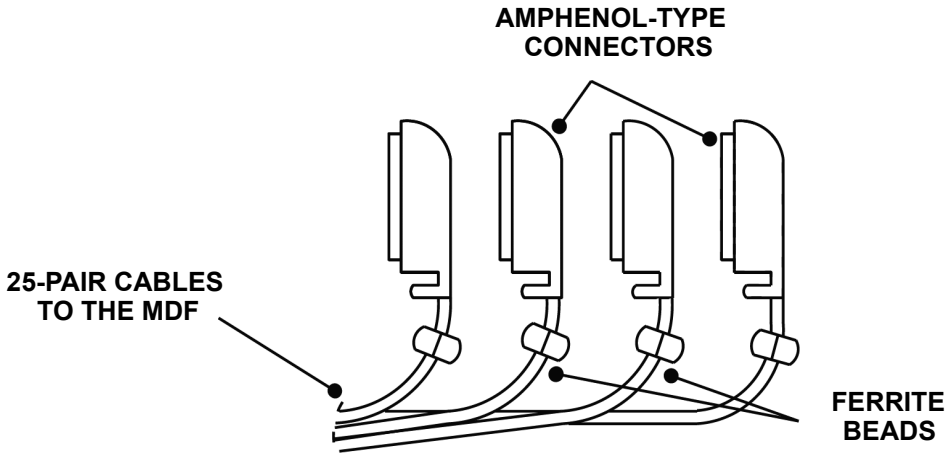
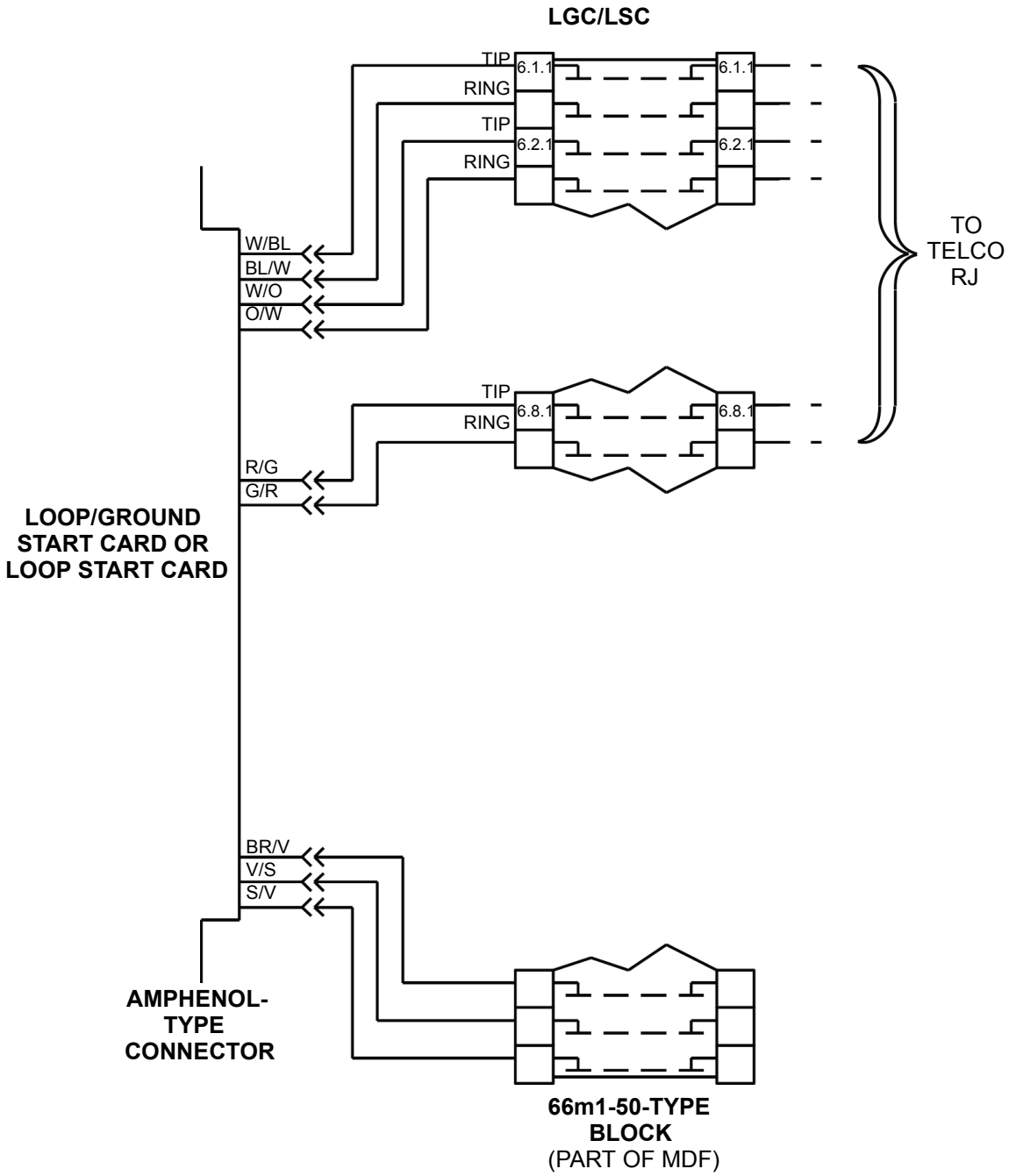


Figure 3-9. *LGC/LSC Block Cable Terminations—Method A (U.S. Only)*



NOTE: Circuits 1-4 (06.01.01-06.04.01) above are used by the LGC/LSC itself. If a Daughter card is attached, it uses circuits 5-8 (06.05.01-06.08.01).

Figure 3-10. *LGC/LSC Block Cable Terminations—Method B (U.S. Only)*

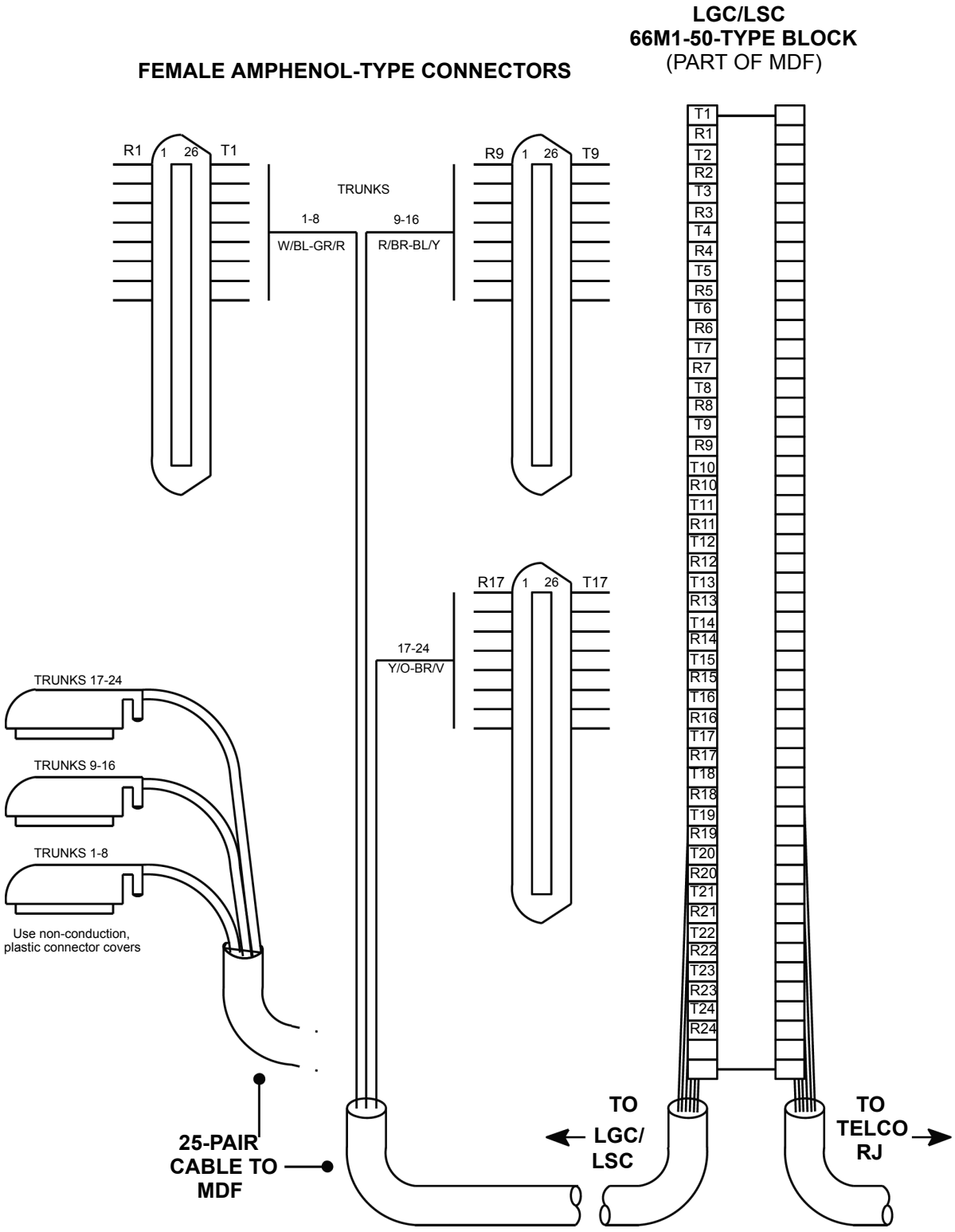
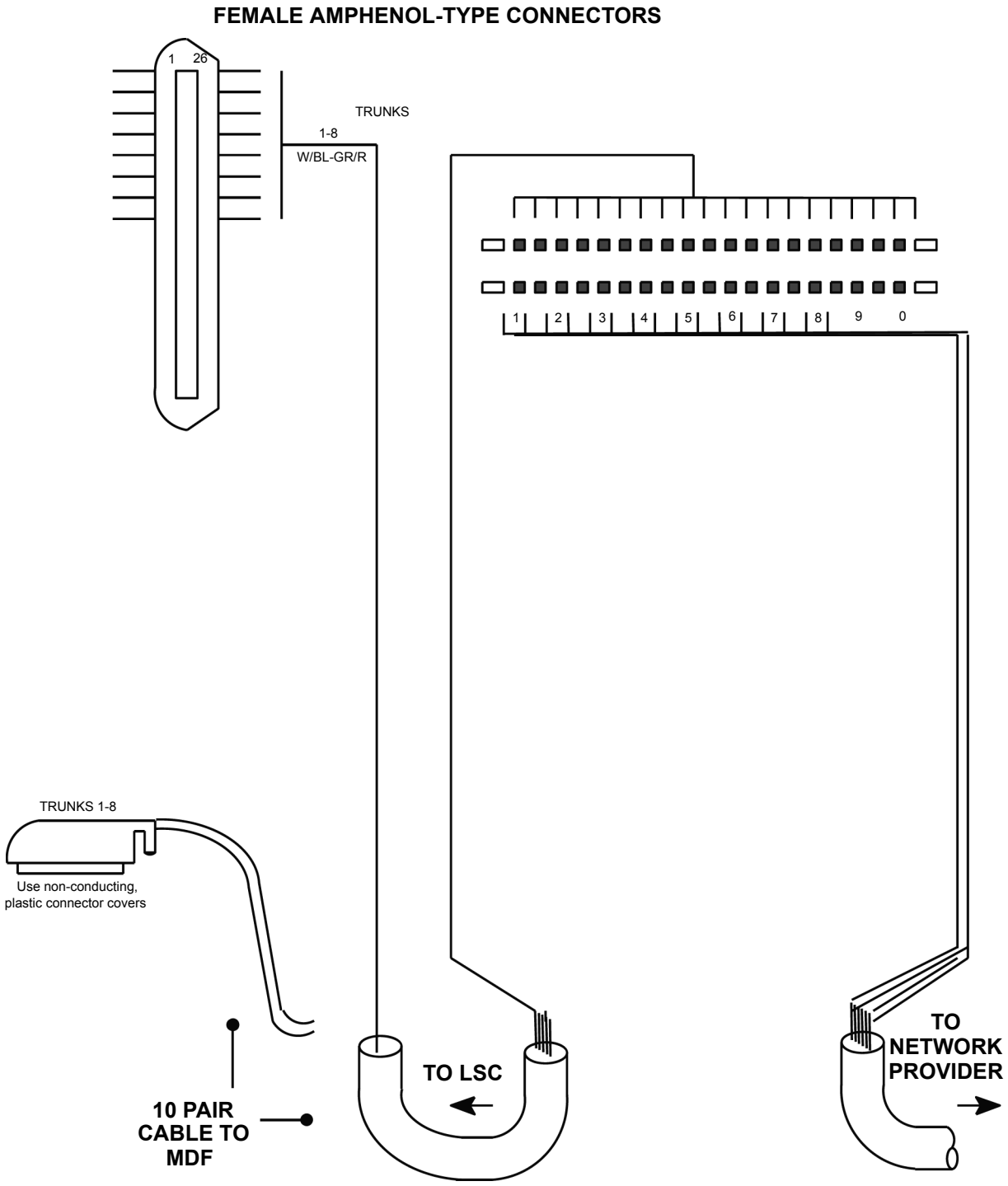
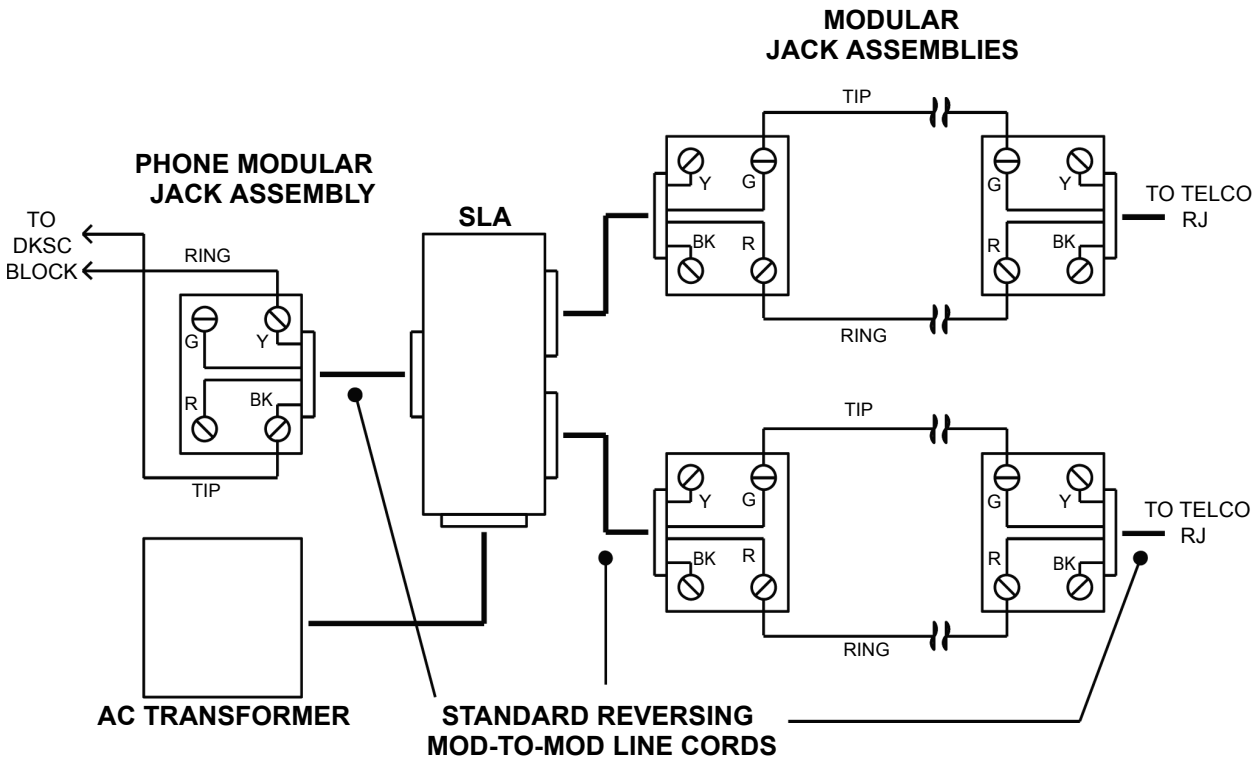


Figure 3-11. *LSC Block Cable Terminations (Europe Only)*



NOTE: Circuits 1-4 (06.01.01-06.04.01) above are used by the LSC itself. If a Daughter Card is attached, it uses circuits 5-8 (06.05.01-06.08.01).

Figure 3-12. *SLA Terminations for DID Trunks (U.S. Only)*



NOTE: If the telephone company termination is near the SLA, it can be plugged directly into the unit.

B. CONNECTING T1/E1 AND PRIMARY RATE INTERFACE (PRI) TRUNKS TO THE MDF

7.11 The installation procedure used to connect T1/E1 and/or PRI trunks at the MDF is dependent on the type of termination used by the telephone company [network provider] and how close it is to the MDF. The T1/PRI or E1/PRI trunks may be terminated on RJ48C modular jacks [RJ45 line jack units] or on RJ-type terminal blocks.

For U.S. Systems

NOTE: See page 3-35 for the installation procedures for a European system.

7.12 Follow one of the three T1/PRI termination methods (A, B, or C) outlined below and on page 3-33.

NOTE: The T1/PRI trunk is typically connected to the standard RJ48C jack provided on the T1 or T1/E1/PRI card. However, in certain situations, it may be more convenient to use the card's DB15 connector instead. See page 3-37 for more information.

7.13 Method A – T1/PRI Trunks Terminated on RJ48C Jacks Near the MDF

7.14 Install T1/PRI trunks terminated on RJ48C jacks near the MDF as outlined below.

Plug one end of a four-pair, non-reversing (straight through), mod-to-mod line cord into each telephone company RJ48C jack. The other end of each line cord is later plugged into the T1 jack on the corresponding T1 or T1/E1/PRI card.

NOTE: Make sure the connectors on the ends of each line cord are properly attached. If any of the wires are cracked or broken, T1/PRI transmission problems can occur.

7.15 Method B – T1/PRI Trunks Terminated on RJ48C Jacks Away from the MDF

7.16 Install T1/PRI trunks terminated on RJ48C jacks away from the MDF as outlined in the following steps. See Figure 3-13 on the next page for a diagram of the complete layout.

1. Mount one eight-conductor modular jack assembly next to each telephone company RJ48C jack.
2. For each modular jack assembly mounted in step 1, mount a corresponding four-conductor modular jack assembly on the MDF backboard.
3. Connect the modular jack assemblies mounted in steps 1 and 2:
   a. Run T1 cable between the corresponding modular jack assemblies.

      NOTE: If the telephone company termination is more than 2000 feet (600 meters) from the MDF, special T1 cable *must* be used (see page 2-54 for cable specifications). For shorter distances, two lengths of standard two-pair voice-frequency cable (one cable for transmit and one cable for receive) may be used instead.

   b. Wire each end of the cables onto their respective modular jack assemblies. See Figure 3-14 on the next page for a diagram of the wiring.
4. At the telephone company termination, plug one end of a four-pair, non-reversing (straight through), mod-to-mod line cord into each RJ48C jack and plug the other end into the corresponding modular jack assembly next to it.

5. At the MDF backboard, plug one end of a four-pair, non-reversing (straight through), mod-to-mod line cord into each modular jack assembly. The other end of each line cord is *later* plugged into the T1 jack on the corresponding T1 or T1/E1/PRI card.

NOTE: Make sure the connectors on the ends of each line cord are properly attached. If any of the wires are cracked or broken, T1/PRI transmission problems can occur.

Figure 3-13. *T1/PRI Trunk Terminations from RJ48C Jacks*

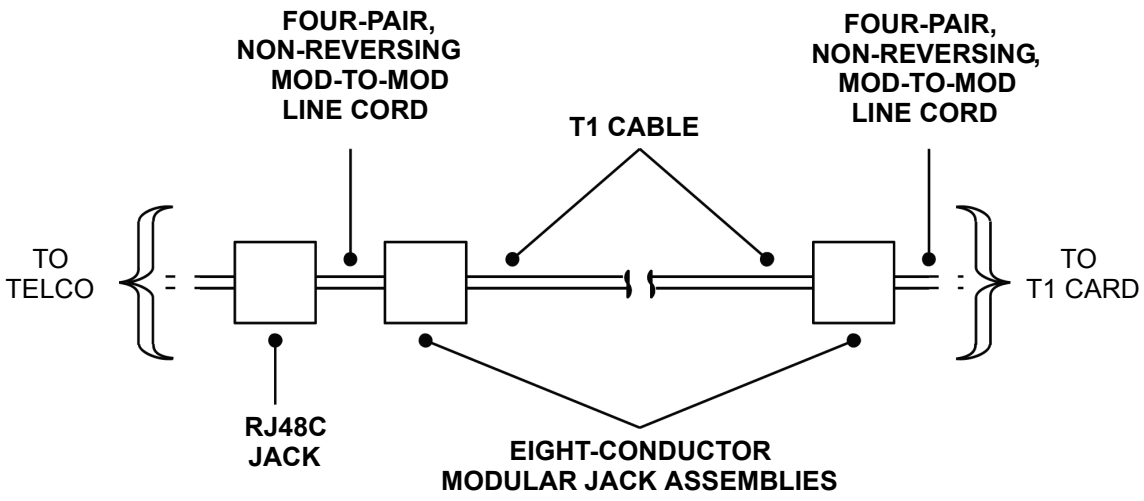
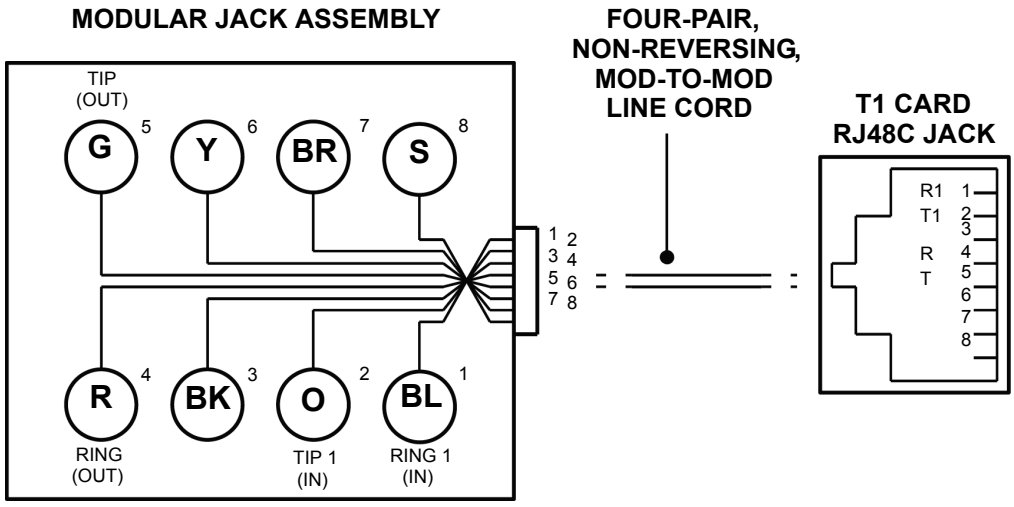


Figure 3-14. *Modular Jack Assembly Wiring for T1/PRI Trunks*



NOTE: OUT = transmit to network; IN = receive from network

7.17 Method C – T1/PRI Trunks Terminated on an RJ-Type Block

7.18 Install the T1/PRI trunks as outlined in the following steps. See Figure 3-15 below for a diagram of the complete layout.

1. On the MDF backboard, mount one eight-conductor modular jack assembly for each T1/PRI trunk termination on the telephone company RJ-type block(s).
2. Connect T1 cable between the telephone company terminations and the corresponding modular jack assemblies as follows:

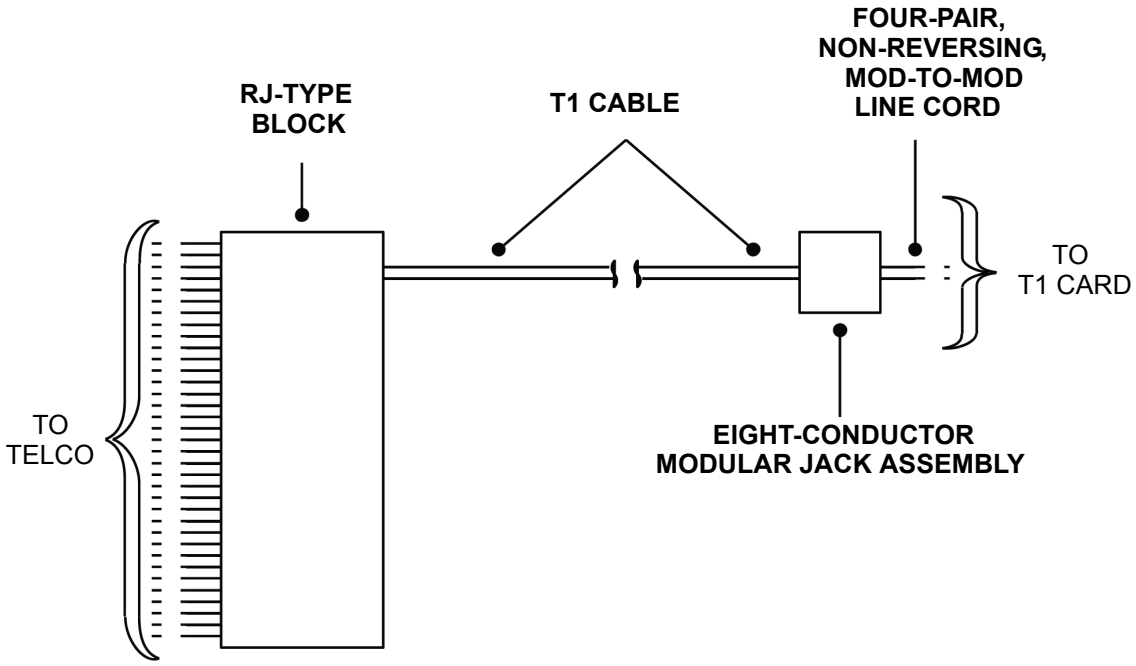
NOTE: If the telephone company termination is more than 2000 feet (600 meters) from the MDF, special T1 cable must be used (see page 2-54 for cable specifications). For shorter distances, two lengths of standard two-pair voice-frequency cable (one cable for transmit and one cable for receive) may be used instead.

   a. Terminate one end of a six-pair cable on each modular jack assembly. See Figure 3-14 on the previous page.
   b. Terminate the other end of the six-pair cable on the *right* side of the T1/PRI RJ-type terminal block.

3. Plug one end of a four-pair, non-reversing (straight through), mod-to-mod line cord into each modular jack assembly (mounted in step 1). The other end of each line cord is *later* plugged into the T1 jack on the corresponding T1 or T1/E1/PRI card

NOTE: Make sure the connectors on the ends of each line cord are properly attached. If any of the wires are cracked or broken, T1/PRI transmission problems can occur.

Figure 3-15. *T1/PRI Trunk Terminations from RJ-Type Blocks*



For European Systems

7.19 Method A – PRI Trunks Terminated On RJ45 Jacks Near The MDF

7.20 Install PRI trunks terminated on RJ45 jacks near the MDF as outlined below.

Plug one end of a four-pair, non-reversing (straight through), mod-to-mod line cord into each network provider RJ45 jack. The other end of each line cord is later plugged into the E1 jack on the corresponding E1/PRI or T1/E1/PRI Card.

NOTE: Make sure the connectors on the ends of each line cord are properly attached. If any of the wires are cracked or broken, E1/PRI transmission problems can occur.

7.21 Method B – PRI Trunks Terminated On RJ45 Jacks Away From The MDF

7.22 Install PRI trunks terminated on RJ45 jacks away from the MDF as outlined in the following steps. See Figure 3-13 on the next page for a diagram of the complete layout.

1. Mount one eight-conductor line jack unit assembly next to each network provider RJ45 jack.
2. For each line jack unit mounted in step 1, mount a corresponding eight-conductor line jack unit assembly on the MDF backboard.
3. To connect the line jack unit assemblies mounted in steps 1 and 2:
   a. Run E1 cable between the corresponding line jack unit assemblies.

   NOTE: If the network provider termination is more than 2000 feet (600 meters) from the MDF, special E1 cable *must* be used (see page 2-54 for cable specifications). For shorter distances, two lengths of standard two-pair voice-frequency cable (one cable for transmit and one cable for receive) may be used instead.

   b. Wire each end of the cables onto their respective line jack unit assemblies. See Figure 3-14 on the next page for a diagram of the wiring.
4. At the network provider termination, plug one end of a four-pair, non-reversing (straight through), mod-to-mod line cord into each RJ45 jack, and plug the other end into the corresponding line jack unit assembly next to it.
5. At the MDF backboard, plug one end of a four-pair, non-reversing (straight through), mod-to-mod line cord into each line jack unit assembly. The other end of each line cord is *later* plugged into the E1 jack on the corresponding E1/PRI or T1/E1/PRI Card.

NOTE: Make sure the connectors on the ends of each line cord are properly attached. If any of the wires are cracked or broken, E1/PRI transmission problems can occur.

Figure 3-16. *PRI Trunk Terminations from RJ45 Jacks*

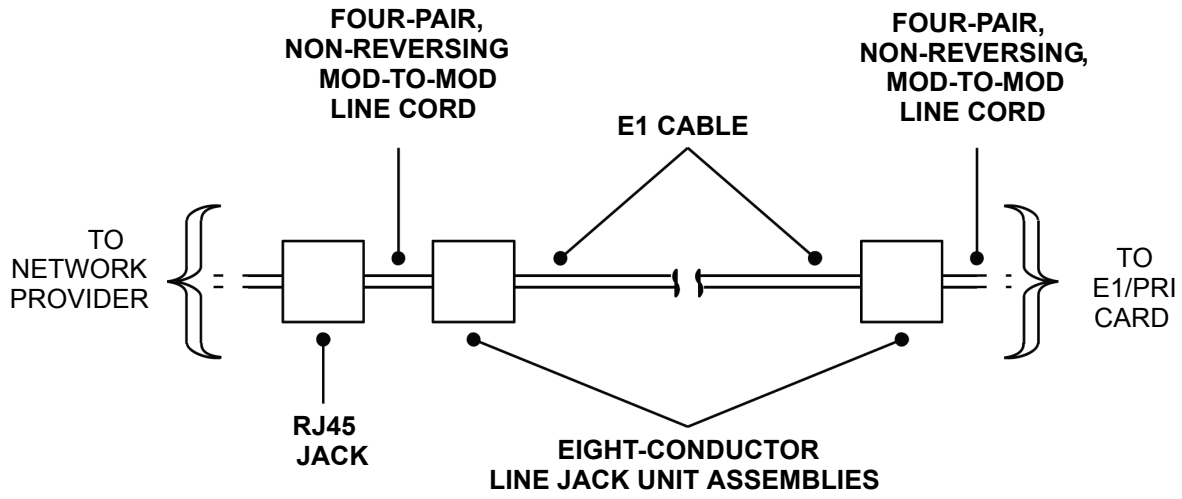
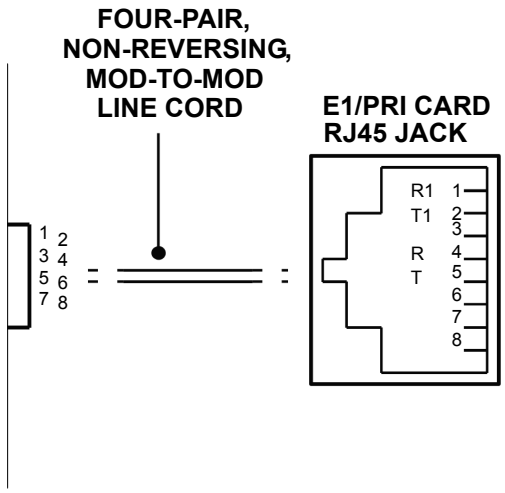


Figure 3-17. *Line Jack Unit Assembly Wiring for PRI Trunks*



NOTE: OUT = transmit to network, IN = receive from network.

Using the DB15 Connector in Place of the RJ48C Jack

7.23 The three T1/PRI or E1/PRI installation methods described on the preceding pages show the T1/PRI or PRI trunk being connected to the RJ48C [RJ45 in Europe] jack on the T1, E1/PRI, or T1/E1/PRI card. However, in certain situations, you may want to use the card's DB15 connector instead.

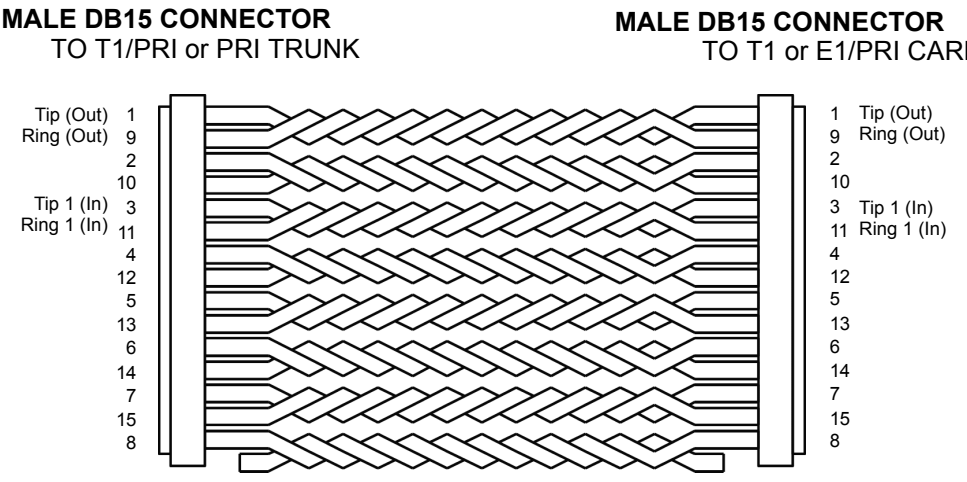
7.24 The pin functions of the T1 or T1/E1/PRI card DB15 connector (15-pin D-subminiature female connector) are as follows:

Table 3-3. *T1C and T1/E1/PRI Card DB15 Connector Pin Assignment*

Signal Name	Function	Pin
Tip (Out)	Transmit Data	1
Ring (Out)	Transmit Data	9
Tip 1 (In)	Receive Data	3
Ring 1 (In)	Receive Data	11

7.25 To connect the DB15 connector on the T1, E1/PRI, or T1/E1/PRI card to a DB15 connector coming from the T1/PRI or PRI trunk, a flat, 28AWG (26 gauge thickness or larger -- smaller number gauge = larger thickness), twisted-pair ribbon cable with a maximum capacitance of 16pF/ft may be used for distances of up to 50 feet (15 meters), as shown in the figure below. For distances longer than 50 feet (15 meters), use the special high-speed data cable described on page 2-54.

Figure 3-18. *Using the T1, E1/PRI, or T1/E1/PRI Card's DB15 Connector*



NOTE: OUT = transmit to network, IN = receive from network

C. CONNECTING BASIC RATE INTERFACE (BRI) TRUNKS TO THE MDF

Connect a Power Supply to the BRS

7.26 Each of the eight BRS ports can be configured to receive line power from the BRS card. If the device connected to a BRS port uses power from the BRS, the jumper for its port must be set over the bottom four pins. In the default position, the jumper is set over the top four pins, which prevents the port from using power from the BRS.

NOTE: The jumper should have an orientation that connects the pins horizontally. For example, in the default position, the jumper connects pins 5 to 6 and 7 to 8. See Figure 3-63 on page 3-133 for jumper positioning.

7.27 Most TE devices use external power sources and do not require power from the BRS card. However, the installer should check the power connection information for the terminating device to determine how to connect it to the BRS. External power can be made available to the devices on the card by connecting an Inter-Tel -48VDC power supply to the card in one of the following ways:

- Connect an Inter-Tel adapter cable (part number 813.1740, included with the card) with the Inter-Tel -48VDC power supply – part number 550.0114 [PS2743 in Europe] to connector J7 on the BRS card. See page 3-133 for an illustration.
- Connect the external power supply to the BRS card at the MDF, using pins 25 and 50 on the Amphenol-type connector. See Figures 3-4 through Figures 3-6 on pages 3-40 through 3-43.

7.28 The BRS limits the amount of current available to each port by using a PTC resettable fuse type device for each port. The PTC allows at least 60 mA of current per port.

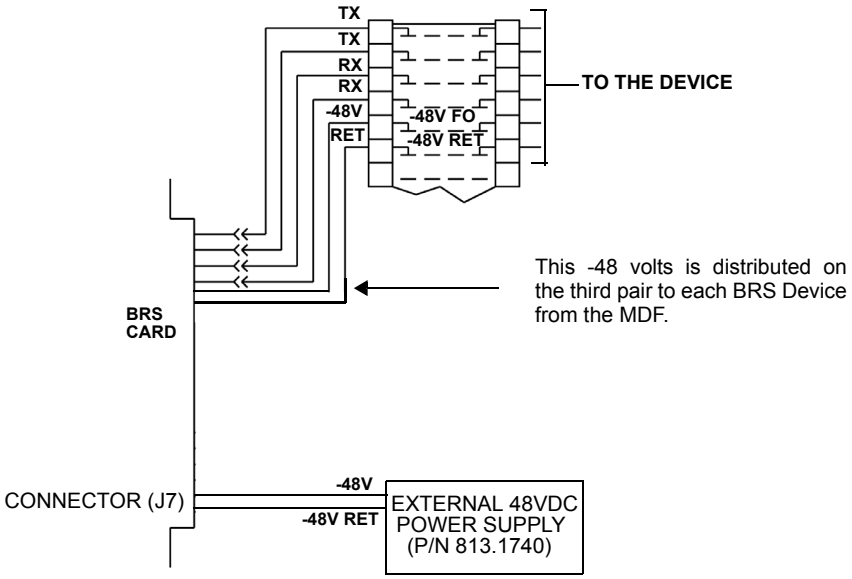
Connect Devices to the BRS Cards

7.29 The installer must check the connection information for the terminating device to determine how to connect it to the BRS.

7.30 There are several ways in which a device can receive power:

- **Line power**: Power can be provided over the device cabling in two ways:
  - — **From the BRS port**: One power supply, connected to the BRS, can provide power to all ports on the card that have their jumpers set on the bottom pins. You can run two-pair or three-pair wire to the devices. (See Figure 3-4 on page 3-40 for a wiring diagram.)
    - Using two-pair wire, the power from the BRS is provided to the device over the transmit and receive pairs.
    - Using three-pair wire, the power is provided over the third pair.
  - — **From an external supply at the MDF**: You can punch down the connection between an external power supply and the device at the MDF. In this case, you would use three-pair wire. Each device would have an individual power supply connected to its third pair, as shown in the following diagram. The power supply is *not* connected to the BRS.

- **External power supply**: This power supply is provided with the device and plugs into a wall socket.



7.31 The BRS transmit pairs connect to the receive pairs of the terminating device, without regard for signal polarity. (The installer must determine which pairs are transmit and receive for the terminating device.)

7.32 Install the BRS cards in the selected slot in the equipment chassis, and connect the external power supply, if required. Also connect the 50-pin Amp connection to the front of the card and terminate the cable to the MDF (Main Distribution Frame).

7.33 Install the BRU card in the selected slot in the equipment chassis, and connect the 50-pin Amp connection to the front of the card, and then terminate the cable to the MDF.

NOTE: The jumper should have an orientation that connects the pins horizontally. For example, in the default position, the jumper connects pins 5 to 6 and 7 to 8. See Figure 3-63 on page 3-133 for jumper positioning.

Table 3-4. *BRS Device Connections to the MDF (U.S.)*

Amphenol Pin	Color	Circuit	BRI Device
1	White/Blue	1	TXN0
26	Blue/White		TXP0
2	White/Orange		RXP0
27	Orange/White		RXN0
3	White/Green		-48V F0
28	Green/White		-48V RET
4	White/Brown	2	TXN1
29	Brown/White		TXP1
5	White/Gray		RXP1
30	Gray/White		RXN1
6	Red/Blue		-48V F1
31	Blue/Red		-48V RET
7	Red/Orange	3	TXN2
32	Orange/Red		TXP2
8	Red/Green		RXP2
33	Green/Red		RXN2
9	Red/Brown		-48V F2
34	Brown/Red		-48V RET
10	Red/Gray	4	TXN3
35	Gray/Red		TXP3
11	Black/Blue		RXP3
36	Blue/Black		RXN3
12	Black/Orange		-48V F3
37	Orange/Black		-48V RET
13	Black/Green	5	TXN4
38	Green/Black		TXP4
14	Black/Brown		RXP4
39	Brown/Black		RXN4
15	Black/Gray		-48V F4
40	Gray/Black		-48V RET
16	Yellow/Blue	6	TXN5
41	Blue/Yellow		TXP5
17	Yellow/Orange		RXP5
42	Orange/Yellow		RXN5
18	Yellow/Green		-48V F5
43	Green/Yellow		-48V RET
19	Yellow/Brown	7	TXN6
44	Brown/Yellow		TXP6
20	Yellow/Gray		RXP6
45	Gray/Yellow		RXN6
21	Violet/Blue		-48V F6
46	Blue/Violet		-48V RET
22	Violet/Orange	8	TXN7
47	Orange/Violet		TXP7
23	Violet/Green		RXP7
48	Green/Violet		RXN7
24	Violet/Brown		-48V F7
49	Brown/Violet		-48V RET
25	Violet/Slate		-48V
50	Slate/Violet	CARD POWER (Input from BRS PSU)	-48V RET

NOTE: Signal names beginning with “TX” are transmit and “RX” are receive, with “P” or “N” denoting positive or negative polarity. For example, “TXP0/TXN0” is the BRS card transmit pair for circuit 1. Likewise, “RXP0/RXN0” is the receive pair for circuit 1. Signal polarity is generally not important.

Table 3-5. *BRU Device Connections to the MDF (U.S.)*

Amphenol Pin	Color	Circuit	BRI Device
1	White/Blue	1	TIP 0
26	Blue/White		RING 0
2	White/Orange	2	TIP 1
27	Orange/White		RING 1
3	White/Green	3	TIP 2
28	Green/White		RING 2
4	White/Brown	4	TIP 3
29	Brown/White		RING 3
5	White/Gray	5	TIP 4
30	Gray/White		RING 4
6	Red/Blue	6	TIP 5
31	Blue/Red		RING 5
7	Red/Orange	7	TIP 6
32	Orange/Red		RING 6
8	Red/Green	8	TIP 7
33	Green/Red		RING 7
9	Red/Brown		NC
34	Brown/Red		NC
10	Red/Gray		NC
35	Gray/Red		NC
11	Black/Blue		NC
36	Blue/Black		NC
12	Black/Orange		NC
37	Orange/Black		NC
13	Black/Green		NC
38	Green/Black		NC
14	Black/Brown		NC
39	Brown/Black		NC
15	Black/Gray		NC
40	Gray/Black		NC
16	Yellow/Blue		NC
41	Blue/Yellow		NC
17	Yellow/Orange		NC
42	Orange/Yellow		NC
18	Yellow/Green		NC
43	Green/Yellow		NC
19	Yellow/Brown		NC
44	Brown/Yellow		NC
20	Yellow/Gray		NC
45	Gray/Yellow		NC
21	Violet/Blue		NC
46	Blue/Violet		NC
22	Violet/Orange		NC
47	Orange/Violet		NC
23	Violet/Green		NC
48	Green/Violet		NC
24	Violet/Brown		NC
49	Brown/Violet		NC
25	Violet/Gray		NC
50	Gray/Violet		NC

NOTE: The “NC” represents No Connection.

Figure 3-19. *On-Site Krone Connections for BRS (Europe Only)*

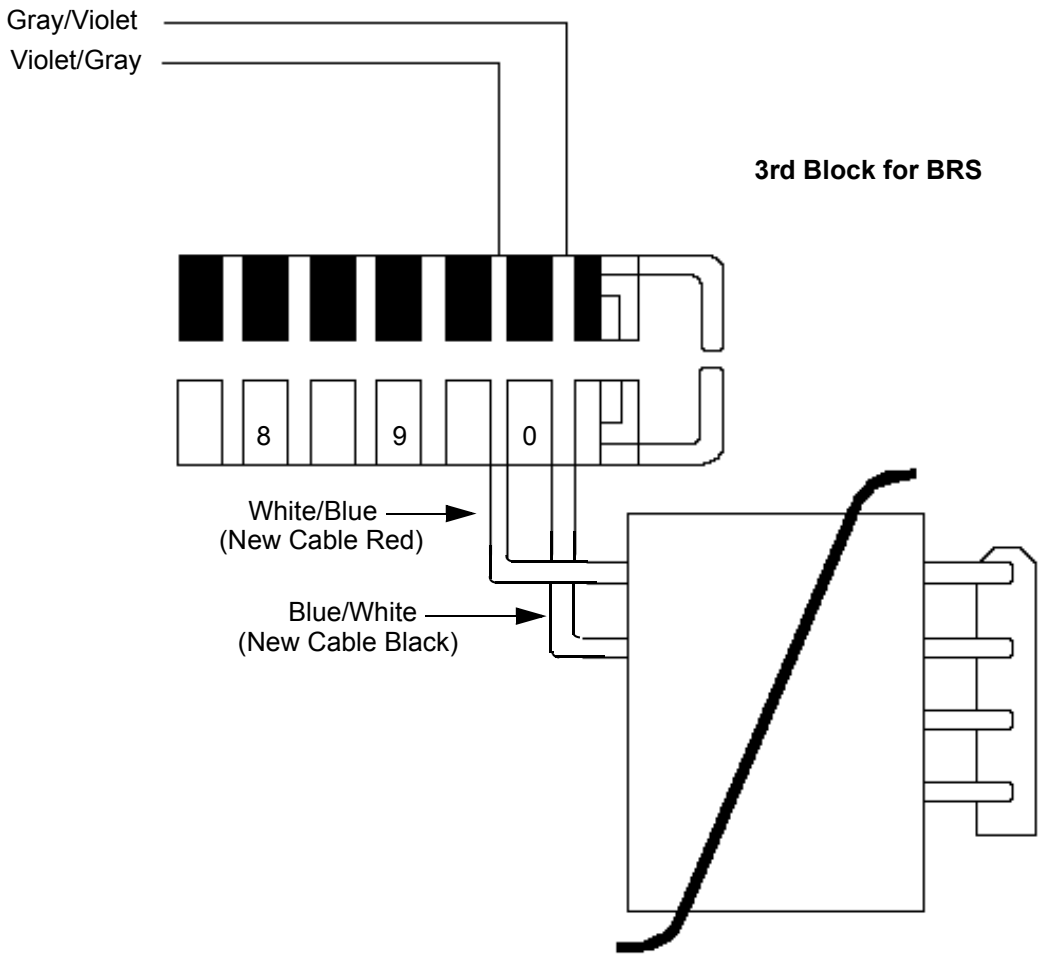


Table 3-6. *BRS Device Connections to the MDF (Europe)*

Amphenol Pin	Color	Circuit	BRI Device
1	White/Blue	1	TXN0
26	Blue/White		TXP0
2	White/Orange		RXP0
27	Orange/White		RXN0
3	White/Green		-48V F0
28	Green/White	CIRCUIT POWER (Only used for older devices)	-48V RET
4	White/Brown	2	TXN1
49	Brown/White		TXP1
5	White/Gray		RXP1
30	Gray/White		RXN1
6	Red/Blue		-48V F1
31	Blue/Red	CIRCUIT POWER (Only used for older devices)	-48V RET
7	Red/Orange	3	TXN2
32	Orange/Red		TXP2
8	Red/Green		RXP2
33	Green/Red		RXN2
9	Red/Brown		-48V F2
34	Brown/Red	CIRCUIT POWER (Only used for older devices)	-48V RET
10	Red/Gray	4	TXN3
35	Gray/Red		TXP3
11	Black/Blue		RXP3
36	Blue/Black		RXN3
12	Black/Orange		-48V F3
37	Orange/Black	CIRCUIT POWER (Only used for older devices)	-48V RET
13	Black/Green	5	TXN4
38	Green/Black		TXP4
14	Black/Brown		RXP4
39	Brown/Black		RXN4
15	Black/Gray		-48V F4
40	Gray/Black	CIRCUIT POWER (Only used for older devices)	-48V RET
16	Yellow/Blue	6	TXN5
41	Blue/Yellow		TXP5
17	Yellow/Orange		RXP5
42	Orange/Yellow		RXN5
18	Yellow/Green		-48V F5
43	Green/Yellow	CIRCUIT POWER (Only used for older devices)	-48V RET
19	Yellow/Brown	7	TXN6
44	Brown/Yellow		TXP6
20	Yellow/Gray		RXP6
45	Gray/Yellow		RXN6
21	Violet/Blue		-48V F6
46	Blue/Violet	CIRCUIT POWER (Only used for older devices)	-48V RET
22	Violet/Orange	8	TXN7
47	Orange/Violet		TXP7
23	Violet/Green		RXP7
48	Green/Violet		RXN7
24	Violet/Brown		-48V F7
49	Brown/Violet	CIRCUIT POWER (Only used for older devices)	-48V RET
25	Violet/Slate		-48V
50	Slate/Violet	CARD POWER (Input from BRS PSU)	-48V RET

D. CONNECTING STATION CABLES TO THE MDF

With a U.S. System

7.34 Each type of station instrument is connected to a specific station card. Digital phones are connected to Digital Keyset Cards (DKSCs) and/or 16-Port Digital Keyset Cards (DKSC-16s). Analog phones are connected to Analog Keyset Cards (AKSCs). Executone IDS and Wave phones are connected to Executone Keyset Cards (EKSCs). Single-line sets and playback devices are connected to Single-Line Cards (SLCs) or 16-Port Single-Line Cards (SLC-16s), or they can be connected to Single-Line Adapters (SLAs) which are in turn connected to available DKSC/DKSC-16 circuits.

NOTE: Besides single-line sets, playback devices, and off-premises extensions, SLAs can also be used to install DID trunks. Any combination using the two circuits on the SLA is possible. See page 3-25 for DID trunk installation instructions.

7.35 After the station cables are run, connect them and the 25-pair cables to the DKSC/DKSC-16 blocks on the MDF as follows:

1. Ensure that both ends of each station cable are labeled with the circuit number of the associated station instrument.
2. Terminate each station cable on the *right* side of the corresponding station block. See Figure 3-20 through Figure 3-7 on the following pages for station cable terminations.

   NOTE: For simplicity, the figures show station instruments being installed using one-pair cable and four-conductor modular jacks. Of course, if three-pair cable and six-conductor modular jacks are used instead, extra terminal blocks and the use of cross-connect wiring techniques are required.

3. Using enough 25-pair cable to run from each block to each station card in the equipment chassis, make the termination cables. Attach a 50-pin female amphenol-type connector to one end of each cable. Label each connector and cable end with the appropriate slot number (for example, SLOT-2). These connectors are attached to the station cards after the circuit cards are installed.
4. *If necessary:* Using standard electrical tape, tape two ferrite split beads around each 25-pair cable just below the amphenol-type connector (as shown in Figure 3-8 on page 3-26).

   NOTE: Ferrite beads may be necessary to help reduce electromagnetic interference (EMI) and radio frequency interference (RFI). The part number for a single split bead is 808.1195.

5. Terminate the other end of each 25-pair cable on the *left* side of the corresponding station block.
6. *DO NOT* attach bridging clips until the loop resistance tests have been performed (as described on page 3-69).

With a European System

7.36 Each type of station instrument is connected to a specific station card. Digital phones are connected to Digital Keyset Cards (DKSCs) and/or 16-Port Digital Keyset Cards (DKSC-16s). Analog phones are connected to Analog Keyset Cards (AKSCs). Single-line sets and playback devices are connected to Single-Line Cards (SLCs) or 16-Port Single-Line Cards (SLC-16s), or they can be connected to Single-Line Adapters (SLAs) which are in turn connected to available DKSC/DKSC-16 circuits.

7.37 After the station cables are run, connect them to the station blocks on the MDF. Ensure both ends of each station cable are labeled with the circuit number of the associated station instrument. The station cables are pre-terminated on the upper half of the Krone block. The line plant wiring extension terminates on the lower half of the Krone block. See Figure 3-20 to Figure 3-19 on the following pages for station cable termination.

DKSC-16 Extended Loop Length Modification

7.38 For longer station loop lengths, an optional customer-supplied, external power supply can be connected to the DKSC-16 termination block. You can use the Inter-Tel 30V power supply for 108 ohms/2100 feet (640 m). Or you can purchase the +36VDC External Power Supply Kit (kit part no. 928.3705).

7.39 This option is available only on DKSC-16s that are at revision level "2.0" or later or that have been modified and labeled with FSM100995. The external power supply is connected to pins 50 (+36V, V/SL) and 25 (GND, SL/V) on the termination block. In addition, the strap on jumper JP1 must be placed over pins 1 and 2 to enable the external power supply.

Installing SLC-16 Power Supplies

7.40 A power supply is required for each SLC-16 because the cards are not designed to pull talk battery, ring, or message lamp power from the system power supply. Power supplies are available that support one card or up to four cards.

7.41 There are two ways to connect the power supply to the SLC-16:

- Connect the power supply directly to the board using the RINGING POWER SUPPLY connector on the SLC-16. See page 3-118.
- Punch down the power supply wires at the MDF. See the diagram in Figure 3-31 on page 3-57 and the chart in Figure 3-7 on page 3-60. The connector on the power supply wires have to be cut off before the power supply can be installed.

NOTE: If a four-card power supply is wall mounted, be sure that the fully enclosed side is on the bottom (facing down).

Figure 3-20. *DKSC Block Cable Terminations for Phones (U.S.)*

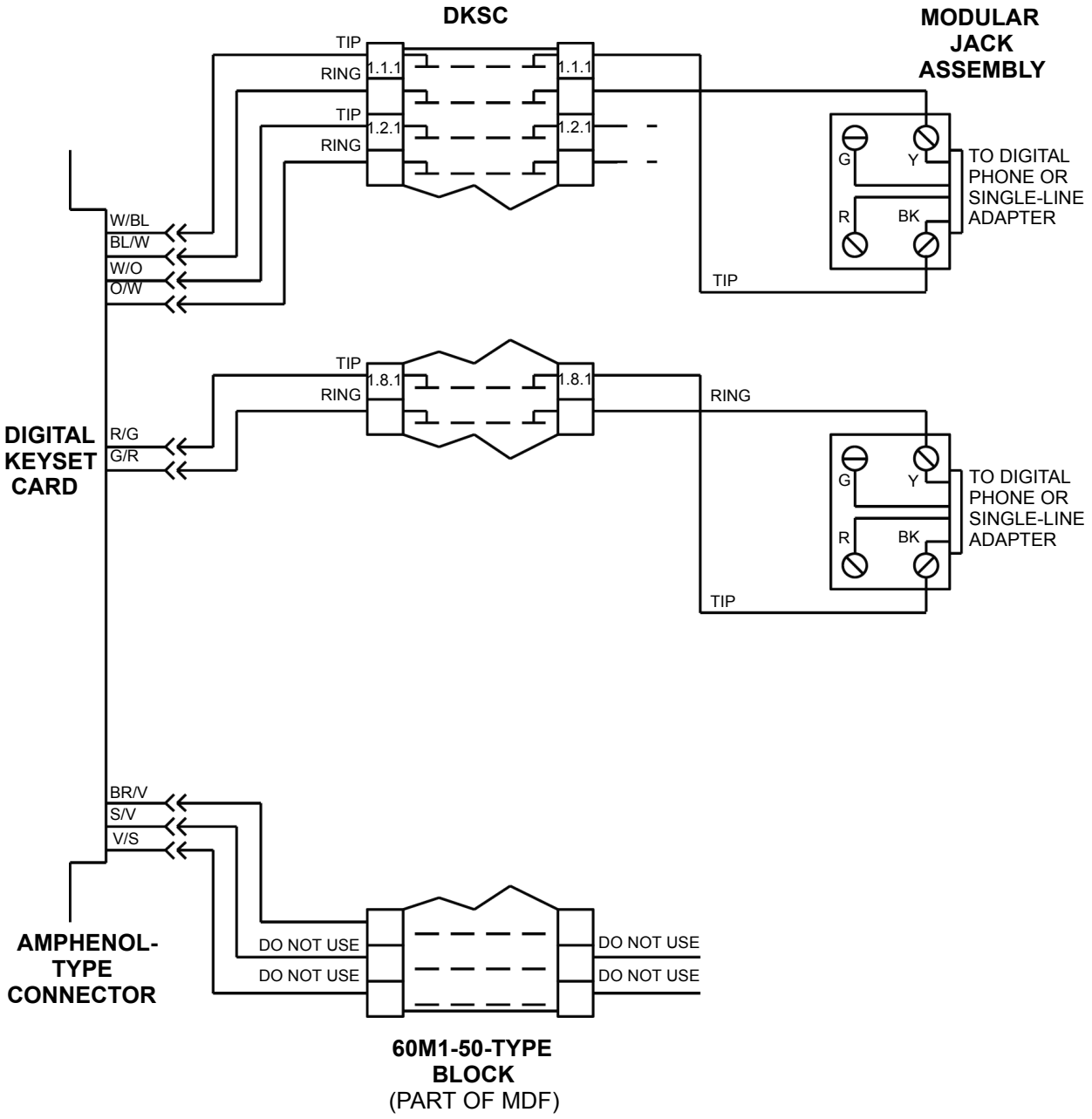


Figure 3-21. *DKSC Block Cable Terminations for Phones (Europe)*

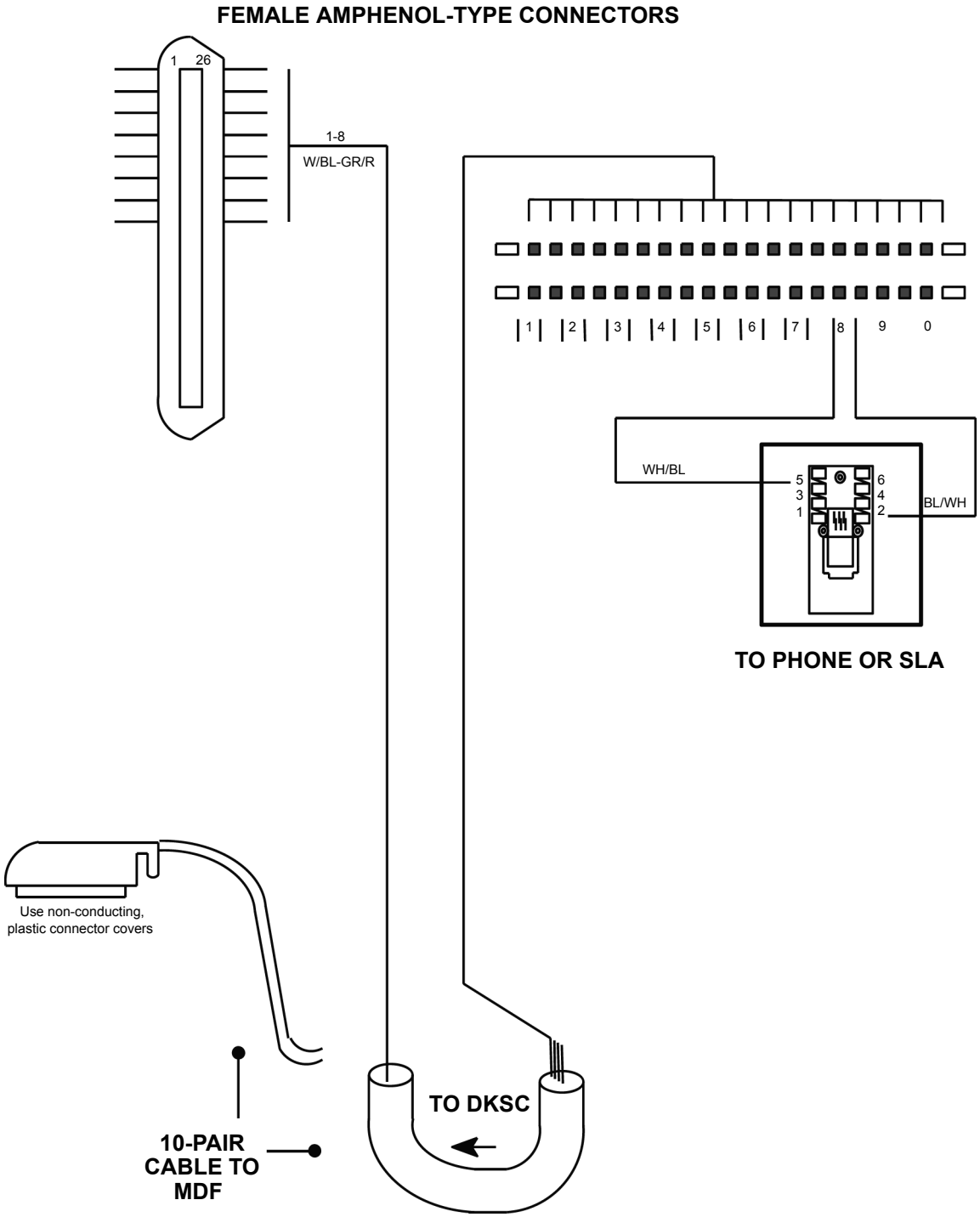
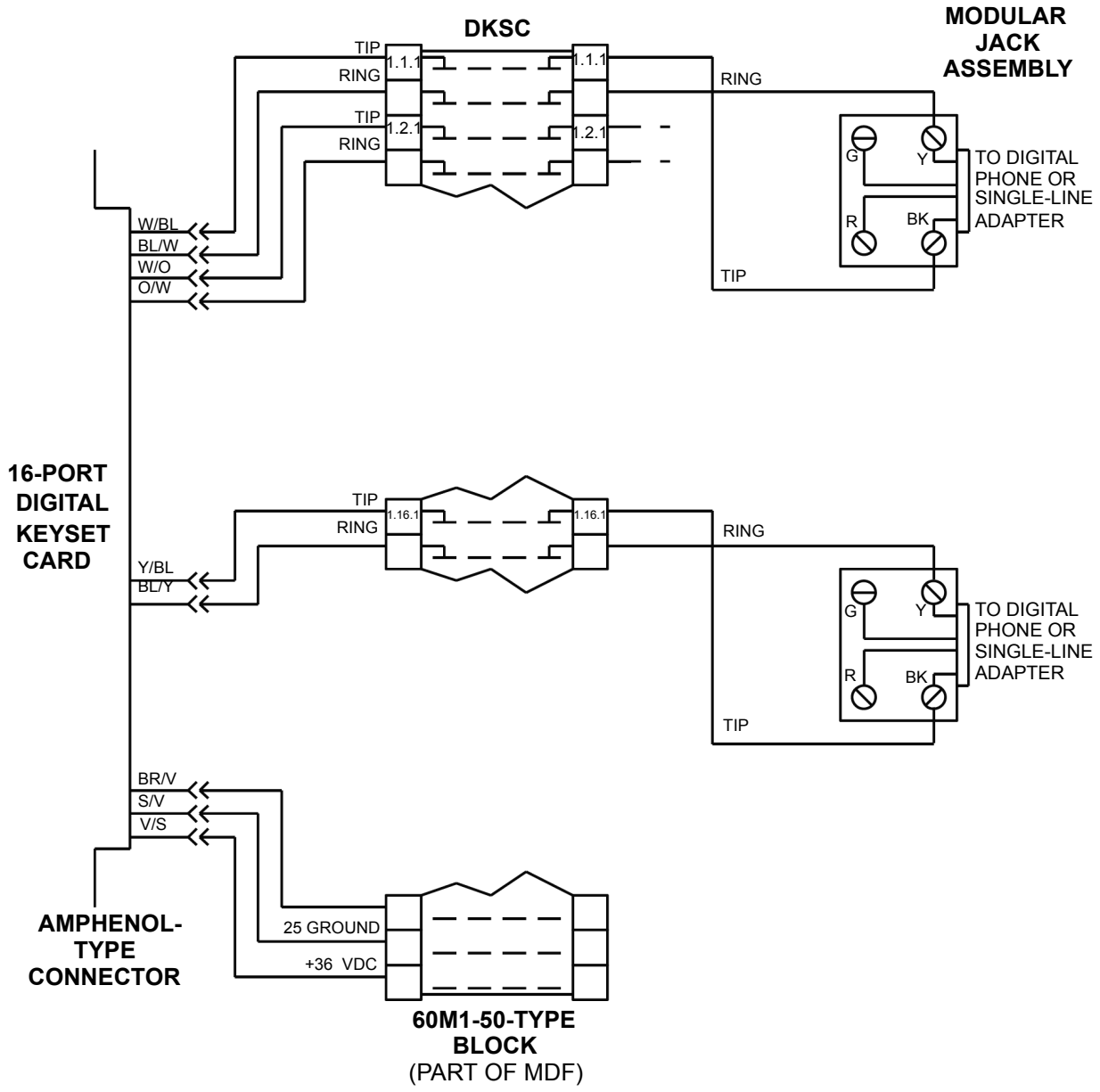


Figure 3-22. *DKSC-16 Block Cable Terminations for Phones (U.S.)*



NOTE: For simplicity, this figure shows digital phones being installed using one-pair cable and four-conductor modular jacks. Of course, if two-pair cable or three-pair cable is used instead, extra terminal blocks and the use of cross-connect wiring techniques are required. Note that tip and ring are connected to the second pair of terminals (black and yellow) on the modular jack.

Figure 3-23. *DKSC-16 Block Cable Terminations for Phones (Europe)*

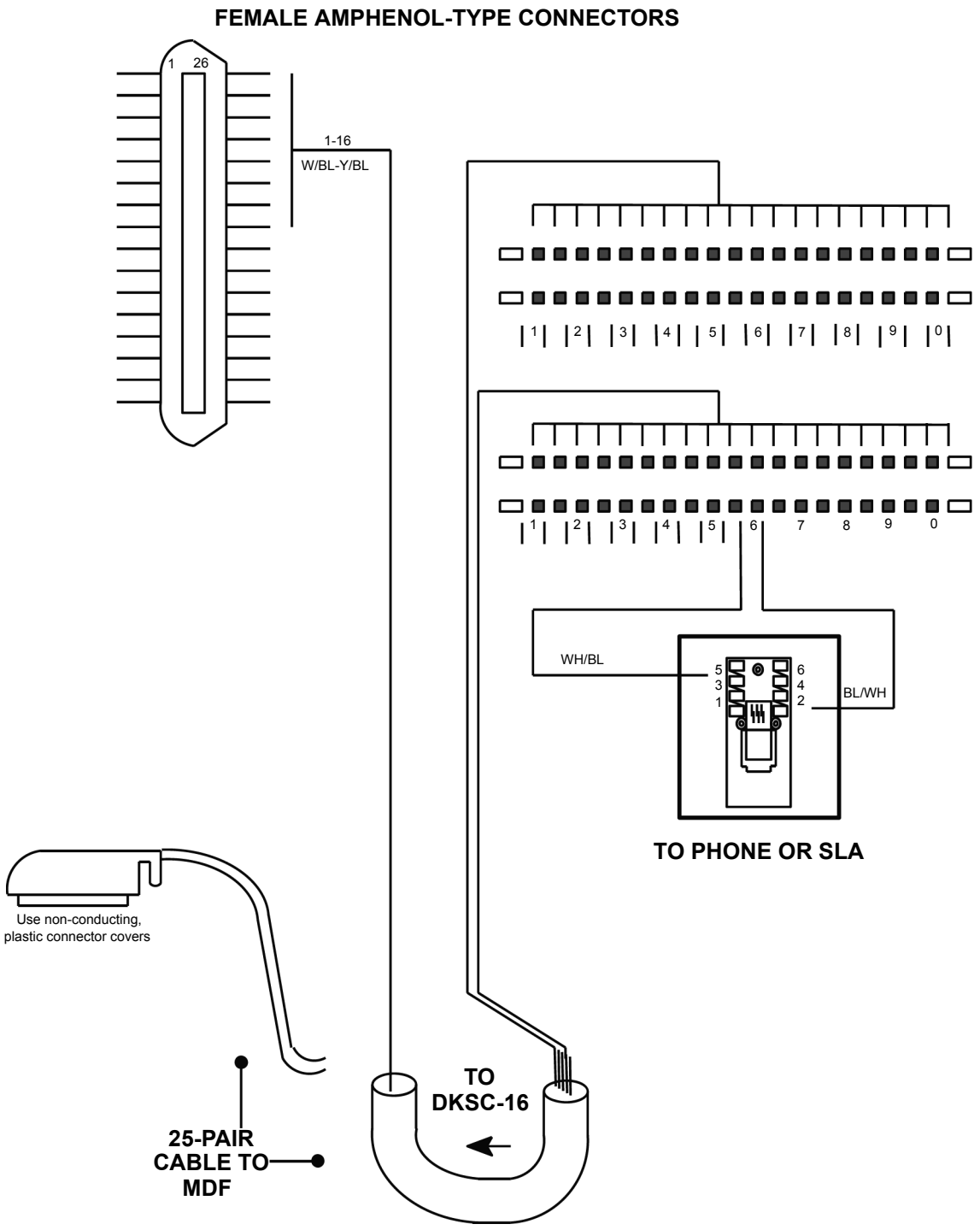


Figure 3-24. *AKSC Block Cable Terminations (U.S.)*

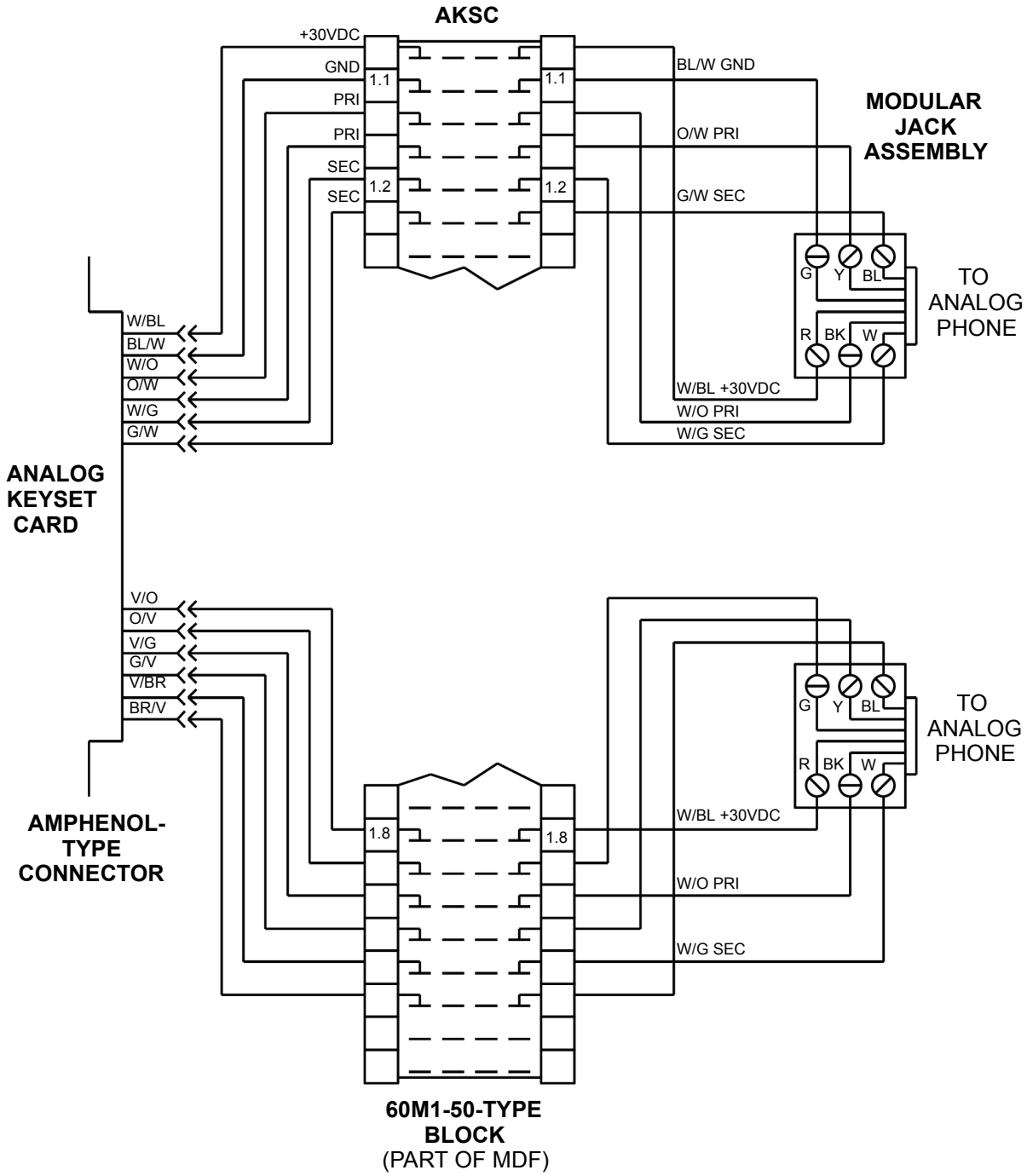


Figure 3-25. *AKSC Block Cable Terminations (Europe)*

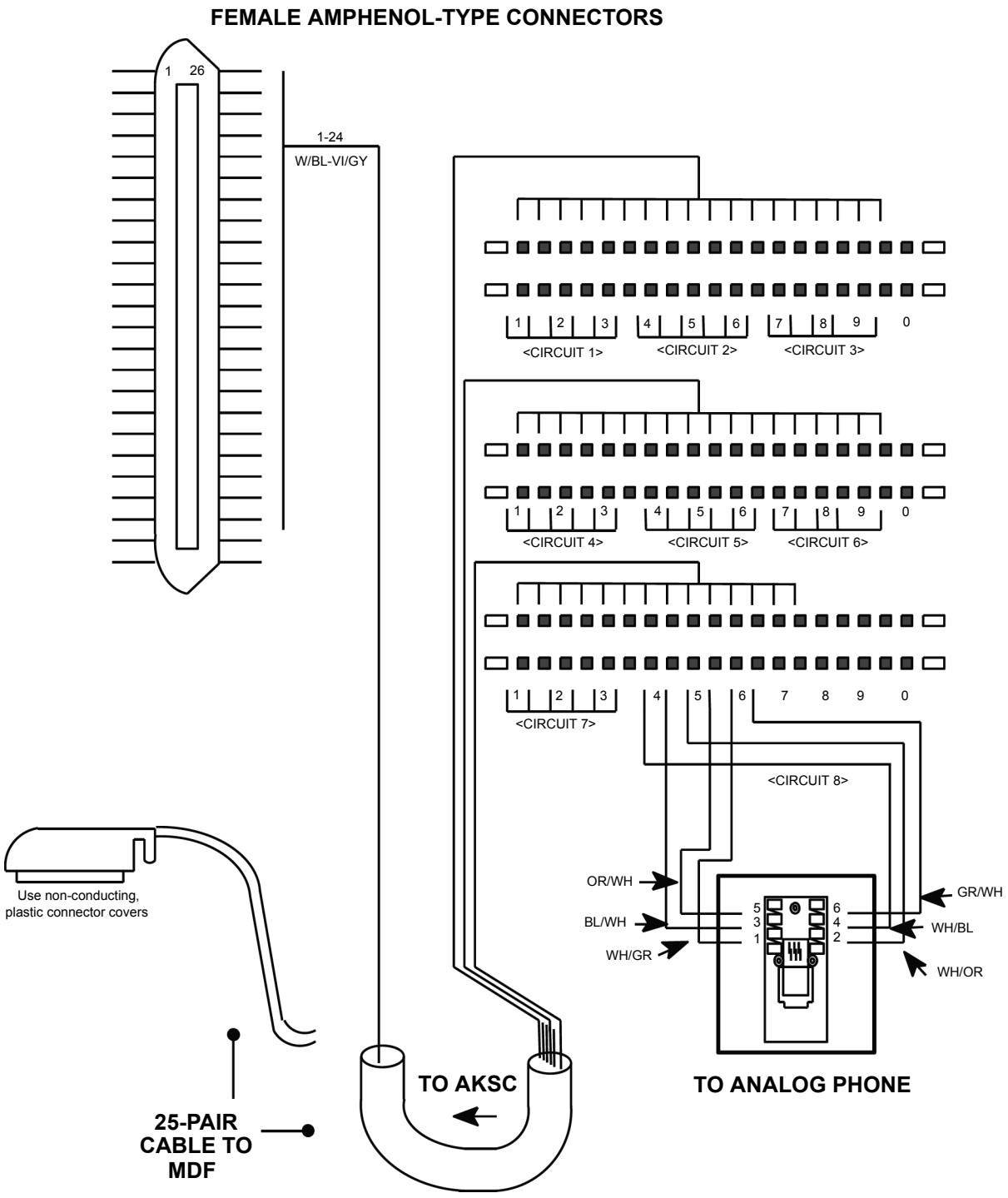


Figure 3-26. *EKSC Block Cable Terminations for Phones (U.S. Only)*

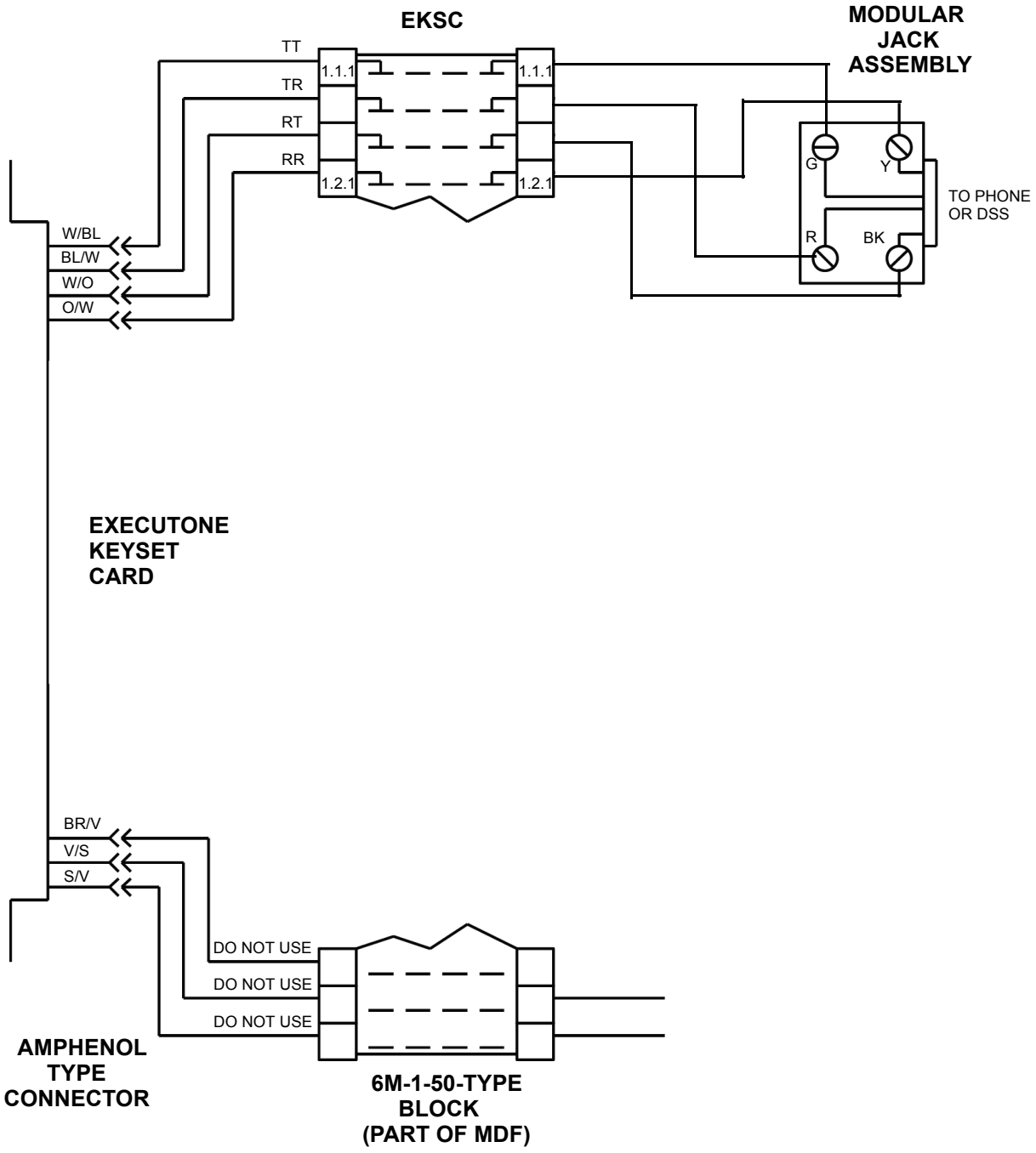
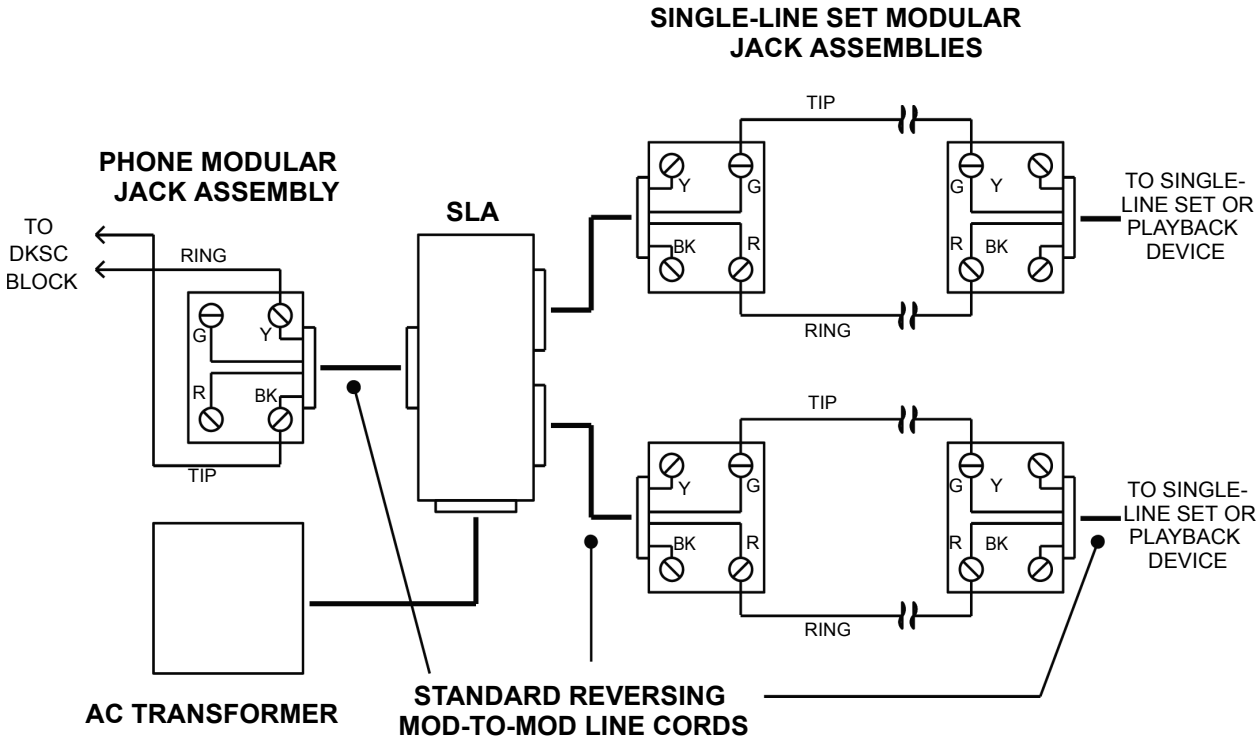
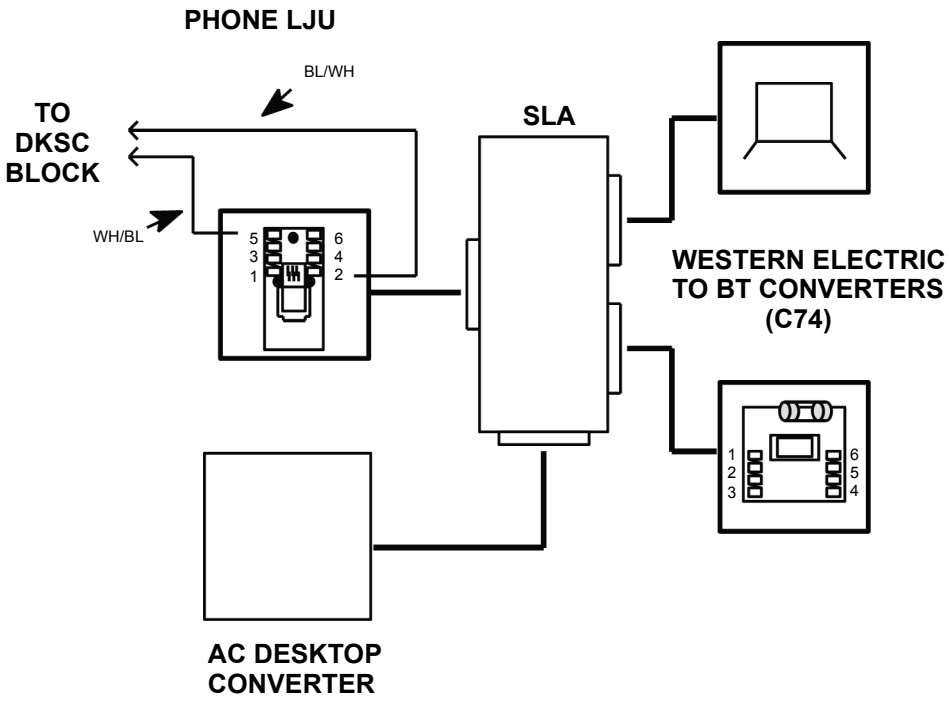


Figure 3-27. *SLA Terminations for Single-Line Sets (U.S.)*



NOTE: If a single-line set or playback device is located near the SLA, it can be plugged directly into the unit.

Figure 3-28. *SLA Terminations for Single-Line Set (Europe)*



7.42 The SLA is supported with two Western Electric to BT converters (part no. C74) for use when the attached devices are near the SLA.

7.43 For longer range connections (up to 4.5Km), an additional Krone block is required. This is connected to the SLA using a standard line cord (part no. C27) and phone line jack unit (LJU).

7.44 Where exposed OPX cabling is utilized, BT14A/1 of BT21A surge arrestors should be used on the OPX lines, and the Krone field should be earthed. The LJU at the far end of the cable should be a line master.

7.45 This entire setup might be part of the main MDF or situated remotely, subject to cabinet-to-SLA cable restrictions.

Figure 3-29. *Off-Premises Connection (Europe Only)*

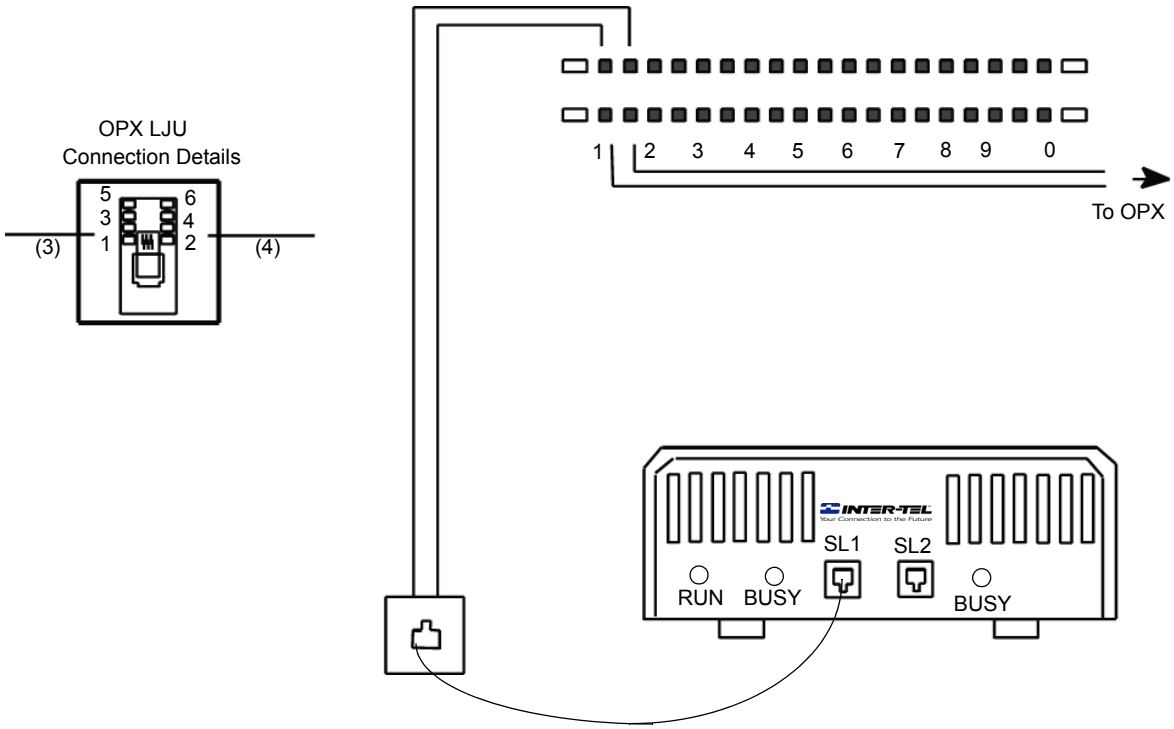
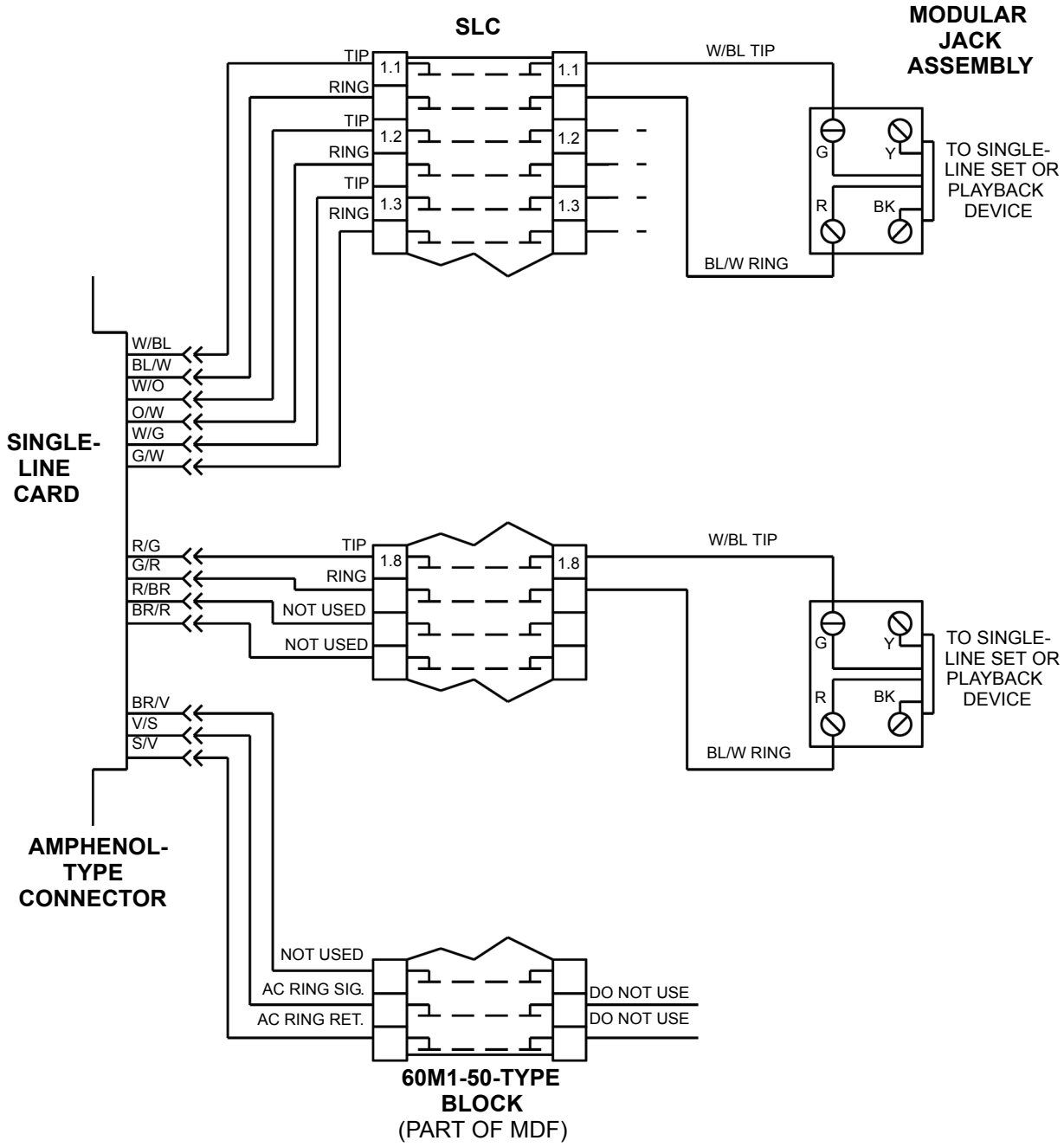
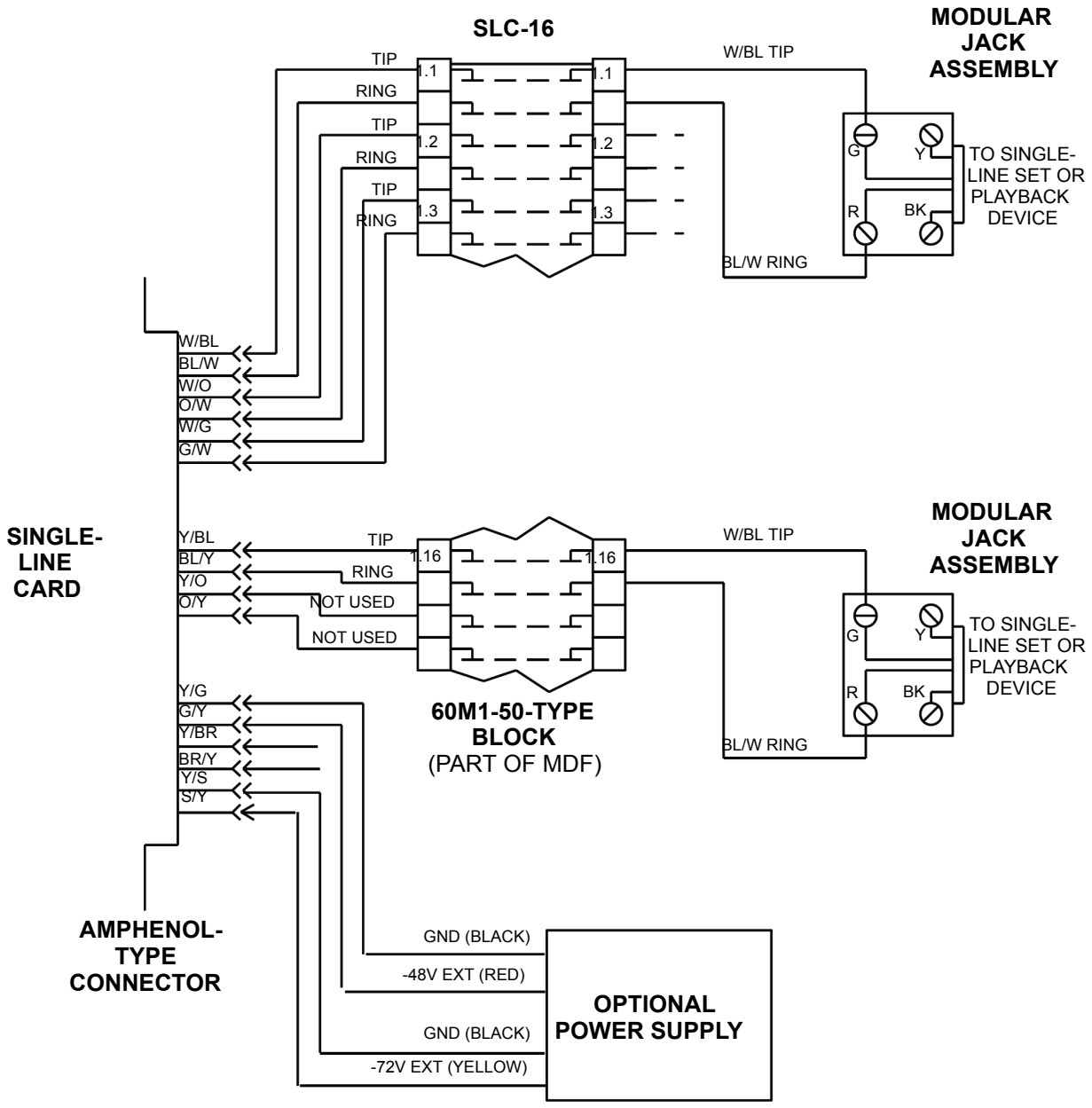


Figure 3-30. *SLC Block Cable Terminations (U.S.)*



NOTE: For simplicity, this figure shows single-line sets and playback devices being installed using one-pair cable and four-conductor modular jacks. Of course, if three-pair cable and six-conductor modular jacks are used instead (as is recommended), extra terminal blocks and the use of cross-connect wiring techniques are required.

Figure 3-31. *SLC-16 Block Cable Terminations (U.S.)*



NOTE: For simplicity, this figure shows single-line sets and playback devices being installed using one-pair cable and four-conductor modular jacks. Of course, if three-pair cable and six-conductor modular jacks are used instead (as is recommended), extra terminal blocks and the use of cross-connect wiring techniques are required.

Figure 3-32. *SLC Block Cable Terminations (Europe)*

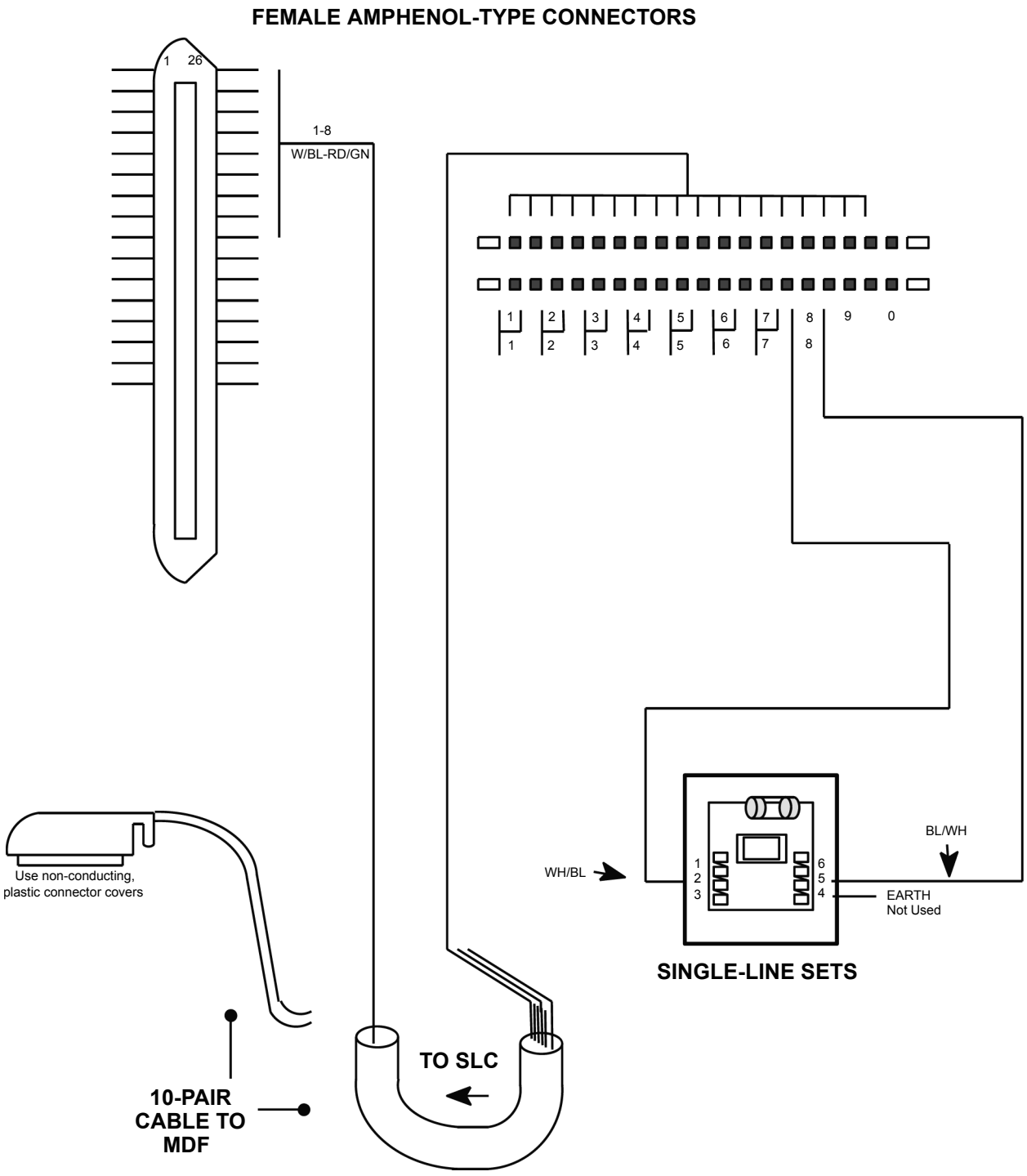


Table 3-7. *Station Cable Terminations on the Station Blocks (U.S.)*

Amphe-nol #	Cable Pair	DKSC	DKSC-16/ DKSC-16+	SLC	SLC-16	Circuit No.*	Ext. No.*
26	W-BL	TIP	TIP	TIP	TIP	01.01.01	1000
1	BL-W	RING	RING	RING	RING		
27	W-O	TIP	TIP	TIP	TIP	01.02.01	1001
2	O-W	RING	RING	RING	RING		
28	W-G	TIP	TIP	TIP	TIP	01.03.01	1002
3	G-W	RING	RING	RING	RING		
29	W-BR	TIP	TIP	TIP	TIP	01.04.01	1003
4	BR-W	RING	RING	RING	RING		
30	W-S	TIP	TIP	TIP	TIP	01.05.01	1004
5	S-W	RING	RING	RING	RING		
31	R-BL	TIP	TIP	TIP	TIP	01.06.01	1005
6	BL-R	RING	RING	RING	RING		
32	R-O	TIP	TIP	TIP	TIP	01.07.01	1006
7	O-R	RING	RING	RING	RING		
33	R-G	TIP	TIP	TIP	TIP	01.08.01	1007
8	G-R	RING	RING	RING	RING		
34	R-BR	NOT USED	TIP	NOT USED	TIP	01.09.01	1008
9	BR-R	NOT USED	RING	NOT USED	RING		
35	R-S	NOT USED	TIP	NOT USED	TIP	01.10.01	1009
10	S-R	NOT USED	RING	NOT USED	RING		
36	BK-BL	NOT USED	TIP	NOT USED	TIP	01.11.01	1010
11	BL-BK	NOT USED	RING	NOT USED	RING		
37	BK-O	NOT USED	TIP	NOT USED	TIP	01.12.01	1011
12	O-BK	NOT USED	RING	NOT USED	RING		
38	BK-G	NOT USED	TIP	NOT USED	TIP	01.13.01	1012
13	G-BK	NOT USED	RING	NOT USED	RING		
39	BK-BR	NOT USED	TIP	NOT USED	TIP	01.14.01	1013
14	BR-BK	NOT USED	RING	NOT USED	RING		
40	BK-S	NOT USED	TIP	NOT USED	TIP	01.15.01	1014
15	S-BK	NOT USED	RING	NOT USED	RING		
41	Y-BL	NOT USED	TIP	NOT USED	TIP	01.16.01	1015
16	BL-Y	NOT USED	RING	NOT USED	RING		
42	Y-O	NOT USED	NOT USED	NOT USED	NOT USED		
17	O-Y	NOT USED	NOT USED	NOT USED	NOT USED		
43	Y-G	NOT USED	NOT USED	NOT USED	GND B2 (BL)		
18	G-Y	NOT USED	NOT USED	NOT USED	-48V EXT (R)		
44	Y-BR	NOT USED	NOT USED	NOT USED	NOT USED		
19	BR-Y	NOT USED	NOT USED	NOT USED	NOT USED		
45	Y-S	NOT USED	NOT USED	NOT USED	GND B1 (B)		
20	S-Y	NOT USED	NOT USED	NOT USED	-72V EXT (Y)		
46	V-BL	NOT USED	NOT USED	NOT USED	NOT USED		
21	BL-V	NOT USED	NOT USED	NOT USED	NOT USED		
47	V-O	NOT USED	NOT USED	NOT USED	NOT USED		
22	O-V	NOT USED	NOT USED	NOT USED	NOT USED		
48	V-G	NOT USED	NOT USED	NOT USED	NOT USED		
23	G-V	NOT USED	NOT USED	NOT USED	NOT USED		
49	V-BR	NOT USED	NOT USED	NOT USED	NOT USED		
24	BR-V	NOT USED	NOT USED	NOT USED	NOT USED		
50	V-S	NOT USED	+36	AC RING SIG.	NOT USED		
25	S-V	NOT USED	GROUND	AC RING RET.	NOT USED		

* The station circuit numbers shown above are for the first station card installed in the equipment chassis. The extension numbers are the default numbers assigned to those circuits.

Table 3-7. *Station Cable Terminations on the Station Blocks with a U.S. System (Continued)*

Amphenol #	Cable Pair	AKSC	Circuit No.*	Ext. No.*
26	W-BL	+30VDC	01.01.01	1000
1	BL-W	GND		
27	W-O	PRIMARY PATH		
2	O-W	PRIMARY PATH		
28	W-G	SECONDARY PATH		
3	G-W	SECONDARY PATH		
29	W-BR	+30VDC	01.02.01	1001
4	BR-W	GND		
30	W-S	PRIMARY PATH		
5	S-W	PRIMARY PATH		
31	R-BL	SECONDARY PATH		
6	BL-R	SECONDARY PATH		
32	R-O	+30VDC	01.03.01	1002
7	O-R	GND		
33	R-G	PRIMARY PATH		
8	G-R	PRIMARY PATH		
34	R-BR	SECONDARY PATH		
9	BR-R	SECONDARY PATH		
35	R-S	+30VDC	01.04.01	1003
10	S-R	GND		
36	BK-BL	PRIMARY PATH		
11	BL-BK	PRIMARY PATH		
37	BK-O	SECONDARY PATH		
12	O-BK	SECONDARY PATH		
38	BK-G	+30VDC	01.05.01	1004
13	G-BK	GND		
39	BK-BR	PRIMARY PATH		
14	BR-BK	PRIMARY PATH		
40	BK-S	SECONDARY PATH		
15	S-BK	SECONDARY PATH		
41	Y-BL	+30VDC	01.06.01	1005
16	BL-Y	GND		
42	Y-O	PRIMARY PATH		
17	O-Y	PRIMARY PATH		
43	Y-G	SECONDARY PATH		
18	G-Y	SECONDARY PATH		
44	Y-BR	+30VDC	01.07.01	1006
19	BR-Y	GND		
45	Y-S	PRIMARY PATH		
20	S-Y	PRIMARY PATH		
46	V-BL	SECONDARY PATH		
21	BL-V	SECONDARY PATH		
47	V-O	+30VDC	01.08.01	1007
22	O-V	GND		
48	V-G	PRIMARY PATH		
23	G-V	PRIMARY PATH		
49	V-BR	SECONDARY PATH		
24	BR-V	SECONDARY PATH		
50	V-S	NOT USED		
25	S-V	NOT USED		

* The circuit numbers shown above are for the first station card in the equipment chassis. The extension numbers are the default numbers assigned to those circuits.

Table 3-7. *Station Cable Terminations on the Station Blocks with a U.S. System (Continued)*

Amphenol #	Cable Pair	EKSC	Circuit No.*	Ext. No.*
26	W-BL	TT	01.01.01	1000
1	BL-W	TR		
27	W-O	RT		
2	O-W	RR		
28	W-G	TT	01.02.01	1001
3	G-W	TR		
29	W-BR	RT		
4	BR-W	RR		
30	W-S	TT	01.03.01	1002
5	S-W	TR		
31	R-BL	RT		
6	BL-R	RR		
32	R-O	TT	01.04.01	1003
7	O-R	TR		
33	R-G	RT		
8	G-R	RR		
34	R-BR	TT	01.05.01	1004
9	BR-R	TR		
35	R-S	RT		
10	S-R	RR		
36	BK-BL	TT	01.06.01	1005
11	BL-BK	TR		
37	BK-O	RT		
12	O-BK	RR		
38	BK-G	TT	01.07.01	1006
13	G-BK	TR		
39	BK-BR	RT		
14	BR-BK	RR		
40	BK-S	TT	01.08.01	1007
15	S-BK	TR		
41	Y-BL	RT		
16	BL-Y	RR		
42	Y-O	TT	01.09.01	1008
17	O-Y	TR		
43	Y-G	RT		
18	G-Y	RR		
44	Y-BR	TT	01.10.01	1009
19	BR-Y	TR		
45	Y-S	RT		
20	S-Y	RR		
46	V-BL	TT	01.11.01	1010
21	BL-V	TR		
47	V-O	RT		
22	O-V	RR		
48	V-G	TT	01.12.01	1011
23	G-V	TR		
49	V-BR	RT		
24	BR-V	RR		
50	V-S	NOT USED		
25	S-V	NOT USED		

* The circuit numbers shown above are for the first station card in the equipment chassis. The extension numbers are the default numbers assigned to those circuits.

Table 3-8. *DKSC/SLC/LSC-D Cable Terminations on the MDF Blocks (Europe)*

Single Block Cables for use with DKSC, SLC, & LSC-D

Cable Pair	Color	Plug Pin No.	Block No.	Block Pair	Circuit
1	White/Blue Blue/White	1 26	1	1	01.01
2	White/Orange Orange/White	2 27		2	02.01
3	White/Green Green/White	3 28		3	03.01
4	White/Brown Brown/White	4 29		4	04.01
5	White/Gray Gray/White	5 30		5	05.01
6	Red/Blue Blue/Red	6 31		6	06.01
7	Red/Orange Orange/Red	7 32		7	07.01
8	Red/Green Green/Red	8 33		8	08.01
9	Red/Brown Brown/Red	9 34		9	Do not use
10	Red/Gray Gray/Red	10 35		10	

Table 3-9. *DKSC-16 Cable Terminations on the MDF Blocks (Europe)*

Twin Block Cables for use with DKSC-16

Cable Pair	Color	Plug Pin No.	Block No.	Block Pair	Circuit
1	White/Blue Blue/White	1 26	1	1	01.01
2	White/Orange Orange/White	2 27		2	02.01
3	White/Green Green/White	3 28		3	03.01
4	White/Brown Brown/White	4 29		4	04.01
5	White/Gray Gray/White	5 30		5	05.01
6	Red/Blue Blue/Red	6 31		6	06.01
7	Red/Orange Orange/Red	7 32		7	07.01
8	Red/Green Green/Red	8 33		8	08.01
9	Red/Brown Brown/Red	9 34		9	09.01
10	Red/Gray Gray/Red	10 35		10	10.01
11	Black/Blue Blue/Black	11 36	2	1	11.01
12	Black/Orange Orange/Black	12 37		2	12.01
13	Black/Green Green/Black	13 38		3	13.01
14	Black/Brown Brown/Black	14 39		4	14.01
15	Black/Gray Gray/Black	15 40		5	15.01
16	Yellow/Blue Blue/Yellow	16 41		6	16.01
17	Yellow/Orange Orange/Yellow	17 42		7	Do not use
18	Yellow/Green Green/Yellow	18 43		8	
19	Yellow/Brown Brown/Yellow	24 25		9	
20	Yellow/Gray Gray/Yellow	49 50		10	

Table 3-10. *DKSC-16+ Cable Terminations on the MDF Blocks (Europe)*

Twin Block Cables for use with DKSC-16 + Cabinet

Cable Pair	Color	Plug Pin No.	Block No.	Block Pair	Circuit
1	White/Blue Blue/White	1 26	1	1	01.01
2	White/Orange Orange/White	2 27		2	02.01
3	White/Green Green/White	3 28		3	03.01
4	White/Brown Brown/White	4 29		4	04.01
5	White/Gray Gray/White	5 30		5	05.01
6	Red/Blue Blue/Red	6 31		6	06.01
7	Red/Orange Orange/Red	7 32		7	07.01
8	Red/Green Green/Red	8 33		8	08.01
9	Red/Brown Brown/Red	9 34		9	09.01
10	Red/Gray Gray/Red	10 35		10	10.01
11	Black/Blue Blue/Black	11 36	2	1	11.01
12	Black/Orange Orange/Black	12 37		2	12.01
13	Black/Green Green/Black	13 38		3	13.01
14	Black/Brown Brown/Black	14 39		4	14.01
15	Black/Gray Gray/Black	15 40		5	15.01
16	Yellow/Blue Blue/Yellow	16 41		6	16.01
17	Yellow/Orange Orange/Yellow	17 42		7	Not used
18	Yellow/Green Green/Yellow	18 43		8	Not used
19	Yellow/Brown Brown/Yellow	24 25		9	Not used GND
20	Yellow/Gray Gray/Yellow	49 50		10	Not used +36V DC

7.46 For longer station loop lengths (up to 154 ohms/3,000 ft.), an optional power supply can be connected to the DKSC-16 termination block. It must be a 36V DC power supply capable of providing 120 mA per phone (1.92A for all 16 circuits). This option is only available on DKSC-16s that are at revision level 2 or later or that have been modified and labeled with FSM100995. The strap on jumper JP1 on the DKSC-16 Card must be placed over pins 1 and 2 to enable the external power supply.

Table 3-11. *AKSC Cable Terminations on the MDF Blocks (Europe)*

Triple Block Cables for use with AKSC

Cable Pair	Color	Plug Pin No.	Block No.	Block Pair	Circuit	
1	White/Blue Blue/White	1 26	1	1		PWR
2	White/Orange Orange/White	2 27		2	01.01	PRIM
3	White/Green Green/White	3 28		3		SEC
4	White/Brown Brown/White	4 29		4	02.01	PWR
5	White/Gray Gray/White	5 30		5		PRIM
6	Red/Blue Blue/Red	6 31		6		SEC
7	Red/Orange Orange/Red	7 32		7		PWR
8	Red/Green Green/Red	8 33		8	03.01	PRIM
9	Red/Brown Brown/Red	9 34		9		SEC
–				10		
10	Red/Gray Gray/Red	10 35	2	1		PWR
11	Black/Blue Blue/Black	11 36		2	04.01	PRIM
12	Black/Orange Orange/Black	12 37		3		SEC
13	Black/Green Green/Black	13 38		4		PWR
14	Black/Brown Brown/Black	14 39		5	05.01	PRIM
15	Black/Gray Gray/Black	15 40		6		SEC
16	Yellow/Blue Blue/Yellow	16 41		7		PWR
17	Yellow/Orange Orange/Yellow	17 42		8	06.01	PRIM
18	Yellow/Green Green/Yellow	18 43		9		SEC
–				10		

Table 3-11. *AKSC Cable Terminations on the MDF Blocks (Europe) (Continued)*

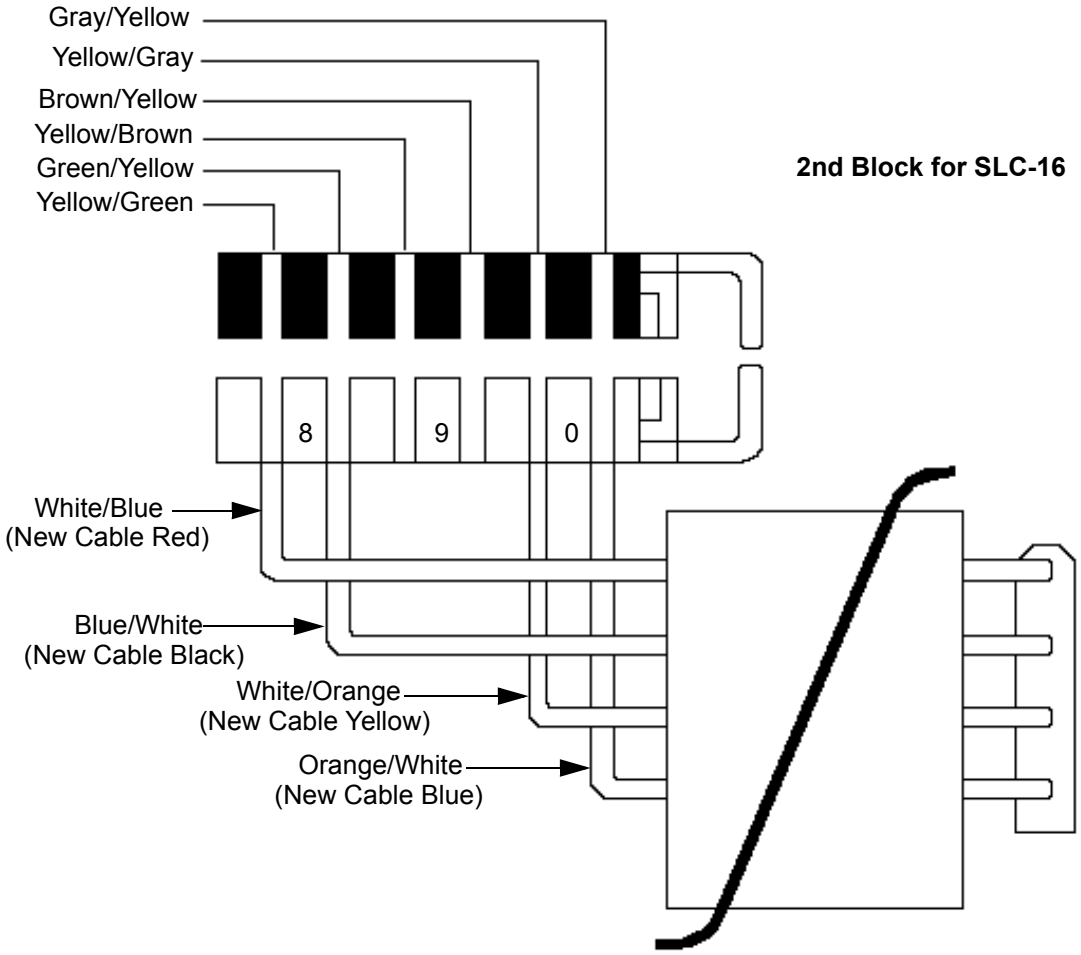
Cable Pair	Color	Plug Pin No.	Block No.	Block Pair	Circuit	
19	Yellow/Brown Brown/Yellow	19 44	3	1	07.01	PWR
20	Yellow/Gray Gray/Yellow	20 45		2		PRIM
21	Violet/Blue Blue/Violet	21 46		3		SEC
22	Violet/Orange Orange/Violet	22 47		4	08.01	PWR
23	Violet/Green Green/Violet	23 48		5		PRIM
24	Violet/Brown Brown/Violet	24 49		6		SEC
25	Violet/Gray Gray/Violet	25 50		7	Do not use	
–				8		
–				9		
–				10		

Table 3-12. *SLC-16 Cable Terminations on the MDF Blocks (Europe)*

Twin Block Cables for use with SLC-16

Cable Pair	Color	Plug Pin No.	Block No.	Block Pair	Circuit
1	White/Blue Blue/White	1 26	1	1	01.01
2	White/Orange Orange/White	2 27		2	02.01
3	White/Green Green/White	3 28		3	03.01
4	White/Brown Brown/White	4 29		4	04.01
5	White/Gray Gray/White	5 30		5	05.01
6	Red/Blue Blue/Red	6 31		6	06.01
7	Red/Orange Orange/Red	7 32		7	07.01
8	Red/Green Green/Red	8 33		8	08.01
9	Red/Brown Brown/Red	9 34		9	09.01
10	Red/Gray Gray/Red	10 35		10	10.01
11	Black/Blue Blue/Black	11 36	2	1	11.01
12	Black/Orange Orange/Black	12 37		2	12.01
13	Black/Green Green/Black	13 38		3	13.01
14	Black/Brown Brown/Black	14 39		4	14.01
15	Black/Gray Gray/Black	15 40		5	15.01
16	Yellow/Blue Blue/Yellow	16 41		6	16.01
17	Yellow/Orange Orange/Yellow	17 42		7	Not used
18	Yellow/Green Green/Yellow	18 43		8	-48V Ext GND for -48V
19	Yellow/Brown Brown/Yellow	19 44		9	Not used
20	Yellow/Gray Gray/Yellow	20 45		10	-72V Ext GND for -72V

Figure 3-33. *On-Site Krone Connections for SLC-16 (Europe Only)*



CAUTION

Please ensure that only the cable supplied with the SLC-16 is used. Cables supplied with other cards may not be suitable. Inter-Tel is not responsible for damage to the SLC-16, PSU, or any other equipment caused by using the wrong cable.

E. STATION LOOP RESISTANCE TEST

NOTE: Do not use shielded cable. Longer loop lengths can be achieved by using heavier gauge cable, such as 22AWG. (For additional station cabling requirements, see page 3-17.)

7.47 Perform the loop resistance test for each station cable individually.

1. With a U.S. system only, ensure that bridging clips have not been installed on the station blocks and that the station instrument is not connected to the modular jack assembly.
2. With a U.S. system, place a short across the BLACK/YELLOW wires (phones) or RED/GREEN wires (single-line sets) on the modular jack assembly.
   With a European system, place a short across the WHITE/BLUE wires (phones) or BLUE/WHITE wires (single-line sets) on the line jack unit assembly.
3. With a U.S. system, at the MDF, measure the resistance across the TIP and RING wires on the *right* (station) side of the station block.
   With a European system, at the MDF, measure the resistance across the WHITE/BLUE wires on the *lower* (station) side of the station block.

   The reading should not exceed the limits (for 24AWG, level-1 or better/24 gauge thickness or larger -- smaller number gauge = larger thickness) listed in the chart below. The resistance values (in ohms) indicate loop measurements; the distance values (in feet/meters) indicate the maximum one-way measurements from the equipment chassis. If necessary, longer loop lengths can be achieved by using a heavier gauge cable, such as 22AWG (22 gauge thickness or larger -- smaller number gauge = larger thickness).

   NOTE: Excessive and/or improperly made connections increase the resistance of a cable, which reduces the allowable cable run length.

4. Remove the short after the test is complete.
5. Repeat this test for each station cable.
6. With a U.S. system only, install bridging clips on the station blocks to complete the cable connections.

Digital Phones And Single-line Stations	Loop Limits
From KSU (EKSC) to Executone IDS or Wave Keysets	62 ohms/1200 ft (366 m)
From KSU (EKSC) with 30V external power supply to Executone Keysets	103 ohms/2000 ft (609 m)
From Chassis (DKSCs or DKSC-16s) to Digital Phones (Axxess® Basic, Eclipse™ Basic, Standard, Associate, Executive, Professional, 8500, 8520, and 8560) or Digital Phones with attached PC and Modem Data Port Modules or Mini-Direct Station Selection Units	62 ohms/1200 ft (366 m)
From Chassis (DKSC-16s) with 30V external power supply to Digital Phones	108 ohms/2100 ft (640 m)
From Chassis (DKSC-16s) with 36V external power supply to Digital Phones	154 ohms/3000 ft (915 m)
From Modem Data Port Modules to single-line sets	195 ohms/3800 ft (1158 m)
From Chassis to Single-Line Adapters (SLAs)	128 ohms/2500 ft (762 m)
From SLAs to single-line sets or playback devices	800 ohms/15560 ft (4742 m)
From Chassis (SLCs) to single-line sets or playback devices	257 ohms/5000 ft (1524 m)
From Chassis (SLC-16s) to single-line sets or playback devices	103 ohms/2000 ft (609 m)

NOTE: For information about analog phone loop limits, see a version 7.0 or earlier *Installation and Maintenance Manual*.

F. CONNECTING OPTIONAL OFF-PREMISES EXTENSIONS (OPX)

7.48 Off-premises extensions are remote single-line DTMF sets that are connected to the Inter-Tel system through telephone company [network provider] OPX circuits (or customer-provided circuits). The circuits are connected to Single-Line Adapter (SLA) circuits, which are connected to available Digital Keyset Card (DKSC) and/or 16-Port Digital Keyset Card (DKSC-16) circuits, or to T1/E1 circuits, which are connected to T1, E1, T1/PRI, or E1/PRI Card circuits.

NOTE: SLAs do not support *visual* message waiting applications. If single-line sets with message waiting lamps are installed on SLAs, the lamps do not function. However, single-line sets can be configured in DB Programming to receive *audible* message waiting indication tones.

7.49 Off-premises station instruments must be AC ringer-equipped. The SLA's onboard high-voltage ringing adapter (HVRA) provides isolation from the telephone company [network provider] OPX circuits as required by FCC regulations.

7.50 Each SLA can support up to two off-premises extensions. (On premises, SLA circuits can be used for installing AC-ringing single-line sets, playback devices, etc.)

7.51 In certain off-premises applications, voice volume levels may not be acceptable. This degradation in voice volume is due to the natural voice frequency range limitations of the telephone company [network provider] or customer-provided circuit. To increase the voice frequency range, installation of a two-wire, negative impedance repeater unit is recommended. (See page 2-86 for specifications.)

7.52 The repeater unit is installed at the MDF between the SLA and the cabling to the off-premises, single-line station. For proper installation with the Inter-Tel system, the following are connected to the appropriate "pins" on the repeater unit (see the manufacturer's instructions for proper pin locations): -Battery (-24 to -56VDC), Ground, System Tip, System Ring, OPX Tip, and OPX Ring. Information on operating and adjusting the repeater unit is included with the product.

7.53 Install the off-premises extensions as follows (see Figure 3-27 on page 3-53):

1. At the MDF, terminate each telephone company [network provider] OPX circuit (or customer-provided circuit) on the desired SLA circuit.
2. At the off-premises location, terminate each telephone company OPX circuit (or customer-provided circuit) on the appropriate single-line set modular jack (line jack unit) assembly.

G. CONNECTING TWO T1 OR E1 CARDS

7.54 T1, E1/PRI or T1/E1/PRI cards are used for connecting system nodes in a network. Each network span requires two T1, E1/PRI, or T1/E1/PRI cards, one for each end of the span. The cards require the most recent firmware to support networking – use upgrade kit part number 828.1506 [SW206 in Europe]. Point-to-point clear channel T1 spans are used to connect the nodes. For each network span, order a clear channel, point-to-point T1 or E1 span from the service provider (not PRI service, even though you are using T1, E1/PRI, or T1/E1/PRI cards).

7.55 T1 Cards can be used to connect two non-networking telephone systems together using customer-provided cable. The systems may be located in separate buildings and the cabling between them may be as long as 6000 feet (1800 m), without repeaters.

7.56 When connecting two systems together, the line cord (or cabling) between the two T1 Cards must be wired so that the receive pair (tip 1 in and ring 1 in) on each card is connected to the transmit pair (tip out and ring out) on the other card, as shown in the figure below. (For information on programming various types of T1 or E1 installations, see page 6-126.)

Figure 3-34. *Connecting Two T1 Cards Together*

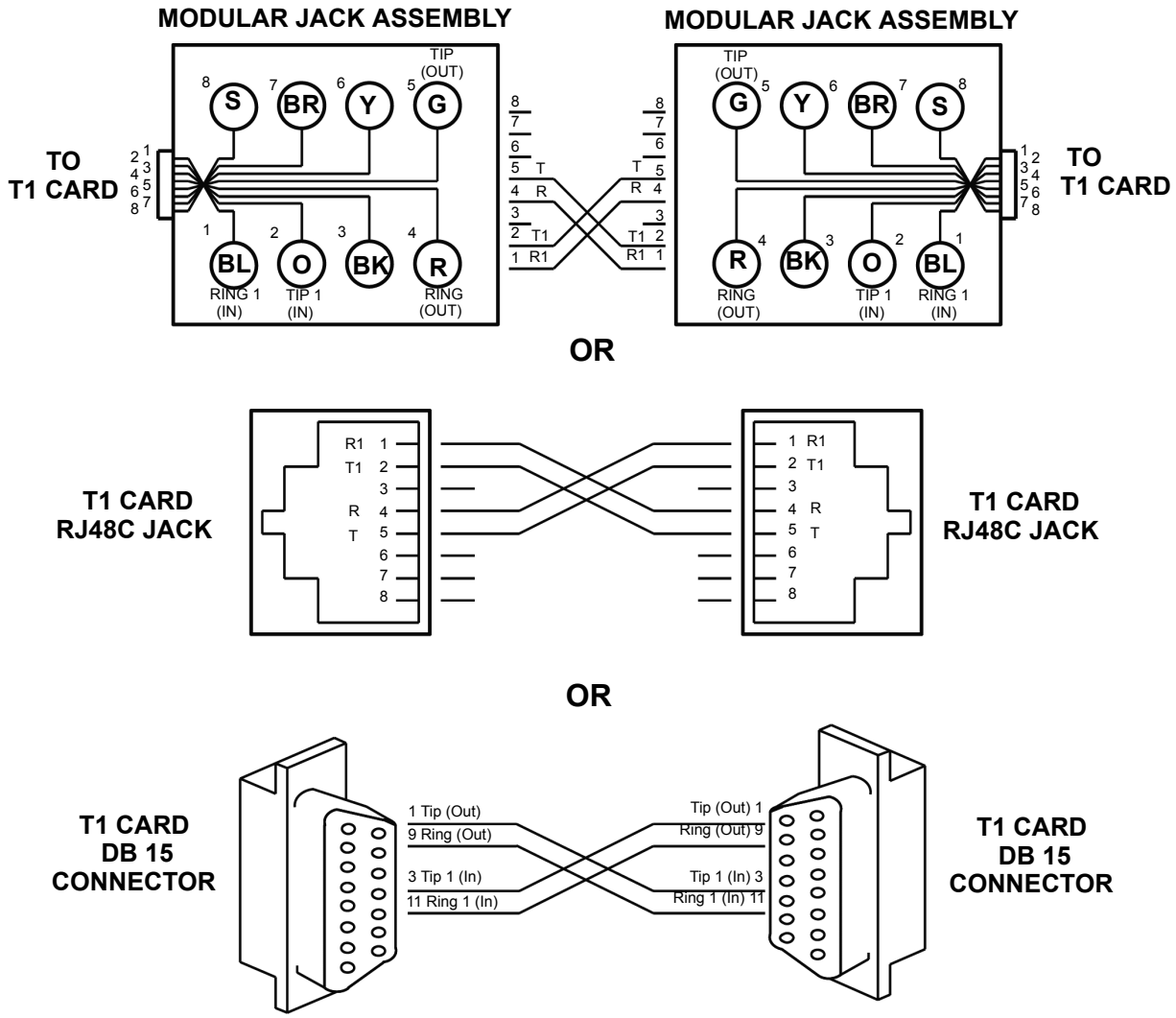
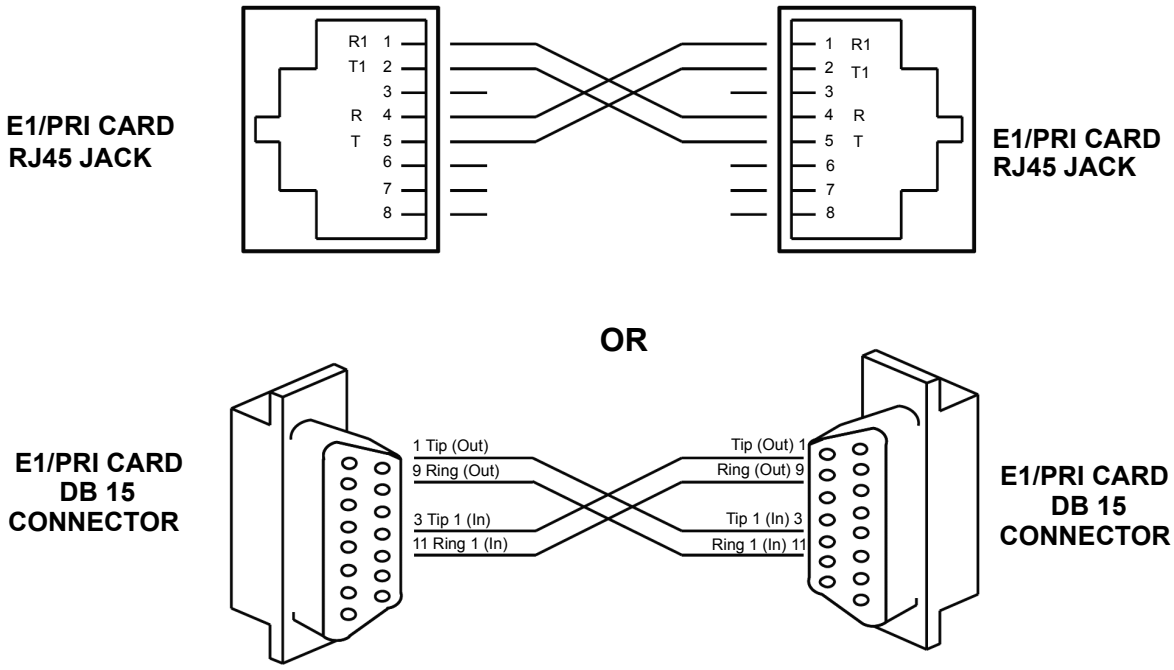


Figure 3-35. *Connecting Two E1 Cards Together*



H. ANALOG FALLBACK SYSTEM (EUROPE ONLY)

7.57 The fallback kit (part no. PS2728-A) contains the following items:

- 12 volt DC power supply (PS2736)
- Fallback Control Module and Control Lead (J36)
- Fallback Module 3-Circuit (J20)
- Charcoal Line Cord (Standard) (C27)

7.58 An additional five fallback modules may also be connected if required.

7.59 The PSU is fitted with a 5-pin din plug which is then attached to CH4 of the control module. The control module is supplied with a standard line cord that connects it to the CPU relay socket.

7.60 The power to each fallback module is jumpered, as shown in Figure 3-36 on page 3-74.

7.61 To use dedicated fallback telephones, the lines to the fallback unit should not be connected to the LSC but to the PSTN connections of the module. The LSC should be jumpered to the COU connections of the module and the dedicated fallback telephone connected to the F/B connections. See Figure 3-37 on page 3-74.

NOTE: The dedicated fallback telephones must be signaling compatible with the lines supplied.

7.62 To use system telephones as fallback telephones, two fallback circuits are required per line. The lines and telephones are not directly connected to the LSC or SLC but as shown in Figure 3-37 on page 3-74.

NOTE: The fallback telephone must be totally compatible with the system and the lines. The user should be made aware of any adjustments and procedures necessary to place calls.

7.63 Fallback operation can be tested by removing the line cord from the CPU. This forces all fallback circuits to operate and may disconnect any calls on these circuits.

Figure 3-36. *Fallback Power and Control*

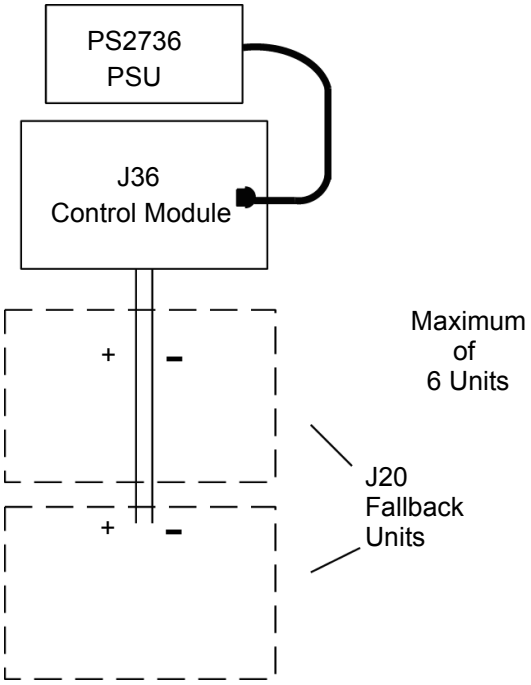
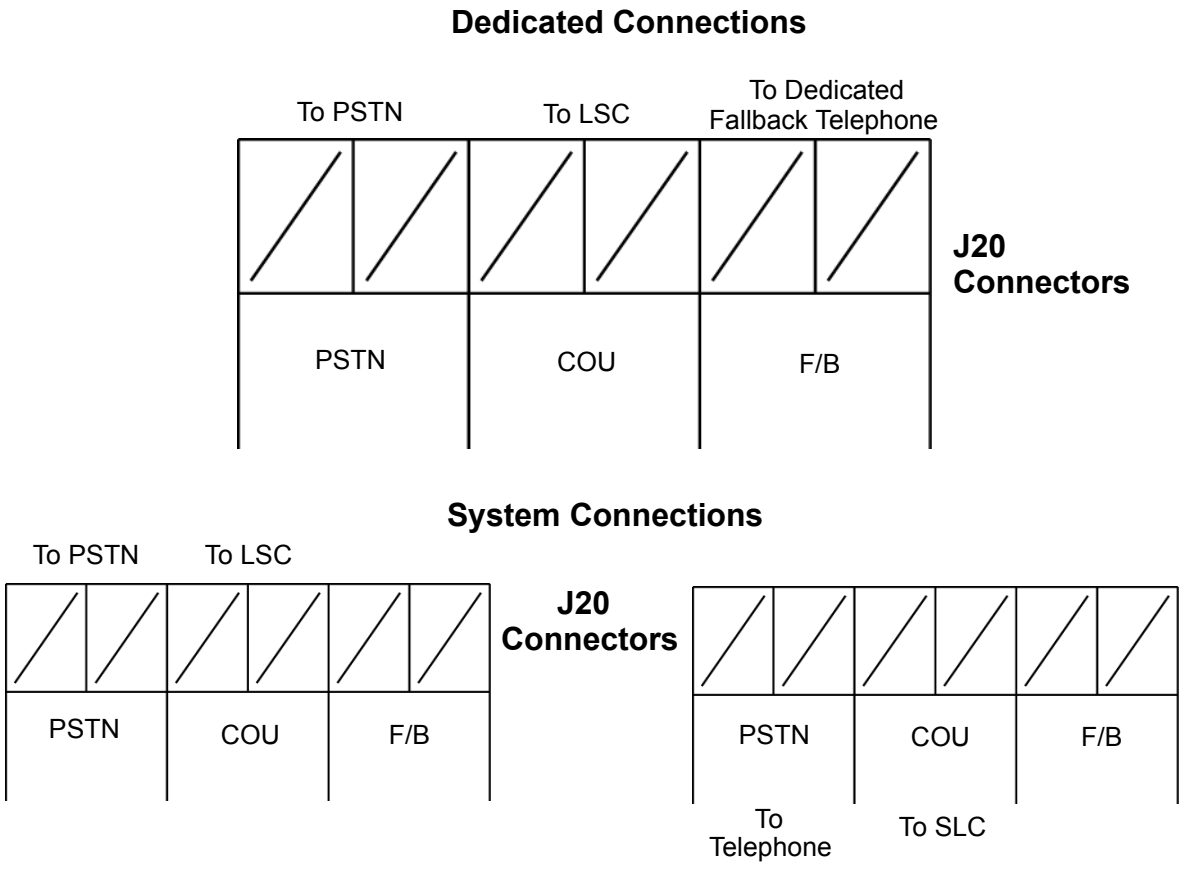


Figure 3-37. *Fallback Connections*



8. ORIGINAL EQUIPMENT CHASSIS INSTALLATION

NOTE: The original equipment chassis is not supported in Europe.

8.1 The original equipment chassis, backplane, and power supply unit have been redesigned. Installation instructions for the original equipment start below. Installation instructions for the new-style equipment start on page 3-84.

A. UNPACK THE EQUIPMENT

8.2 Unpack the equipment and inspect the equipment chassis and all accompanying parts following these guidelines:

1. Check the parts against the packing slip and inspect them for damage. If any item is missing or damaged, contact Customer Support.
2. Check all circuit card assemblies as follows:

   NOTE: The cards contain static-sensitive components. Handle all cards by the edges only; however, avoid touching the gold contacts on the side of the card that plugs into the chassis. (For additional precautionary information, see the CAUTION on page 3-93.)

   a. Check each card to ensure there is no shipping foam or tape attached.
   b. Inspect each card for shorted components.
   c. Return all cards to their protective bags until they are ready to be installed in the equipment chassis.

B. CHECK THE CHASSIS PRE-ASSEMBLY

8.3 The equipment chassis is shipped fully assembled. To ensure that none of the items have been damaged or have become unattached during shipment, follow these steps.

1. Position the chassis near the MDF backboard.

   NOTE: The chassis can be placed on a table or other secure horizontal surface, or it can be mounted on the wall as outlined in section E. Special feet are provided in Kit part number 828.1490. These feet must be used on any flat-surface installation.

2. Loosen the four screws securing the panel on the front of the chassis and set the panel aside.
3. Remove the four screws securing the panel on the top of the chassis and set the panel aside.
4. Check to see that the backplane is securely installed in the rear of the chassis.
5. Check to see that the power supply unit is securely installed in the top of the chassis.
6. Ensure that all cables are securely attached to their appropriate connectors.
7. *DO NOT* plug in the system AC power cable until the power supply electrical test, as outlined on page 3-83 has been performed.
8. *If installing a dual-chassis system,* connect the two basic chassis together as outlined in section C.
9. Wall mount the chassis, if desired (see page 3-86).
10. Ground the system according to the instructions on page 3-81.

C. ATTACH TWO UNITS TOGETHER, IF NECESSARY

8.4 To create a dual-chassis system, connect two basic chassis together as follows:

NOTE: If necessary, an original chassis and a new-style chassis can be connected together to create a dual-chassis system. If you are rack mounting the chassis or stacking them vertically, instead of following these procedures, use the longer expansion cable (part no. 813.1636-1B) to connect the chassis.

1. On one equipment chassis, remove the two screws securing the small access panel on the *right* side of the chassis and set the panel aside.
2. On the other chassis, remove the two screws securing the small access panel on the *left* side of the chassis and set the panel aside.
3. With the cutouts to the middle, place the two chassis side by side.
4. Using the screws removed in steps 1 and 2, connect the two chassis together as shown in Figure 3-38 on page 3-78. (The screws are inserted through the chassis on the left into the chassis on the right.)
5. Remove the resistor from connector RN2 on the backplane in the second and fourth chassis (right-hand, as viewed from the front) and attach it to connector RN5 on the *same* backplane. (See Figure 3-39 on page 3-79.) In the left-hand chassis, the resistor should remain in location RN2.

   NOTE: In tri-chassis and quad-chassis installations, the location of the resistor in the third chassis should match the location of the resistor in the first chassis. Similarly, the location of the resistor in the fourth chassis should match the location of the resistor in the second chassis.

6. Attach one end of the 60-pin chassis expansion cable (part no. 813.1576) to connector J10 on the backplane in the left-hand chassis.
7. Route the free end of the equipment chassis expansion cable through the cutout into the other equipment chassis and attach it to connector J9 on the backplane in the right-hand chassis.

NOTE: An ATM system supports up to four single- or dual-chassis. It does not support tri- or quad-chassis systems.

D. ATTACH THREE OR FOUR UNITS TOGETHER, IF NECESSARY

8.5 To create a tri-chassis system, simply connect one dual-chassis system to a single-chassis system using a fiber-optic cable between the two PCM-F cards. To create a quad-chassis system, simply connect two dual-chassis systems together using a fiber-optic cable between the two PCM-F cards. (See pages 2-11 and 2-12 for additional/other multiple-chassis system configuration information, and see page 3-104 for the actual fiber-optic connection instructions.) If necessary, the fiber-optic cable between the first (master) chassis and the third (slave) chassis can be up to one kilometer (.6 miles) in length. If running version 8.0 or later, you must use a CP Server for tri-/quad-chassis systems.

NOTE: A 8.5-foot (2.5-meter) cable is available using part no. 813.1628 (U.S.). Or, if necessary, contact Inter-Tel's CommSource™ division for information on purchasing longer cable lengths. Custom cables can also be built using industry-standard ST-type connectors.

E. WALL MOUNT THE CHASSIS, IF DESIRED

8.6 The basic or expanded chassis may be mounted on the MDF backboard as outlined in the following steps.

NOTE: If the two portions of a tri-/quad-chassis system are installed on the same vertical plane on the same MDF, the first (master) pair of chassis must be separated from the second (slave) pair of chassis by at least one foot to provide proper air circulation.

1. With the top cover removed, position the chassis on the MDF backboard and mark the location of the *top* two mounting screw holes. (Be sure to allow sufficient air circulation around the chassis.) Set the equipment chassis aside.
2. Drive a screw into the center of each mounting hole marking, allowing the heads of the screws to protrude slightly.

   NOTE: Use screws of sufficient strength to support the equipment chassis.

3. Hang the chassis on the two screws. If necessary, adjust the screws to ensure that the chassis is held firmly in place.
4. To further secure the chassis to the backboard, insert screws into the *bottom* two mounting holes from inside the chassis.

Figure 3-38. *Connecting Two Equipment Chassis Together*

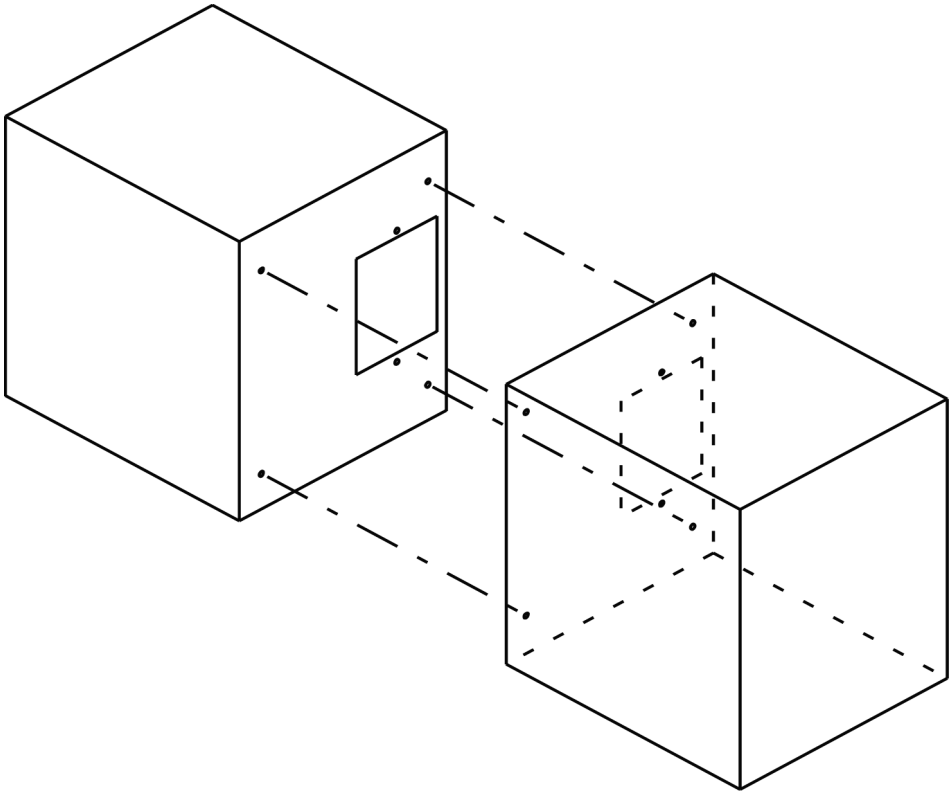
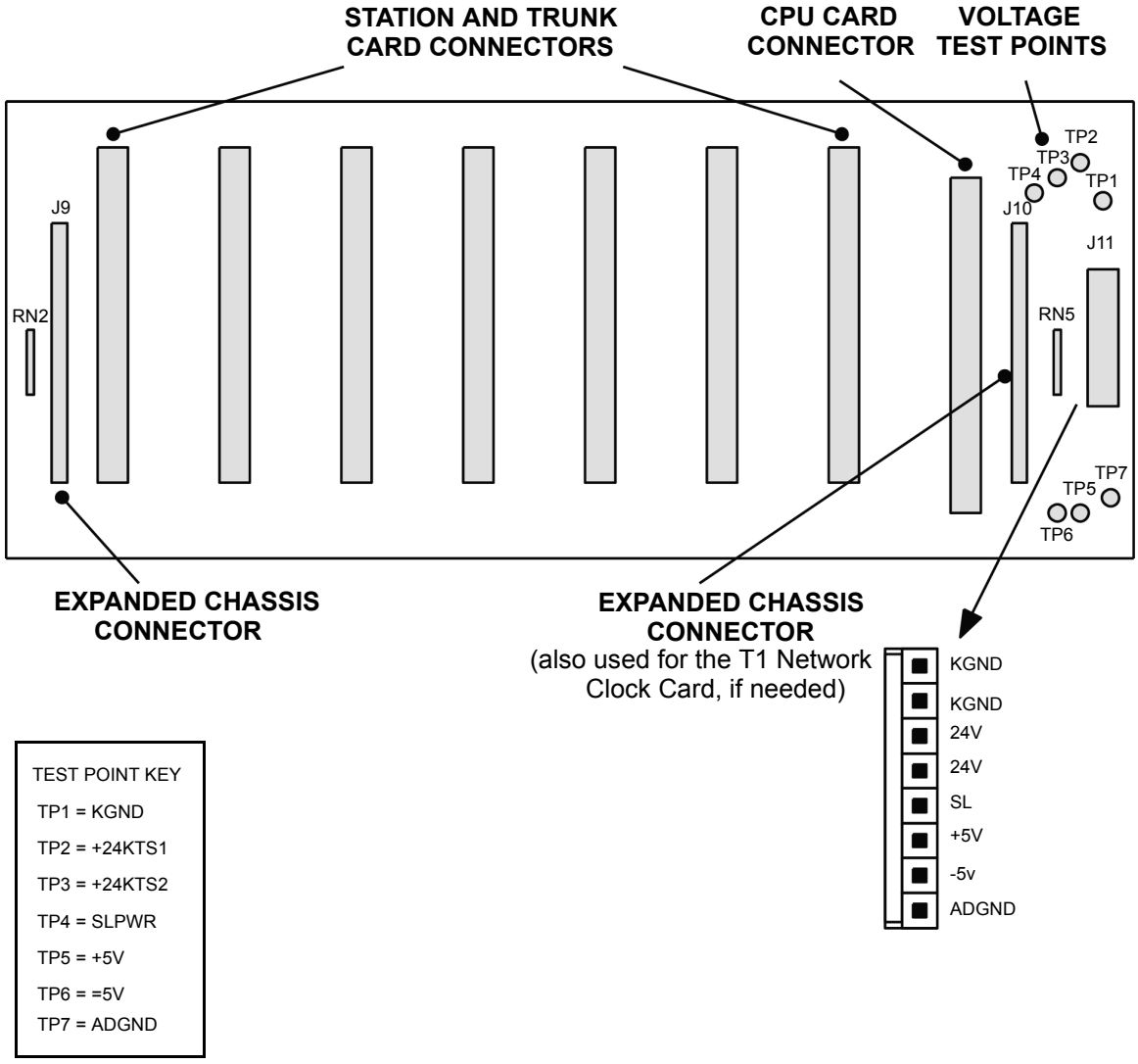


Figure 3-39. *Original Equipment Chassis Backplane*



F. REPLACE THE ORIGINAL POWER SUPPLY UNIT, IF NECESSARY

CAUTION

HIGH VOLTAGE. When the system power is turned on, the power supply unit contains 110VAC. Before inserting or removing the power supply unit, make sure that the AC POWER switch has been turned OFF for at least five minutes. Failure to allow the high voltage to dissipate can cause bodily injury.

8.7 If the power supply is defective, replace the unit as follows (see Figure 3-39 on the following page):

1. Remove the four screws securing the panel on the front of the equipment chassis and set the panel aside.
2. Turn OFF the system AC POWER switch and wait at least five minutes.
3. Disconnect the AC power cable and the ground wire from the front of the power supply unit.
4. Remove the four screws securing the panel on the top of the equipment chassis and set the panel aside.
5. Disconnect the backplane-to-power supply interface cable from connector J11 on the chassis backplane.
6. Remove the three screws securing the power supply unit to the top of the chassis cardfile (see Figure 3-39 on the next page).
7. Lift the power supply unit up out of the chassis and set it aside.
8. Insert the new power supply unit and replace the three screws.
9. Connect the new backplane-to-power supply interface cable to connector J11 on the chassis backplane.
10. Replace the top panel on the chassis.
11. Connect the AC power cable and the ground wire to the front of the new power supply unit.
12. Remove all the circuit cards from the chassis, then perform the power supply electrical test as outlined on page 3-83.

CAUTION

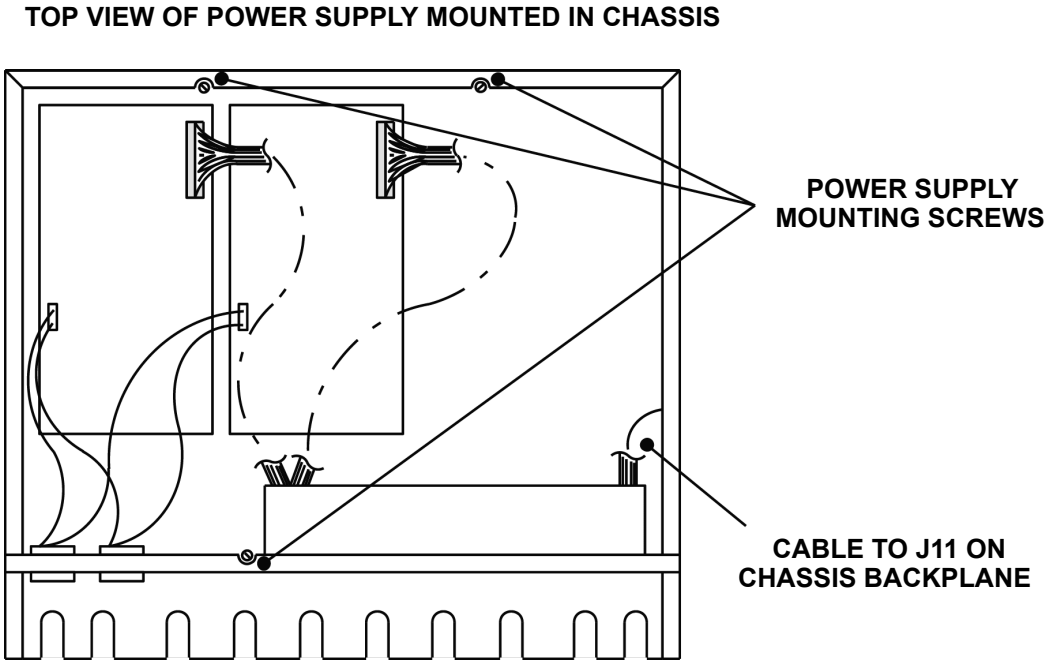
The power supply must be properly tested and grounded to prevent possible system damage.

13. Ensure that the power supply unit is properly grounded by performing one of the following tests:
    - Use a digital multimeter to measure the difference of potential between the chassis ground point and the power neutral, and between the chassis ground point and the third wire ground. If the voltage measurement exceeds 0.5V, noise may develop on the system. If this occurs, call an electrician.
    - Use a Megger to test the ground point. (A Megger should be available through the local power company.) The reading should be 5 ohms or less.

NOTE: For additional information on grounding requirements, see page 3-81.

14. Turn OFF the system AC POWER switch, replace the circuit cards in the equipment chassis, and then turn the system AC POWER switch back ON.
15. Replace the front panel on the chassis.
16. To receive proper credit for the advance replacement, return the defective power supply unit to Inter-Tel.

Figure 3-40. *Original Power Supply Unit Installation*



Grounding Requirements

8.8 The equipment chassis must be plugged into a properly grounded AC power source. (See page 3-10 for additional AC power requirements.) Some good ground points include the building's frame, a ground rod (driven a minimum of 8 ft/2.4 m), and a copper (not PVC) cold water pipe. For proper grounding methods, please see the diagram on the following page.

8.9 In addition, all optional Single-Line Adapters, trunk gas discharge tubes, and UPS/SPS battery compartments that are used must be properly grounded to the chassis AC outlet. Inter-Tel's CommSource division or Inter-Tel Europe sells a surge/spike protector (model DTK-3GTP) that has a built-in ground lug just for this purpose.

8.10 The grounding wire should be 10AWG (10 gauge thickness or larger -- smaller number gauge = larger thickness) or larger and should not exceed 25 feet (7.5 meters) to help provide RFI/EMI protection, or should not exceed 10 feet (3 meters) to help provide both RFI/EMI and lightning protection.

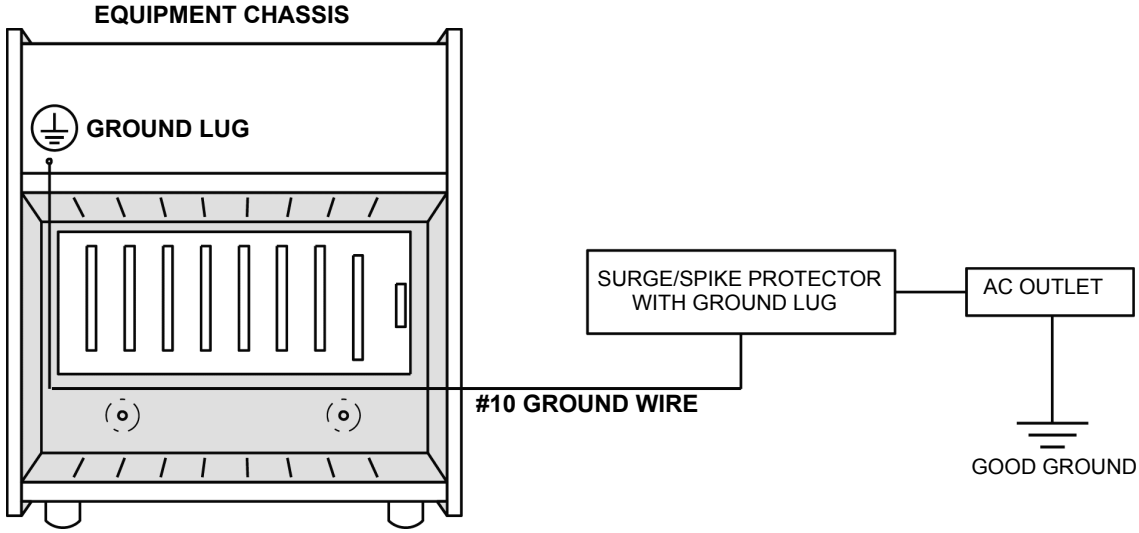
8.11 To ground the system (see Figure 3-41 below):

1. Ensure that the system's AC power source is properly grounded. *DO NOT* defeat (or leave floating) the ground wire in the AC power cord.
2. Install an AC surge/spike protector with a built-in ground lug and connect it to:
   a. The ground lug (indicated by the symbol ⏚) on the front of the power supply unit.
   b. The ground lug (indicated by the symbol ⏚) on each Single-Line Adapter, if used.
   c. Battery compartments, if used.
   d. Trunk gas discharge tubes, if used.

8.12 If AC power-related problems appear on the system while in operation, one of the following three methods can be used to test the ground.

- Use a digital multimeter to measure the difference of potential between the equipment chassis ground point and the power neutral, and between the chassis ground point and the third wire ground. If the voltage measurement exceeds 0.5V, noise may develop on the system. If this occurs, call an electrician.
- Use a Megger to test the ground point. (A Megger should be available through the local power company.) The reading should be 5 ohms or less.
- Unplug the AC power cord from the outlet and insert a ground isolation plug into the outlet. Plug the AC power cord into the ground isolation plug. If the problem ceases, call an electrician. When finished, remove the ground isolation plug to restore ground protection.

Figure 3-41. *Equipment Chassis Grounding*



Power Supply Electrical Test

8.13 Perform the following electrical check on the power supply.

1. Ensure that the system AC power cable is not plugged in and the AC POWER switch on the front of the power supply is OFF.
2. Plug the recommended surge/spike protector provided into a 105-125 VAC, 57-63 Hz, 15A, single-phase commercial power source.

NOTE: This MUST be an isolated, dedicated circuit. DO NOT use an extension cord when installing the power supply. Also, DO NOT plug any other equipment into the same AC outlet.

3. Plug the system AC power cable into the surge/spike protector and turn ON the AC POWER switch.
4. Observe the four LEDs on the front of the power supply unit. All four LEDs (+5V, -5V, +24V1, and +24V2) should be lit. *If any of the LEDs are blinking or are unlit*, the power supply is defective and the entire unit must be replaced.

NOTE: A blinking LED indicates that an overly high voltage has been detected. A lit LED merely indicates that there is some voltage present (as shown in the table at the right); it does not, however, indicate whether the voltage level is acceptable. To determine the exact voltage level present, perform the next step.

Voltage	LED	Output
+5VDC	OFF ON BLINKING	1V >1V & <6.5V 6.5V
-5VDC	OFF ON BLINKING	-1V <-1V & >-6.5V -6.5V
+24VDC	OFF ON BLINKING	1V >1V & <29V 29V

5. Referring to Figure 3-38 on page 3-78, measure the following system voltages on the chassis backplane. A digital voltmeter of ±0.5% accuracy is required. Insert the "common" voltmeter probe into the ground point and insert the other probe into the desired voltage test point (TP1 is ground for test points 2, 3, and 4; TP7 is ground for test points 5 and 6). The measurements should be within the limits listed in the following chart.

Voltage	Tolerance
-5VDC	±5%
+5VDC	±5%
+24VDC	±5%
SLPWR (+24VDC)	±5%.

6. *If the voltages are within the specified limits,* turn OFF the AC POWER switch and proceed to the next section.
7. *If voltage measurements are not within the specified limits,* replace the defective power supply unit.

CAUTION

DO NOT attempt to adjust the power supply voltages. Damage caused by unauthorized voltage adjustment is not covered by the warranty.

9. NEW-TYPE EQUIPMENT CHASSIS INSTALLATION

9.1 The original equipment chassis, backplane, and power supply unit have been redesigned. Installation instructions for the new-style equipment start below. Installation instructions for the original equipment start back on page 3-75.

A. UNPACK THE EQUIPMENT

9.2 Unpack the equipment and inspect the equipment chassis and all accompanying parts following these guidelines:

1. Check the parts against the packing slip and inspect them for damage. If any item is missing or damaged, contact Customer Support immediately.
2. Check all circuit card assemblies as follows:

NOTE: The cards contain static-sensitive components. Handle all cards by the edges only; however, avoid touching the gold contacts on the side of the card that plugs into the equipment chassis. (For additional precautionary information, see the CAUTION on page 3-93.)

   a. Check each card to ensure there is no shipping foam or tape attached.
   b. Inspect each card for shorted components.
   c. Return all cards to their protective bags until they are ready to be installed in the equipment chassis.

B. CHECK THE CHASSIS PRE-ASSEMBLY

9.3 The equipment chassis is shipped fully assembled. To ensure none of the items have been damaged or have become unattached during shipment, follow these steps.

1. Position the chassis near the MDF backboard.

NOTE: The chassis can be placed on a table or other secure horizontal surface, or it can be mounted on the wall as outlined in section E.

2. Loosen the four screws securing the panel on the front of the chassis and set the panel aside.
3. Ensure that the backplane is securely installed in the rear of the chassis.
4. *DO NOT* plug in the system AC power cable until the power supply electrical test, as outlined on page 3-83, has been performed.
5. *If installing a dual-chassis system,* connect the two basic chassis together as outlined in section C.
6. To wall mount the chassis, if desired, see page 3-77.
7. Ground the system according to the instructions on page 3-90.

C. ATTACH TWO UNITS TOGETHER, IF NECESSARY

9.4 To create a dual-chassis system, connect two basic chassis together as follows:

NOTE: If necessary, an original equipment chassis and a new-style chassis can be connected together to create a dual-chassis system. If you are rack mounting the chassis or stacking them vertically, instead of following these procedures, use the longer expansion cable (part no. 813.1636) to connect the chassis.

1. On one equipment chassis, remove the two screws securing the small access panel on the *right* side of the chassis and set the panel aside.
2. On the other equipment chassis, remove the two screws securing the small access panel on the *left* side of the chassis and set the panel aside.
3. With the cutouts to the middle, place the two chassis side by side.
4. Using the screws removed in steps 1 and 2, connect the two chassis together as shown in Figure 3-42 on page 3-87. (The screws are inserted through the chassis on the left into the chassis on the right.)
5. Remove the resistor from connector RN2 on the backplane in the right-hand chassis (as viewed from the front) and attach it to connector RN5 on the *same* backplane. (See Figure 3-43 on page 3-88.) In the left-hand chassis, the resistor should remain in location RN2.

   NOTE: In tri-chassis and quad-chassis installations, the location of the resistor in the third chassis should match the location of the resistor in the first chassis. Similarly, the location of the resistor in the fourth chassis should match the location of the resistor in the second chassis.

6. Attach one end of the 60-pin chassis expansion cable – part no. 813.1576 [C99 in Europe] to connector J10 on the backplane in the left-hand chassis.
7. Route the free end of the chassis expansion cable through the cutout into the other chassis and attach it to connector J9 on the backplane in the right-hand chassis.

NOTE: An ATM system supports up to four single- or dual-chassis. It does not support tri- or quad-chassis.

D. ATTACH THREE OR FOUR UNITS TOGETHER, IF NECESSARY

9.5 To create a tri-chassis system, simply connect one dual-chassis system to a single-chassis system using a fiber-optic cable between the two PCM-F cards. To create a quad-chassis system, simply connect two dual-chassis systems together using a fiber-optic cable between the two PCM-F cards. (See page 2-12 for additional/other multiple-chassis system configuration information, and see page 3-104 for the actual fiber-optic connection instructions.) If necessary, the fiber-optic cable between the first (master) chassis and the third (slave) chassis can be up to one kilometer (.6 miles) in length. If running version 8.0 or later, you must use a CP Server for tri-/quad-chassis systems.

NOTE: A 8.5-foot (2.5 m) cable is available using part no. 813.1628 [C118 in Europe]. Or, if necessary, contact Inter-Tel's CommSource division or Inter-Tel Europe for information on purchasing longer cable lengths. Custom cables can also be built using industry-standard ST-type connectors.

E. WALL MOUNT THE CHASSIS, IF DESIRED

9.6 The basic or expanded chassis may be mounted on the MDF backboard as outlined in the following steps.

NOTE: If the two portions of a tri-/quad-chassis system are installed on the same vertical plane on the same MDF, the first (master) pair of chassis must be separated from the second (slave) pair of chassis by at least one foot to provide proper air circulation.

1. With the front cover removed, position the chassis on the MDF backboard and mark the location of the *top* two mounting screw holes. (Be sure to allow sufficient air circulation around the chassis.) Set the chassis aside.
2. Drive a screw into the center of each mounting hole marking, allowing the heads of the screws to protrude slightly.

   NOTE: Use screws of sufficient strength to support the equipment chassis.

3. Hang the chassis on the two screws. If necessary, adjust the screws to ensure the chassis is held firmly in place.
4. To further secure the chassis to the backboard, insert screws into the *bottom* two mounting holes from inside the chassis.

Figure 3-42. *Connecting Two Equipment Chassis Together*

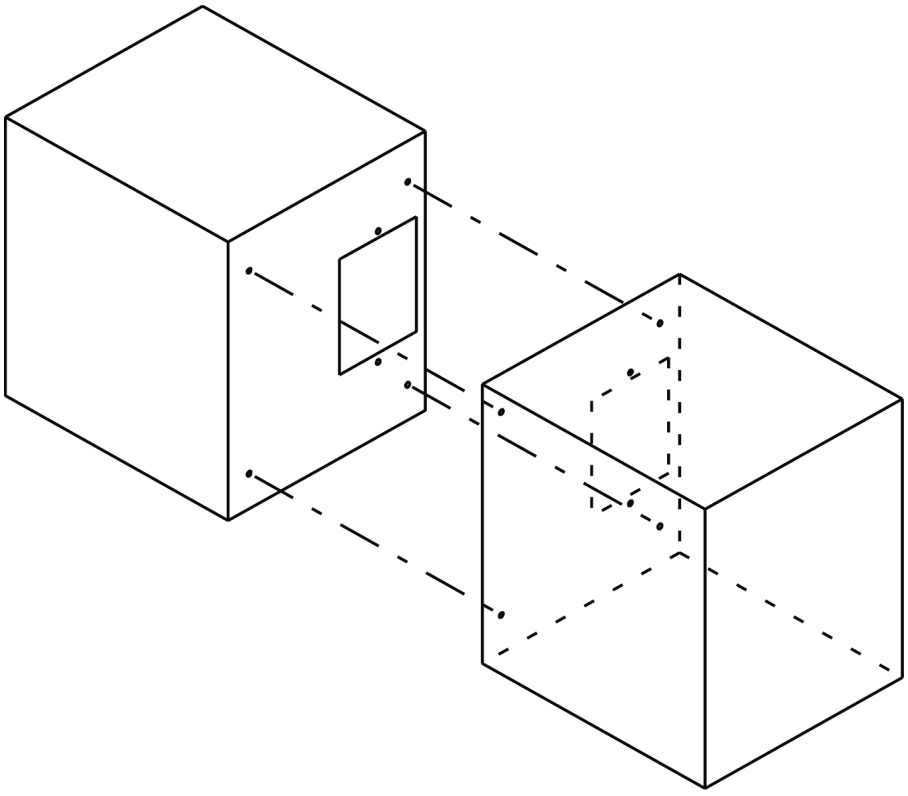
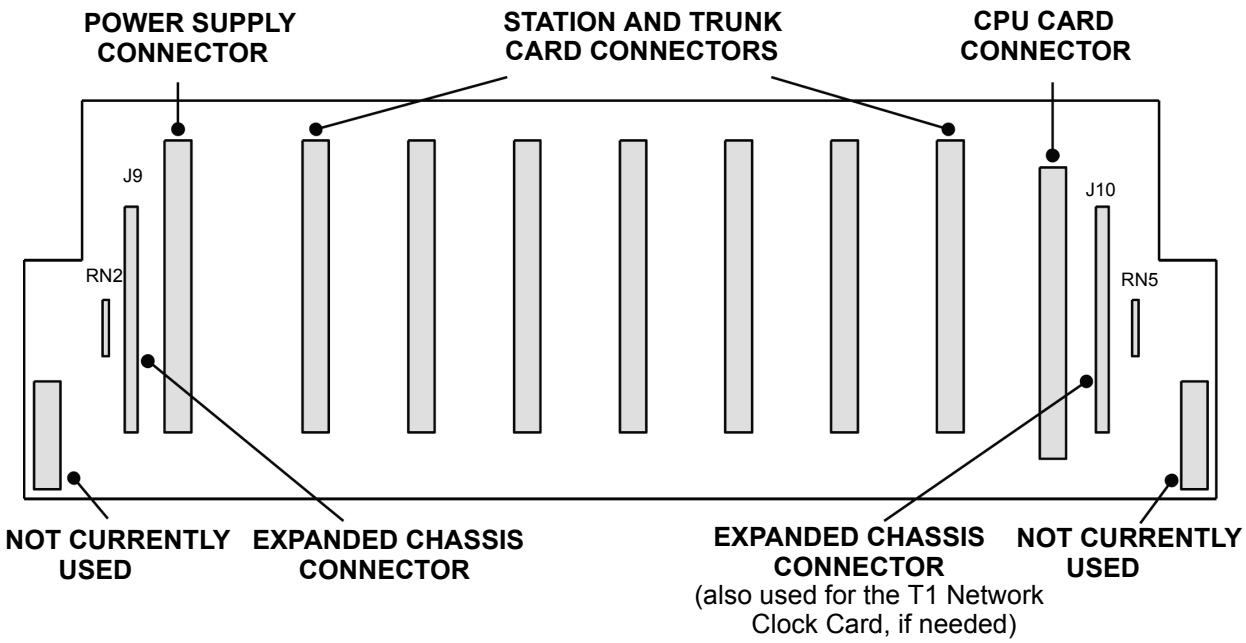


Figure 3-43. *New-Style Equipment Chassis Backplane*



F. INSTALL THE POWER SUPPLY UNIT

CAUTION

HIGH VOLTAGE. When the system power is turned ON, the power supply unit contains 110VAC. Before inserting or removing the power supply unit, make sure the AC POWER switch has been turned OFF for at least five minutes. Failure to allow the high voltage to dissipate can cause bodily injury.

9.7 Install the power supply unit in the chassis as outlined in the following steps (see Figure 3-44 below):

1. Remove the four screws securing the panel on the front of the chassis and set the panel aside.
2. Carefully slide the power supply unit into the far left slot in the chassis until it locks securely into place.

NOTE: The 6-amp power supply unit is not supported in Europe.

3. Connect the AC power cable to the front of the power supply unit.

Figure 3-44. *9-Amp or 6-Amp Power Supply Unit*

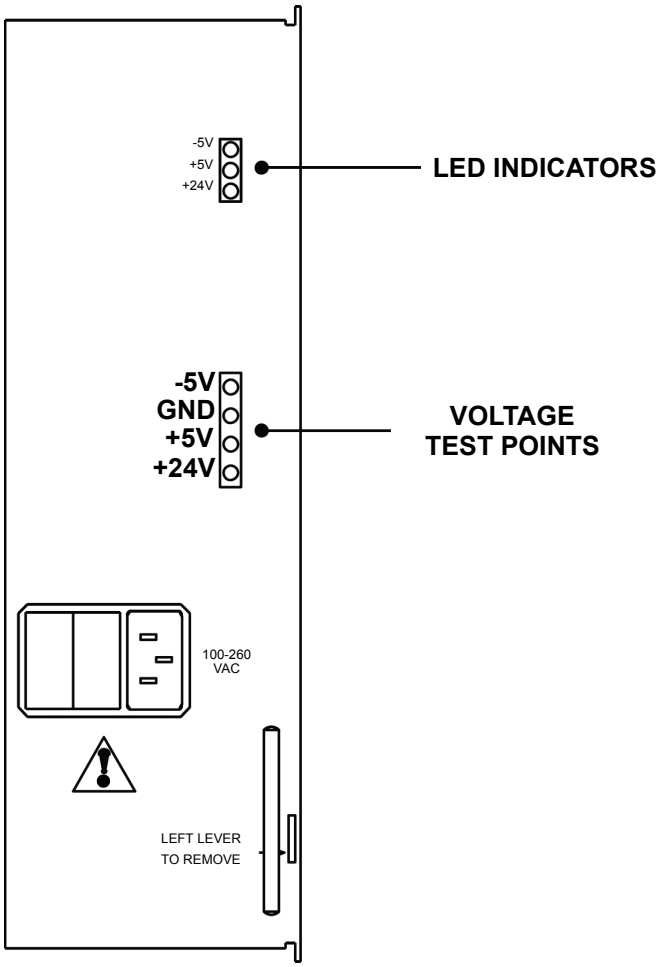
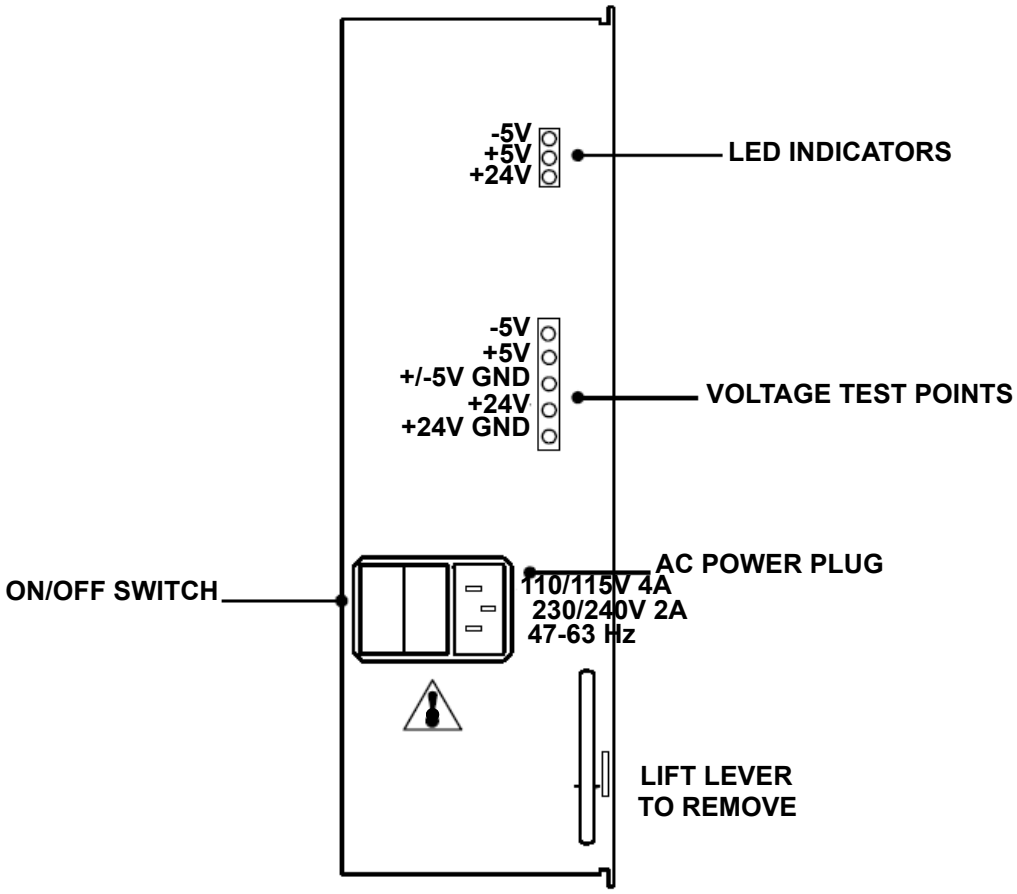


Figure 3-45. *4-Amp Power Supply Unit*



Grounding Requirements

9.8 With a U.S. system, the equipment chassis must be plugged into a properly grounded AC power source. (See page 3-10 for additional AC power requirements.) Some good ground points include the building's frame, a ground rod (driven a minimum of 8 ft), and a copper (not PVC) cold water pipe.

9.9 With a European system, the equipment chassis must be plugged into a properly grounded AC power source. (See page 3-10 for additional AC power requirements.) The cabinet and Krone field should also be hardwire earthed to the building earth.

9.10 In addition, all optional Single-Line Adapters, trunk gas discharge tubes, and UPS/SPS battery compartments that are used must be properly grounded to the chassis AC outlet. Inter-Tel's CommSource division or Inter-Tel Europe sells a surge/spike protector that has a built-in ground lug just for this purpose.

9.11 The grounding wire should be 10AWG (10 gauge thickness or larger -- smaller number gauge = larger thickness) or larger and should not exceed 25 feet (7.5 meters) to help provide RFI/EMI protection, or it should not exceed 10 feet (3 meters) to help provide both RFI/EMI and lightning protection.

9.12 To ground the system (see Figure 3-46 on the following page):

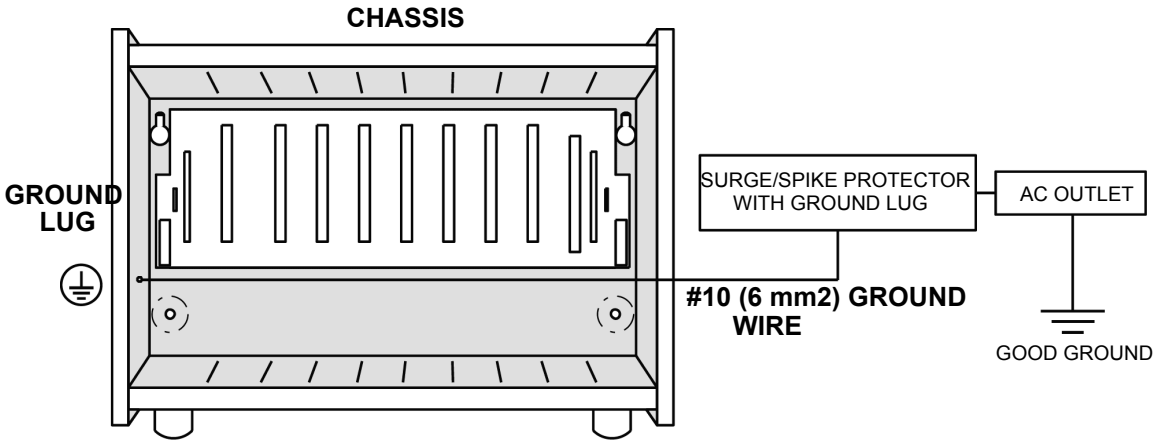
1. Ensure that the system's AC power source is properly grounded. *DO NOT* defeat (or leave floating) the ground wire in the AC power cord.
2. For a U.S. system only, install an AC surge/spike protector with a built-in ground lug and connect it to:
   a. The ground lug (indicated by the symbol ⏚) on the left, inside wall of the equipment chassis.
   b. The ground lug (indicated by the symbol ⏚) on each Single-Line Adapter, if used.
   c. Battery compartments, if used.
   d. Trunk gas discharge tubes, if used.

9.13 If AC power-related problems appear on the system while in operation, one of the following three methods can be used to test the ground.

- Use a digital multimeter to measure the difference of potential between the chassis ground point and the power neutral, and between the chassis ground point and the third wire ground. If the voltage measurement exceeds 0.5V, noise may develop on the system. If this occurs, call an electrician.
- Use a Megger to test the ground point. (A Megger should be available through the local power company.) The reading should be 5 ohms or less.
- *(For a U.S. system only)* Unplug the AC power cord from the outlet and insert a ground isolation plug into the outlet. Plug the AC power cord into the ground isolation plug. If the problem ceases, call an electrician. When finished, remove the ground isolation plug to restore ground protection.

NOTE: For information about how to properly ground Inter-Tel telephone systems or updated ground information, refer to the *Inter-Tel System Equipment Grounding Guidelines* (part no. 835.2845).

Figure 3-46. *Equipment Chassis Grounding*



Power Supply Electrical Test

9.14 Perform the following electrical check on the power supply.

1. Ensure the system AC power cable is not plugged in and the AC POWER switch on the front of the power supply is OFF.

   With a U.S. system, plug the recommended surge/spike protector provided into a 105-125 VAC, 57-63 Hz, 15A, single-phase commercial power source.

   With a European system, plug the recommended surge/spike protector provided into a 230 VAC, 50 Hz, 13A, single-phase commercial power source.

   NOTE: This MUST be an isolated, dedicated circuit. DO NOT use an extension cord when installing the power supply. Also, DO NOT plug any other equipment into the same AC outlet.

2. Plug the system AC power cable into the surge/spike protector and turn ON the AC POWER switch.

3. After waiting five seconds, observe the three LEDs on the front of the power supply unit. All three LEDs +5V, -5V, and +24V) should be lit. *If any of the LEDs are blinking or are unlit*, the power supply is defective, and the entire unit must be replaced.

   NOTE: A blinking LED indicates that an overly high voltage has been detected. A lit LED merely indicates that there is some voltage present (as shown in the table at the right); it does not, however, indicate whether the voltage level is acceptable. To determine the exact voltage level present, perform the next step.

Voltage	LED	Output
+5VDC	OFF	1V
	ON	>1V & <6.5V
	BLINKING	6.5V
-5VDC	OFF	-1V
	ON	<-1V & >-6.5V
	BLINKING	-6.5V
+24VDC	OFF	1V
	ON	>1V & <29V
	BLINKING	29V

4. Referring to Figure 3-44 on page 3-89, measure the following system voltages on the front of the power supply unit. A digital voltmeter of ±0.5% accuracy is required. Insert the "common" voltmeter probe into the ground point and insert the other probe into the desired voltage test point. The measurements should be within the limits listed in the chart at the right.

Voltage	Tolerance
-5.3VDC	±5%
+5.3VDC	±5%
+24.6VDC	±5%

5. *If the voltages are within the specified limits,* turn OFF the AC POWER switch and proceed to the next section.

6. *If voltage measurements are not within the specified limits,* replace the defective power supply unit.

CAUTION

DO NOT attempt to adjust the power supply voltages. Damage caused by unauthorized voltage adjustment is not covered by the warranty.

10. CIRCUIT CARD INSTALLATION

10.1 The information in this section includes instructions about how to install the various circuit cards in the cabinet. With the exception of the CPC, CPU cards, and the Network Clock Card, the circuit cards can be installed in any card slot.

CAUTION

The cards and components are static-sensitive. Handle the cards by the edges only and keep them in their protective bags until they are to be installed. Do not bend or touch the pins of the components or subject them to a static discharge. When working with the cards, use a properly terminated anti-static wrist strap and cover the work surface with the bag that protected the card during shipping. Any static discharge (no matter how small) must be discharged from the body before touching the cards or components. *The warranty for this equipment does not cover damage caused by static or mishandling. Cards or components damaged in such a manner cannot be replaced.*

A. NETWORK CLOCK CARD (NCC)

NOTE: If upgrading from a CPU64 or CPU128 to a CPC, you must first remove any Network Clock Cards (NCCs) from the chassis's backplane before installing the CPC. If you do not remove NCCs from the backplane, the system continually resets. Also, because the CPC and CPU020 cards have built-in NCC functionality, NCCs are not required for version 8.0 and later systems.

10.2 If any optional T1, E1/PRI, or T1/E1/PRI cards are to be installed in an equipment using the CPU64/128 card configuration, a NCC must also be attached to the chassis backplane. To allow adequate room for installation, it is best to install the NCC before installing the CPU card.

NOTE: Do not install an NCC in chassis equipped with a CPU020/PCM card or CPC; it could cause T1 or E1 timing problems. If one is accidently installed, the programming PC displays a warning.

1. Ensure all integrated circuits and components on the card are seated securely in their sockets and that no pins are bent.
2. *Make sure the power supply's AC POWER switch is turned OFF. Installing the NCC with the power on may damage the circuitry on the card.*
3. Install the NCC with the components *facing left* in the expanded chassis connector slot (J10) on the far right side of the chassis backplane. (See backplane drawing on page 3-79.)

   NOTE: When two basic chassis are attached together to form a dual-chassis system, the NCC is installed only in the second (right-hand) chassis, as viewed from the front.

4. Turn OFF the AC POWER switch.

B. CALL PROCESSING CARD (CPC)

10.3 A CPC supports only systems with 64, 128, or 256 devices. For 512 devices and higher, a CP Server is required. Before you install the CPC, however, you must consider the following:

- Because the CPC has an IP port, there is a risk of security violations. Inter-Tel strongly recommends that you install the CPC behind a firewall or NAT box.
- If upgrading from a CPU64 or CPU128, you must first remove any Network Clock Cards (NCCs) from the chassis's backplane before installing the new CPC. If you do not remove NCCs from the backplane, the system continually resets.

10.4 To install the CPC in the chassis:

1. Make sure all power supply AC POWER switches are turned OFF. Installing the CPC with the power on may damage the circuitry on the card.
2. Measure the rechargeable battery voltage. The voltage test points are located on the top of the battery itself (+) and underneath the battery, on the opposite side of the card (-).

   To test the voltage:

   a. Insert the "common" voltmeter probe into the ground point (the hole underneath the battery).

   b. Place the other probe on the voltage test point (+).

   If the voltage is not greater than 3.3VDC, the battery may need recharging. Allow the battery to recharge for at least another five hours and then test it again. If the voltage is still not greater than 3.3VDC, replace the battery.
3. Install the CPC with the components *facing right* in the last (8th) card slot in the chassis.

   NOTE: Dual-chassis systems require only one CPC to control the entire system. However, the CPC must be installed in the eighth slot of the left-hand chassis (as viewed from the front).

4. Turn ON the AC POWER switch and observe the LEDs on the front edge of the card. The top four LEDs should sequentially flash in an up and down pattern for approximately 10 seconds. Then the CPU Active LED should light.

   NOTE: When a new system is installed, the DATABASE ERROR LED remains lit until any changes are made to the system database. The MINOR ALARM LED may also be lit and can later be cleared from an administrator station using the Clear System Alarm feature code (default is 9850).

5. Upload the software license (see page 6-45).

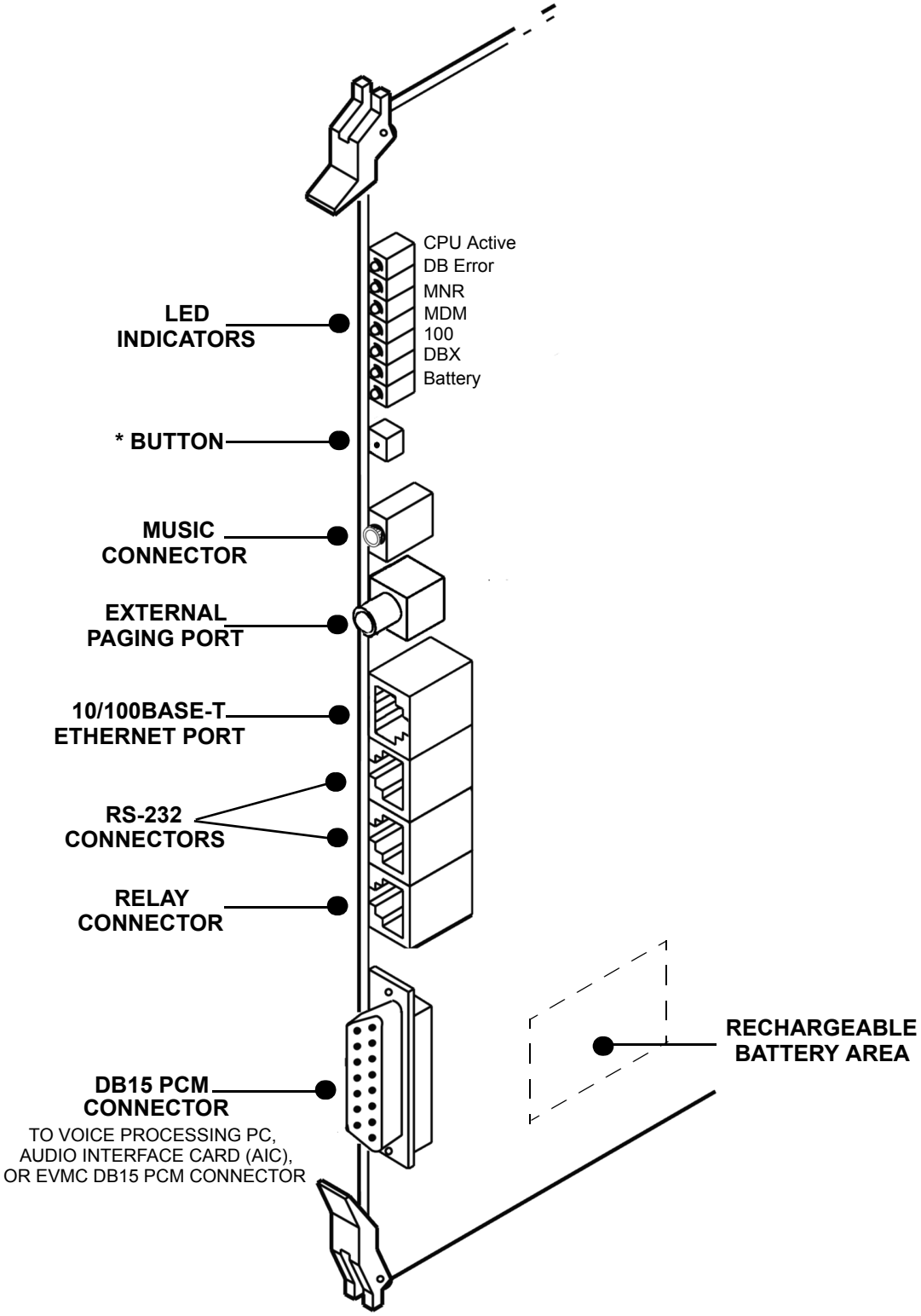
   NOTE: Inter-Tel highly recommends that you upload the software license before attempting to program the database. If you do not upload the software license, Alarm 125 is generated, and "License Error" appears on administrator phones. In addition, the system resets every four hours until a valid license is uploaded (see page 6-45).

6. *If the CPC does not have the appropriate system software already installed in flash memory,* use the Board Utility to upload the software from the programming PC to the card.

   NOTE: If using the IP port for the Board Utility, the Web Server flag must be enabled (see page 6-179).

7. After installing the CPC, the optional music source, external paging equipment, Voice Processing PC, EVMC, power failure transfer equipment, output device, etc. may be connected to the CPC at any time.

Figure 3-47. *Call Processing Card (CPC)*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front end of the card.

C. CPU64 OR CPU128 CARD

NOTE: The CPU64 and CPU128 cards cannot be used on a version 8.0 or later system. You must install a CPC for 64, 128, and 256 systems and a CP Server for 512 and higher systems.

10.5 **CPU128**: The Central Processing Unit 128 (CPU128) card can be used only in single-chassis or dual-chassis systems. In tri-chassis or quad-chassis systems, the CPU020-Q/PCM-F card must be used in the first (master) chassis and the CPU020-X/PCM-F card must be used in the third (slave) chassis. See page 2-27 for more information on the various CPU020/PCM combinations.

10.6 CPU128 with Memory Daughter Card: The CPU128 card (part no. 550.2017) comes equipped with a memory daughter card (part no. 550.2016) to support increased system features. A daughter card adds extra RAM to the existing CPU128 card. Additional capacity enables all available features on a system to function properly. This card replaces an existing CPU128 (part no. 550.2015). See Figure 3-49 on page 3-101 for a diagram of the CPU128 with the daughter card.

10.7 **CPU64**: The CPU64 card is identical to the CPU128 card, except that it only supports up to 64 devices and it can be used only in a single-chassis system. Like the CPU128, the CPU64 card uses flash memory circuitry. Other CPU64 features include the following:

- It can be used with the 4-amp, 6-amp, or 9-amp power supplies. (However, the EVMC cannot be used with the 6-amp power supply.)
- It can be used with any of the voice mail platforms (EVMC or Windows-based Voice Processing Unit).

  NOTE: The CPU can only support the EVMC if you are using version 5.270 software or later.

- It can be used in a node in a phone system network. However, the node should not be a hub and its voice mail system should not support any other nodes.
- If any T1 or E1 Cards are to be installed in an equipment chassis using the CPU64 card configuration, a single T1 or E1 Network Clock Card must also be installed. Then, instead of using the clock circuitry on the CPU card, the NCC becomes the clock for the entire system. The NCC provides the smooth, error-free clock transitions required for T1 or E1 applications. You can use only one PRI-equipped T1 or E1 card with the CPU64.

10.8 Install the CPU64 or CPU128 as follows:

1. Check the part number and socket number printed on the feature-unit PAL. They must match the software configuration purchased by the customer and should be labeled with the correct part number and installed in socket number U4. (The boot ROM and memory map PAL are already installed on the card.)
2. Place the CPU card on a clean, flat, anti-static, non-conductive surface.
3. Carefully install the feature unit PAL in the socket labeled “Feature Unit PAL” on the card. (See Figure 3-48 on page 3-100.) The notch (or small dot in the corner) indicating pin 1 must match the notch of the outline silk-screened on the card.

   NOTE: The label on the chip does not necessarily indicate its orientation.

4. Check that all components on the card are seated securely in their sockets and that no pins are bent.

10.9 Install the CPU card in the chassis as follows. See Figure 3-48 on page 3-100 for an illustration of the card.

1. Ensure all integrated circuits and components on the card are seated securely in their sockets and no pins are bent.
2. Ensure the mu-law/a-law jumper is in the correct position for your installation.
3. Place the battery back-up (BATTERY) jumper strap in the ON position to enable database battery back-up. If the jumper strap is placed in the OFF position, the BATTERY OFF LED lights when the card is installed and AC power is on.

NOTE: The BATTERY OFF LED does not light if the jumper strap is missing.

4. While pressing the battery check button, measure the database back-up battery voltage. The voltage test points (TP1 to TP2) are located on the front edge of the card. Insert the "common" voltmeter probe into the ground point (TP2) and insert the other probe into the voltage test point (TP1). If the voltage is not greater than 3.3VDC, replace the battery as described on page 3-112. (See Figure 3-48 on page 3-100 for back-up battery voltage test point locations.)
5. Make sure all power supply AC POWER switches are turned OFF. Installing the CPU card with the power on may damage the circuitry on the card.
6. Install the CPU card with the components *facing right* in the last (8th) card slot in the equipment chassis.

NOTE: Dual-chassis systems require only one CPU card to control the entire system. However, the CPU card must be installed in the eighth slot of the left-hand chassis (as viewed from the front).

7. Turn ON the AC POWER switch and observe the LEDs on the front edge of the card for the indications at the right. *If they are incorrect,* recheck the system voltages as outlined on page 3-92, and then contact Customer Support. (See Figure 3-48 on page 3-100 for LED indicator locations.)

Name	Correct Indication
BATTERY OFF	Off (Red)
CPU ACTIVE	Lit (Green)
DATABASE ERROR	Off* (Red)
EXT. CLOCK ACTIVE	Off (Green)
REMOTE SYNC	Off (Green)
LOCAL SYNC	Off (Green)

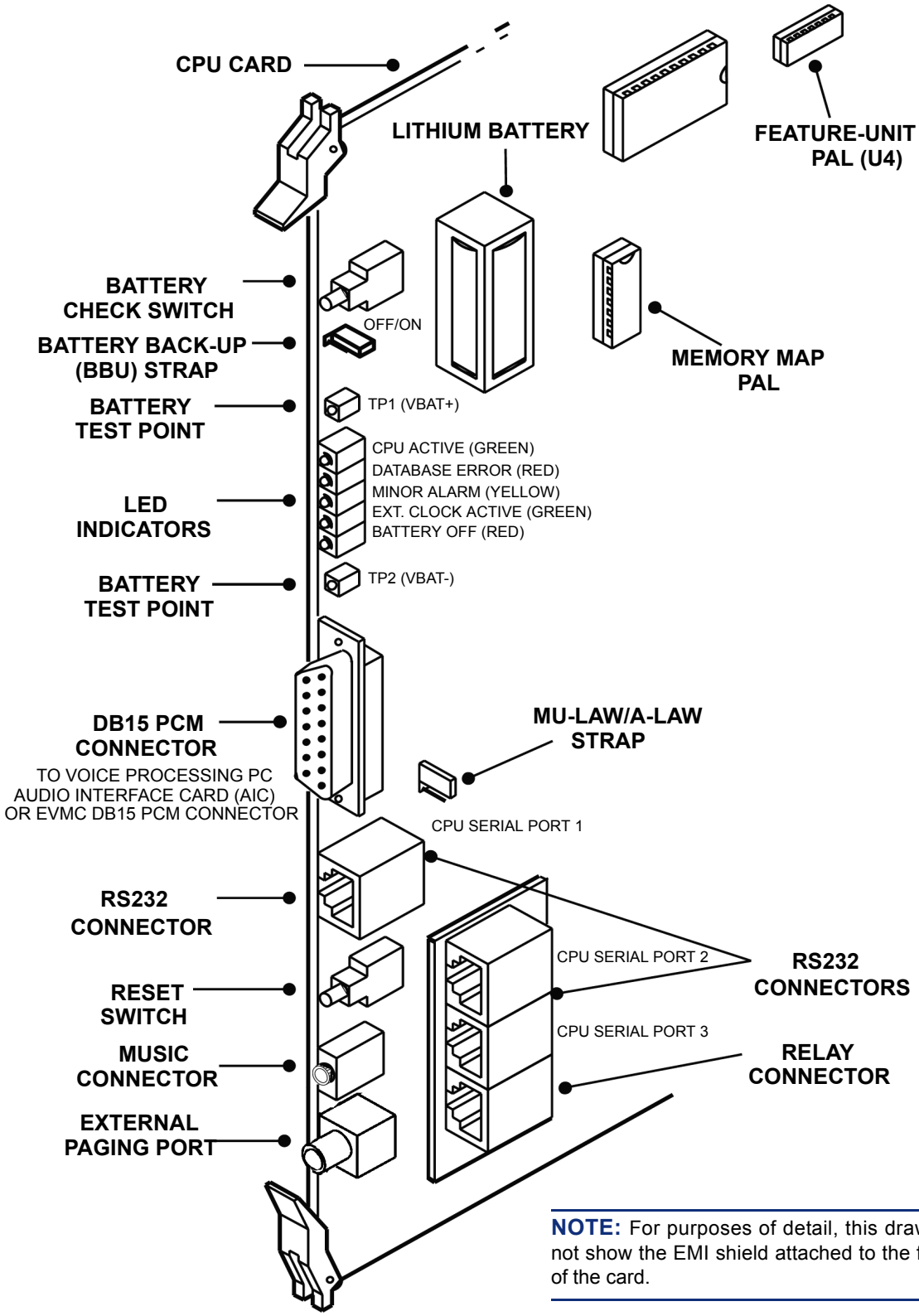
* When a new system is installed, the DATABASE ERROR LED remains lit until the system database has been programmed. The MINOR ALARM LED may also be lit and can later be cleared from an administrator station using the Clear System Alarm feature code (default is 9850).

8. *If the CPU card does not have the appropriate system software already installed in flash memory,* see page 3-173 to upload the software from the programming PC to the card.

9. After installing the CPU card, the optional music source, external paging equipment, Voice Processing PC, EVMC, power failure transfer equipment, output device, etc. may be connected to the CPU card at any time. (See page 3-213 for Voice Processing PC installation instructions and to page 3-240 for external music source installation instructions.)

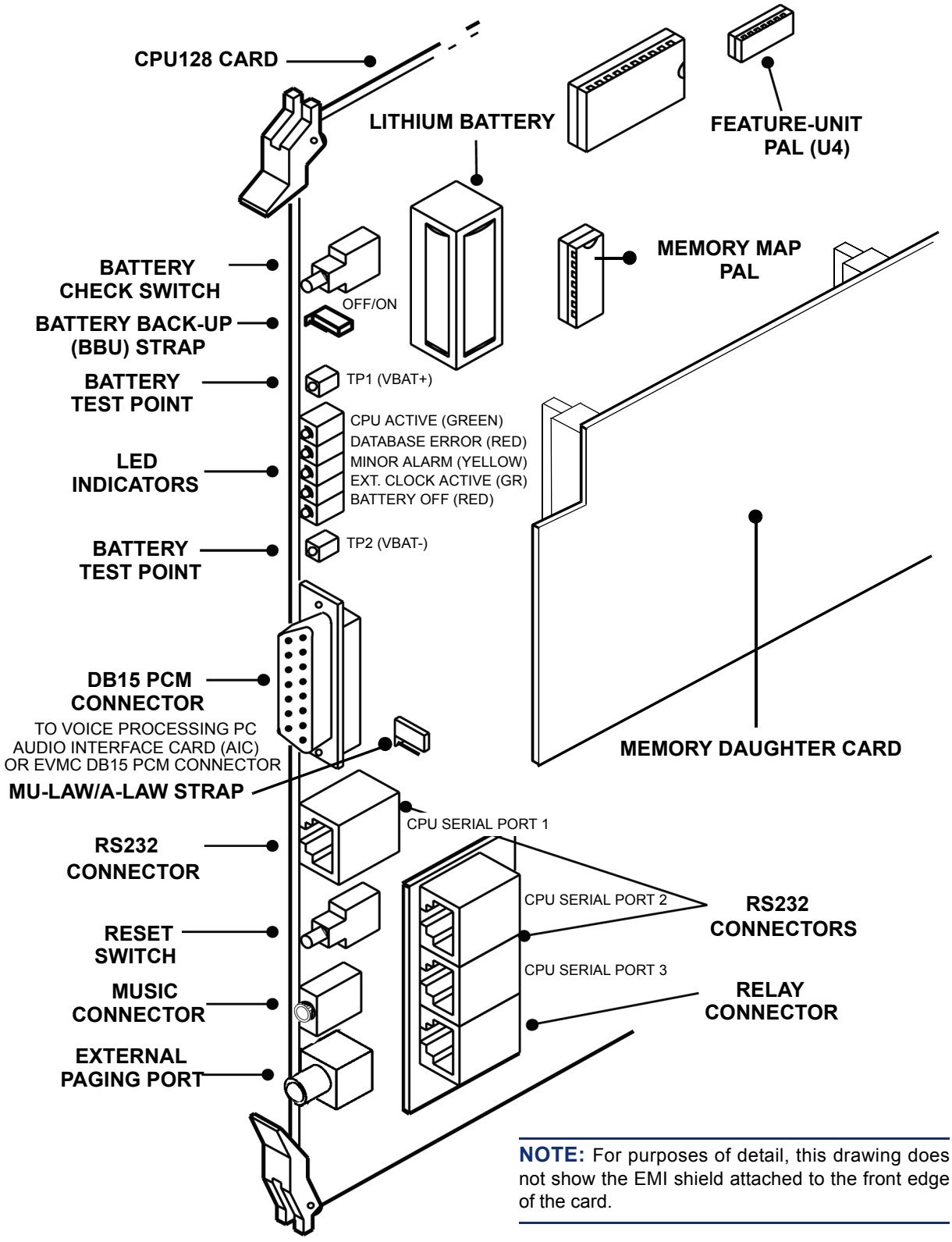
10. Turn OFF the AC POWER switch.

Figure 3-48. *CPU64 or CPU128 Card*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

Figure 3-49. *CPU128 with Memory Daughter Card*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

D. CPU020 CARDS

10.10 CPU020/PCM or CPU020/PCMA card combination kits are available. Each kit includes a CPU020 card, a PCM or PCMA card, the proper boot ROM and memory map PAL, and any additional memory components required. The combined cards are shipped pre-assembled, with the current system software already loaded. See page 2-27 for more information on the various CPU020/PCM and CPU020/PCMA combinations.

NOTE: As of version 8.0, a CPC or CP Server is required for all installations. To use the CP Server, a CPU020/PCM/EAC is required.

- **CPU020-D/PCM-D Card**: The CPU020-D/PCM-D (a.k.a. CPU 256) can be used only in single-chassis or dual-chassis systems.
- **CPU020-Q/PCM-F Card and CPU020-X Card/PCM-F Card**: In tri-chassis and quad-chassis system configurations, a CPU020-Q/PCM-F card (a.k.a. CPU 512 Master) must be used in the first (master) chassis, and a CPU020-X/PCM-F card (a.k.a. CPU 512 Slave) must be used in the third (slave) chassis. See pages 2-13 and 2-14 for drawings showing multiple-chassis system configurations.
- **CPU020-A/PCMA-F Card, CPU020-A/PCMA-FS Card, and CPU020-A/PCMA-C Card**: For an ATM system, a CPU020-A/PCMA card is required to connect to the ATM Switch and CP Server via CAT5 or MMF cables. The type of card required depends on the type of connectors available on the ATM Switch.

NOTE: The PCMA-C does not have power cross and lightning protection and should be used only for on-premises applications. The PCMA-F or PCMA-FS version may be used for off-premises applications.

CPU020 Card with EAC Connector

10.11 The newer CPU020 cards have an 80-pin connector that attaches to Ethernet Adapter Card (EAC). In addition, the PCM faceplate has an extra cutout for the RJ45 connector on the EAC.

10.12 The EAC supports 10Base-T and can be connected to the CP Server via an industry-standard category 5 crossed patch cable (813.1724). The cable is 14 feet (4.2 meters) in length allowing some freedom of placement. Per IEEE 802.3 10Base-T specification, this cable can be up to 328 feet (100 meters) long. For complete information about the CP Server, see Appendix C — CP Server.

NOTE: As of version 8.0, a CP Server is required for 512 and higher systems.

Installing CPU020/PCM or CPU020/PCMA Cards

10.13 If the CPU020 card and the PCM or PCMA card are purchased separately, you must install the special software components and connect the two cards as outlined below.

NOTE: The CPU card employs flash memory circuitry. This technology allows the technician to install and upgrade software through the programming PC, rather than having to replace the software ROMs on the card. For information on uploading the system software to the CPU card, see page 3-173.

1. Check the part numbers and socket numbers printed on the labels of the software components. The boot ROM and memory map PAL should be labeled with the part numbers and installed in socket numbers U25 and U41 respectively. If installing a PCMA card for an ATM-based system, make sure an ATM PAL is present on the board (U13).

   NOTE: The v7.0 ATM PAL is not compatible with v6.0 software. Make sure the ATM PAL in U13 is the correct part number (827.9683 for 7.0 and 827.9413 for 6.0). Version 8.0, however, supports any ATM PAL.

2. Place the CPU020 card on a clean, flat, anti-static, non-conductive surface.

3. Carefully install the software components in the corresponding sockets on the card. (See Figure 3-50 on page 3-107.) The notch (or small dot in the corner) indicating pin 1 on each component must match the notch of the outline silk-screened on the card.

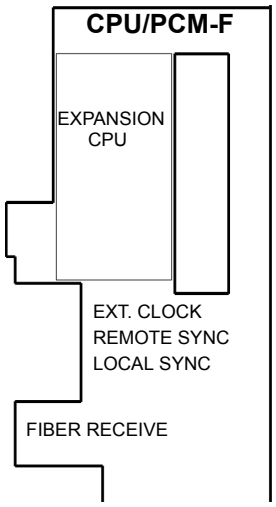
   NOTE: The labels on the chips do not necessarily indicate orientation.

4. Check that all components on the card are seated securely in their sockets and that no pins are bent.

5. Attach the CPU020 card to the PCM or PCMA card as shown in the figures on the following pages. Make sure the connectors and the standoffs are properly aligned and seated. If applicable, secure the cards together using the supplied screws.

   NOTE: If the components on the cards are not properly seated and/or if the two parts of the CPU020/PCM or CPU020/PCMA card are not properly attached and seated, the CPU ACTIVE LED may flash when the card is installed, and the card may not operate properly.

6. *For CPU020-X/PCM-F cards only:* Place the "EXPANSION CPU" sticker included with the CPU020-X card on the EMI shield (attached to the front edge of the combined card) as shown below.



10.14 Install the CPU020/PCM or CPU020/PCMA card in the chassis as follows. See the figures on the following pages for illustrations of the cards.

1. Ensure all integrated circuits and components on the card are seated securely in their sockets and no pins are bent. For PCMA cards, the ATM PAL must be present. If the ATM PAL is not present, the CP Server generates a warning message (see page 7-19) and does not bring that particular chassis online. The ATM CPU operating system, therefore, only acknowledges ATM PALs and ignores any standard set PALs.

   NOTE: The v7.0 ATM PAL is not compatible with v6.0 software. Make sure the ATM PAL in U13 is the correct part number (827.9683 for 7.0 and 827.9413 for 6.0). Version 8.0, however, supports any ATM PAL.

2. Place the battery back-up (BATTERY) jumper strap in the ON position (over the bottom two pins) to enable database battery back-up. If the jumper strap is placed in the OFF position (top two pins), the BATTERY OFF LED lights when the card is installed and AC power is on.

   NOTE: The BATTERY OFF LED does not light if the jumper strap is missing. Make sure **all** power supply AC POWER switches are turned OFF. Installing the CPU020/PCM or CPU020/PCMA card with the power on may damage the circuitry on the card.

3. Install the CPU020/PCM or CPU020/PCMA card with the components *facing right* in the last (8th) card slot in the equipment chassis. (See Figure 2-1 on page 2-13 for a multiple-chassis illustration.)

   NOTE: Dual-chassis systems require only one CPU020/PCM or CPU020/PCMA card to control the entire system. However, the CPU020/PCM or CPU020/PCMA card must be installed in the 8th slot of the left-hand chassis (as viewed from the front).

4. *For CPU020-X/PCM-F cards only:* Connect the fiber-optic cable between the PCM-F card in the first (master) chassis and the PCM-F card in the third (slave) chassis.

NOTE: The FIBER TRANSMIT jack on each card must be connected to the FIBER RECEIVE jack on the other card. A 8.5-foot (2.5 m) cable is available using part no. 813.1628 [C118 in Europe]. Or, if necessary, contact Inter-Tel's CommSource division or Inter-Tel Europe for information on purchasing longer cable lengths. Custom cables can also be built using industry-standard ST-type connectors.

5. *For CPU020-A/PCMA cards only:* Connect the fibre-optic or CAT5 cable between the PCMA card and the ATM Switch. If installing multiple chassis, make sure you connect them to different interfaces on the ATM Switch (see page 2-9 for details).

6. Turn ON the AC POWER switch and observe the LEDs on the front edge of the card for the indications below. If installing an ATM system, turn ON the power to the ATM Switch. *If the LEDs are incorrect,* recheck the system voltages as outlined on page 3-92, and then contact Customer Support.

CPU020-D/PCM-D: See Figure 3-50 on page 3-107 for LED indicator locations.

Table 3-13. *CPU020-D/PCM-D LED Indications*

Name	Correct Indication
Battery Off	Off (Red)
CPU Active	Lit (Green)
Database Error	Off* (Red)
Minor Alarm	Off* (Yellow)
External Clock Active	Off (Green)
Local Synch	Off (Green)

CPU020-Q/PCM-F: See Figure 3-51 on page 3-108 for LED indicator locations.

Table 3-14. *CPU020-Q/PCM-F LED Indications*

Name	Correct Indication
Battery Off	Off (Red)
CPU Active	Lit (Green)
Database Error	Off* (Red)
Minor Alarm	Off* (Yellow)
External Clock Active	Off (Green)
Remote Synch	Off (Green)
Local Synch	Off (Green)

CPU020-X/PCM-F: See Figure 3-52 on page 3-109 for LED indicator locations.

Table 3-15. *CPU020-X/PCM-F LED Indications*

Name	Correct Indication
CPU Active	Lit (Green)
Minor Alarm	Off* (Yellow)
External Clock Active	Off (Green)
Remote Synch	Off (Green)
Local Synch	Off (Green)

CPU020-A/PCMA-F/PCMA-FS/PCMA-C: See Figures 3-53 and 3-54 on pages 3-110 and 3-111 for LED indicator locations.

Table 3-16. *CPU020-A/PCMA-F/PCMA-FS/PCMA-C LED Indications*

Name	Correct Indication
Battery Off	Off (Red)
CPU Active	Lit (Green)
Database Error	Off* (Red)
Minor Alarm	Off* (Yellow)
External Clock Active	Off (Green)
ATM Link Bad	Off (Red)

* When a new system is installed, the DATABASE ERROR LED remains lit until the system database has been programmed. The MINOR ALARM LED may also be lit and can later be cleared from an administrator station using the Clear System Alarm feature code (default is 9850).

7. While power is ON, press the battery check button on the CPU020-D/PCM-D or CPU020-A/PCMA card to measure the database back-up battery voltage. The voltage test points (TP1 to TP2) are located on the front edge of the card. Insert the "common" voltmeter probe into the ground point (TP2) and insert the other probe into the voltage test point (TP1). If the voltage is not greater than 3.3VDC, replace the battery as described on page 3-112. (See Figure 3-50 on page 3-107 for back-up battery voltage test point locations.)

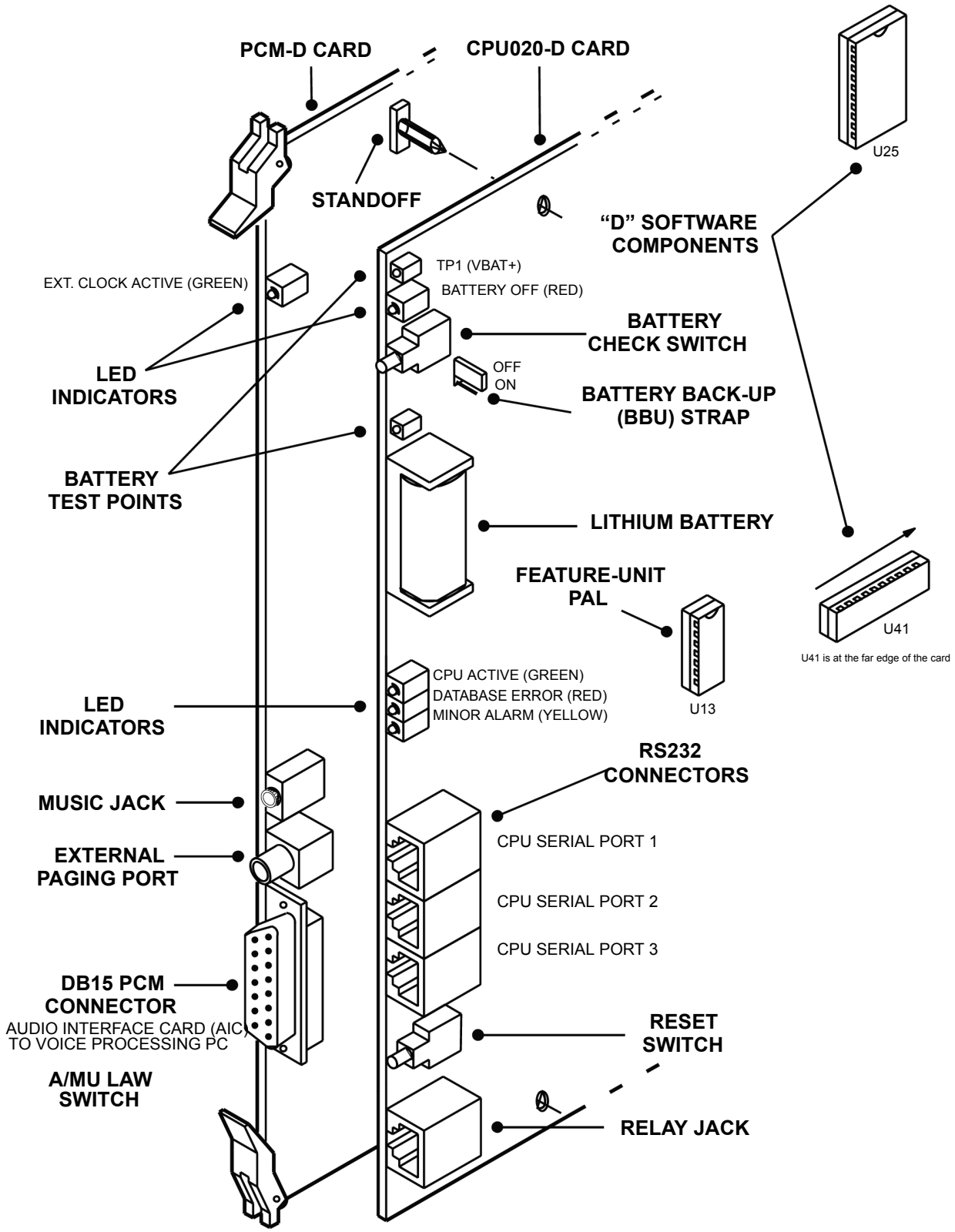
8. *If the CPU020/PCM or CPU020/PCMA card does not have the appropriate system software already installed in flash memory,* see page 3-173 to upload the software from the programming PC to the card.

9. After installing the CPU020/PCM or CPU020/PCMA card, the optional music source, external paging equipment, Voice Processing PC, power failure transfer equipment, output device, etc. may be connected to the CPU020/PCM or CPU020/PCMA card at any time. (See page 3-213 for Voice Processing PC installation instructions, to page 3-240 for external music source installation instructions, and to page 3-242 for external paging equipment installation instructions.)

NOTE: The fiber-optic cable is attached to the FIBER TRANSMIT and FIBER RECEIVE jacks after the PCM-F card in the third (slave) chassis is installed. See page 3-104 for details.

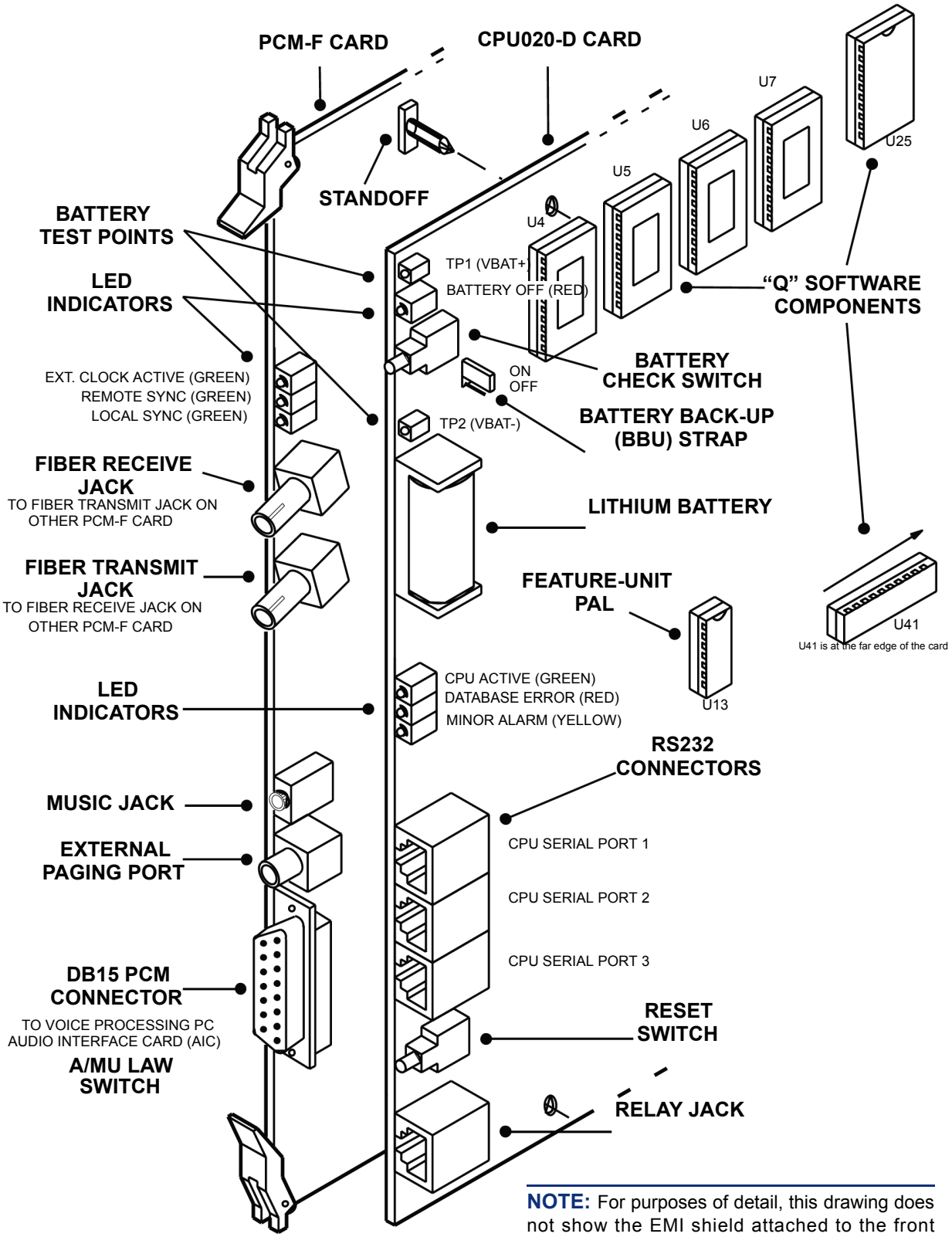
10. Turn OFF the AC POWER switch.

Figure 3-50. *PCM-D Card and Attached CPU020-D Card*



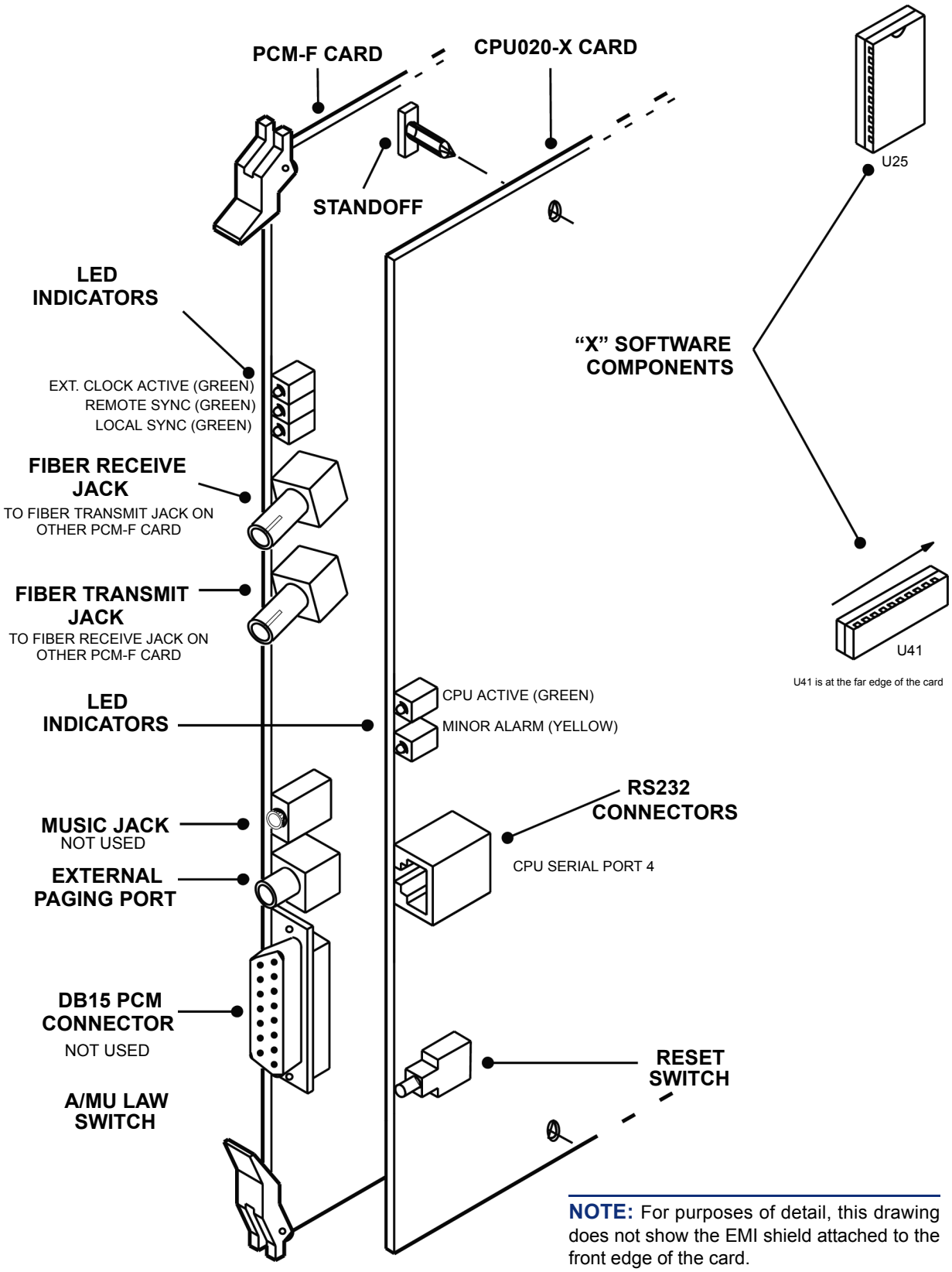
NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

Figure 3-51. *PCM-F Card and Attached CPU020-Q Card*



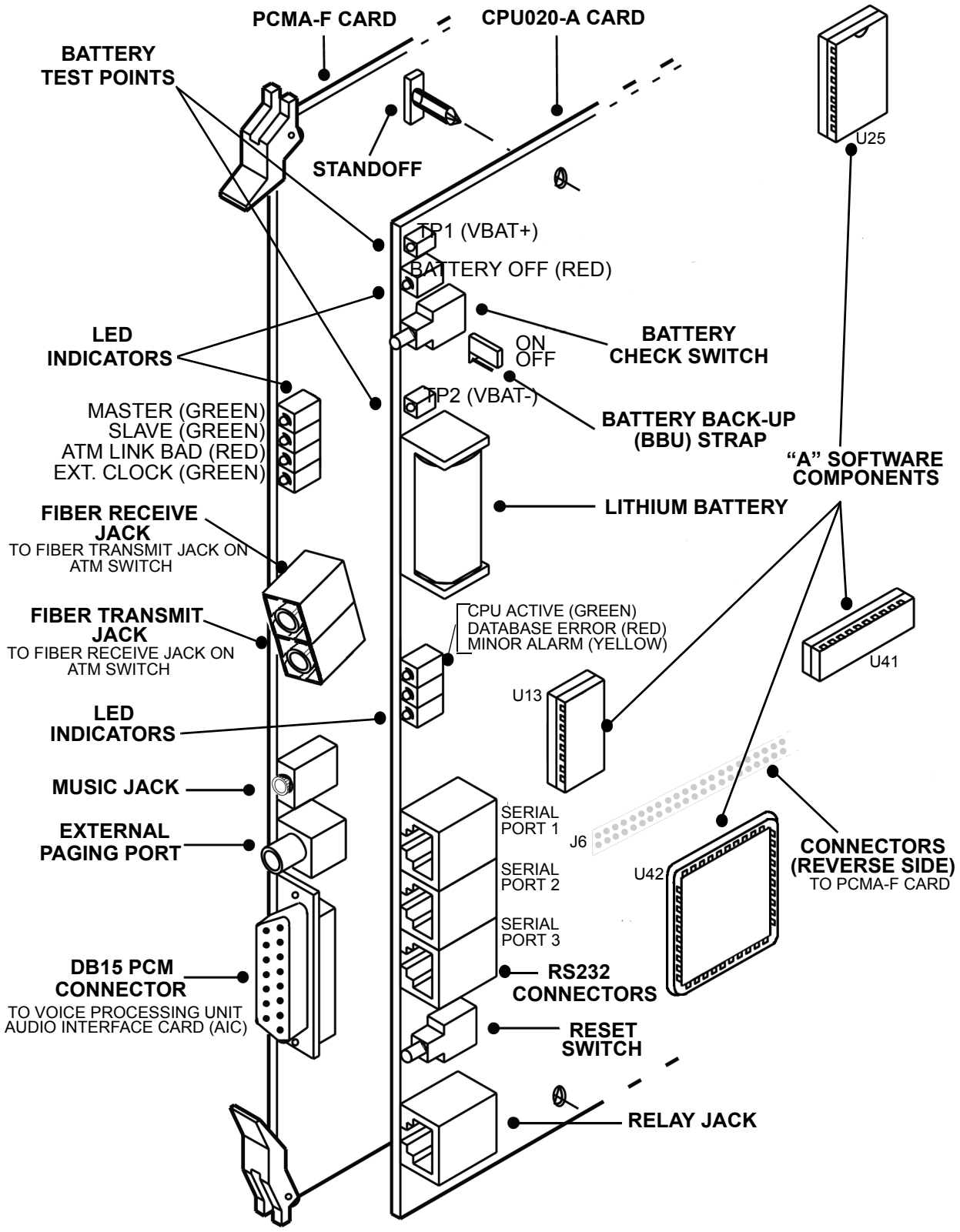
NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

Figure 3-52. *PCM-F Card and Attached CPU020-X Card*



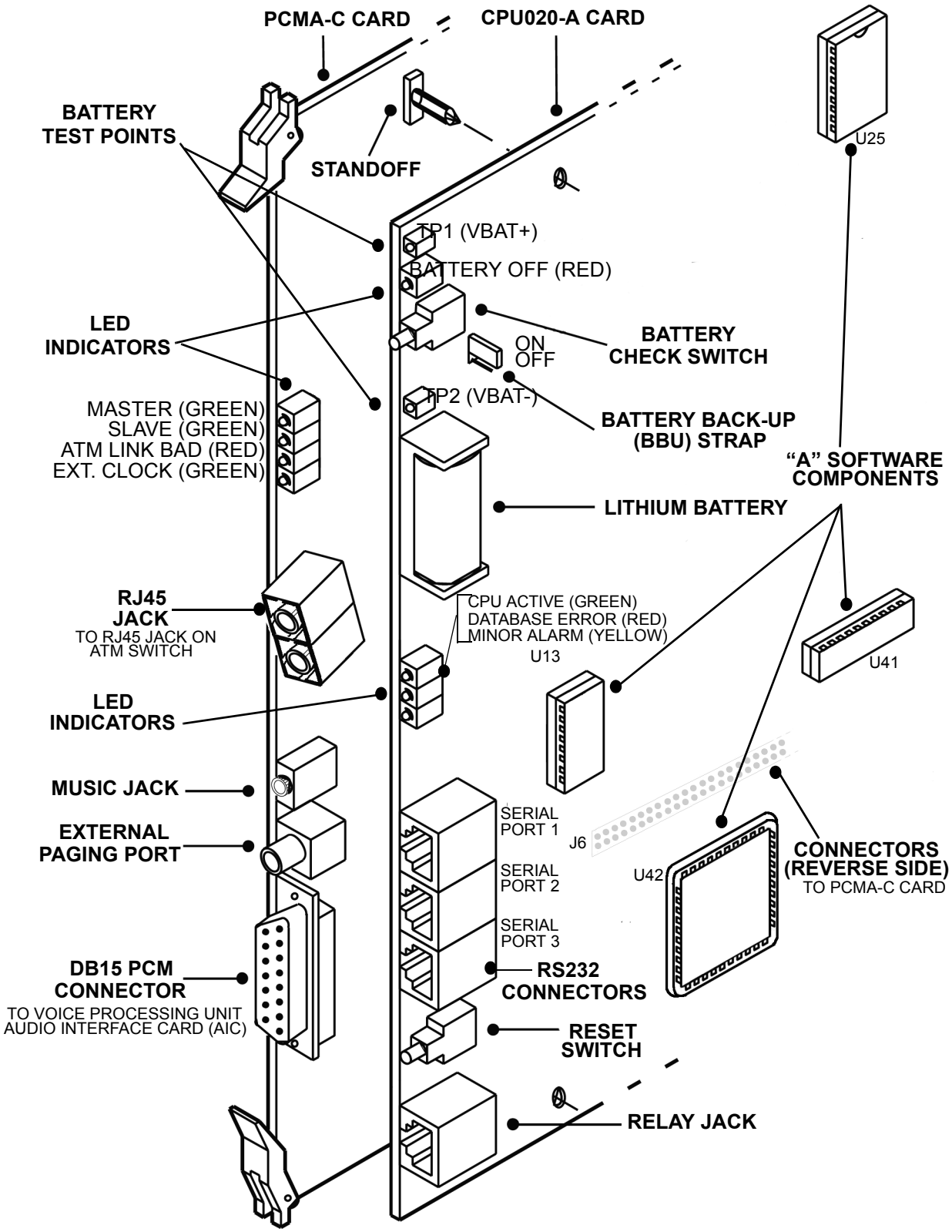
NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

Figure 3-53. *PCMA-FS Card and Attached CPU020-A Card*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

Figure 3-54. *PCMA-C Card and Attached CPU020-A Card*



Replacing the CPU Battery, if Necessary

NOTE: Save the database before you replace the CPU battery (see page 6-65 for details).

10.15 The CPU128, CPU020/PCM, and CPU020/PCMA cards are each equipped with a 3.6V, ¾AA, lithium battery (part no. 815.1011) that can support the database for at least two months of accumulated system down time.

10.16 If the battery's voltage is not greater than 3.3VDC (as measured in step 4 on page 3-98 or 3-106), replace the battery as outlined in the following steps:

1. Turn OFF the system AC POWER switch, remove the card from its slot in the equipment chassis, and place it on a clean, flat, anti-static, non-conductive surface. *If necessary,* remove the plastic cover housing the existing battery (see Figure 3-50 on page 3-107 or Figure 3-51 on page 3-108).
2. Remove the existing battery and dispose of it properly.
3. Observing the proper polarity, install the new battery.
4. *If a plastic cover was removed in step 2,* replace the cover.
5. While pressing the battery check button on the card, measure the database back-up battery voltage. The voltage test points (TP1 to TP2) are located on the front edge of the card. Insert the "common" voltmeter probe into the ground point (TP2) and insert the other probe into the voltage test point (TP1). If the voltage is not greater than 3.3VDC, try another battery.
6. Replace the card in its appropriate slot in the equipment chassis and turn ON the system AC POWER switch.

E. CALL PROCESSING (CP) SERVER

10.17 The Call Processing (CP) Server is a type of CPU for the Inter-Tel system. The call processing software has been moved from the system CPU to a Windows NT-, 2000-, or 2003-based server that communicates with the system via a 10Base-T Ethernet connection or via the ATM Switch. As of version 8.0, all 512 or higher systems require a CP Server. For an ATM-based system, the CP Server PC must be running Windows 2000 or later.

10.18 In v11.0, Windows Server 2003 (Telco version) operating system (OS) with Service Pack 2 (SP2) is supported for the Call Processing Server (CPS). Field upgrades must be done by certified technicians only. See page C-3 for details.

10.19 In a non-ATM system, the PC and CPU/PCM/EAC are connected directly via a cross-connect Ethernet cable and use the UDP protocol to transfer data across a direct connection or a LAN connection. In an ATM-based system, the PC and the ATM Switch are connected directly via a UTP5 or multi-mode fiber (MMF) cable. The UTP5 and OC-3 connectors are provided on the ATM Adapter card which is installed in the CP Server PC. An EAC is not required when using the ATM Switch. For more information and specifications of the CP Server, see Appendix C — CP Server.

10.20 Each CP Server also has a traditional Ethernet Network Interface Card (NIC) and an internal modem. You should configure the CP Server to match the end user's Ethernet network.

NOTE: If the CP Server uses the UTP5 Adapter, the CP Server must be placed within 100 m of the ATM Switch. It is also recommended that the CP Server not be off-site because the UTP5 cable is unshielded.

F. DIGITAL KEYSET CARD (DKSC)

CAUTION

Do not install an analog phone on a Digital Keyset Card, or vice versa. Installing a phone on the wrong type of station card can damage both the phone and the card.

10.21 Install the Digital Keyset Cards (DKSC) and/or 16-port Digital Keyset Card (DKSC-16 or DKSC-16+) as described below. See Figure 3-55 on page 3-114 for an illustration.

10.22 See page 6-121 for instructions about how to convert a DKSC-16 to a v8.2 or later firmware IPRC.

NOTE: When inserting or removing the CPU card, CPC, or Network Clock Card, the system power must be turned off to prevent damage to the card. All other cards can be inserted or removed with power on.

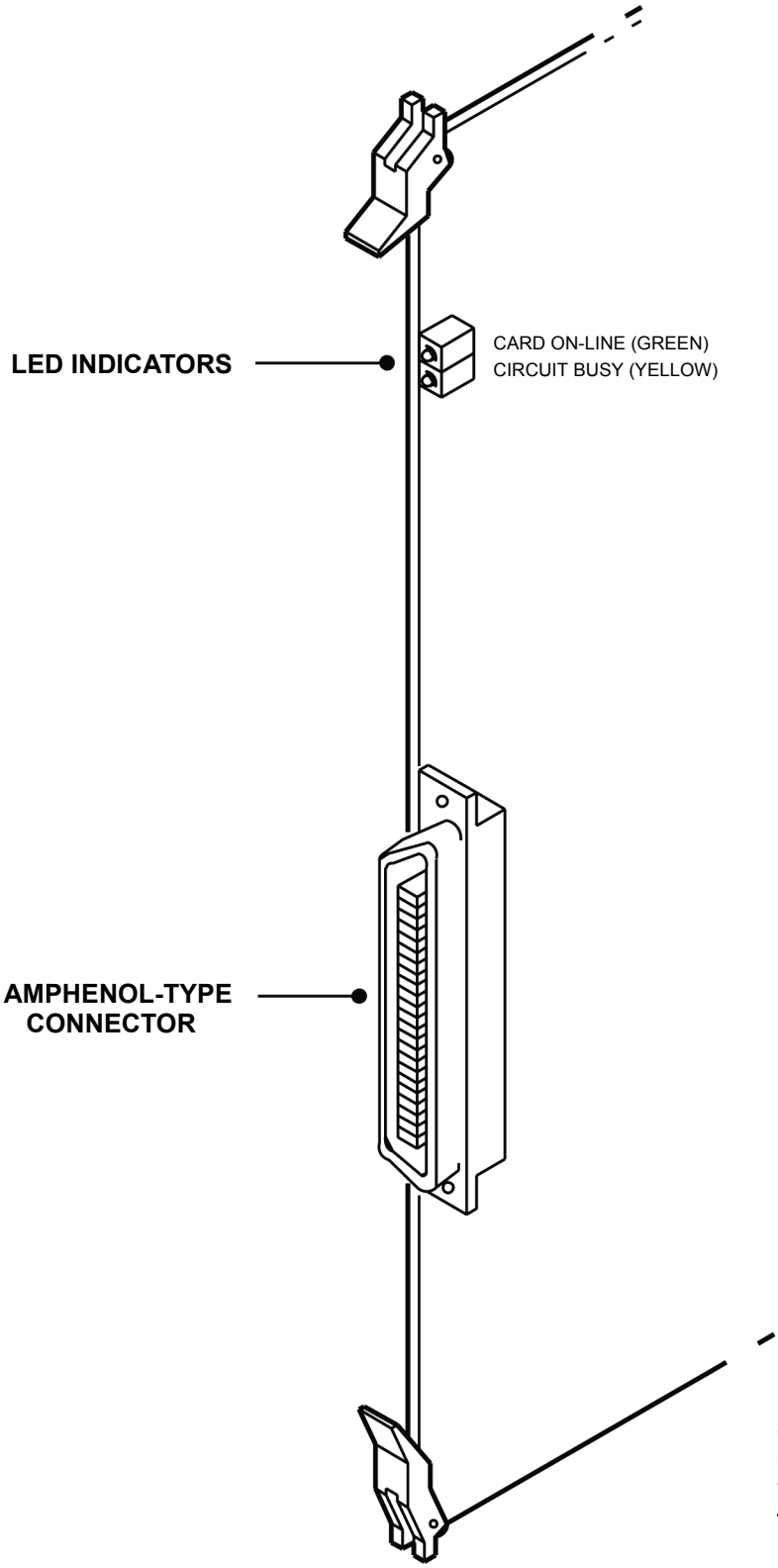
1. Make sure that all components on the cards are seated securely in their sockets and that no pins are bent.

   NOTE: If installing any single-line devices or off-premises extensions, see page 3-70 before installing the DKSCs or DKSC-16s.

2. Install the DKSCs and/or DKSC-16s one at a time in the slots to be used for station connections. The component sides of the cards must *face right*. Any type of card may be installed in any of the first seven slots (in any chassis), and slots may be skipped, if desired.

   NOTE: If, when first powered up, the actual cards installed do not exactly match the default configuration, adjustments must be made in DB Programming. Circuit 1.1.1 (a DKSC circuit) is preset as an administrator and as the primary attendant. If changing card types (e.g., changing from a DKSC to an LSC), see the programming procedures on page 6-114 and to the information on pages 3-16 and 3-17.

Figure 3-55. *DKSC or DKSC-16 and DKSC-16+*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

G. ANALOG KEYSET CARD (AKSC) AND EXECUTONE KEYSET CARD (EKSC)

CAUTION

Do not install an analog phone on a Digital Keyset Card, or vice versa. Installing a phone on the wrong type of station card can damage both the phone and the card.

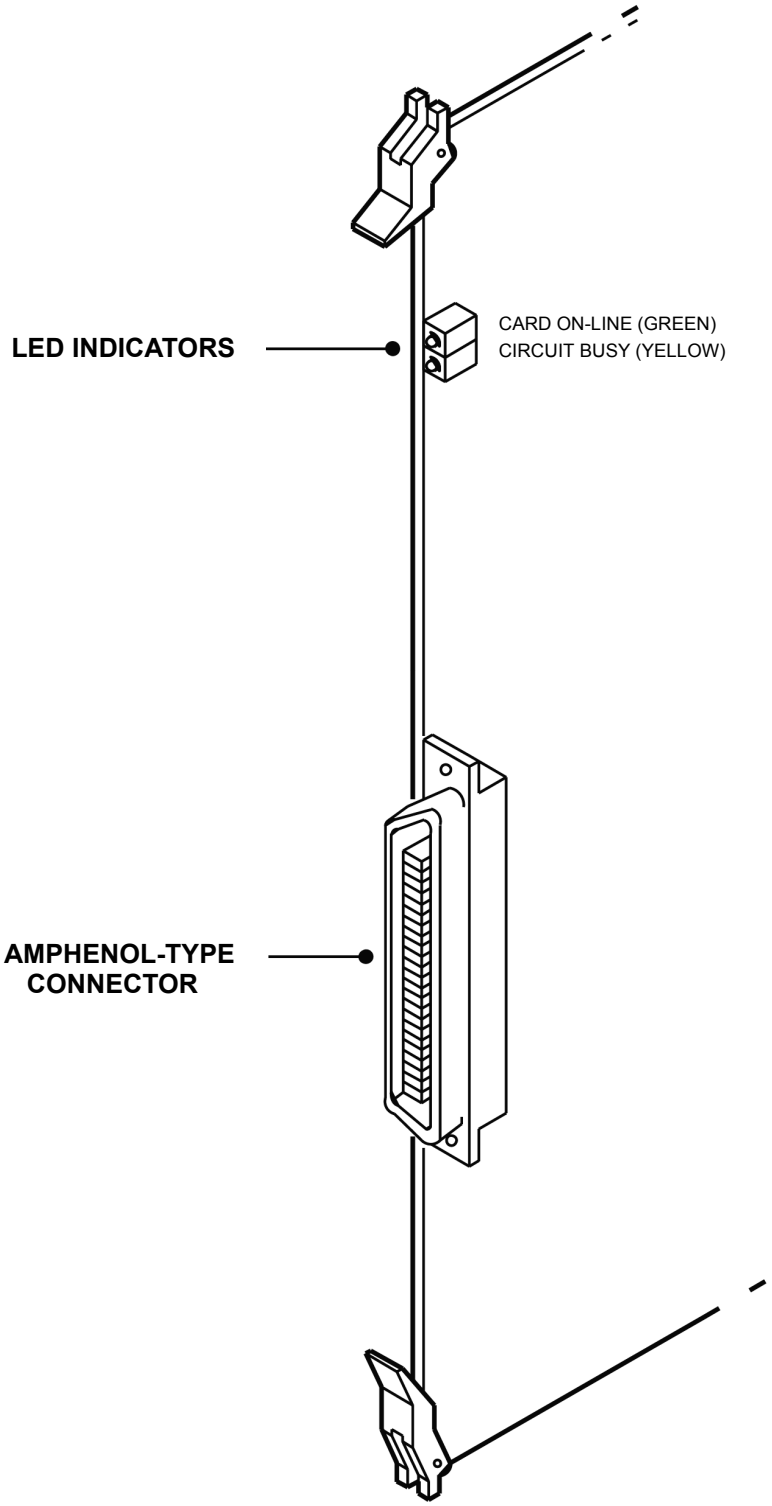
10.23 Install the AKSCs/EKSCs as described below. See Figure 3-56 on page 3-116 for an illustration.

NOTE: When inserting or removing the CPU Card, CPC, or Network Clock Card, the system power must be turned off to prevent damage to the card. All other cards can be inserted or removed with power on.

1. Make sure that all components on the cards are seated securely in their sockets and that no pins are bent.
2. Install the AKSCs/EKSCs one at a time in the slots to be used for analog phone connections. The component sides of the cards must face right. Any type of card may be installed in any of the first seven slots (in any chassis), and slots may be skipped, if desired.

   NOTE: If, when first powered up, the actual cards installed do not exactly match the default configuration, adjustments must be made in DB Programming. Circuit 1.1.1 (a DKSC circuit) is preset as an administrator and as the primary attendant. If changing card types (e.g., changing from an AKSC or EKSC to an LSC), see the programming procedures on page 6-114 and to the information on pages 3-16 and 3-17.

Figure 3-56. *AKSC/EKSC*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

H. SINGLE-LINE CARD (SLC) AND OPTIONAL LAMP OPTION CARD (LOC)

10.24 Install the Single-Line Cards (SLCs) and 16-port Single-Line Cards (SLC-16s) as described below. Part numbers are listed in REPLACEMENT PARTS. See Figure 3-57 on page 3-119 for an illustration.

10.25 An SLC-16 cannot be installed in a card slot that is programmed for an SLC. The ports must be specifically programmed for an SLC-16. However, if port programming is changed from SLC to SLC-16, the existing eight circuits retain their individual programming, and the new circuits stay in the default state. If the port is changed from an SLC-16 to an SLC, all circuits retain their programming, but only the first eight are available.

10.26 Newer SLCs (with artwork number 826.4062-9 stamped on the card) have two onboard ring generators and therefore have two jumpers that must be set when using an external ring generator. Jumper J701 controls circuits 1-4, and jumper J702 controls circuits 5-8.

10.27 The SLC-16 and LOC-16 can be used only with version 4.3 (or later) system software.

NOTE: When inserting or removing the CPC, CPU card. or Network Clock Card, the system power must be turned OFF to prevent damage to the card. However, station and trunk cards can be inserted or removed with power on.

1. Make sure that all components on the cards are seated securely in their sockets and that no pins are bent.
2. *If an external AC ring generator is used instead of the onboard ring generator*, place the ring generator selection strap(s) (J701 and J702) in the external position (see Figure 3-57 on the next page).
3. *If installing the optional Lamp Option Card (LOC)*:
   a. Carefully, remove the solid, 8-position jumper strap from jumper location J7 (see Figure 3-57).
   b. Attach the LOC to the SLC as shown in Figure 3-57. Make sure the connectors and the standoffs are properly aligned and seated.
   c. When programming the database, be sure to enable the message waiting lamps in station programming.
4. *If installing the optional LOC-16*:
   a. Carefully remove the two, solid, 16-position jumper straps from jumper locations J3 and J4 on the SLC-16.
   b. Place jumper strap (J6) on the SLC-16 in the LOW position (over the top two pins).
   c. Carefully attach the LOC-16 to jumper locations J3, J4, and J5 on the SLC-16. Make sure the connectors are properly aligned and seated.
   d. When programming the database, be sure to enable the message waiting lamps in station programming.

NOTE: Unlike the CPU combinations, the SLC and LOC are attached with the component sides of the cards facing each other.

5. Install SLCs and SLC-16s in the slots to be used for AC-ringing single-line set and AC playback device connections. The component sides of the cards must *face right*. Any type of card may be installed in any of the first seven slots (in any chassis), and slots may be skipped, if desired.

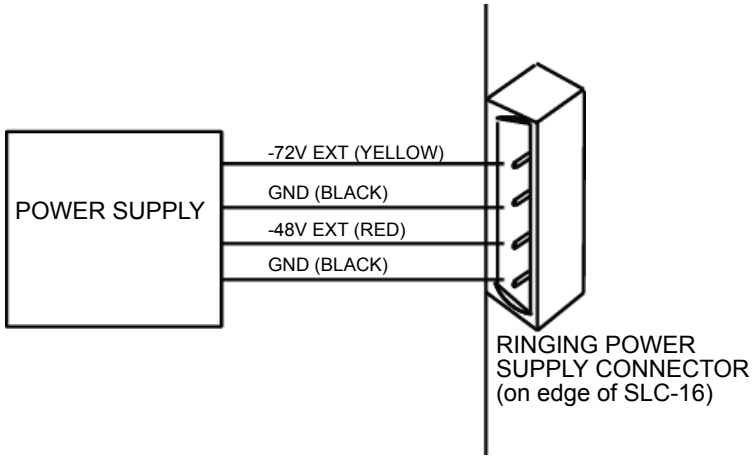
NOTE: If, when first powered up, the actual cards installed do not exactly match the default configuration, adjustments must be made in DB Programming. Circuit 1.1.1 (a DKSC circuit) is preset as an administrator and as the primary attendant. If changing card types (e.g., changing from an SLC to a DKSC), see the programming procedures on page 6-114 and to the information on pages 3-16 and 3-17.

SLC-16 Power Supplies

10.28 A power supply is required for each SLC-16 because the cards are not designed to pull talk battery, ring, or message lamp power from the system power supply. Power supplies are available that support one card or up to four cards.

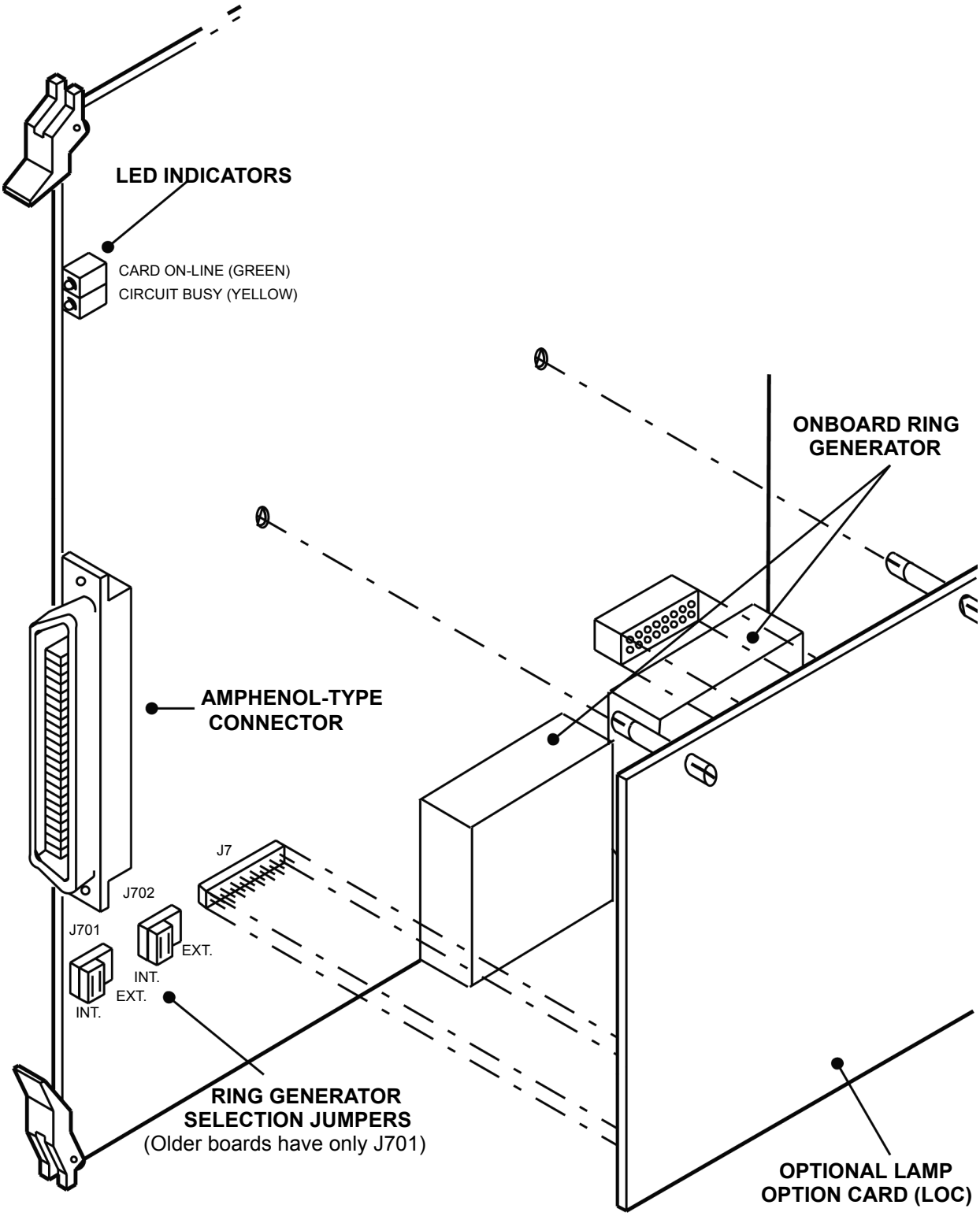
10.29 There are two ways to connect the power supply to the SLC-16:

- Connect the power supply directly to the board using the RINGING POWER SUPPLY connector on the SLC-16. To connect the power supply to the SLC-16, simply insert the connector on the power supply to the Ringing Power Supply connector on the card. Note that the shape of the connector only allows you to insert the power supply connector in the proper direction. Figure 3-58 on page 3-120 shows the location of the Ringing Power Supply connector. The diagram below shows the pinout and wire colors for the power connector and power supply.



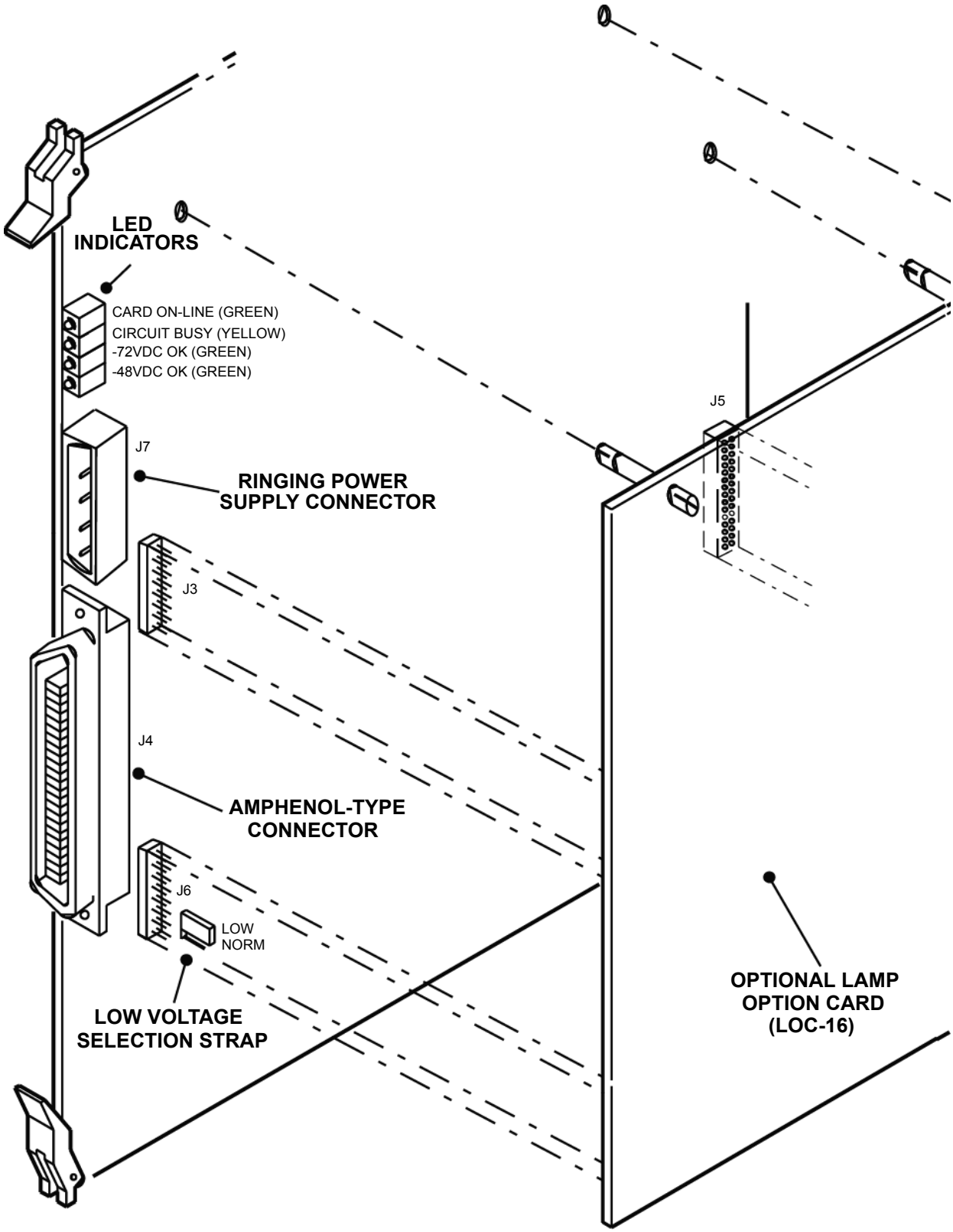
- Punch down the power supply wires at the MDF, as described on page 3-45.

Figure 3-57. *SLC and Optional LOC*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

Figure 3-58. *SLC-16 or SLC-16+ and Optional LOC-16*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

I. LOOP/GROUND START OR LOOP START CARD (LGC/LSC, LGC-D/LSC-D)

CAUTION

With a European system, new Loop Start Cards (LSC – part no. 552.2340) and LSC Daughter Cards (LSC-D – part no. 552.2341) have been modified to meet a regulatory compliance requirement for EU Telco Directive 99/05/EEC. However, these new cards cannot be paired with older LSC/LSC-D cards due to mechanical interference with some of the components between the new and old cards. More specifically, new LSC-D revision level cards (artwork version 826.5422-5) only work with new LSC revision level cards (artwork version 826.5421-5), they cannot be used with earlier artwork version LSCs. Additionally, new LSC revision level cards (artwork version 826.5421-5) only work with new LSC-D revision level cards (artwork version 826.5422-5); they cannot be used with earlier artwork version LSC-Ds.

10.30 Install each LGC, LGC Daughter Card (LGC-D), LSC, and LSC Daughter Card (LSC-D) as described below. See Figure 3-59 on page 3-124 and Figure 3-60 on page 3-125 for illustrations.

NOTE: When inserting or removing the CPC, CPU card, or Network Clock Card, the system power must be turned OFF to prevent damage to the card. All other cards can be inserted or removed with power on.

1. Check to see that all components on the card are seated securely in their sockets and that no pins are bent.
2. Check the LSC fuses (LGCs do not have fuses). There are four specially-designed fuses on each card that cannot be replaced in the field (see page 2-47 for more information).
3. *If installing a Daughter card,* attach the LGC-D or LSC-D as shown in Figure 3-59 on page 3-124 and Figure 3-60 on page 3-125. Make sure the connectors and the standoffs are properly aligned and seated.

   NOTE: Unlike the CPU combinations, these cards are attached with the component sides of the cards facing each other. Also, LGC-Ds cannot be installed on LSCs, and LSC-Ds cannot be installed on LGCs.

4. Install the LGCs and LSCs one at a time in the slots to be used for trunk connections. The component sides of the LGC/LSCs must *face right*. Any type of card may be installed in any of the first seven slots (in any chassis), and slots may be skipped, if desired.

   NOTE: If, when first powered up, the actual cards installed do not exactly match the default configuration, adjustments must be made in DB Programming. If changing card types (e.g., changing from an LGC to a DKSC), see the programming procedures on page 6-114 and to the information on pages 3-16 and 3-17.

LGC and LGC-D Modification

10.31 Under certain conditions, previous versions of the LGC (part no. 550.2309) and the LGC-D (part no. 550.2310) were not able to always properly detect the "ground start" signal provided by the central office (CO). A modification to the LGC and LGC-D has been implemented to correct this problem.

10.32 More specifically, the previous versions of the LGC and LGC-D did not function properly in ground-start mode when terminated on a CO trunk that had a constant current battery feed of less than approximately 25mA. The new cards have a built-in 50VDC power supply and therefore no longer rely on the current from the CO to detect the "ground start" signal.

Problem Identification

10.33 A customer site may have this problem if a phone user's display shows "LINE IS UNPLUGGED" when attempting to access a ground start trunk on an LGC or LGC-D. To determine if the card really does have a problem detecting the ground start signal from the CO, follow these steps:

1. Ensure that the appropriate trunk circuit on the card is programmed for ground start mode. (Loop start mode is not subject to this problem.)
2. Verify that the card is properly installed and connected.
3. Verify that tip and ring for the trunk circuit are properly connected. Below is a simple method that works for most cases:
   a. At the associated CO block, remove the bridging clips for the trunk.
   b. On the telco side of the block, attach a test set (butt set) in monitor mode (on-hook) to tip and ring.
   c. Apply ground to the ring terminal.
   d. *If you hear dial tone (only while ground is applied),* tip and ring are properly connected. Replace the bridging clips and check to make sure the card can still receive dial tone.

      If you do not hear dial tone, apply ground to the tip terminal. If you now hear dial tone, tip and ring are reversed.

> **NOTE:** Be sure the test set is in monitor mode. If it is in off-hook mode, sometimes the line can be seized regardless of the polarity, and the test is invalid.

Modified Cards Identification

10.34 Remove the LGC and/or LGC-D and look at the part number stenciled on the card. Immediately after the part number, there should be a dash (-) and a single letter contained within a box. This denotes the revision level of the card. The following chart shows the revision level of the current, modified cards:

Table 3-17. *LGC/LGC-D Revision Levels*

Card	Part No.	ECN No.	Revision
LGC	550.2309	4813	826.5223-4D (or later) 826.5223-5D (or later) 826.5223-6 (any letter)
LGC-D	550.2310	4814	826.5224-3B (or later) 826.5224-4B (or later) 826.5224-5 (any letter)

Compatibility

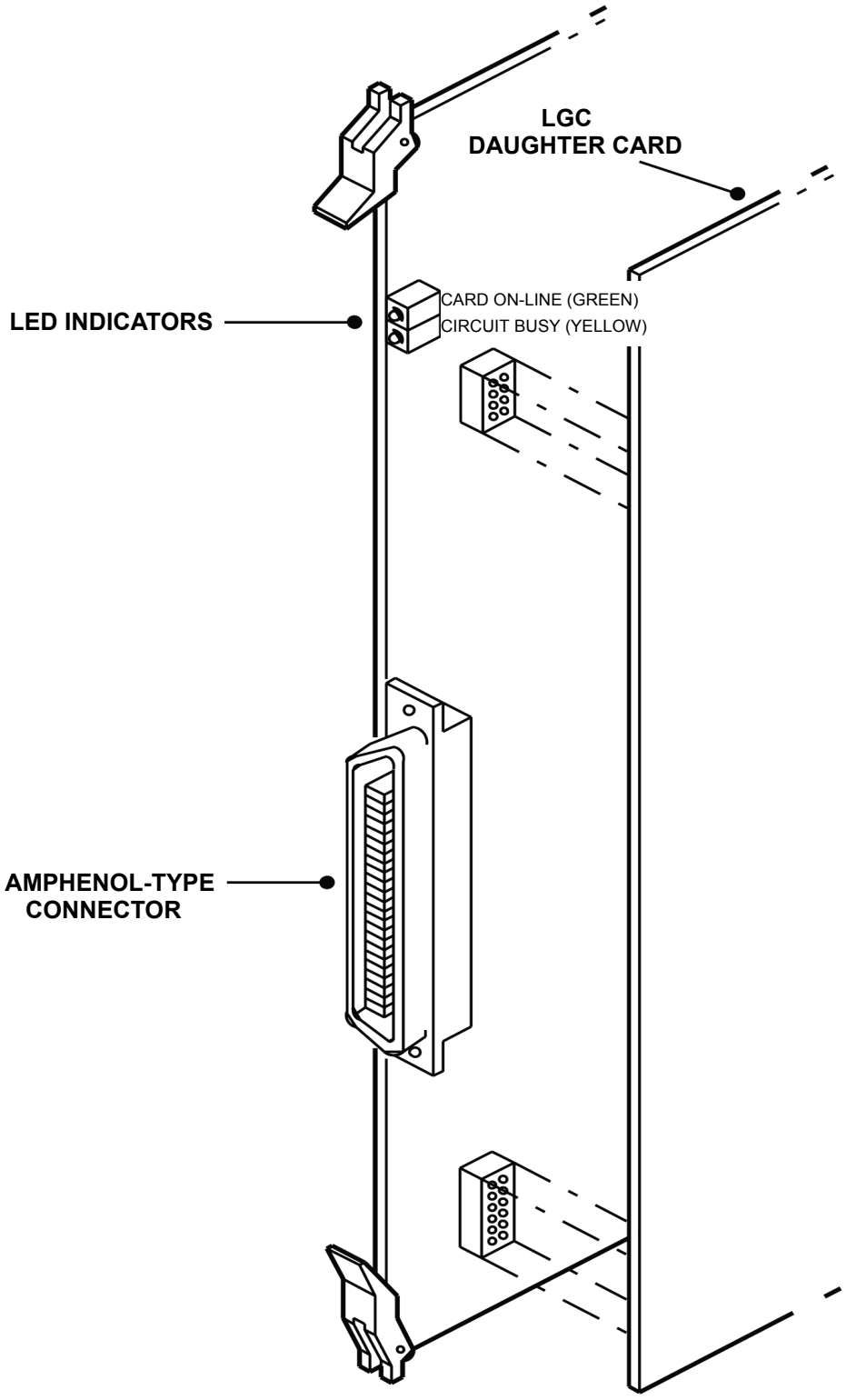
10.35 Improperly paired cards do not work together and the trunks attached to these cards do not function.

10.36 LGC -6 cards only work with LGC-D -5 cards (and vice versa). They are not backwards compatible. LGC -4 and -5 cards only work with either LGC-D -3 and -4 cards.

Resolution

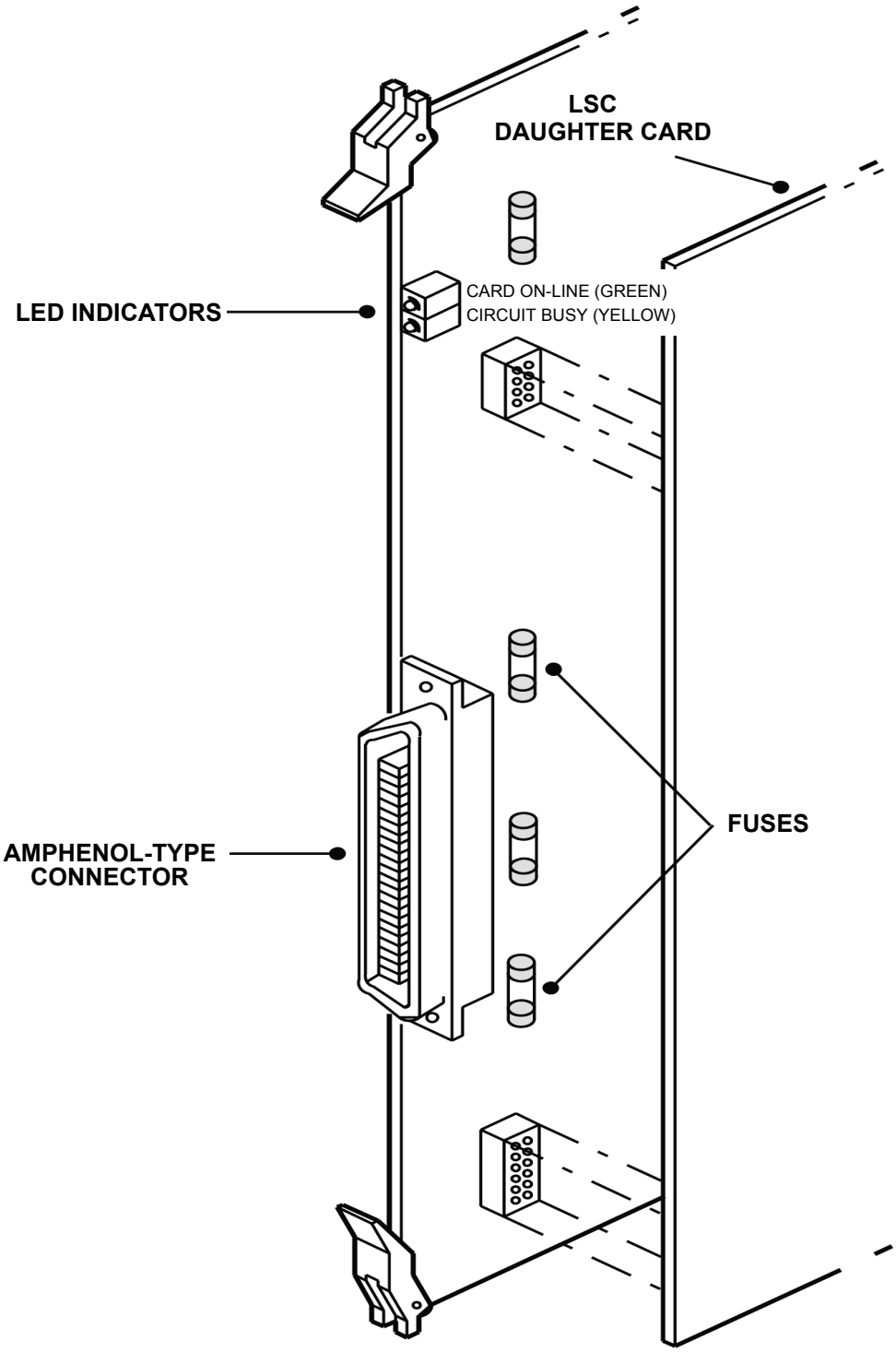
10.37 If a system is experiencing this problem ***and*** if it is equipped with older, unmodified LGC and/or LGC-D cards, the card(s) may be replaced by contacting Inter-Tel Technical Support.

Figure 3-59. *LGC and Optional LGC-D*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

Figure 3-60. *LSC and Optional LSC-D*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

J. T1 OR E1 CARD (T1C/E1C) AND PRIMARY RATE INTERFACE (PRI) DAUGHTER CARD

NOTE: In tri-/quad-chassis installations, the master T1C or E1/PRI (if one exists) should be located in the master dual-chassis.

10.38 See page 2-51 in SPECIFICATIONS to ensure that the T1 or E1 Card (T1C or E1C) has the most current firmware.

NOTE: Like the other station and trunk cards, T1 or E1/PRI cards can be inserted or removed with the system power turned ON, as long as the T1/PRI or E1/PRI trunk is not connected to the card. However, *before removing a T1 or E1/PRI Card while power is on*, first press the make-busy switch on the card and allow the system to "busy out" the T1/E1 and/or PRI trunk (wait for the CIRCUIT BUSY LED to go out to avoid dropping calls; you may have to press the make-busy switch more than once). Then unplug the T1/E1 and/or PRI trunk from the connector on the card (see page 2-56 for more information on the make-busy switch).

10.39 Install each T1C or E1C and, if applicable, attached Primary Rate Interface (PRI) Daughter card as described in the following steps (see REPLACEMENT PARTS for the part numbers). See Figure 3-61 on page 3-127 for an illustration.

1. Ensure that each T1C or E1C to be installed is at the proper revision level and has the appropriate software components installed (see Figure 3-61 on page 3-127 for component locations).
2. *If a Primary Rate Interface trunk is connected to the T1 or E1 Card,* attach the PRI Daughter card as shown in Figure 3-61 on page 3-127. Make sure the connectors and the standoffs are properly aligned and seated.

   NOTE: Unlike the CPU combinations, these cards are attached with the component sides of the cards facing each other.

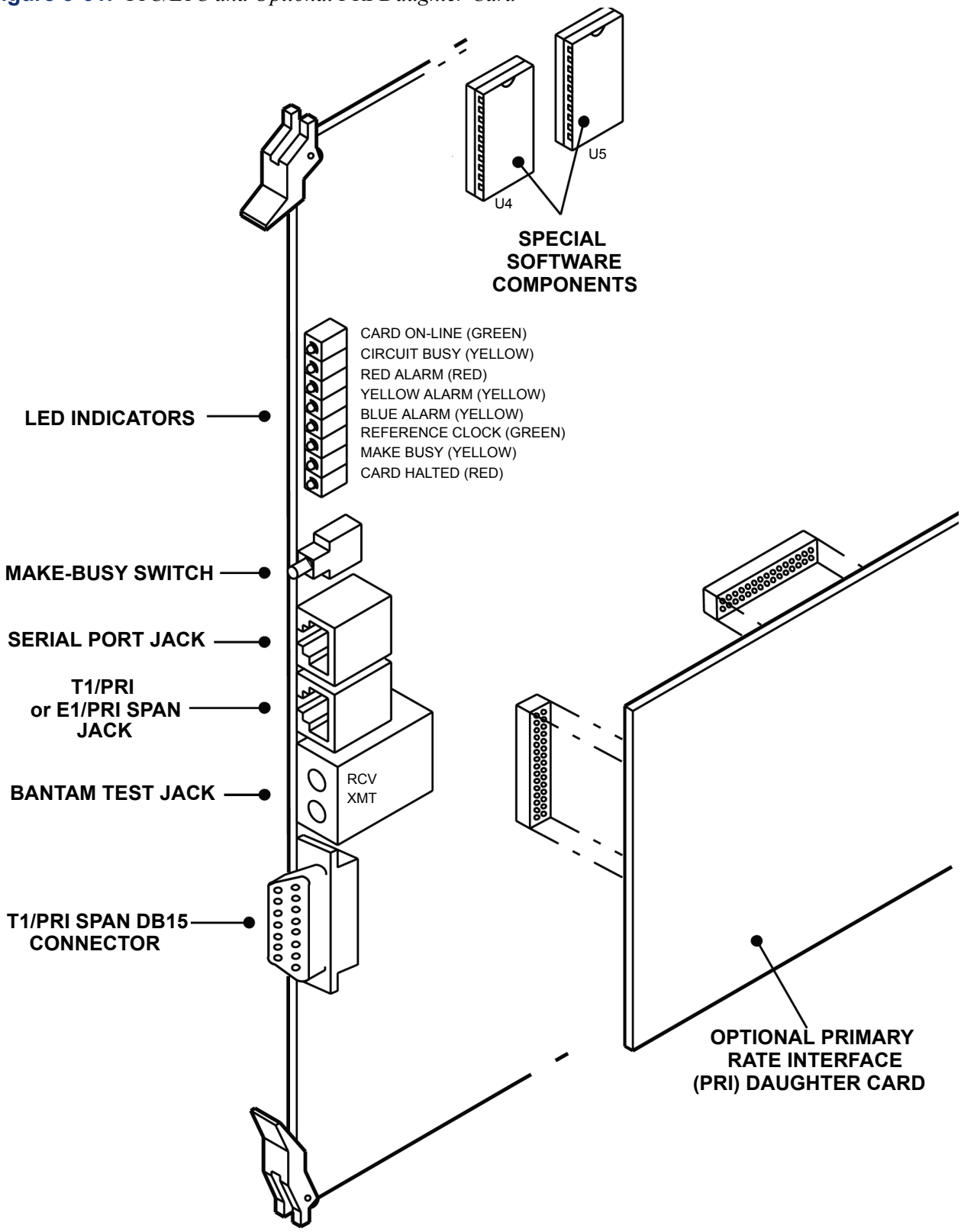
3. Check to see that all components on the card(s) are seated securely in their sockets and that no pins are bent.
4. Install the T1Cs or E1/PRIs one at a time with the components *facing right*. The cards may be installed in any of the first seven slots (in any chassis), and slots may be skipped, if desired.

   NOTE: If, when first powered up, the actual cards installed do not exactly match the default configuration, adjustments must be made in DB Programming. If changing card types (e.g., changing from an LSC to a T1C or E1/PRI), see the programming procedures on page 6-114 and to the information on pages 3-16 and 3-17.

10.40 If experiencing excessive E&M gain levels on a T1 Card that is not equipped with a PRI Daughter card, two special software components and a TGC Daughter card (kit part no. 828.1397) can be attached to the card to provide 6 dB of attenuation.

10.41 To use the TGC Daughter card, the T1 Card must be at revision level "2.G" or later.

Figure 3-61. *T1C/E1C and Optional PRI Daughter Card*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

K. T1/E1/PRI CARD

10.42 The T1/E1/PRI card, shown on page 3-130, is compatible with any version of the phone system that supports T1/ E1 and PRI.

10.43 The T1/E1 board is converted for Primary Rate Interface (PRI) use by adding a feature PAL on the card. There is no daughter board for PRI.

10.44 The T1/E1/PRI card supports programmable gain control, as described on page 6-257. However, it also has DIP switches to program gain control for older versions of software, as described in the installation document shipped with the card. (If the phone system is upgraded to a version of software that supports programmable gain control, the programmed setting in the database overrides the settings programmed by the DIP switch.) All trunks, regardless of the board they are on, have default transmit and receive gains of 0 dB in the database. The programmable range for the Transmit or Receive gain is -16 to +12 dB.

10.45 The T1/E1/PRI card uses flash memory, in place of EPROMS, to determine the PRI software version. The PRI software is included on the Inter-Tel edGE Web site and can be downloaded if necessary.

10.46 Use the following procedure to install a T1/E1/PRI card.

1. Place the T1/E1/PRI card on a clean, flat, anti-static surface.

   NOTE: T1/E1/PRI cards can be inserted or removed with the system power turned on, as long as the T1/E1 span is not connected to the card. However, before removing a T1/E1/PRI card while power is on, first press the make-busy switch on the card and allow the system to "busy out" the T1/E1 span, then unplug the T1/E1 span from the connector on the card.

2. Jumper J14, in the lower right corner of the board, does not need to be installed. However, if it is, it must be over the bottom two pins, as shown on the next page. (This is the write-protect jumper. If it is not installed or is over the bottom two pins, the card's "boot sector" is protected from being overwritten. The boot sector is the card's basic code that provides flash programming functionality. It does not change from version to version, but may require future upgrades.)

   NOTE: Jumpers J4 and J5 normally are not used. If required by local telecom [network provider] rules, jumpers over the lower two pins of J4 and the lower two pins of J5 apply frame ground to the transmit and receive ring leads.

3. Jumpers J10 and J15 select the T1/E1 termination impedance. The setting is generally country specific. For T1 applications, set it to 100 ohms. For E1, set it to 120 or 75 ohms, depending on the requirements of the local provider. The jumper positions are indicated on the board and in the figure on the next page.

4. To convert the card for PRI use, carefully install the PRI PAL in socket U14, see Figure 3-62 on page 3-130. The notch (or small dot in the corner) indicating pin 1 on the PAL must match the notch of the outline silk-screened on the card. Check to see that the PAL (part number 827.8877) is seated securely in its socket and that no pins are bent.

5. Install the T1/E1/PRI card in the equipment chassis. The PRI software is uploaded automatically, as described on the next page.

10.47 In system versions 5.1 and later software, the T1/PRI information is included in the CPU software. This allows the T1/E1/PRI card software to be uploaded automatically, through the backplane, when the cards are installed. This automatic upload occurs when a T1/E1/PRI card is first plugged in, whether it is a new installation or the card is being added to an existing installation. Here is what happens:

- The CPC or CPU checks the card and determines whether it needs to be updated. If it does, the yellow and red LEDs on the T1/E1/PRI card flash alternately and the upload begins.
- While the information is being uploaded, both the yellow and red LEDs flash at once.
- The card then resets and the green LED lights to indicate that the T1/E1/PRI card is ready for use.

10.48 Some other special considerations concerning the internal update include the following:

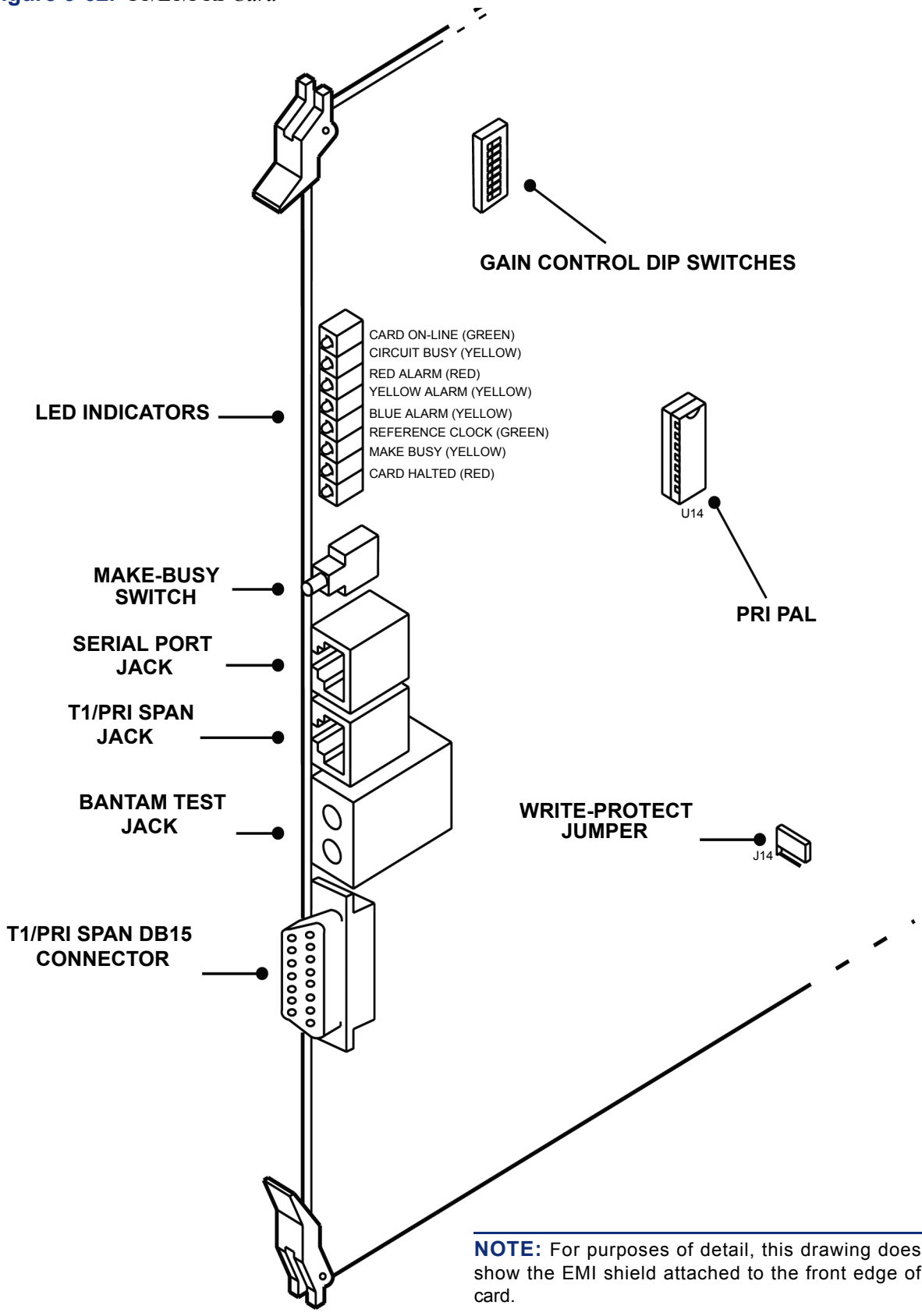
- In a new installation, the equipment chassis slot does not have to be pre-programmed as a T1/E1/PRI card. The slot assignment is automatically updated after the CPC or CPU uploads the software.
- If you ever need to downgrade the software version, you must do it manually, using the instructions on page 3-173. The CPC or CPU never loads an older version of software onto the T1/E1/PRI card.
- If you change to a newer version of CPC or CPU software (that contains newer T1/E1/PRI software), the system performs the automatic upload to bring the T1/E1/PRI card(s) up to the latest version.

10.49 If an automatic upload fails, it could be that the T1/E1/PRI card's boot sector is corrupted. Contact Inter-Tel Customer Support for replacement.

10.50 **Manual PRI Software Upload Procedure**: If Inter-Tel issues upgrade software for the T1/E1/PRI card or if you are installing the T1/E1/PRI in a phone system with any other version, you must upload different software to the card. See page 3-173 for uploading instructions.

NOTE: System software v9.0 and later provides a networking wizard to simplify the required programming for T1/E1 PRI networking. See page page 6-87 for more information about the networking wizard.

Figure 3-62. *T1/E1/PRI Card*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

L. BASIC RATE INTERFACE CARDS (BRS AND BRU)

10.51 Follow these steps to install either type of BRI card, shown on pages 3-133 and 3-134. Detailed instructions for equipping and programming the ports and circuits are presented after this outline.

NOTE: BRU cards are not supported in Europe.

Installation Procedure Overview

1. Make sure the **Advanced CO Interface** is licensed. If this feature is not licensed, the warning message "Online Ignored for Board - Check Advanced CO Interfaces" appears in Message Print when the card is installed. If this occurs, upload a software license that supports the Advanced CO Interface premium feature.
2. Choose a slot and equip it for either a BRS or BRU card using System\Cabinets in DB Programming. (see page 6-136 in PROGRAMMING)
3. In DB Programming, equip the ports to be either station or trunk ports. Leave unused ports set to **None**.
4. Program the ports for either BRI station or BRI trunks depending on the device being installed.
5. Program the established BRI station and trunk ports.
6. Program the individual BRI devices.

   NOTE: Refer to the manufacturer's manual for the external device being installed.

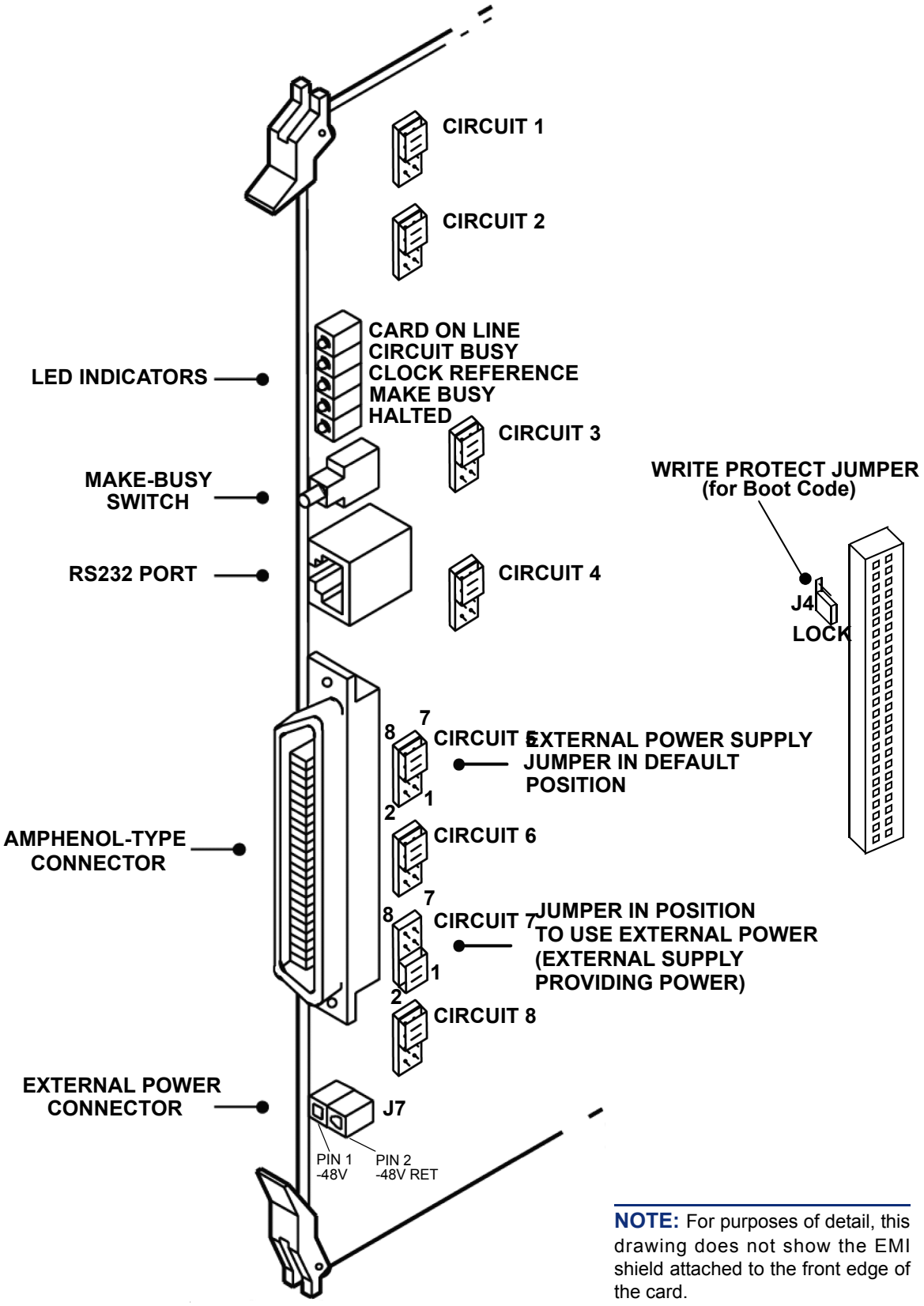
7. Program the reference clock information for each of the BRI trunk ports in System\Cabinets\Reference Clock List.
8. Install the BRS cards in the selected slot in the equipment chassis, and connect the external power supply, if required. Also connect the 50-pin Amp connection to the front of the card and terminate the cable to the MDF (Main Distribution Frame).
9. Jumper J4, in the lower right corner of the board, is a write-protect jumper and does not need to be installed. However, if it is, it must be over the bottom two pins (locked position), as shown in Figures 3-63 and 3-64 on the following pages. If it is not installed or is over the bottom two pins, the card's "boot sector" is protected from being overwritten. The boot sector is the card's basic code that provides flash programming functionality. It does not change from version to version, but may require future upgrades. To upload boot code, see page 3-177.
10. Install the BRU card in the selected slot in the equipment chassis, and connect the 50-pin Amp connection to the front of the card, and then terminate the cable to the MDF.
11. Connect the individual BRI devices to the chosen ports.

    NOTE: If a BRS or BRU card is installed on a system using CPU 128 or CPU 64, a Network Clock Card (NCC) must also be installed. If the system uses a CPC, however, an NCC is not required.

Equip the BRI Cards

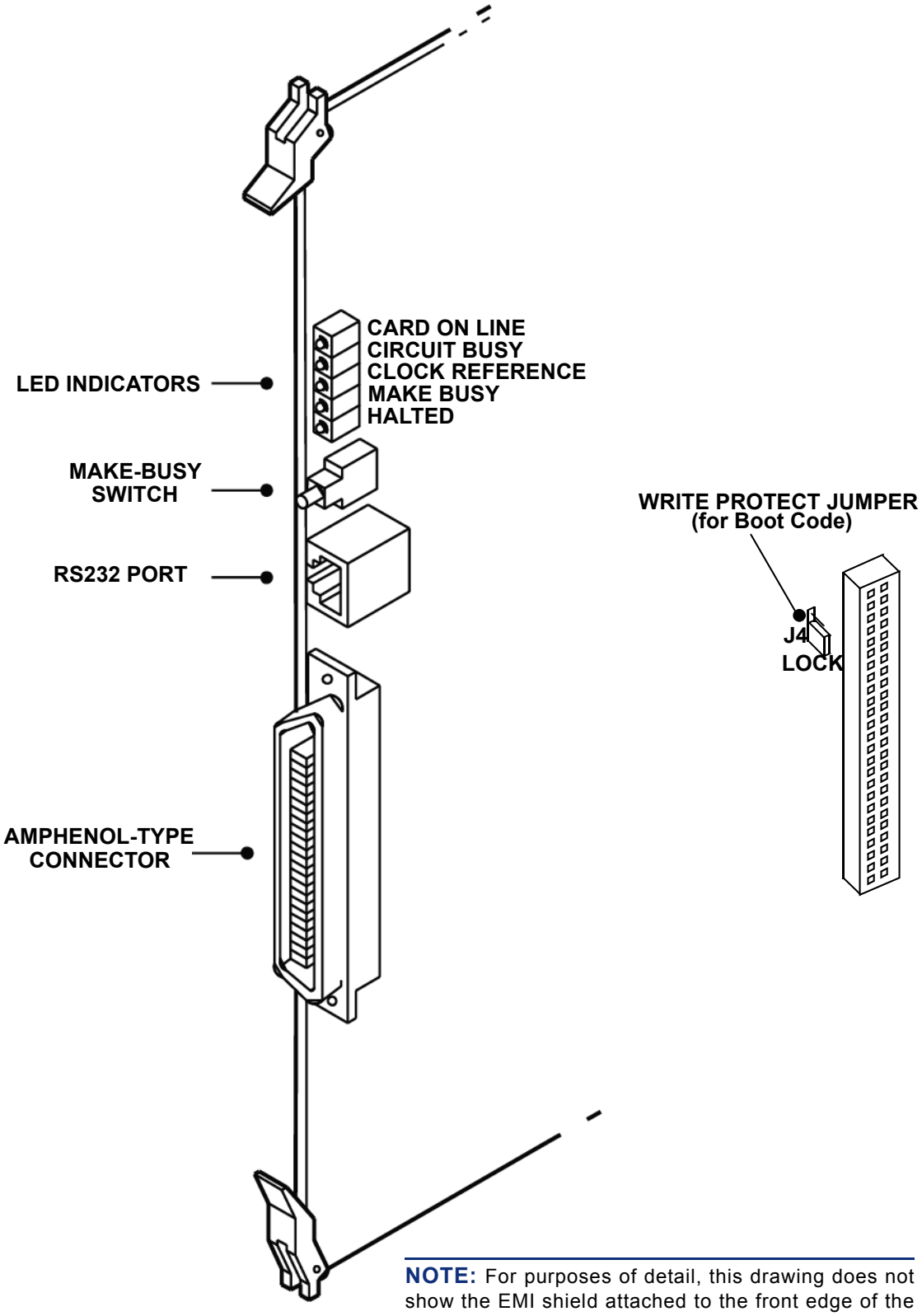
10.52 Both types of BRI cards are equipped in DB Programming under System\Cabinets. Select the card slot and then choose either Basic Rate S/T Interface or Basic Rate U Interface from the drop-down list of card types. Either card type can be equipped in any programmable slot. If the country chosen for the system installation does not support U interface from the Central Office, the Basic Rate U Interface option does not appear on the drop-down list of card types. At this time, only North American systems support U interface. If the card is installed in a previously unused slot, the system automatically recognizes the card type and equip the slot accordingly if DB Programming is not active. If a DB Programming session is active, the card must be equipped in DB Programming.

Figure 3-63. *BRS Card*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

Figure 3-64. *BRU Card*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

Equip the BRI Ports

10.53 After the card is equipped, the installer can equip the eight individual ports by choosing the Ports sub-folder under each card's folder, located under System\Cabinets. The default device type is None. Both BRS and BRU ports can be equipped as station or trunk ports. The choices for the ports on the drop-down list are Station/Station, Trunk/Trunk, and None. Ports may be unequipped by setting the port type to None. See below for more details.

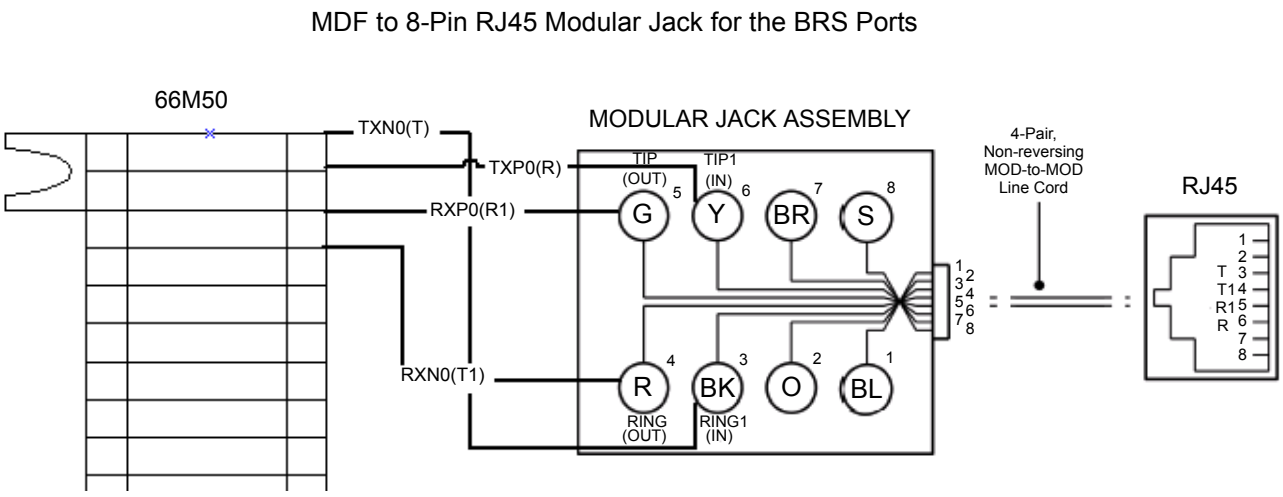
10.54 When a trunk port is equipped, the installer must choose an extension for each newly equipped B-channel trunk. The default extension number of a B-channel trunk on a BRI trunk port is 94XXX, where XXX is determined by the formula [(B-1)*32 + (P-1)*2 + D], where B indicates board number, P indicates port number, and D denotes device number. The hardware address of each trunk has the form NN:BB:PP:CC, where NN is the node number, BB is the board number, PP is the port number, and CC is the circuit number.

10.55 An extension must also be chosen for each equipped station port. The default extension number of a B-channel station on a BRI station port is 1XXX, following the same rules as normal station extension numbers. The hardware address of each station is NN:BB:PP:CC, just as for trunk circuits. The BRS and BRU ports can be equipped as follows:

BRS Ports Programmed as Trunk Ports

10.56 The following is an example of the termination between a BRS (T) trunk port and an NT1 device using four-wire connection. The BRS ports are programmed or equipped as trunk ports that connect to an NT1.

Figure 3-65. *BRS Trunk/Trunk Port to NT1*



NOTE: See page 3-38 for more information on the BRS card connection.

BRS Ports Programmed as Station Ports

10.57 The following is an example of the termination between a BRS (S/T) station port and a BRI ISDN (S/T) device using four-wire connection.

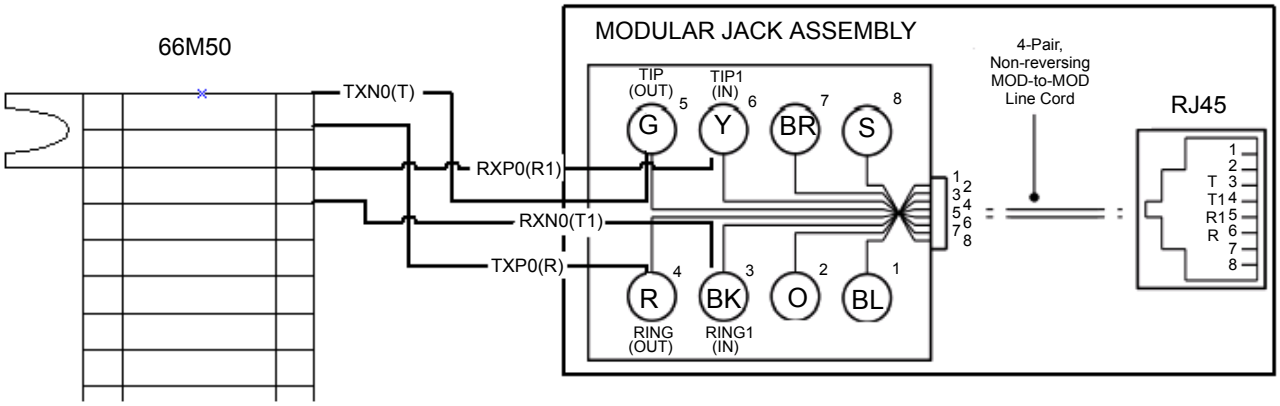
- The BRS ports are programmed or equipped as station ports. The station ports are connected to an ISDN device that has an S/T interface, such as ISDN telephone or video conferencing device.

NOTE: The device must have an S/T interface to work with the BRS card.

- A BRS port is programmed or equipped as a station port that provides dial tone to the other ISDN device (BRI station to BRI trunk) using an S/T interface

Figure 3-66. *BRS Station/Station Port to a BRI ISDN Device.*

MDF to 8-Pin RJ45 Modular Jack for the BRS Ports

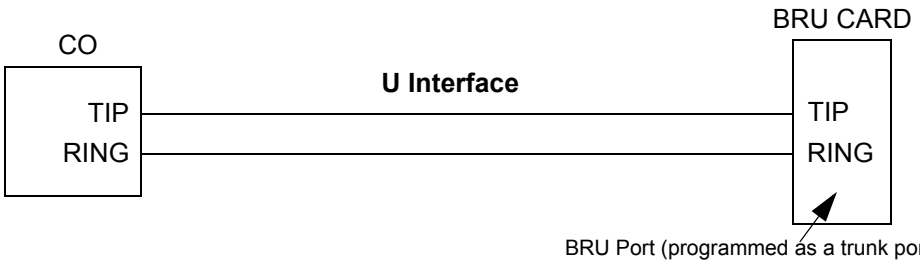


NOTE: See page 3-38 for more information on the BRS card connection.

BRU Ports Programmed as Trunk Ports

10.58 The following is an example of the termination between a BRU (U) trunk port and an incoming BRI circuit from the Central Office. The BRU ports are programmed or equipped as trunk ports that connect to the incoming BRI circuit from the Central Office or ISDN provider.

Figure 3-67. *BRU Trunk/Trunk Port to CO*



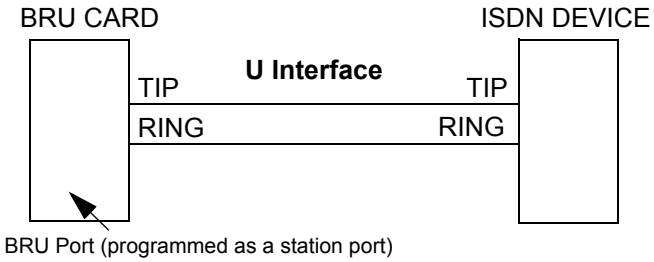
NOTE: BRI U interface is not polarity sensitive. BRU card has a built-in NT1 on each port. Therefore, an external NT1 is not required.

BRU Ports Programmed as Station Ports

10.59 The following is an example of the termination between a BRU (U) station port and a BRI ISDN (U) device using two-wire connection.

- The BRU ports are programmed or equipped as station ports. The station ports are connected to an ISDN device that has an built-in NT1 or U interface.
- A BRU Port is programmed or equipped as a station port that provides dial tone to the other ISDN device (BRI station to BRI trunk) using a U interface.

Figure 3-68. *BRU Station/Station Port to a BRI ISDN Device*



NOTE: BRI U interface is not polarity sensitive. BRU card has a built-in NT1 on each port. Therefore, an external NT1 is not required.

Terminal Endpoint Identifier (TEI)

10.60 With BRI ISDN, the Terminal Endpoint Identifiers (TEIs) are a number that identifies a specific connection endpoint within a service access point. In other words, a unique number assigned to identify an ISDN device on the ISDN bus. The configuration of the TEIs is very important when using a point-to-multi point and a point-to-point.

Table 3-18. *Terminal Endpoint Identifier (TEI)*

TEI Value	User TE Type
0-63	Non-Automatic TEI Assignment for the Equipment
64-126	Automatic TEI Assignment for the Equipment
127	BroadCast TEI

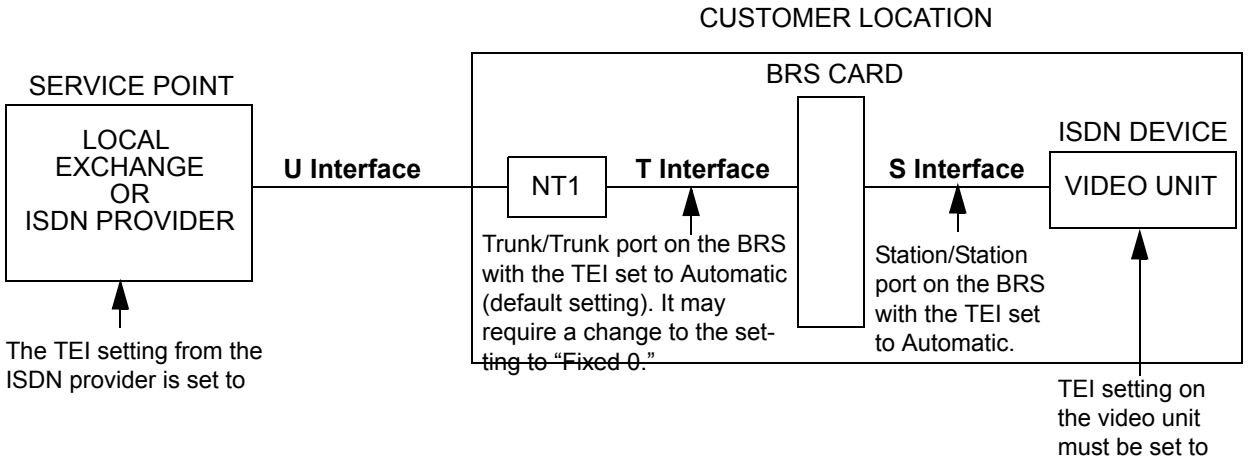
NOTE: The system supports point-to-point and point-to-multipoint applications only. For information purposes, a brief description of passive bus configuration is provided below to help with explaining the TEIs.

Point-to-Point and Point-to-Multipoint

10.61 The system supports point-to-point and point-to-multipoint (BRS trunk side only).

- **Point-to-Point Setup**: *(Supported)* This is a configuration of a single BRI ISDN circuit provided by the ISDN Provider to an ISDN device. One terminal or user system is connected through the S Interface (4-Wire) to the network termination (NT1). The length of the bus is limited to 3333 ft (1 km) of cable with a maximum cable attenuation of 6 dB at 96 kHz. In a point-to-point scenario only a single TEI value of ZERO is used as the connection is assumed always on. See below for an example of a Point-to-Point setup.

Figure 3-69. *Example of a Point-to-Point Setup Using a BRS Card*

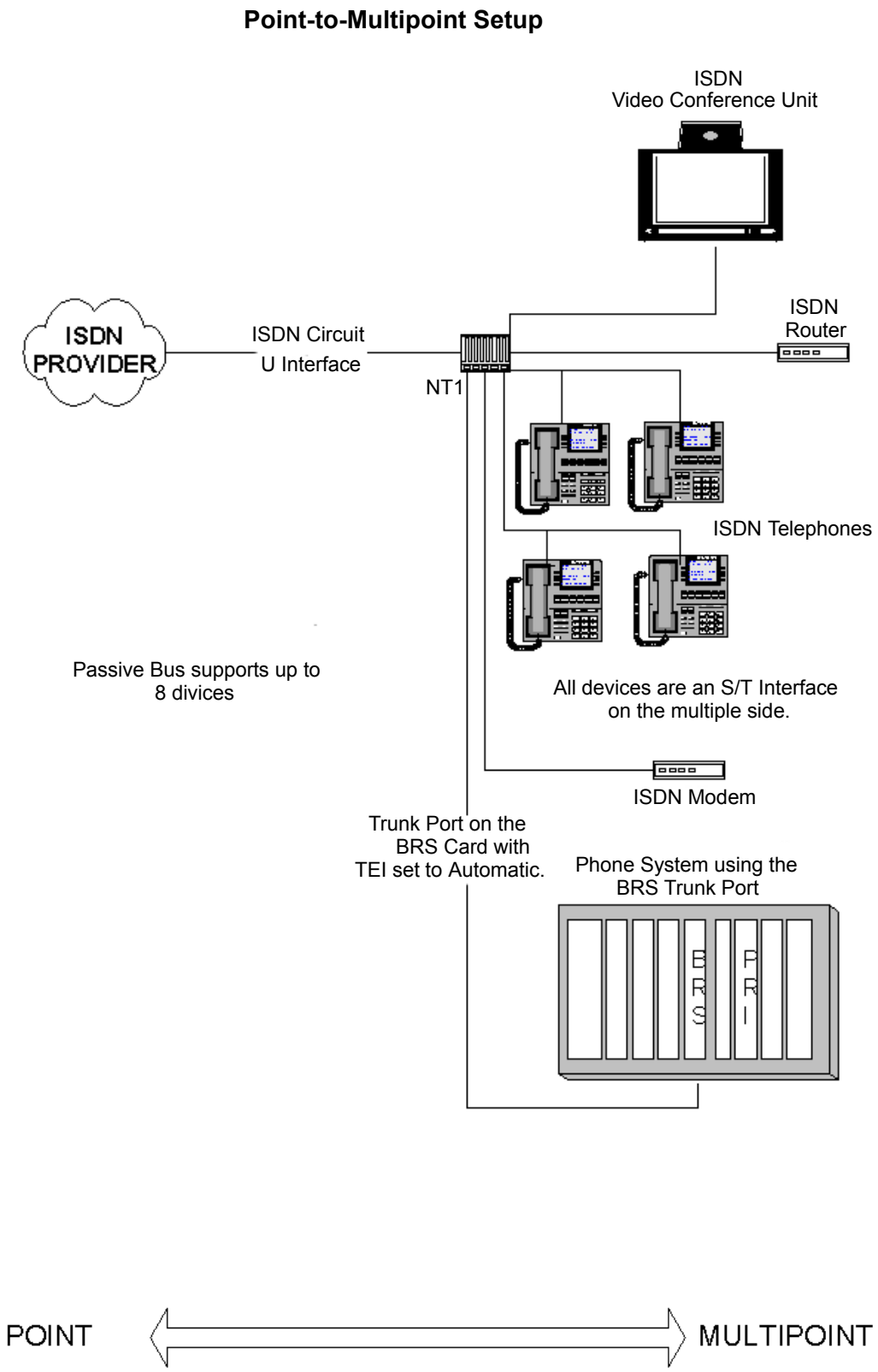


- **Point-to-Multipoint Setup (Passive Bus)**: *(Supported on BRS Trunk Side only)* The Point-to-Multipoint setup allows multiple connections from the NT1 to multiple ISDN devices. In other words, one or more ISDN devices (up to eight devices) can share a single BRI ISDN circuit. This type of configuration is called passive bus configuration. Each ISDN device on a multiple point configuration has a unique TEI number so it can be identified. The passive Bus setup is more than one ISDN device connected through the S (S/T) interface (4-wire) to the same network termination (NT1), sharing the access to the two B-channels and to the D-channel. But only two terminals can communicate simultaneously on the two B-channels in this application. Using the Inter-Tel BRS trunk port, it connects to the S (S/T) interface with other ISDN devices.

  The length of the passive bus is limited to 667 ft (200 m) of cable in a short passive bus configuration, or to 3333 ft (1 km) of cable in an extended passive bus configuration. In an extended passive bus configuration, the terminals must be clustered in the last 83 to 167 feet (25 to 50 meters) of the bus. In a point-to-multipoint (PMP) scenario multiple auto assigned TEI values are used to communicate with the devices. The devices themselves are generally on most PBXs. They are programmed only to respond to their extension numbers when they see the numbers in the subaddress field in the setup message broadcast down the bus. See Figure 3-70 on the following page for an example of a Point-to-Multipoint Setup.

NOTE: The Passive bus configuration is only supported on the Inter-Tel system with the BRS card programmed as a Trunk port. The Passive bus setup is not supported on BRS station port.

Figure 3-70. *Example of a BRS Card Trunk Side Passive Bus Setup*



Station/Station (Station-to-Station) Ports for BRS Cards

10.62 When the BRS ports are configured for Station/Station ports in the DB Programming, the TEI is configured as follow:

- The TEI is set to Automatic in the system software, and the TEI option does not appear in the DB Programming screen for the port.
- The TEI setting is non-programmable.
- The BRS ports only support a single connection to a device or a circuit.
- Each B-channel is assigned a TEI number during a call.

Trunk/Trunk (Trunk-to-Trunk) Ports for BRS Cards

10.63 When the BRS ports are configured for Trunk/Trunk ports in the DB Programming, the TEI is configured as follow:

- The TEI is programmable and the default setting is set to Automatic. The system automatically assigns a TEI once connected to an ISDN provider.
- The setting can be changed in the Trunk/Trunk port programming under the TEI settings. The programmable settings for TEI are Fixed 0 through Fixed 63 and Automatic (the automatic settings are 64 through 126).
- The BRS ports only support a single connection to a device or a circuit.
- Each B-channel is assigned a TEI number during a call.

NOTE: Local ISDN providers in Europe support a point-to-point connection, and the TEI uses Fixed 0 from the Service Point [Local Exchange or ISDN provider] to the End Point (CPE, Customer Provided Equipment). Therefore, it may require a change to the TEI setting from Automatic to Fixed 0.

Single Interface-to-Multiple Devices

10.64 Multiple Devices on a Single Interface: *(Supported)* The Inter-Tel system can have one or more ISDN circuit connected via a BRS, BRU, or PRI, then the various devices connected to the Inter-Tel system can place Data calls and/or Voice calls on this one ISDN interface or trunk port. The principal application remains the same if the incoming ISDN circuit is connected to a BRS, BRU, or PRI card. The Inter-Tel system is an ISDN PBX managing the multiple devices on a single ISDN connection. See Figure 3-72 through Figure 3-77 on the following pages for examples of Point-to-Multi-devices Setups.

Figure 3-71. *Example of a Point-to-Multi-Devices Setup Using a BRS Card*

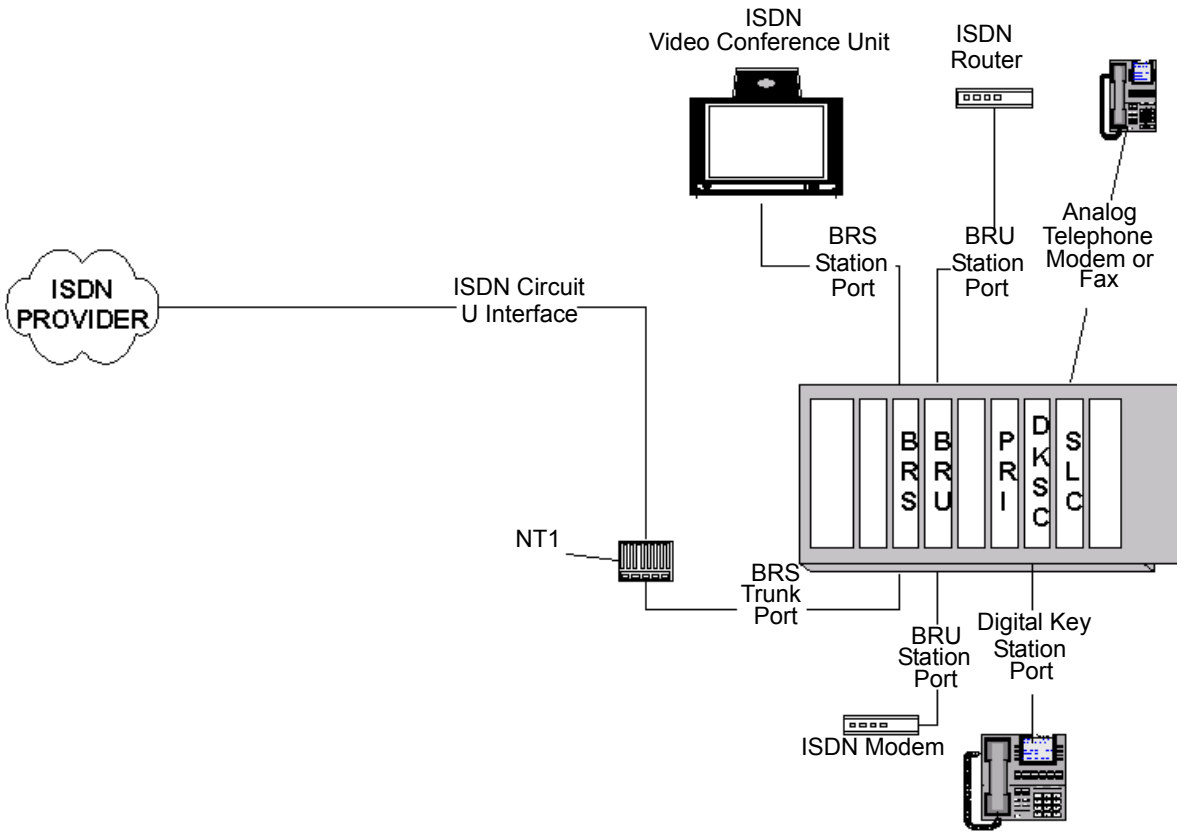


Figure 3-72. *Example of a Point-to-Multi-Devices Setup Using a BRU Card*

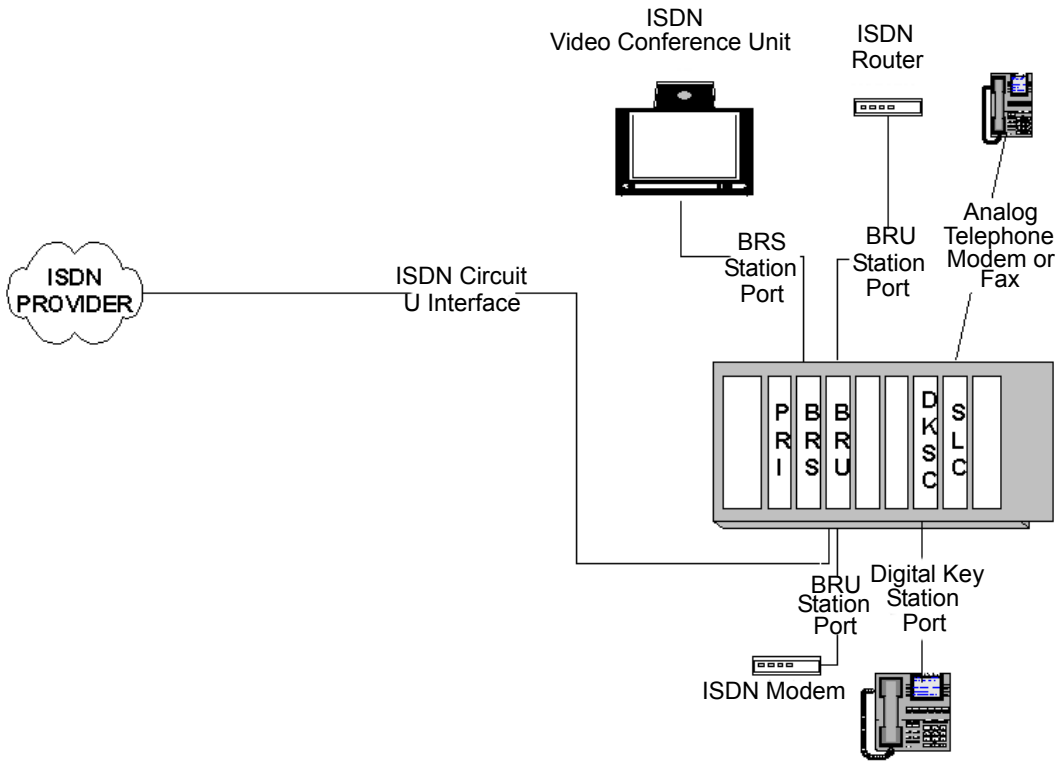


Figure 3-73. *Example of a Point-to-Multi-Devices Setup Using ISDN PRI*

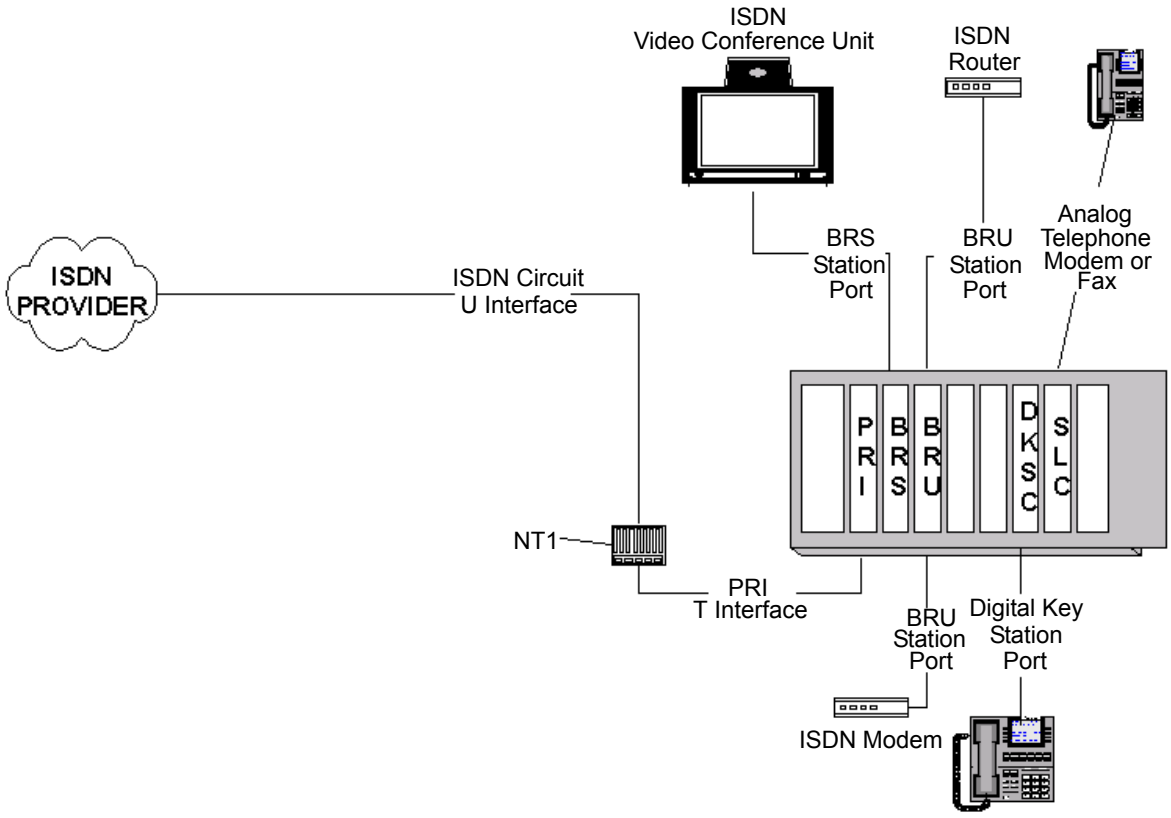


Figure 3-74. *Example of a BRI Point-to-Multi-Devices Setup without a Phone System*

NOTE: This example shows one BRI ISDN line being shared with three ISDN devices (terminals). This is a basic example of a station ISDN installation without a system. This is how the ISDN devices are installed without a telephone system.

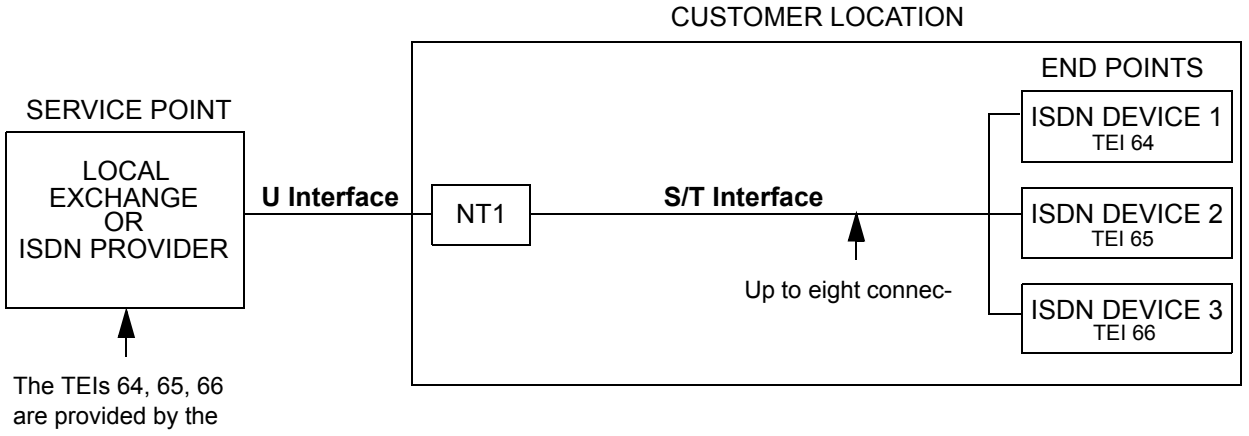


Figure 3-75. *Example of a Multi-Site Setup Using PRI and BRI*

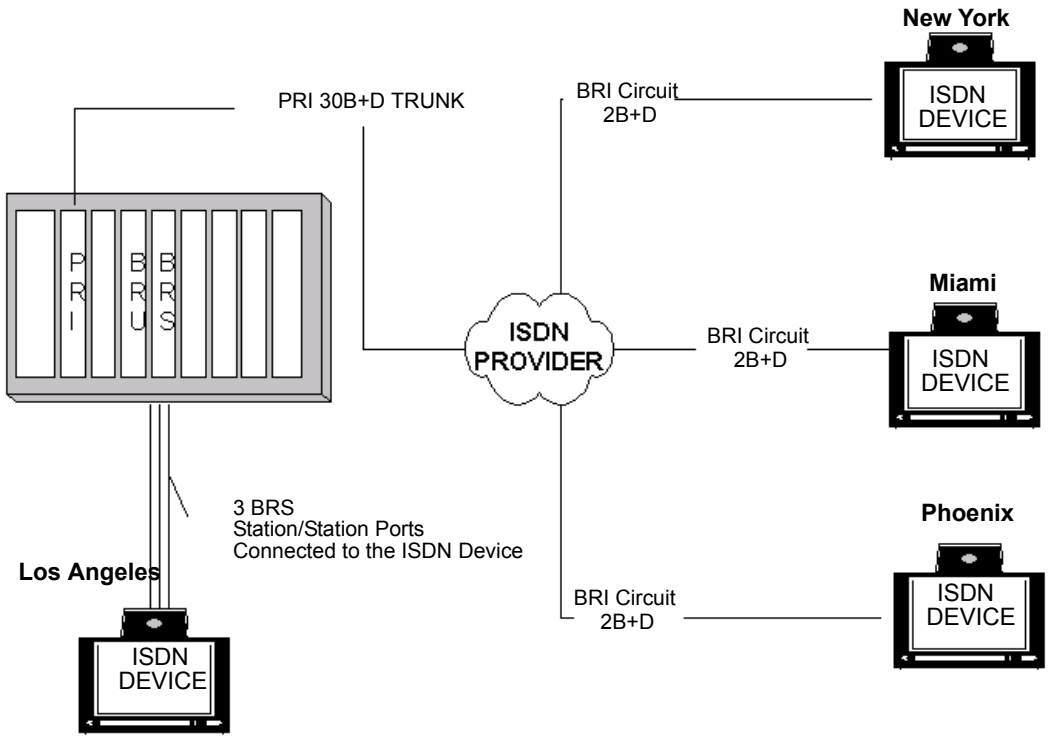


Figure 3-76. *Example of a Multi-Site Inter-Tel Systems Setup Using PRI and BRI*

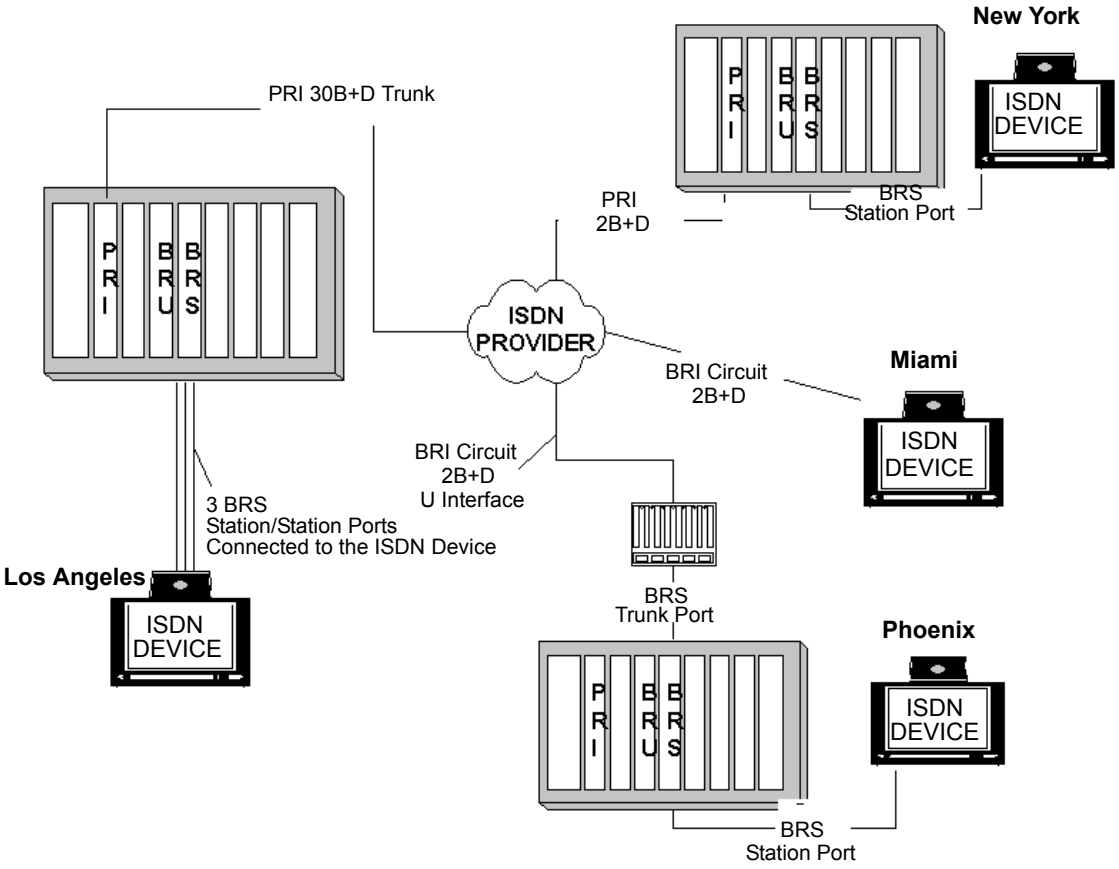
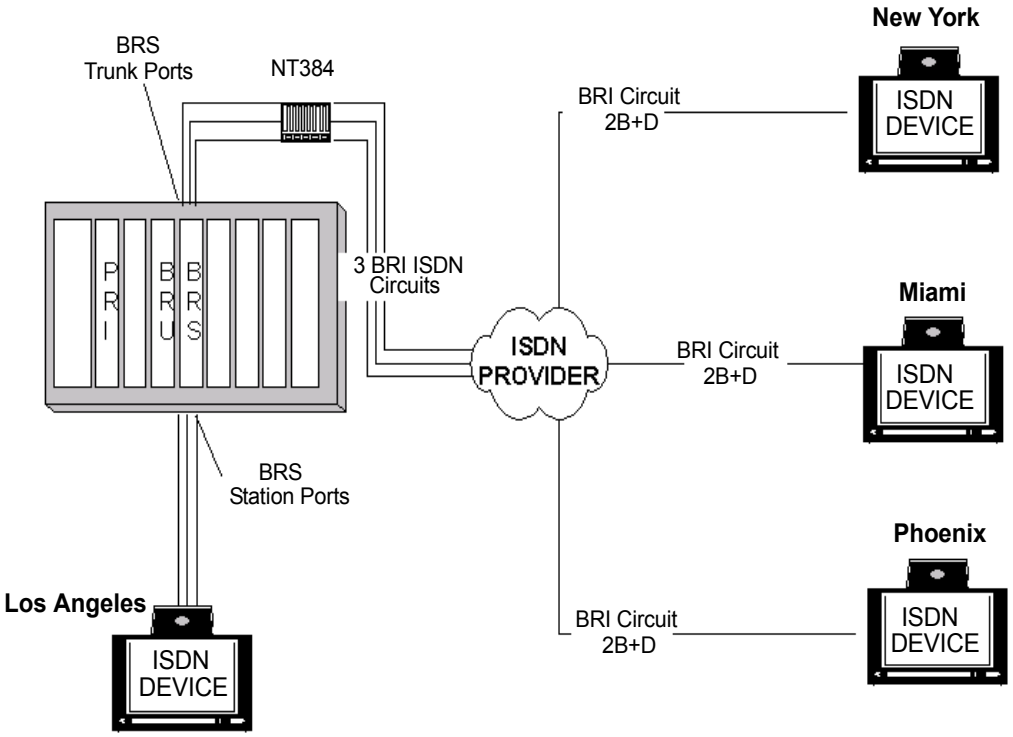


Figure 3-77. *Example of a Multi-Site Setup Using BRS Station and Trunk Ports*



The multi-site applications are multiple ISDN devices calling one central ISDN device and bonding (connecting) all three calls on one unit.

The single ISDN device on the Inter-Tel system requires multi-site capability to place or receive three different calls to the three different sites. The bonding of the three BRS or BRU ports is done in the BRI ISDN device.

The Inter-Tel system can be programmed to route the incoming calls to the ISDN device by placing all the BRS or BRU station ports in a hunt group and directing the incoming calls to this hunt group.

The examples remain the same if using either BRS or BRU ports.

M. INTERNET PROTOCOL RESOURCE CARD (IPRC)

CAUTION

Security is the responsibility of the customer's network administrator. Inter-Tel is not responsible for network problems due to security violations involving the IP address exposure of the IPRC. This includes, but is not limited to, toll fraud and interrupted network service.

10.65 Prior to v9.0, Internet Protocol Resource Cards (IPRC) were configured to be networking IPRCs (used for private networking) or device IPRCs (used for IP devices). Version 9.0 or later system software introduces an improved IPRC that supports **both** IP devices and private networking on the same card. This functionality allows for a cost reduction where one IPRC provides sufficient resources for both the IP endpoints and IP networking.

10.66 The new IPRC configuration replaces both the networking IPRC and 8.x firmware device IPRC configurations from previous releases. When version 8.1.x or later IPRC firmware is used with a v9.0 or later Axxess system, the IPRC allows an Axxess system to share IP endpoint and IP networking capabilities on the same IPRC. When upgraded to v9.0 or later, the system prevents IPRCs running with firmware older than version 8.1 to come online.

10.67 In v11.0, the IPRC "enhanced" daughter card offers new technology for echo cancellation on all calls routed through the IPRC, as well as enhanced diagnostics for troubleshooting echo problems. When the enhanced daughter card is installed and programmed, the new technology on the daughter card replaces the echo cancellers on the IPRC, resulting in improved echo cancellation. There are two types of enhanced daughter cards available:

- **IPRC-DE (Daughter Enhanced)**, part no. 550.2277: Provides new echo cancellation technology.
- **IPRC16-DE**, part no. 550.2276: Provides new echo cancellation technology with the DSP circuitry for up to 16 additional VoIP channels.

10.68 For more information, refer to the *IPRC Enhanced Daughter Card Installation Instructions* (document part no. 835.3195) and the *IPC/IPRC and IP Endpoint Firmware Notice* (document part no. 835.2720-9).

10.69 Private Networking: To use a v9.0.0 IPRC for private networking, at least one of the VoIP resources must be programmed for networking. System software v9.0 or later provides a networking wizard (see page 6-87), which provides you with a step-by-step process to quickly and efficiently program private networking on the IPRC.

NOTE: The IPRC does not support modem connections over the IP network.

10.70 IP Devices: Each IPRC supports up to 32 IP devices (16 IP devices without an IPRC daughter card), MGCP trunks, SIP trunks, multi-protocol devices, Model 8602 IP softphone, or IP networking ports. The v9.0.0 firmware IPRC does not support the IP SoftPhone. The IP SoftPhone requires an IPRC running firmware v1.x. The Model 8602 IP softphone requires IPRC firmware v9.0.0 or later. For information about IP Devices, see the latest *IP Devices Installation Manual*.

NOTICE

The IP SoftPhone requires v1.x firmware and is not supported by IPRCs that have v8.1 or later firmware. Other IP endpoints, such as the Model 8660, however, require a v8.1 or later firmware IPRC. For these reasons, Inter-Tel recommends that you do not connect IP SoftPhones and other IP endpoints to the same IPRC. If you do, you may lose feature or phone functionality.

10.71 IPRC Circuit Busy LED: When an IP endpoint comes online, the system immediately allocates a VoIP resource to the endpoint. In previous versions, the IPRC's Circuit Busy LED would light when the system allocated a VoIP resource to an IP device. The LED would also light during a networking call. In v9.0, the Circuit Busy LED only lights when the VoIP resources are in use. For IP devices, the LED is lit when an endpoint is using VoIP resources. For IP networking calls, the LED is lit when a networking call is in progress. The Circuit Busy LED turns off immediately when no VoIP resources are in use.

10.72 Converting a Pre-9.0.0 IPRC to a Version 9.0.0 or Later IPRC: When you upgrade the system to v9.0 or later system software (see page E-1) the upgrade process converts all pre-9.0.0 IPRCs (Networking and 32-Device) to the v9.0.0 IPRC configuration. When these cards have been converted, VoIP resources programmed on a Networking IPRC are converted to networking VoIP resources and VoIP resources programmed on a 32-device IPRC are converted to IP endpoint VoIP resources.

10.73 Programming: Although the physical components of the IPRC have not changed for 9.0.0, the programming interface has changed slightly. Because the IPRC now supports **both** IP devices and networking, the associated programming fields for the 32-device IPRC and the networking IPRC have been combined for the v9.0.0 IPRC (see page 6-145).

10.74 VoIP Resources: All IP endpoints use one VoIP resource on the IPRC and each VoIP resource reserved for networking also uses a resource on that IPRC in the v9.0 and later configuration. All IP endpoints are allowed to come online until the VoIP-resource limit for the IPRC is reached.

10.75 Camping on for VoIP Resources: When an IP endpoint comes online, the system may not have sufficient VoIP resources to allocate to the endpoint. If this happens, the endpoint camps on for a VoIP resource until it is available. While camping on for the resource, the endpoint's displays **WAITING FOR RESOURCES.** This display replaces the original **VOIP RESOURCE IS UNAVAILABLE** message.

10.76 Web and Telnet Password/Server: An IPRC includes Web-based and telnet configuration interfaces (see page 6-154). However, the system does not propagate changes made using these interfaces to call processing, and call processing may quickly override such changes. Therefore, you should only use these interfaces for troubleshooting purposes when Call Processing is not communicating with the IPRC. When you attempt to save any configuration changes using these interfaces, the IPRC warns you that those changes are overwritten the next time the card communicates with Call Processing. The firmware operation is as follows:

- **Companding Type**: An IPRC transmits audio data using the system-wide companding type.
- **DTMF**: When DTMF tone is detected, the system uses the DTMF encoding settings dictated by DB Programming (see page 6-154 for more information about the DTMF encoding settings).
- **Minor and Major Resets**: When an IPRC encounters a software fault (minor reset), the system attempts to reconnect all conversations and data connections. After a major reset, the system cannot re-establish existing connections.

10.77 Troubleshooting: For troubleshooting information specific to the v9.0.0 IPRC, see page 7-59.

IPRC Installation

10.78 Although the v9.0.0 IPRC supports both IP devices and networking, the physical components of the card have not changed for v9.0. The original IPRC has two serial ports (see Figure 3-78) while the newer IPRC has one (see Figure 3-79). The daughter card can attach to either IPRC. In addition, the v8.x or later firmware IPRC does not use PALs to determine the number of ports it supports. Instead, it relies on the software license.

10.79 This section details the procedures for installing a v9.0.0 IPRC, which supports both networking and IP devices on the same card.

10.80 To install the IPRC:

1. *If you plan to use one or more VoIP resources for private networking*, verify that the software license supports the IP Networking premium feature. Systems running 7.0 or earlier software also require the networking PAL. Versions 8.0 and later do not require this PAL.

2. *If using a v1.x firmware IPRC,* make sure the appropriate PAL is installed on the IPRC. The IPRC provides up to 16 ports, and the IPRC-D provides up to an additional 16 ports (32 ports total), depending on the PAL. The available networking PALs are provided in the following table.

Table 3-19. *Networking PALs for a v1.x Firmware IPRC*

Pal P/N (U.S./Europe)	IPRC/IPRC-D	Available Ports
4 port (827.9595/DI234)	IPRC	4
8 port (827.9596/DI235)	IPRC	8
16 port (827.9597/DI236)	IPRC	16
24 port (827.9598/DI237)	IPRC	16
	IPRC-D	24
32 port (827.9599/DI238)	IPRC	16
	IPRC-D	32

If using a v8.x or later firmware IPRC, make sure the appropriate software license is loaded. The available software licenses are provided in the following table.

Table 3-20. *Software Licenses for a v8.x or Later Firmware IPRC*

License P/N (U.S./Europe)	IPRC/IPRC-D	Available Ports
4 port (840.0115/SW0115)	IPRC	4
8 port (840.0116/SW0116)	IPRC	8
16 port (840.0117/SW0117)	IPRC	16
32 port (840.0118/SW0118)	IPRC	16
	IPRC-D	32

3. Upgrade the CPU card(s) using the proper upgrade kit. Refer to the following table for the appropriate kit part number.

Table 3-21. *CPU Card Kit Part Numbers*

CPU Card	Kit Part # (U.S./Europe)
CPU020/PCM-X	828.1456/SW204
CPU020/PCM-D	828.1457/SW205
CPU020/PCM-Q	828.1458/SW203

NOTE: You do not need to upgrade a CPC.

4. *If you plan to use one or more VoIP resources for private networking,* verify that the desired number of DSPs are installed on the IPRC. The IPRC supports up to eight DSPs capable of encoding and decoding data for transmission on IP networks. Each DSP is capable of providing up to two, 2-way audio connections. Audio connections include:

   - **Media Types**: The audio connections the DSPs provide are capable of encoding and decoding voice and DTMF digits.
   - **Vocoders**: The audio connections the DSPs provide are capable of using the G.711, G.729, and G.729B vocoders for encoding and decoding data.

5. *If you plan to use one or more VoIP resources for IP devices,* verify that the IPRC does not reside behind a firewall or NAT.

6. In DB Programming (System\Cabinets), select **Internet Protocol Resource Card** as the card type for the slot (see page 6-145). When prompted for a confirmation, click **Yes**. Then select the IP extension used to access the IP port on the card and click **OK**. This extension must be unique.

NOTE: Inter-Tel strongly advises installers to create a universal numbering plan that includes IP connections for forward compatibility. One such plan is to use the convention of "PSnnc" where P is the pause digit, S is the number 6 or 7, nn is the node number, and c is a number between 0-9 identifying the specific IPRC to which the IP connection corresponds.

7. Insert the IPRC in its appropriate slot in the equipment chassis. (The slot must be programmed as IPRC as discussed in the previous step.) The component sides of the card must face right. Verify that all components of the card are seated properly and that no pins are bent.

NOTE: The IPRC might halt (red "HALT" LED remains lit) on hot-card insertion. If so, simply wait about 4-5 seconds for the CPU card or CPC to bring the card out of the halted state, rather than attempt to reseat the IPRC.

8. Connect one end of the Ethernet/10Base-T/LAN cable to the RJ-45 connector labeled "LAN" on the IPRC. Connect the other end of the cable to the RJ-45 connector on the LAN or network hub.

NOTE: Do not connect to the LAN until the IP card is programmed. Some hubs have a straight-through/crossover switch. The switch must be selected for straight-through operation.

9. Upload firmware to the IPRC using the Board Utility program (see page 3-178 for details). There is no auto-download for the IPRC software from Call Processing. The Board Utility supports an RS-232 method as well as a TFTP method over IP for uploading the firmware onto the IPRC. Inter-Tel recommends using the TFTP method for uploading the firmware.

10. Configure all IPRC settings, other than those specified by the PAL or software license, using the DB Programming interface. The Voice Resources field (see page 6-146) allows you to program the 32 channels for networking or for IP devices. The Devices field (see page 6-147) allows you to configure IP devices that terminate at the IPRC.

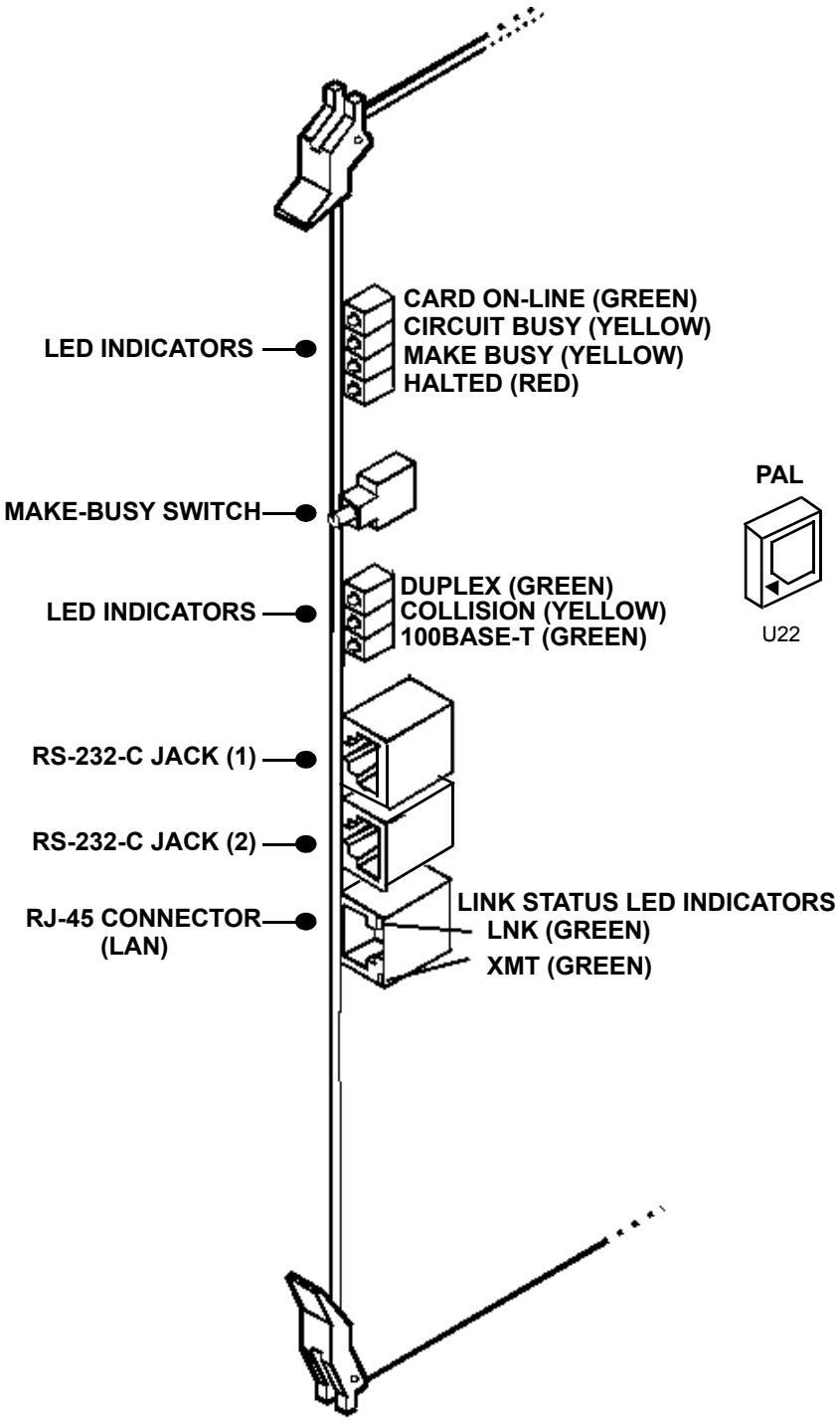
10.81 VoIP Resources: All IP endpoints use one VoIP resource on the IPRC and each VoIP resource reserved for networking also uses a resource on that IPRC in the v9.0 and later configuration. All IP endpoints are allowed to come online until the VoIP-resource limit for the IPRC is reached.

10.82 Camping on for VoIP Resources: When an IP endpoint comes online, the system may not have sufficient VoIP resources to allocate to the endpoint. If this happens, the endpoint camps on for a VoIP resource until it is available. While camping on for the resource, the endpoint's displays **WAITING FOR RESOURCES.** This display replaces the original **VOIP RESOURCE IS UNAVAILABLE** message.

10.83 Web and Telnet Password/Server: An IPRC includes Web-based and telnet configuration interfaces (see page 6-154). However, the system does not propagate changes made using these interfaces to call processing, and call processing may quickly override such changes. Therefore, you should only use these interfaces for troubleshooting purposes when Call Processing is not communicating with the IPRC. When you attempt to save any configuration changes using these interfaces, the IPRC warns you that those changes are overwritten the next time the card communicates with Call Processing. The firmware operation is as follows:

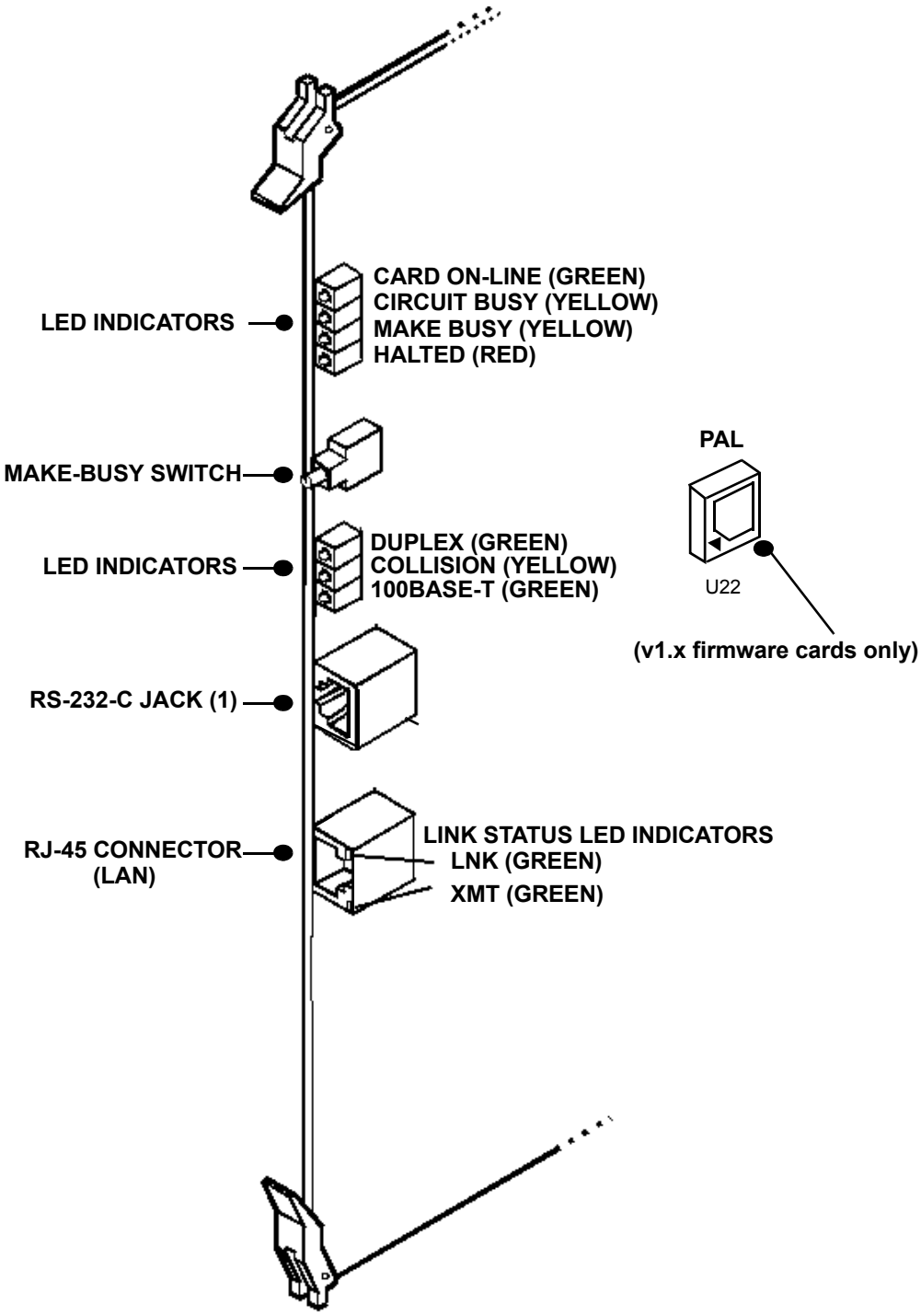
- **Companding Type**: An IPRC transmits audio data using the system-wide companding type.
- **DTMF**: When DTMF tone is detected, the system uses the DTMF encoding settings dictated by DB Programming (see page 6-154 for more information about the DTMF encoding settings).
- **Minor and Major Resets**: When an IPRC encounters a software fault (minor reset), the system attempts to reconnect all conversations and data connections. After a major reset, the system cannot re-establish existing connections.

Figure 3-78. *Original IP Resource Card (IPRC)*



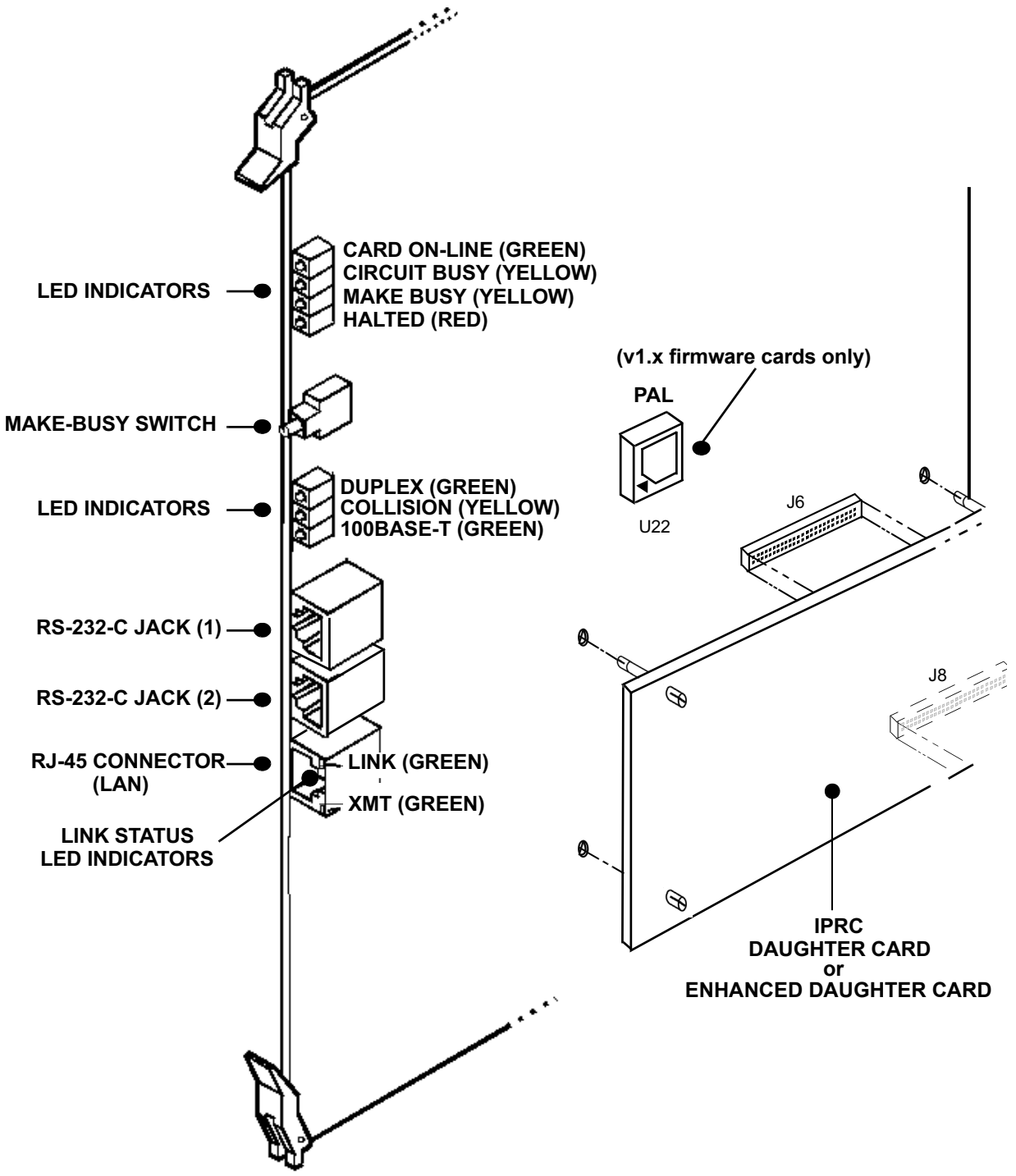
NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front of the card.

Figure 3-79. *Newer IP Resource Card (IPRC)*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front of the card.

Figure 3-80. *IPRC-D or IPRC-DE*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front of the card.

IPRC Firmware Upgrade (Version 1.x Firmware to Version 8.2 or Later Firmware)

NOTICE

Inter-Tel recommends that you upgrade the firmware on the IP endpoints BEFORE you upgrade the IPRC. If you upgrade the IPRC first, the endpoints cannot connect to the IPRC, but they continuously try. During these connection attempts, the Upload Utility cannot connect with the endpoints.

10.84 You can upgrade any existing v1.x firmware IPRC to an IPRC with v8.2 or later firmware. If IP SoftPhones terminate at the IPRC, however, you should not upgrade the card. IP SoftPhones require v1.x firmware and do not support version 8.1 firmware.

NOTE: The new v9.0.0 IPRC firmware replaces the existing v8.2.0 firmware.

10.85 Because the programming for the IPRCs has moved from the Web pages to DB Programming, it is important that you have a copy of the v1.x IPRC programming. Before you upgrade the IPRC, save the IPRC database and use its Web pages as a reference while entering the device information in DB Programming. You can also print the v1.x IPRC Web pages that contain the current IPRC and device programming information.

10.86 To upgrade a v1.x firmware IPRC to a v8.2 or later firmware IPRC:

1. Make sure the software license supports the number of multi-protocol endpoints that terminate on the IPRC.
2. Make sure the IPRC does not reside behind a firewall or NAT.
3. From the IPRC Web page, save the existing database in the event it needs to be restored. This is stored as a **`.txt`** file.
4. Remove the card from the chassis and remove the PAL (located in slot U22).
5. *If necessary,* install the daughter card.
6. Place the card in the correct slot in the chassis.
7. Run the Upload Utility and upload the 8.1 or later firmware to the IPRC and the IP devices that terminate at the card.
8. Under System\Cabinets, change the card type to **Internet Protocol Resource Card**.
9. Under System\Cabinets\<IPRC>, program the card settings using the settings saved from the Web page as a guide. (See step 3.)
10. Under System\Cabinets\<IPRC>\Devices, program the IP devices, using the settings saved from the Web page as a guide. (See step 3.)

10.87 You can now program the IP devices to terminate at the card (see page 6-147).

N. OPTIONS CARD (OPC)

10.88 If desired, additional Digital Signal Processor (DSP) resources (for DTMF receivers, conferencing circuits, Caller ID [CLIP] receivers, Caller ID [CLIP] transmitters, and shared speakerphones) can be added to the system by installing one or more Options Cards (OPCs) as described below. See Figure 3-81 on page 3-158 for an illustration.

10.89 Each Options Card (OPC) is equipped with the two ROMs (from kit part no. 828.1279) that allow the card to work with version 2 or later system software. In addition, the part numbers of the two ROMs in the kit have been changed from 827.8078 (U3) and 827.8073 (U29) to 827.8590 and 827.8599 respectively. See SPECIFICATIONS, page 2-67, and PROGRAMMING, page 6-117 for more information.

NOTE: When inserting or removing the CPU card, CPC, or Network Clock Card, the system power must be turned OFF to prevent damage to the card. All other cards can be inserted or removed with power ON.

1. *If any additional DSPs are to be installed:*
   a. Place the OPC on a clean, flat, anti-static, non-conductive surface.
   b. Remove the existing PAL from socket U22. (See Figure 3-81 on page 3-158.)
   c. Carefully install the new PAL (that allows the desired number of DSPs to be installed) in socket U22 (see the following chart). The notch (or small dot in the corner) indicating pin 1 on the component must match the notch of the outline silk-screened on the card.

Table 3-22. *DSPs*

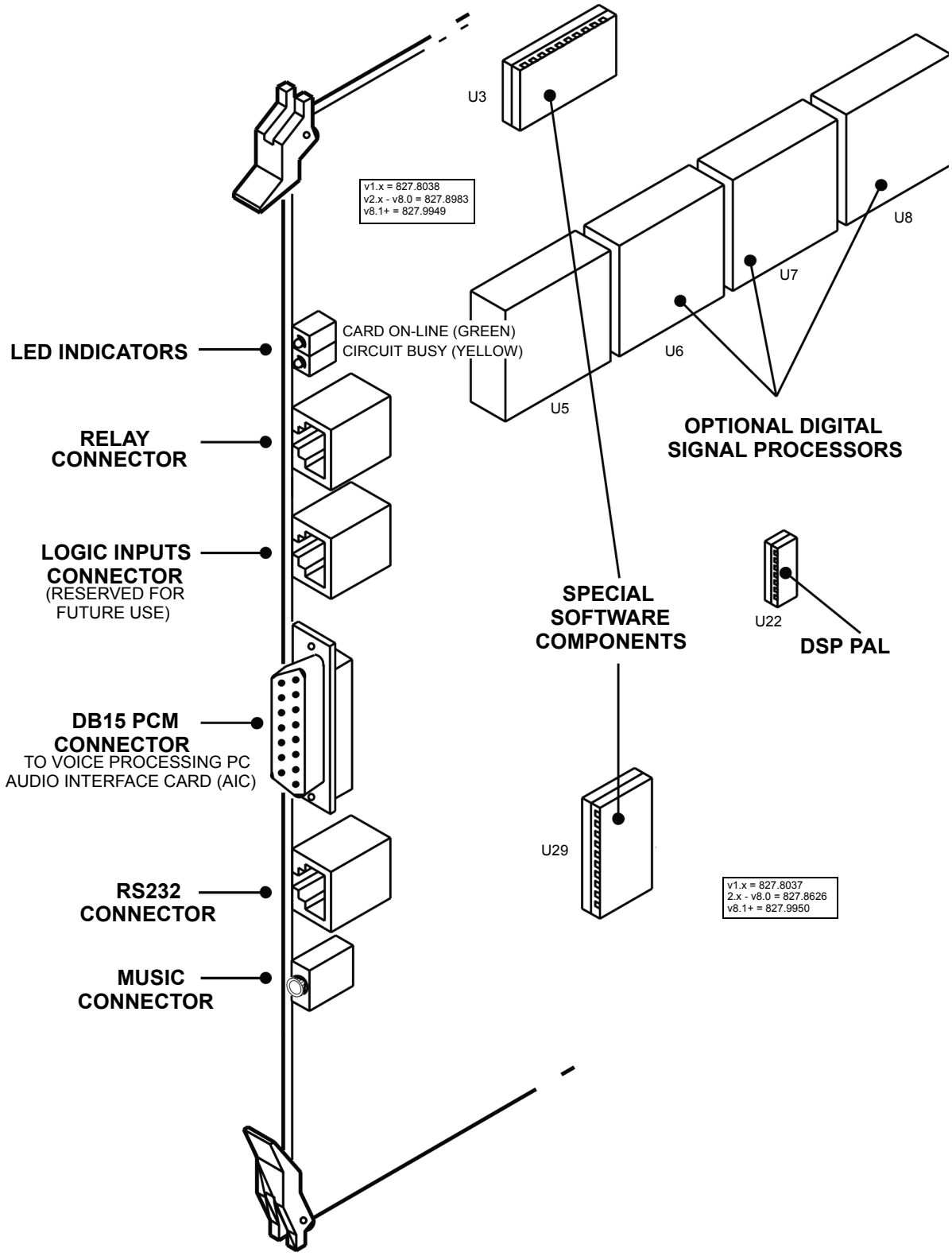
No. of DSPs	PAL P/N
1	827.8023
2	827.8024
3	827.8025
4	827.1020

   d. Carefully install the DSPs in the corresponding sockets on the OPC. Install the first additional DSP in socket U6, the second in U7, and the third in U8. (See Figure 3-81 on page 3-158.) The notch (or small dot in the corner) indicating pin 1 on each DSP must match the notch of the outline silk-screened on the card.
2. Check to see that all components on the card are seated securely in their sockets and that no pins are bent.
3. Install the OPCs one at a time in any unused card slots. The component sides of the OPCs must *face right*.

   NOTE: In tri-/quad-chassis installations where the third (slave) chassis is located away from the first (master) chassis, be sure to install the OPC in the master dual-chassis if it is used to provide additional Voice Processing channels.

4. After installing the OPC, the night transfer relay equipment, output device, Voice Processing PC port, etc. may be connected to the card at any time.

Figure 3-81. *OPC*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

O. EMBEDDED VOICE MAIL CARD (EVMC)

CAUTION

When handling the EVMC be very careful not to damage the hard drive. It is sensitive to vibration, pressure, and shock.

10.90 The Embedded Voice Mail Card (EVMC) was recently modified. Although both EVMCs function in the same way, the available ports differ. The older EVMC has two RS232 serial ports, however, only the first RS232 port is operational at this time. The newer EVMC has one RS232 serial port and one DB9 serial port. In addition, the newer EVMC no longer has the Reset button, which was for development purposes only and should not be used.

EVMC Installation

10.91 To install the EVMC, see Figures 3-82 and 3-83 on pages 3-161 and 3-162 and do the following:

1. Make sure the proper security PAL is installed in socket U23. (The PAL determines the number of available mailboxes.)

   NOTE: Currently, the 4-port EVMC is shipped with the 32-mailbox PAL.

2. *If installing a second DSPM card,* attach the *DSPM* card in the lower socket, below the existing DSPM card. Make sure that the connectors and the standoffs are properly aligned and seated.
3. Insert the EVMC into slot 7 of the equipment chassis.
4. Observe the LEDs on the front of the EVMC. The LED indications are as follows. (The left LED is labeled "CARD ON-LINE," and the right LED is labeled "OK TO REMOVE CARD.")

Table 3-23. *EVMC LED Indications*

LED Indication	Meaning
Both LEDs off	If the system equipment chassis is powered up, the operating system is starting.
Left LED blinking slowly	Voice Processing Unit is attempting to connect to the CPU or CPC.
Left LED blinking rapidly	The operating system and the Voice Processing Unit driver have started.
Left LED lit solid	Voice Processing Unit is initialized and connected to the CPU or CPC.
Right LED blinking rapidly	Voice Processing Unit is shutting down.
Right LED blinking slowly	Voice Processing Unit is shut down and the operating system is shutting down.
Right LED lit solid	Voice Processing Unit and operating system are shut down. It is now safe to turn off power and remove the EVMC from the equipment chassis.

You may see one of the following error indications.

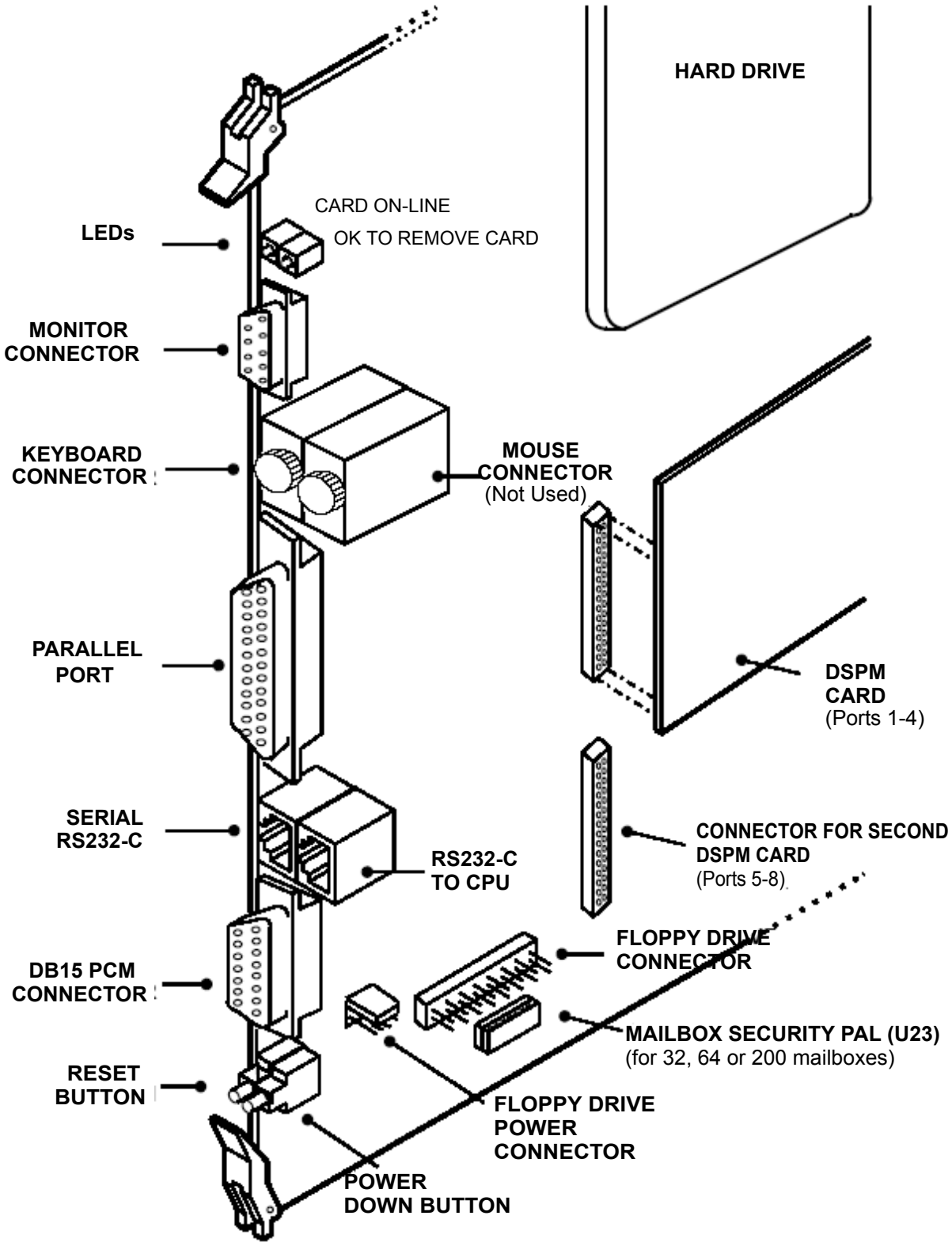
Table 3-24. *EVMC LED Error Indications*

LED Indication	Meaning
Left LED blinking rapidly, right LED blinking three on/one off	No DSPM cards are installed and Voice Processing Unit is shutting down. Press the POWER DOWN button and wait for the right LED to light before turning off the system chassis. Then check the EVMC. If there is at least one DSPM card, make sure it is seated properly. If there are no DSPM cards, install one or two. If using only one DSPM card, make sure it is in the top socket, below the hard drive.
Left LED lit, right LED flashing slowly	The EVMC PAL is not compatible with the CPU or CPC. Press the POWER DOWN button and wait for the right LED to light before turning off the system chassis. Then upload a CPC64 license (or replace the CPU with a CPU64).
Left LED lit, right LED flashing very slowly	No PAL on the EVMC. Press the POWER DOWN button and wait for the right LED to light before turning off the system chassis. Then install the appropriate PAL on the EVMC.

5. To set up the RS232 interface between the CPC (or CPU) and the EVMC, plug one end of the mod-to-mod line cord into the RS232-C-TO-CPU jack. Then plug the other end of the cord into an RS232 port on the system CPC (or CPU card). (The RS232-2 port is the preferred location.) Do *not* use the Options Card port.

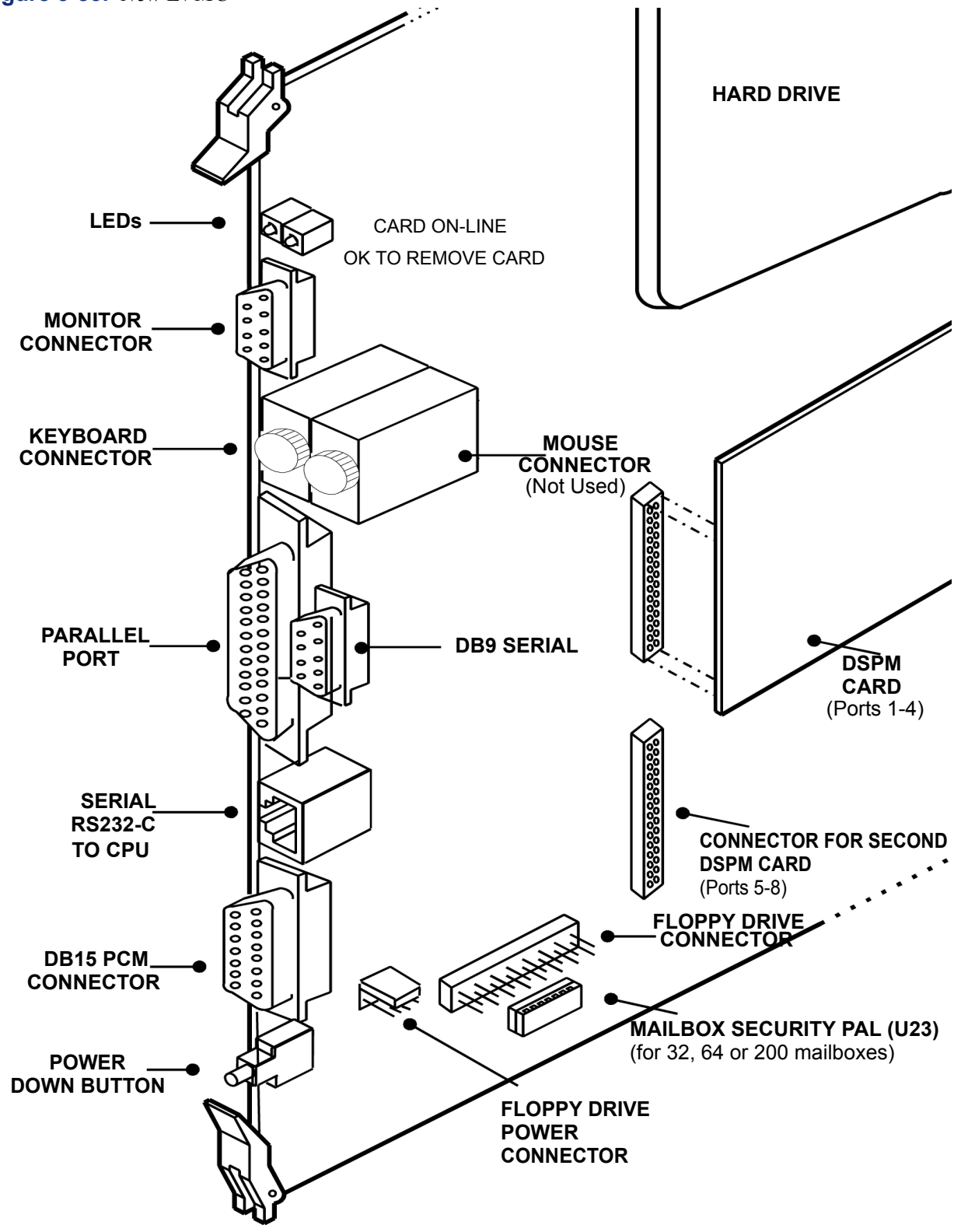
6. To enable voice path for the EVMC ports, attach one end of the DB15 interface cable (813.1739) to the DB15 PCM connector on the CPC (or CPU card). Then attach the other end of the cable to the PCM connector on the front of the EVMC.

Figure 3-82. *Old EVMC*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

Figure 3-83. *New EVMC*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

EVMC Voice Processing Software Installation

10.92 The EVMC is shipped with OS/2 or Linux and the Voice Processing applications software already installed on the hard disk drive. As a back-up, a CD containing the software is also available.

NOTE: If the software is upgraded at a later time, the new versions are available on the Inter-Tel *edGE* Web site.

10.93 To upload the software to the EVMC you need:

- A set of Voice Processing software disks (make copies from the CD)
- An external floppy drive (available in the Floppy Drive Kit 828.1486)
- A keyboard and monitor

10.94 If you are using secondary language prompts, you also need to do the following:

- Copy the secondary language prompt files to floppy disks. This requires five disks. The prompt files are located on the CPU64 software CD in the Axxess_64\CPU-SOFT\EVMCSOFT\PROMPTS directory.
- Make sure the Multilingual Support premium feature is licensed and select Secondary Language in DB Programming.

10.95 Use the following procedure to install the Voice Processing applications software via floppy drive:

CAUTION

If the EVMC is not shut down properly before it is removed from the equipment chassis, the database probably becomes corrupted, and the EVMC goes into an endless rebooting cycle.

1. Shut down the EVMC by using a small object to press the Power Down button on the front edge of the card (see illustration on the previous page for its location). The EVMC goes through a shutdown process and the "OK To Remove Card" LED flashes. When shutdown is complete, the LED is solidly lit.

   NOTE: When the "OK To Remove Card" LED turns green, it is safe to remove the EVMC from the equipment chassis for the next ***ten seconds.*** If you do not remove the card within that amount of time, allow the card to reboot and wait for the "Card On-Line" LED to light. Then you can repeat the shutdown process and remove the EVMC.

2. Remove the EVMC from the equipment chassis and lay it on a clean, flat, anti-static surface.

   CAUTION

   When handling the EVMC be very careful not to damage the hard drive. It is sensitive to vibration, pressure, and shock.

3. Attach the floppy drive cable to the floppy connector on the EVMC.

   NOTE: When plugging in the floppy drive or power connector ribbon cable, the red stripe goes on the side closest to the power supply.

4. Attach the larger connector on the floppy drive power cable to the floppy drive.

5. Attach the other (smaller) connector on the floppy drive power cable to the floppy drive power connector on the EVMC.
6. Temporarily install the EVMC in slot 6 of the equipment chassis (Or any slot that has a vacant space to the right, to allow room for the floppy ribbon cable. The slot you choose does not need to be programmed for an EVMC for this to work properly.)
7. Attach the monitor to the 15-pin connector on the front of the EVMC. Then, plug in the monitor's AC power cord and turn on the monitor's AC power switch.
8. Attach the keyboard to the keyboard connector on the front of the EVMC.
9. After some initial power-up information, the monitor eventually displays "Waiting 5 seconds. Press <ESC> to escape to the command line." Press **ESC** before the five seconds expire.
10. A flashing prompt shows "Shutting down in 5 seconds." Press **ESC** before the five seconds expire. The monitor displays the OS/2 command prompt "[OS/2 C:\AVDAP]".

NOTE: If you miss this five-second opportunity let the EVMC power up all the way up and then press the POWER DOWN button on the EVMC Card. Wait until you see "Waiting 5 seconds. Press <ESC> to escape to the command line." Then press **ESC** on your keyboard. The monitor displays the OS/2 command prompt "[OS/2 C:\AVDAP]".

11. Insert the EVMC Software Disk 1 in the PC's floppy disk drive.

NOTE: All disks must be installed. For optional secondary language voice prompts, additional disks are also required.

12. At the [OS/2 C:\AVDAP] command prompt, enter "`a:setup`" and press **ENTER**. Then follow the installation instructions displayed on the monitor.
13. **To install secondary language prompts:** Insert the Secondary Language Disk 1 in the PC's floppy disk drive. At the [OS/2 C:\AVDAP] command prompt, enter "`a:pmtinst`" and press **ENTER**. Then follow the installation instructions displayed on the monitor.

NOTE: All five language disks must be installed.

14. When the installation is complete, remove the last disk.
15. Shut down the EVMC by using a small object to press the Power Down button on the front edge of the card. The EVMC goes through a shutdown process and the "OK To Remove Card" LED flashes. When shutdown is complete, the LED is solidly lit.

NOTE: When the "OK To Remove Card" LED turns green, it is safe to remove the EVMC from the equipment chassis for the next ***ten seconds.*** If you do not remove the card within that amount of time, allow the card to reboot and wait for the "Card On-Line" LED to light. Then you can repeat the shutdown process and remove the EVMC.

16. Remove the EVMC and disconnect the floppy drive.
17. Re-install the EVMC. The EVMC reboots automatically when placed in the equipment chassis and the "Card On-Line" LED lights.
18. Program the Voice Processing Unit as described in PROGRAMMING.

P. CHECK THE CIRCUIT CARDS AND COMPLETE THE CHASSIS INSTALLATION

10.96 After all of the circuit cards have been installed in the equipment chassis, check and complete the chassis installation by performing the following steps.

1. Turn ON the AC POWER switch and observe the top (green) LED on each DKSC, DKSC-16, AKSC, EKSC, SLC, SLC-16, LGC, LSC, T1C, E1/PRI, T1/E1/PRI, IPC, IPRC, and OPC installed in the equipment chassis. *If, after one minute, the top LED on every card is lit,* proceed to step 2. *If not,* follow these steps:

   a. Remove all cards that do not have a lit green LED and check them for shorted or loose components, bent pins, etc.

   NOTE: *DO NOT* remove or insert the CPC or CPU card with the system power turned on.

   b. Re-insert the cards one at a time and check the top LED again. *If the LED is not lit after one minute,* replace the defective card.

2. Connect each labeled female amphenol-type connector to the 50-pin male connector on the corresponding card. Use caution when tightening the screws. If the screw attaching the connector to the card is too tight, the connector is not seat properly. With a U.S. system, you can use connector clips in place of the screw as an alternative. These clips are available from Inter-Tel in packs of 25 (part no. 828.1475). With a European system, ensure supplied locking clip is fitted.

3. For T1/PRI or E1/PRI trunks, connect each labeled line cord or ribbon cable to the T1 or E1 jack or DB15 connector on the corresponding T1, E1/PRI, or T1/E1/PRI card.

   NOTE: Before connecting any T1/PRI or E1/PRI trunks to the T1, E1/PRI, or T1/E1/PRI cards in the chassis, always make sure the system power is turned ON. All T1/PRI or E1/PRI trunks should always be disconnected from their cards before turning the system power ON or OFF.

4. *If attaching an output device or other peripheral device to the equipment chassis,* connect the cables to the corresponding card(s). If attaching a Voice Processing PC, see the instructions beginning on page 3-213.

5. Secure each cable to the chassis with a cable tie, leaving enough slack so that the card can be removed or replaced if necessary.

6. Replace the chassis top and front panels.

11. ATM SWITCH INSTALLATION

11.1 The ATM system configuration builds upon the dual-chassis configuration of the current phone systems. With the ATM Switch serving as the backbone, up to four, dual-chassis systems can be interconnected to create a high-capacity system. Each system chassis (or set of chassis) is then controlled by a centralized Call Processing (CP) Server, which also connects directly to the ATM Switch (see Figure 2-2 on page 2-13 for a sample configuration).

NOTE: The ATM Switch supports single- or dual-chassis systems only. For documentation purposes, the term *chassis* is used to indicate both types.

A. UNPACK THE SWITCH

11.2 When you receive the ATM Switch, it should already be configured and contain the required cards. Before installation, unpack the Switch and make sure the applicable cards are properly seated according to the specifications in the manual included with the switch.

B. DEFAULT THE ATM SWITCH

11.3 Although the ATM Switch comes pre-configured, you may need to default and then program the Switch if directed to do so by Technical Support.

11.4 To default a Switch:

1. Connect a 9-pin female/female Null Modem cable to the ATM Switch and to an available serial port on your PC.
2. Open a terminal session on that serial port.
3. Press **ENTER** to get the **`login:`** prompt.
4. Type **`ami`** and press **ENTER**.
5. Enter the password or press **ENTER** when prompted for a password.
6. At the root menu, type **`system cdb init`** and press **ENTER**.
7. Answer `y` to the "Do you wish to continue?" question and press **ENTER**.

11.5 The Switch returns to the default state. You can then program it as required.

C. PROGRAM THE ATM SWITCH

11.6 The programming differs for the *ForeRunner* LE155 and ASX-200BX switches.

ForeRunner LE155

11.7 To program a ForeRunner LE ATM switch from default:

1. Establish a terminal session on the serial port and access the root menu (see steps 1-5 in the previous section).
2. At the root menu, type `system startup` and press **ENTER**.
3. Answer the following questions as indicated.

   What name do you want to give the switch? [ATM SWITCH]

   Type `ICP ATM SWITCH` and press **ENTER**.

   What is this switch's software license key? [Base]

   Press **ENTER**, The switch responds with "The software license remains unchanged."

   IP interface to configure (none, lane) [lane]:

   Press **ENTER** to select the default.

   What IP address do you want to give the switch?

   Type `192.168.200.204` and press **ENTER**.

   And what network mask?

   Type `255.255.255.0` and press **ENTER**. The switch responds with "The switch now has IP address 192.168.200.204 with mask 255.255.255.0 on el0."

   What is the address for the gateway router?

   Press **ENTER** to select none. The switch responds with "Gateway has not been set. Continuing..."

   Would you like to select a new password? (yes, no) [no]

   Type `yes` and press **ENTER**.

   Enter the new password:

   Type `attitude` and press **ENTER**. The switch does not echo any characters.

   Please enter it again:

   Type `attitude` and press **ENTER**. The switch does not echo any characters, but it responds with "The password has been changed."

   What is the address of your ForeView management console?

   Press **ENTER** to select none. The switch responds with "ForeView management console has not been set. Continuing..."

   Only permit access from the ForeView management console? (yes, no) [no]

   Press **ENTER** to select the default. The switch responds with "ForeView management console access has not been set. Continuing..."

   Would you like to run LAN Emulation services on this switch? (yes, no) [no]

   Type `yes` and press **ENTER**. The switch responds with "Your switch has now been configured."

11.8 After you have ensured the ATM Switch is programmed correctly, you can connect the chassis together via the ATM Switch to form a large capacity node. See page 2-9 for details about connector locations.

ASX-200BX

11.9 To program an ASX-200BX switch from default:

1. Establish a terminal session on the serial port and access the root menu (see steps 1-5 in the previous section).
2. At the root menu, type `system` and press **ENTER**.
3. Type `startup` and press **ENTER**. The screen displays a welcome message.
4. Answer the following questions as indicated.

   What name do you want to give the switch? [Node X ATM SWITCH]

   Type `ICP ATM SWITCH` and press **ENTER**.

   What is this switch's software license key? [Base]

   Press **ENTER**. The switch responds with "The software license remains unchanged."

   IP interface to configure (none, lane, ethernet) [ethernet]:

   Type `lane` and press **ENTER**.

   What IP address do you want to give the switch?

   Type `192.168.200.204` and press **ENTER**.

   And what network mask?

   Type `255.255.255.0` and press **ENTER**. The switch responds with "The switch now has IP address 192.168.200.204 with mask 255.255.255.0 on el0."

   What is the address for the gateway router?

   Press **ENTER** to select none. The switch responds with "Gateway has not been set. Continuing..."

   Configure Domain Name System (DNS) service? (yes, no) [yes]

   Type `no` and press **ENTER**.

   Would you like to select a new password? (yes, no) [no]

   Type `yes` and press **ENTER**.

   Enter the new password:

   Type `attitude` and press **ENTER**. The switch does not echo any characters.

   Please enter it again:

   Type `attitude` and press **ENTER**. The switch does not echo any characters, but it responds with "The password has been changed."

   Configure SNMPv1/v2? (yes, no) [yes]

   Type `no` and press **ENTER**.

   Configure SNMPv3? (yes, no) [no]

   Press **ENTER**.

   Configure SNMP event logging station? (yes, no) [yes]

Type `no` and press **ENTER**.

Configure IP filtering (If you are running "startup" remotely, you should add the IP address of your host first. Otherwise, you will lose IP connectivity to the switch)? (yes, no) [no]

Type `no` and press **ENTER**.

Depending on the previous ELAN settings, the following prompt may be displayed:

Would you like to configure LAN Emulation services on this switch? (yes, no) [no]

If this prompt appears, type `yes` and press **ENTER**. The switch responds with a few lines of text, including notices and warnings, followed by "Your switch has now been configured." You are returned to the command prompt.

11.10 If desired, you can connect the switch to a LAN using the Ethernet port. You can then assign an IP address to this port so that you can program the switch remotely.

CAUTION

Security is the responsibility of the customer's network administrator. Inter-Tel is not responsible for network problems due to security violations involving the IP address exposure of the ATM switch. This includes, but is not limited to, toll fraud and interrupted network service.

NOTE: Inter-Tel recommends that you use a static IP address for the Ethernet port. Consult the customer's network administrator for IP address, gateway, and subnet mask information.

11.11 To program the Ethernet port for an ASX-200BX switch:

1. Establish a terminal session on the serial port and access the root menu (see steps 1-5 under "Defaulting the Switch").
2. At the root menu, type `system` and press **ENTER**.
3. Type `startup` and press **ENTER**. The screen displays a welcome message.
4. Answer the following questions as indicated.

   What name do you want to give the switch? [Node X ATM SWITCH]

   Type `ICP ATM SWITCH` and press **ENTER**.

   What is this switch's software license key? [Base]

   Press **ENTER**. The switch responds with "The software license remains unchanged."

   IP interface to configure (none, lane, ethernet) [ethernet]:

   Press **ENTER**.

   What IP address do you want to give the switch?

   Type the IP address to assign to the port and press **ENTER**.

   And what network mask?

   Type the subnet mask for the port and press **ENTER**.

   What is the address for the gateway router?

   Press **ENTER** to select none. The switch responds with "Gateway has not been set. Continuing..."

Configure Domain Name System (DNS) service? (yes, no) [yes]

Type `no` and press **ENTER**.

Would you like to select a new password? (yes, no) [no]

Type `yes` and press **ENTER**.

Enter the new password:

Type `attitude` and press **ENTER**. The switch does not echo any characters.

Please enter it again:

Type `attitude` and press **ENTER**. The switch does not echo any characters, but it responds with "The password has been changed."

Configure SNMPv1/v2? (yes, no) [yes]

Type `no` and press **ENTER**.

Configure SNMPv3? (yes, no) [no]

Press **ENTER**.

Configure SNMP event logging station? (yes, no) [yes]

Type `no` and press **ENTER**.

Configure IP filtering (If you are running "startup" remotely, you should add the IP address of your host first. Otherwise, you will lose IP connectivity to the switch)? (yes, no) [no]

Press **ENTER**. The switch responds with "Default ELAN exists. Your switch has now been configured."

11.12 You can now connect to the switch using a telnet session.

D. CONNECT THE ATM SWITCH TO THE CHASSIS AND CP SERVER

11.13 The interface and architecture of the ATM switch types differ. This affects the way the switch physically connects to the chassis and CP Server, as described in the following sections.

ForeRunner LE155

11.14 The *ForeRunner* LE155 switch houses three (or four if the expansion slot is populated) different interfaces, each labeled A-C (or D). Each of these interfaces contains four ports (1-4), for a total of 12 ports (16 with the expansion slot). Each individual port is then identified by the interface letter and port number. For example, the third port under Interface B is considered Port B3.

11.15 The architecture of the ATM Switch allows each interface to function independently. Using the Element Manager, see the ATM Switch documentation on the Ericsson® (Marconi) CD, you can reset one group of ports without affecting the others. For this reason, use a separate interface for each chassis connection when possible. For example, if you have four chassis, connect the first to Port A1; the second to Port B1; the third to Port C1; and the fourth to Port A2 (or D1 if populated). When configured this way, you can reset ATM connections almost entirely on a per-chassis basis. For example, if chassis two has a bad connection but chassis three is running, you can reset Interface B without bringing down the third chassis.

NOTE: The chassis PCMAs must have the same connector types (UTP5, MMF, or SMF) as those provided on the ATM Switch.

11.16 To connect the ForeRunner LE155 switch to the PCMA cards and the CP Server:

1. Connect CAT5 or MMF cables to the appropriate PCMA card and ATM Switch connectors. Connect each equipment chassis to a different interface on the ATM Switch, if possible. For more information on the PCMA Groups and how they connect to the Switch, see page 3-14.
2. Connect CAT5 or MMF cables to the PCMA card and ATM Adapter card on the CP Server. If using a redundant system, connect each CP Server to a different interface on the ATM Switch.

NOTE: The CP Server is responsible for ensuring the chassis in an ATM system are running compatible operating system (OS) or CP versions. When the CP Server first signs on to a chassis, the chassis sends a version response. If the version is not compatible, the CP Server does not allow that particular CPU to come online, and message print M5134 is submitted.

ASX-200BX

11.17 The architecture of the ASX-200BX switch allows each interface to function independently. Using the ATM Management Interface (AMI), you can reset one group of ports without affecting the others, see the ATM switch documentation on the Ericsson (Marconi) CD. For this reason, use a separate interface for each chassis connection when possible. For example, if you have three chassis, connect the first to Port 1A1; the second to Port 1B1; and the third to 1C1. When configured this way, you can reset ATM connections almost entirely on a per-chassis basis. For example, if chassis two has a bad connection but chassis three is running, you can reset Interface B without bringing down the third chassis.

NOTE: The chassis PCMA cards must have the same connector types (UTP5, MMF, or SMF) as those provided on the ATM switch.

11.18 To connect the ASX-200BX switch to the PCMA cards and the CP Server:

1. Connect CAT5 or MMF cables to the appropriate PCMA cards and to the ATM switch connectors. Connect each equipment chassis to a different interface on the ATM switch if possible.
2. Connect a CAT5 or MMF cable to the ATM Adapter card on the CP Server and to the ATM switch connector. If using a redundant system, connect each CP Server to a different interface on the ATM switch.

NOTE: The CP Server is responsible for ensuring the chassis in an ATM system are running compatible operating system (OS) or CP versions. When the CP Server first signs on to a chassis, the chassis sends a version response. If the version is not compatible, the CP Server does not allow that particular CPU to come online, and message print M5134 is generated.

12. UPLOADING SOFTWARE

12.1 The Board Utility program, supplied with the software can be used for loading software on the CPC, CPU, T1/E1/PRI cards, BRS/BRU cards, and IP devices.

A. RS232 CONNECTIONS

1. Ensure that the programming PC is equipped with the DB Programming software. (See page 6-17 in PROGRAMMING for details.)
2. Make the RS232 connection between the circuit card and the programming PC, as follows. The necessary connectors can be purchased separately or in the Universal RS232 Kit (part number 828.1282).

   For CPC, CPU, and T1/E1/PRI cards:

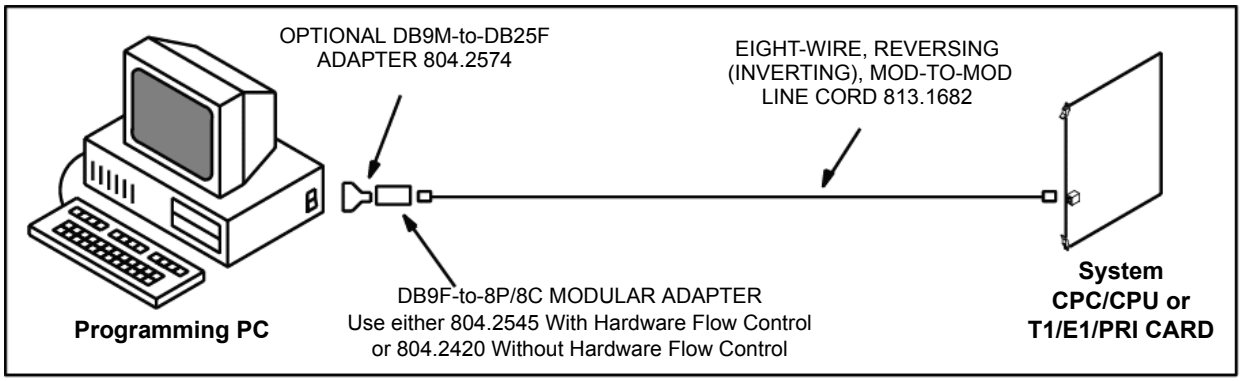
   a. Attach one end of an 8-wire reversing mod-to-mod line cord to the circuit card.

      - CPC or CPU: Use the *top* RS232 port (labeled CPU SERIAL PORT 1) on the CPC or CPU card. See Figure 3-50 on page 3-107 for a drawing showing the serial ports.
      - T1/E1/PRI: Use the T1/E1/PRI serial port jack.

   b. Attach the other end of the line cord to a DB9F-to-8P/8C modular adapter with the proper flow control. (This application does not require flow control unless the bps rate exceeds 9600.)

   c. If the PC does not have an available DB9M COM port but does have an available DB25M COM port, attach the DB9M-to-DB25F converter to the DB9F end of the modular adapter.

   d. Connect the cable to the appropriate COM port on the programming PC.



For IPC/IPRC/IPRC-Ds:

a. Connect one end of the Ethernet/10Base-T/LAN cable to the RJ45 connector labeled “LAN” on the card.

b. Connect the other end of the cable to the RJ45 connector on the LAN or network hub.

B. UPLOADING CALL PROCESSING CARDS

NOTE: Although the CPC supports a TCP/IP network connection and a direct serial connection (using an RS-232 cable), Inter-Tel recommends using the TCP/IP connection when uploading the software.

1. Ensure that the programming PC is equipped with the DB Programming software. (See page 6-17 in PROGRAMMING for details.)
2. Press the * button while powering up the CPC to place the card in download mode. The CPU Active and DB LEDs alternately lights.
3. Connect one end of the Ethernet cable from the LAN or network hub to the IP port on the CPC. (See Figure 3-47 on page 3-96 for a drawing showing the IP port.)
4. Select Start - Inter-Tel DB Programming - **Board Utility**. (The main upload screen appears.)
5. Click **New** or select **New** from the File menu to create an entry.
6. Enter a description, up to 40 characters, in the **Settings For** field that allows you to identify the connection type. For example, if creating a setting for the CPC on node 2, enter `Node 2 CPC`.
7. Select **CPU/CPC - Call Processing Cards** as the Board type.
8. Enter the appropriate upload source file name in the File box or use the **Browse** button to locate the file (the file ends with **`.bin`**). Make sure the complete file name appears in the box.
9. Enter the IP Address or hostname of the CPC. The IP address is the device's current IP address or the default IP address that is programmed into the device's local database in flash memory.
10. Enter the Username and Password required to access the CPC. The default username is `CPU`, and the default password is `itpassw`.
11. If necessary, enter the port number that corresponds to the CPC Web Listening Port field in DB Programming (see page 6-179). The Port value must match the Web listening port number; otherwise, the upload fails. By default, this is 80.
12. If the System has just been powered up, wait at least three minutes to allow the system to stabilize and finish printing power-up messages before performing the next step.
13. Click **Start** (or select Start from the File menu). The software is automatically uploaded into flash memory on the CPC.

    If the upload fails to start, the CPC LEDs indicate that the card is still in the download state, and the PC displays the reason for the failure. The card remains in download mode until the upload is attempted again. If an upload is not attempted before the download state times out, the system resets.
14. When the upload is complete, the system resets and the PC displays a message indicating the operation was successful. Click **OK** and then exit the program.

C. UPLOADING CPU CARDS

NOTE: As long as both the first (master) and third (slave) chassis in tri-/quad-chassis installations are properly connected and on line, the system software only needs to be uploaded to the *master* CPU020-Q/PCM-F card. However, if for some reason the third (slave) chassis is not connected or is off line, the system software must also be uploaded to the *slave* CPU020-X/PCM-F card.

1. Ensure that the programming PC is equipped with the DB Programming software. (See page 6-17 in PROGRAMMING for details.)
2. *If connected to a CP Server,* stop the Call Processing Service (see page C-9 for details). You cannot upload software if the CP Server is attempting to communicate with any of the cards.
3. Attach an RS232 cable from an available COM port on the PC to the *top* RS232 port (labeled CPU SERIAL PORT 1) on the CPU card. (See Figure 3-50 on page 3-107 for a drawing showing the serial ports.)
4. Select Start - Inter-Tel DB Programming - **Upload Utility**. (The main upload screen appears.)
5. Click **New** or select **New** from the File menu to create an entry.
6. Enter a description, up to 40 characters, in the **Settings For** field that allows you to identify the connection type. For example, if creating a setting for the CPU128 on node 2, enter `Node 2 CPU128`.
7. Select **CPU/CPC - Call Processing Cards** as the Board type.
8. Select the appropriate "Port" and the desired "Speed" (bit per second rate). Estimated upload times are listed in the following table.

Table 3-25. *Estimated Upload Time*

BPS RATE	APPROX. UPLOAD TIME
9600	15 Minutes
19200	12 Minutes
38400	6 Minutes
57600	4 Minutes
115,000	2 Minutes

NOTE: Be sure to select a bit rate that is compatible with the PC's capabilities. For example, only PCs equipped with high-speed serial communications chips (e.g., 16550 UART) can use the 38400 and 57600 bit rates.

9. Enter the appropriate upload source file name in the File box or use the **Browse** button to locate the file (the file ends with **`.bin`**). The file name usually depicts the country, version number, and capacity of the system. For example, **`us_52F9_prod_cpu_512.bin`** indicates the file required for a CPU512 card that is running the production version of 5.2F9 for the U.S. Make sure the complete file name appears in the box.
10. If the system has just been powered up, wait at least three minutes to allow the system to stabilize and finish printing power-up messages before performing the next step.
11. Click **Start** (or select Start from the File menu).

12. The software is automatically uploaded into flash memory on the CPU card. A warning message appears if you attempt to load CPU 512 Master software without a CPU 512 Slave card installed and communicating. You can continue the upload by clicking **Yes** or abort the upload by clicking **No**. (If you click **Yes**, you must install software on the CPU 512 Slave card before it functions.)

13. *If the upload fails to start,* press the reset switch on the CPU card six times within three seconds, then retry the upload. If the upload fails while in progress, the red and yellow LEDs on the CPU card blink rapidly (ten times per second), and the PC displays the reason for the failure. If possible, correct the problem and retry the upload.

14. When the upload is complete, the system resets and the PC displays a message indicating the operation was successful. Click **OK** and then exit the program.

D. CHECKING SOFTWARE ON A CPC OR CPU

12.2 You can use the following procedure to check the validity of software on the CPC or CPU card:

1. Attach a terminal (or a PC using terminal emulation software) to the IP port for a CPC or to SERIAL PORT 1 for a CPU. (Communication parameters for the serial port are 9600 bps, no parity, 8 bits, and 1 stop bit.)

   NOTE: If the system contains an EVMC, you must first shut down the EVMC properly before turning off the Power Supply.

2. Turn off the Power Supply on the first chassis and the third chassis (if used).

   NOTE: Be sure not to unplug CPCs or CPU card(s) with any Power Supply still ON.

3. Turn on the Power Supply on the first chassis or the third chassis while watching the power-up messages on the terminal.
   You should see:
   Boot Version 8.X
   Inter-Tel 8.X Call Processing

4. If you do not see the correct information, upload the premium software from the programming PC to the CPC or CPU card using the Board Utility. To do so, select the **Board Utility** menu from the Inter-Tel DB Programming.

E. UPLOADING T1/E1/BRU/BRS CARDS

1. Connect a programming PC to the card's serial port jack.
2. Click **New** or select **New** from the File menu to create an entry.
3. Enter a description, up to 40 characters, in the **Settings For** field that allows you to identify the connection type. For example, if creating a setting for the BRU Card on node 2, slot 7, enter `Node 2 BRU slot 7`.
4. Select **T1/E1/BRU/BRS** as the Board type.
5. Enter the appropriate upload source file name in the File box or use the **Browse** button to locate the file (the file ends with ".ss"). The file name usually depicts the connection type (T1 vs. E1), the country, and the version of the system. For example, **`t1_us6_0.ss`** indicates the file required for a T1 Card that is running version 6.0 for the U.S. Make sure the complete file name appears in the box.
6. Set the proper "Port" and "Speed" for the programming PC.
7. Click **Start** (or select Start from the File menu).
8. When the software is fully loaded, the card is ready for use.

12.3 Uploading the Boot Code: The boot code controls the upload process. The boot code is the first code that runs on the card when it's powered up, then the control of the card is passed to the card specific code.

12.4 The boot code is provided with the BRS and BRU upload files (Boot.ss) and should only be uploaded if the board is not functioning or is having problems uploading the BRS.ss or BRU.ss upload files. The Boot.ss file for the T1/E1 card is different from the Boot.ss file that comes with the BRS and BRU cards. Therefore, be careful not to copy over the files.

12.5 Uploading the boot.ss file on the BRS or BRU card:

The write protect jumper (J4) must be installed over the top two pins (program position). See Figures 3-63 and 3-64 on pages 3-133 and 3-134. Then using the Board Utility, upload the boot.ss file.

NOTE: Use the DBRD (Display Board Information) command to verify software versions on the BRU and BRS cards. For more details, refer to the *ISDN Troubleshooting Manual*.

F. UPLOADING IP DEVICES

NOTICE

Inter-Tel recommends that you upgrade the firmware on the IP endpoints BEFORE you upgrade the IPRC. If you upgrade the IPRC first, the endpoints cannot connect to the IPRC, but they continuously try. During these connection attempts, the Upload Utility cannot connect with the endpoints.

12.6 The latest version of the Upload Utility (previously called the Board Utility) allows you to upgrade multiple IP devices at once. This feature was implemented to assist installers who need to upgrade the IP devices to take advantage of the version 8.1 and later features (i.e., peer-to-peer audio).

12.7 With the Upload Utility, you can also save your settings so that you do not have to configure information for every upload. For example, you can create an entry used strictly for IPRCs on Node 2. Or, you can create one for all hard IP endpoints, such as the Model 8660, in your network.

12.8 The Inter-Tel Upload Utility application allows you to upload object code to upgrade the firmware in IP devices.

NOTE: If uploading boot code to the IPC, make sure to move the boot-sector protection jumper to the top two pins for the upload.

1. Click **New** or select **New** from the File menu to create an entry.
2. Enter a description, up to 40 characters, in the **Settings For** field that allows you to identify the connection type. For example, if creating a setting for the IP devices on node 2, enter `Node 2 IP Devices`.
3. Select **IPP+/IP SLA - (IP Phones)** as the "Board" type. The Username field is automatically populated (IPT) in the Network tab.
4. Click **Browse** in the Network tab. The Upload Utility queries the network for a list of IP addresses for Inter-Tel devices on the network. This information is then displayed in the Browse IP Address screen and includes the following:
   - **IP Address**: The IP address of the board or device found on the network.
   - **Hostname**: The hostname assigned to the board or device.
   - **Description**: The description used to identify the board or device in DB Programming. For IP phones, this is the station description.
5. Highlight the IP address of the desired device and click **Add**. To add all IP addresses or hostnames, click **Add** without highlighting any items. The selected IP addresses are added to the list box in the main screen.

   NOTE: To add multiple addresses/hostnames, use the standard Windows **SHIFT** + click or **CTRL** + click method.

6. Click **Save** to save the settings. At the prompt, click **Yes** to confirm the save.

7. Specify the file name that you want to upload in the "File" field, or click **Browse** to select a file from the dialog box. The source file names differ based on the device. In general, the file names are **`ipp+_X_Y_Z.hex`** for the IP Phone+, **`ipp8660_X_Y_Z.bin`** for the Model 8660 Phone, and **`ip_sla_X_Y_Z.hex`** for the IP SLA. For these file names, *X_Y_Z* represents the version number (e.g., **`ipp+_8_1_0.hex`** is for version 8.10).

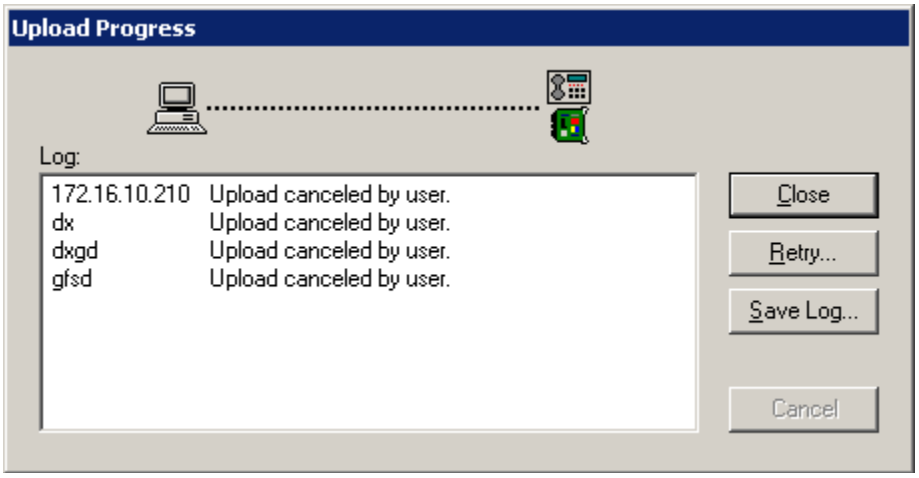
NOTE: If using an IPRC, make sure you select the correct source file based on whether the card is used for networking (iprc_ipnc.hex) or IP devices (iprc_iksc.hex).

8. Enter the target device's "Username" and "Password." The default usernames and passwords are shown below.

Table 3-26. *IP Devices Default Usernames and Passwords*

Device	Username	Password
IP Card (IPC or IPRC)	IPC IPRC	ipcpassw itpassw
IP PhonePlus or IP Single-Line Adapter	IPT	iptpassw

9. After entering all the fields, click **Start** (or select **Start** from the File menu). If you entered the correct IP address, username, and password, the upload process begins, and an Upload Progress dialog box appears. This dialog box indicates the IP address or hostname of the affected devices and the status of the upload.



While in this screen, you have the following options:

- **Close**: Closes the screen.
- **Retry**: Allows you to attempt the upload again for failed operations. To retry the upload, highlight the IP address or hostname of the devices that did not get upgraded and click **Retry**. To attempt the upload again for all devices that did not get upgraded, do not select any IP addresses or hostnames before clicking **Retry**.
- **Save Log**: Saves the log information to a text file (`.txt`).
- **Cancel**: Terminates the upload operation.

CAUTION

If you click **Cancel** while uploading files to IP devices, the Upload Utility generates a warning message. This message indicates that the IP address for the IP devices may return to factory default, which is the same for all IP devices. In other words, all IP devices have the **SAME** IP address. If this occurs, you must either reprogram the IP address for each device or upload the file to each device one at a time.

10. Click **Close** when the upload is complete.
11. After the device accepts the file, the device indicates this change of state.
    - — The IPC or IPRC flashes the green Card On-Line LED and the yellow Circuit Busy LED once per second while it erases and programs its flash memory.
    - — The LCD on the IP PhonePlus displays ERASING FLASH, followed by PROGRAMMING CODE, and followed by DOWNLOAD COMPLETE.
    - — The IP SLA flashes the On-Line LED and the SL Status LED once per second while it erases and programs its flash memory.

NOTE: Do not forget to replace the boot-sector protection jumper on the IPC to the lower two pins once the upload completes.

13. STATION INSTALLATION

NOTE: For information about installing analog and Executone™ phones, see a version 7.0 or earlier *Installation and Maintenance Manual*.

13.1 Digital phones (and any attached DSS/BLF Units) are connected to Digital Keyset Cards (DKSCs) and/or 16-Port Digital Keyset Cards (DKSC-16s). Analog phones are connected to Analog Keyset Cards (AKSCs). Executone IDS and Wave phones are connected to Executone Keyset Cards (EKSCs). Single-line DTMF or pulse-tone sets and playback devices are connected to Single-Line Cards (SLCs) or 16-Port Single-Line Cards (SLC-16s), or they can be connected to Single-Line Adapters, which are in turn connected to available DKSC/DKSC-16 circuits. (Version 2.x or later software is required for pulse-tone sets.)

CAUTION

Connecting a station instrument to the wrong type of station card can damage both the instrument and the card.

A. DIGITAL PHONE INSTALLATION

NOTICE

The digital phones are proprietary phones and are to be used **only** with Inter-Tel Phone systems.

13.2 Before installing the digital phones, unpack each one and check for damage. If items are damaged or missing, contact Customer Support. Each phone should have the following:

- Baseplate
- One six-foot, three-pair line cord
- One handset
- One four-conductor coiled handset cord
- Slide-out directory card *(not available on Model 8500, 8520, or 8560 Phones)*
- Extra feature key templates

13.3 To install digital phones:

1. Before mounting the modular jack (line jack unit) assembly and connecting the phone, measure the voltage on the YELLOW terminal of the modular jack (line jack unit) assembly with respect to the BLACK terminal. (The "common" probe of the voltmeter is placed on the BLACK terminal.) It must measure +24VDC (±2.0VDC). If a negative voltage is measured, check the cabling for a reversed pair.

   CAUTION

   If tip and ring are reversed, the phone does not function.

2. Mount the modular jack (line jack unit) assembly on the wall.
3. Attach the coiled handset cord to the handset and to the handset jack on the bottom of the phone. Place the handset on hook.

NOTE: The bottom of the Model 8500, 8520, and 8560 Phones have molded channels designed to hold the handset and line cords. For best results when connecting a handset, uncoil a portion of the handset cord and push the cord into the channel so that the flat side of the cord is against the phone.

4. Plug one end of the line cord into the wall-mounted modular jack (line jack unit) assembly. Plug the other end into the line cord jack on the bottom of the phone.

NOTE: To aid in installation and troubleshooting, display phones show their station circuit number, extension number, and assigned user name for several seconds when power is turned ON and the line cord is first plugged in. This display also appears after a system reset, and whenever the line cord is removed and replaced while power is ON. The station identification displays for several seconds, then the phone changes to the appropriate display, depending on its current status.

5. *If installing an Executive Display, Professional Display, Standard Display, Associate Display, Axxess Basic Digital, or Eclipse Basic Digital Phone,* perform the digital phone self-test as follows:

   a. While pressing the asterisk (*) and pound [hash in Europe] (#) buttons, unplug and replace the phone line cord.

   b. Release the buttons. The phone rings momentarily. (Display phones show a dark display – all pixels lit.)

   c. Lift and replace the handset. The phone rings momentarily. (Display phones show a blank display – no pixels lit.)

   d. Lift and replace the handset. The phone rings momentarily and all LED-equipped buttons light. (Display phones show DKTS SELF TEST LED MATRIX.) If any of the LEDs do not light, return the phone for repair.

   e. Lift and replace the handset. The phone rings momentarily and all of the LEDs go out. (Display phones show DKTS SELF TEST KEY MATRIX 00.)

   f. One at a time, press each of the buttons on the phone in any order. If the button is functioning properly, a key tone (or a DTMF tone for Executive and Professional keypad buttons) is heard and, on display phones, a specific code is shown. (The proper code for each button is diagrammed in Figure 3-84 on page 3-185.) If the signal is not heard or the displayed code is incorrect, the button is faulty; return the phone for repair.

   g. Lift and replace the handset. The speaker tones are broadcast over the speakerphone speaker. (Display phones show DKTS SELF TEST SPEAKER TONES.)

   h. Lift and replace the handset. Tones of various volume levels, from softest to loudest, are broadcast over the speakerphone speaker. (Display phones show DKTS SELF TEST SPEAKER VOLUME.)

   i. Lift the handset, then press and release the hookswitch. The handset tones are broadcast over the handset receiver. (Display phones show DKTS SELF TEST HANDSET TONES.)

   j. Place the handset back in its cradle. The phone rings momentarily and it takes approximately five seconds for the phone to return to normal operation. (Display phones show DKTS SELF TEST COMPLETED! for several seconds. Then, as described in the NOTE to step 4, the phone's identification displays for several seconds.)

NOTE: If the phone watchdog timer fails, the DKTS SELF TEST COMPLETED! display is followed by a DKTS SELF TEST WATCHDOG ERROR display and eight short bursts of ring tone. Replace the defective phone.

k. Replace the phone if faulty.

If installing a Model 8500, 8520, or 8560 Phone, perform the digital phone self-test as follows:

a. While pressing the asterisk (*) and pound [hash] (#) buttons, unplug and replace the phone line cord.

b. Release the buttons. The phone rings momentarily. (Display phones show a dark display – all pixels lit. If all pixels are not lit, return the phone.) The LEDs should not light. If an LED is lit, the button may be stuck.

c. Lift and replace the handset. The phone sounds an alerting tone. If no tone is heard, return the phone. (Display phones show a blank display – no pixels lit. If any pixel is lit, return the phone.)

d. Lift and replace the handset. The phone rings momentarily. (Display phones show a display filled with Os.)

e. Lift and replace the handset. The phone rings momentarily. (Display phones show a display filled with Ms.)

f. Lift and replace the handset. The phone rings momentarily, and all red, LED-equipped buttons light. (Display phones show TESTING LED MATRIX.) If any of the LEDs do not light, return the phone for repair.

g. Lift and replace the handset. The phone rings momentarily, and all green, LED-equipped buttons light. (Display phones show TESTING LED MATRIX.) If any of the LEDs do not light, return the phone for repair.

h. Lift and replace the handset. The phone rings momentarily. (Display phones show TESTING KEY MATRIX.)

i. One at a time, press each of the buttons on the phone in any order. If the button is functioning properly, a key tone is heard, the LED lights red, and a specific code is shown on the display. (The proper code for each button is diagrammed in Figure 3-85 on page 3-186.) If the signal is not heard or the displayed code is incorrect, the button is faulty; return the phone for repair.

j. Lift and replace the handset. The speaker tones are broadcast over the speaker-phone speaker. (Display phones show TESTING SPEAKER TONES.)

k. Lift and replace the handset. Tones of various volume levels, from softest to loudest, are broadcast over the speakerphone speaker. (Display phones show TESTING SPEAKER VOLUME.)

l. Lift the handset, then press and release the hookswitch. The handset tones are broadcast over the handset receiver. (Display phones show TESTING HANDSET TONES.)

m. Put on the headset *(if one is attached)* and place the handset back in its cradle. The headset tones are broadcast over the headset. (Display phones show TESTING HEADSET TONES.)

n. Lift and replace the handset. The phone rings momentarily, and it takes approximately five seconds for the phone to return to normal operation. Display phones show TESTNG COMPLETED! for several seconds.)

o. Replace the phone if faulty.

NOTE: If the phone watchdog timer fails, the TESTING COMPLETED! display is followed by a WATCHDOG ERROR display and eight short bursts of ring tone. Replace the defective phone.

6. The phone ring tone can be changed by performing the steps described in the FEATURES section on page 5-145.

7. *If necessary,* replace the phone's paper template with one labeled to match the programmed feature buttons.

NOTE: The Model 8000 digital phones allow you to adjust the viewing angle. If you position the phone's viewing angle to 60° or greater, flip the handset hanger to the horizontal position.

Figure 3-84. *Digital Display and Basic Digital Phones Self-Test Key Matrix*

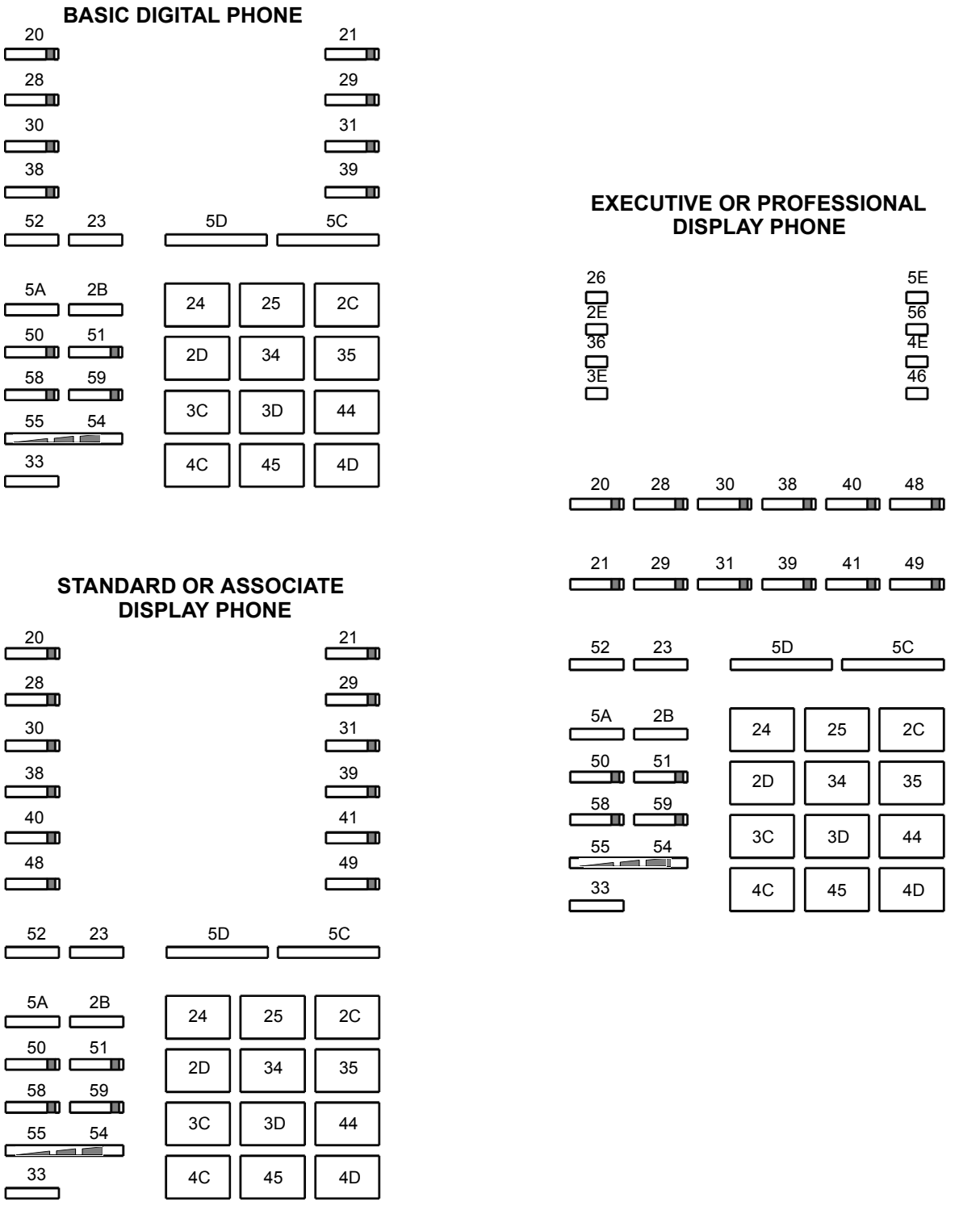
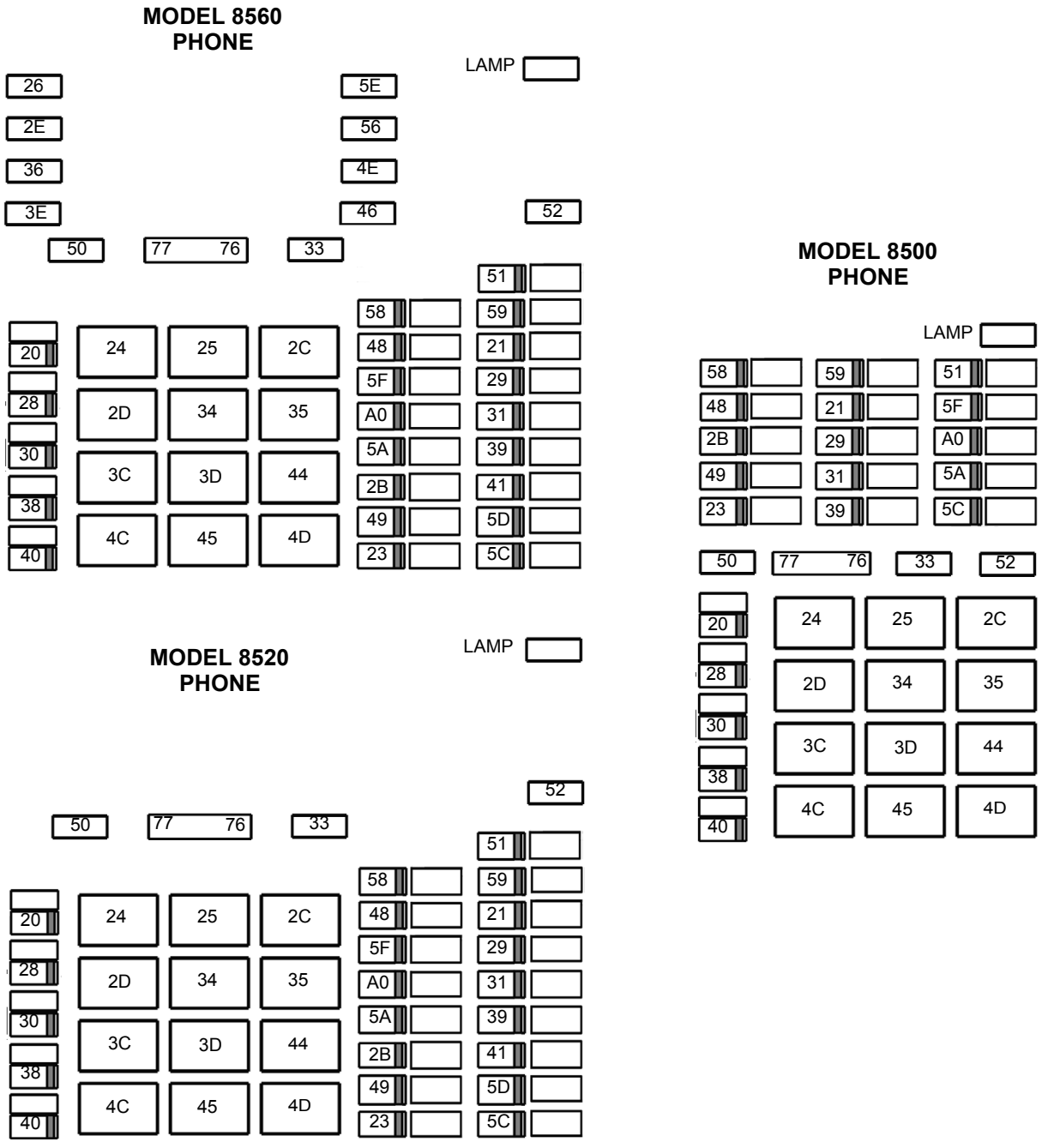


Figure 3-85. *Model 8500, 8520, and 8560 Phones Self-Test Key Matrix*



Adjustable Baseplate

13.4 The Professional, Associate, and Eclipse Basic Phones have adjustable baseplates that allow the users to set the angle of the phone. To change the baseplate position:

1. Gently press on one edge of the baseplate and lift it off the phone.
2. If you have a PCDPM installed, route the cables so that they are under the baseplate.
3. Position the baseplate over one of the three sets of holes and snap it into place. The curved edge should be toward the top of the keyset.

Wall Mounting Digital Display and Basic Digital Phones

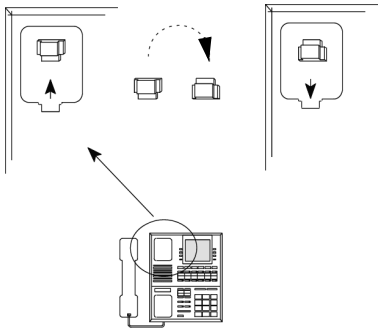
13.5 To mount the phone on a wall, use an industry-standard wall-mount bracket (sometimes called a wall jack assembly).

13.6 If a wall-mount bracket is not available, use the two-screw method outlined below:

1. Remove the phone baseplate from the top half of the back of the phone.
2. Rotate the baseplate 180° and attach it to the bottom half of the back of the phone.

> **NOTE:** If a PCDPM is already installed inside the baseplate, first open the phone and reroute the PCDPM-to-keyset interface cable through the knockout in the bottom half of the phone (see Figure 3-87 on page 3-200).

3. Using a ruler, mark the location of the phone mounting holes on the wall. (The centers of the two perpendicular mounting holes on the back of the phone are four inches apart.)
4. Drive a screw into the center of each mounting hole marking. Allow the head of the screw to protrude approximately ¼ inch.
5. To hold the handset in place while the phone is wall-mounted, slide the reversible cradle hook out, rotate it 180°, and reinsert it as shown at right.
6. Position the phone mounting holes over the screws and slide the phone into position on the wall. (If necessary, adjust the screws to ensure the phone is held firmly in place.)



Wall Mounting Model 8500, 8520, and 8560 Phones

NOTE: Wall mounting the Model 8000 digital phones is not recommended if the user is connecting and disconnecting cords (handset, headset, line) and/or cables (Ethernet, USB, power supply) frequently. To connect and disconnect these cords and cables from a wall-mounted phone, the user needs to remove the phone from the wall.

13.7 Because there are two types of bases used on the Model 8000 digital phones, two sets of instructions for wall mounting the phones are included here. Determine which type of base the phone has and follow the appropriate instructions.

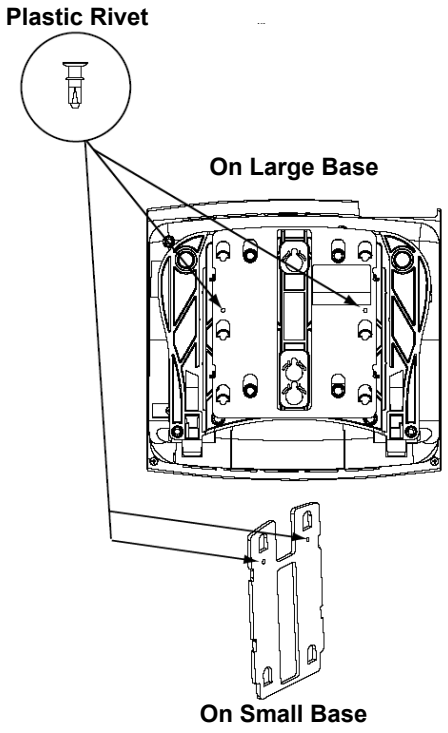
13.8 For best results when wall mounting the Model 8000 digital phones, Inter-Tel strongly recommends that you use the two plastic rivets shipped with the phone. After installed, the plastic rivets lock the components into place, providing additional stability for wall-mounted phones.

13.9 Model 8000 digital phones are shipped with a plastic adhesive label that covers the rectangular wall-mount knockout on the base of the phone. A portion of the adhesive label must be removed to use the wall-mount knockout.

13.10 To wall mount the Model 8000 digital phones, you need an industry-standard wall-mount bracket (sometimes called a wall jack assembly) and a four-inch line cord. Both of these items are available through the Inter-Tel CommSource™ division or Inter-Tel Europe.

13.11 To prepare the phone for wall mounting (regardless of base type):

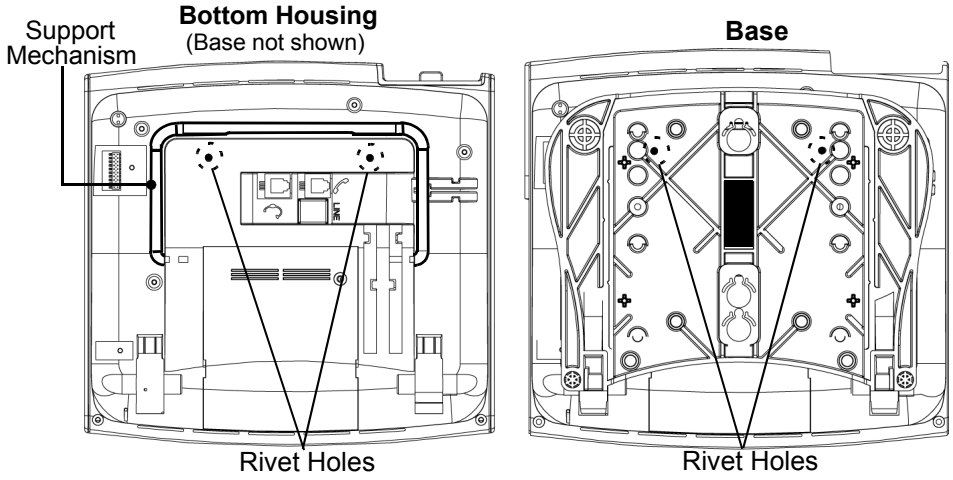
1. Remove the metal plate from the base of the phone by removing the two plastic rivets to unlock it from the back.
2. Slide the cover off. The two plastic rivets are not included on phones built prior to May 2005.
3. Use a sharp blade to cut out the rectangular portion of the adhesive label covering the wall-mount knockout (see diagram at right).



13.12 To wall mount the Model 8000 digital phones that have a support mechanism:

NOTE: The following instructions apply to phones that have a support mechanism (see illustration below) that fits into holes in the base.

1. Collapse the support mechanism completely and secure it against the bottom housing of the phone, as shown below.

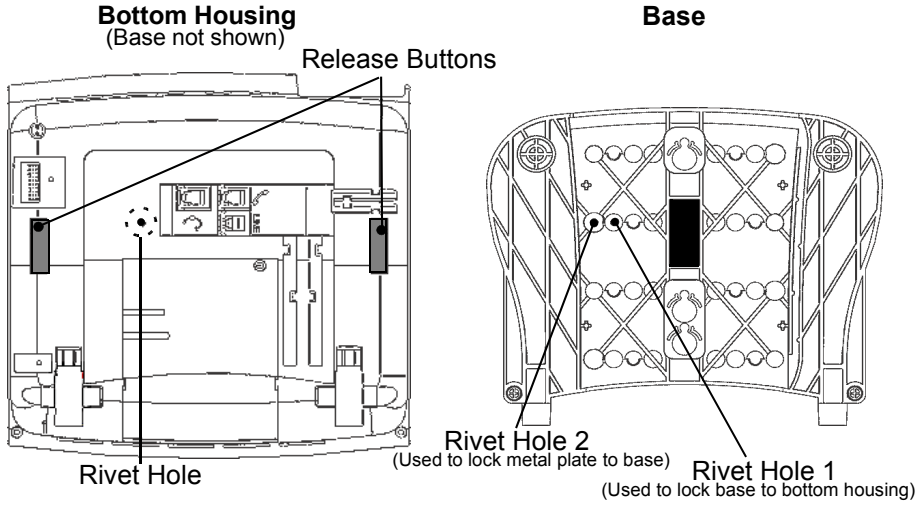


2. Close the base completely so that it lies flat against the phone. (The two rivet holes on the base align with the two rivet holes on the bottom housing of the phone.)
3. Plug the four-inch line cord into the jack on the back of the phone and thread the line cord through the wall-mount knockout on the base.
4. Insert the two plastic rivets through the base and press them firmly into the holes on the bottom housing of the phone. This locks the base to the phone.
5. Test the base to verify that it is securely locked to the phone.
6. Re-attach the metal plate to the phone base.
7. Plug the line cord into the jack on the wall-mount bracket and position the phone securely on the wall-mount bracket.
8. Flip the handset hanger down to the horizontal position and lock it into place. This holds the handset in place.

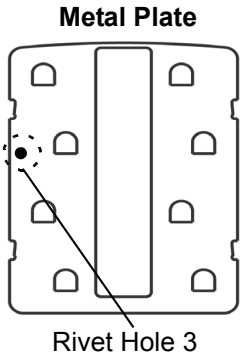
13.13 To wall mount the Model 8000 digital phones that have release buttons located on the bottom housing:

NOTE: The following instructions apply to phones that have release buttons located on the bottom housing of the phone. To disengage the locking mechanism in the base, push in both buttons. To lock the base in position, release the buttons.

1. Close the base completely so that it lies flat against the phone. (Rivet hole 1 on the base aligns with the rivet hole on the bottom housing of the phone.)



2. Plug the four-inch line cord into the jack on the back of the phone and thread the line cord through the wall-mount knockout on the base.
3. Insert one of the plastic rivets in rivet hole 1 on the base (see illustration above) and press it into the rivet hole on the phone. This locks the base to the phone.
4. Test the base to verify that it is securely locked to the phone.
5. Re-attach the metal plate to the phone base.
6. Insert the two plastic rivets through the metal plate and press them firmly into the holes on the base of the endpoint. This locks the metal plate to the base.
7. Insert the remaining plastic rivet through rivet hole 3 (as shown at right) in the metal plate and press it firmly into rivet hole 2 in the base. This locks the metal plate to the base.



8. Plug the line cord into the jack on the wall-mount bracket and position the phone securely on the wall-mount bracket.
9. Flip the handset hanger down to the horizontal position and lock it into place. This holds the handset in place.

Optional Headsets

13.14 See SPECIFICATIONS, page 2-75, for headset requirements.

13.15 To attach a headset to an Executive Display, Professional Display, Standard Display, Associate Display, Axxess Basic Digital, Eclipse Basic Digital, or Model 8500 Phone:

1. Remove the coiled handset cord from the handset jack on the bottom of the phone. Leave the handset in the cradle.
2. Insert the headset modular plug into the jack. Plug in the headset power source, if used.
3. If the headset has an ON/OFF switch, turn ON the headset.
4. On the phone, press the Special button and enter the Headset Enable feature code (default value is 315).
5. *To disable the headset*, press the Special button and enter the Headset Disable feature code (default value is 316). Then unplug the headset and reconnect the handset.

13.16 To attach a headset to a Model 8520 or 8560 Phone:

1. Insert the headset modular plug into the jack located underneath the phone. Plug in the headset power source, if used.
2. If the headset has an ON/OFF switch, turn ON the headset.
3. On the phone, press ∞ and enter the Headset Enable feature code (default value is 315).
4. *To disable the headset*, press ∞ and enter the Headset Disable feature code (default value is 316). Then unplug the headset and reconnect the handset.

Optional Personal Computer Data Port Modules (PCDPMs)

13.17 For off-hook voice announce receive capability, and to allow an optional DSS/BLF Unit and/or an optional Modem Data Port Module to be installed, any Standard Display, Associate Display, Executive Display, Professional Display, Model 8520, or Model 8560 Phone may be equipped with an optional PC Data Port Module. For the Executive Display and Standard Display Phones, PCDPM part no. 550.3018 [PB2585 in Europe] or 550.3014 [PB2548-A] is required. For the Professional Display and Associate Display Phones, PCDPM part no. 550.3018 is required. For the Model 8520 and 8560 Phones, the PCDPM required is part no. 550.8400.

13.18 The original model PCDPM – part no. 550.3014 [PB2548-A in Europe] for the Executive Display and Standard Display Phones has been replaced with a new model – part no. 550.3018 [PB2585]. The new PCDPM (part no. 550.3018) and the Model 8520/8560 PCDPM (part no. 550.8400) do not require an external AC transformer (see Figure 2-7 on page 2-78 for a diagram). Refer to the revised installation instructions included with the new unit. If a DSS/BLF Unit is not installed on the module's RS232 port, the port can be used to connect an SMDR/error message output device or a personal computer. See page 3-191 for instructions on installing the optional PC Data Port Module.

NOTE: RS232 has undergone several revisions over the last 30 years. The most current version of RS232 is EIA/TIA-232-E (July 1991), which includes the alternate use of RTS and CTS as character-by-character hardware flow control signals. The Inter-Tel system serial ports conform to EIA/TIA-232-E. Older versions of RS232, such as EIA-232-D (1987) and RS232-C (1969) are a subset of EIA/TIA-232-E. Hence, serial ports which conform to EIA/TIA-232-E are compatible with RS232-C serial ports. However, if the RS232-C serial port does not support RTS/CTS flow control, then the connection cannot use RTS/CTS hardware flow control.

13.19 PCDPMs can be used for a variety of purposes, including:

- Off-hook voice announce (OHVA) receive capability
- Optional Modem Data Port Module (MDPM) connection
- Optional DSS/BLF Unit connection
- CTI applications requiring a desktop OAI connection
- Low-speed (4800 bps or slower) RS232 connection to a PC for system serial port applications, such as SMDR and message print.

NOTE: For all applications requiring RS232 connections, the PCDPM supports only hardware flow control; it does not support software flow control.

13.20 The following three areas must be addressed to properly install and configure the PCDPM:

- Hardware Installation (see below)
- System DB Programming (see page 3-196)
- RS232 Setup and Configuration (see page 2-76)

13.21 Install the PC Data Port Module for an Executive Display or Standard Display Phone as outlined below. For diagrams of the installation, see Figure 3-86 on page 3-199 and Figure 3-87 on page 3-200.

1. Unplug the phone line cord and the handset from the modular jack (line jack unit) on the back of the phone.
2. Remove the phone baseplate.
3. On the back of the phone, use a pair of cutting pliers to remove the plastic knockout covering the PC Data Port Module connector (J3) on the phone control board (see page 3-199).

   NOTE: Be careful not to let the plastic knockout fall inside the phone. If the phone is wall mounted, use the knockout in the bottom half of the phone instead (see Figure 3-86).

   Some of the newer-style digital phones equipped with both red and green LEDs do not have a plastic knockout in the upper portion of the back of the phone to allow easy connection to the phone control board. Instead, you must open the back of the phone (secured by four screws) and route the cable through the plastic knockout in the lower portion of the back housing, just as you would if you were wall-mounting the phone.
4. Align the PC Data Port Module over the four posts protruding from the inside of the baseplate and insert the screws (do not over tighten).

5. *If not already installed*, attach one end of the PCDPM-to-keyset interface cable to the connector on the back of the PCDPM.

NOTE: ***If installing the PCDPM on an older-style digital phone equipped with red LEDs only,*** use the appropriate end of the new PCDPM to ***old*** keyset interface cable (part no. 813.1633). Note that the connector is keyed so that it can only be installed one way.

If installing the PCDPM on one of the newer-style digital phones equipped with both red and green LEDs, use the appropriate end of the new PCDPM to ***new*** keyset interface cable (part no. 813.1632). Pin 1 of the cable (indicated by the stripe on the edge of the cable) must correspond to pin 1 on the KEYSET connector (indicated by the silkscreen on the PCDPM). Note that the connector is keyed so that it can only be installed one way.

6. Plug the free end of the PCDPM-to-keyset interface cable into connector J3 on the phone control board (where the plastic knockout was previously removed).

NOTE: Use a pencil (eraser end) or other suitable rod-like instrument to press each end of the cable connector into connector J3. Apply pressure, one end at a time, until the connector clicks securely into place.

7. Depending on the PCDPM's intended use, do one of the following:

- **To connect to a DSS/BLF:**

  Attach one end of the DSS-to-DSS interface cable (part no. 813.1519; included with the DSS/BLF Unit) to the RS-232 connector (J2) on the PCDPM (see Figure 3-86). (The other end of the cable is attached to the DSS/BLF Unit after it is installed.) If using an older PCDPM (part no. 550.3014), plug the main AC transformer unit into an available AC power source. DO NOT use the outlet for the equipment chassis. (Newer PCDPMs do not require an external AC power source.)

  NOTE: Pin 1 of the cable (indicated by the arrow on the connector) must correspond to pin 1 on the RS-232 connector (indicated by the silkscreen on the PCDPM). Also, note that the PCDPM-to-DSS interface cable (part no. 813.1595; included with the DSS/BLF Unit) is not used with the new model PCDPM; use the DSS-to-DSS interface cable instead.

- **To connect to a PC/output device:**

  Attach the appropriate end of the new PCDPM-to-RS-232 interface cable (part no. 813.1631) to the RS-232 connector (J2) on the PCDPM. (The other end of the cable is attached to the PC or output device after it is installed.) For complete RS-232 cable pinout requirements, see the information beginning on page 2-99.

  NOTE: Pin 1 of the cable (indicated by the arrow on the connector) must correspond to pin 1 on the RS-232 connector (indicated by the silkscreen on the PCDPM).

- **To connect to an MDPM:**

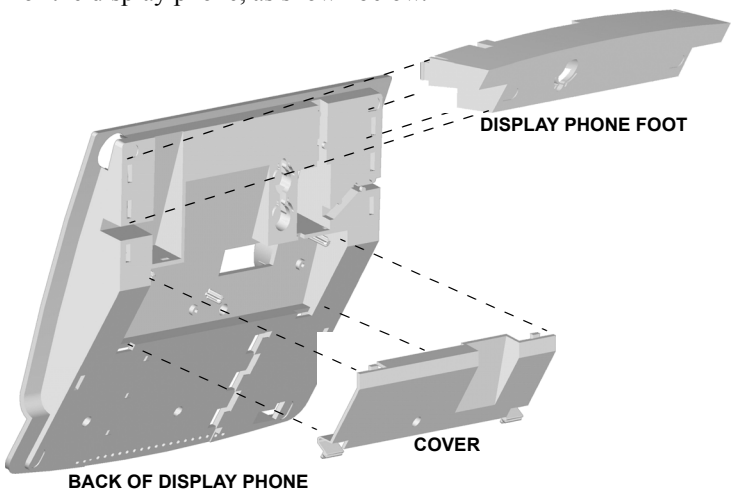
  Plug the appropriate end of the PCDPM-to-MDPM interface cable (part no. 813.1566; included with the MDPM) into the MDPM connector (J1) on the PCDPM. (The other end of the cable is attached to the MDPM after it is installed.)

NOTE: Pin 1 of the cable (indicated by the stripe on the edge of the cable) must correspond to pin 1 on the MDPM connector (indicated by the notch silkscreened on the PCDPM). Some older PCDPM-to-MDPM cables may have the stripe on the wrong edge. If the MDPM doesn't respond properly, try reversing the connector on the PCDPM.

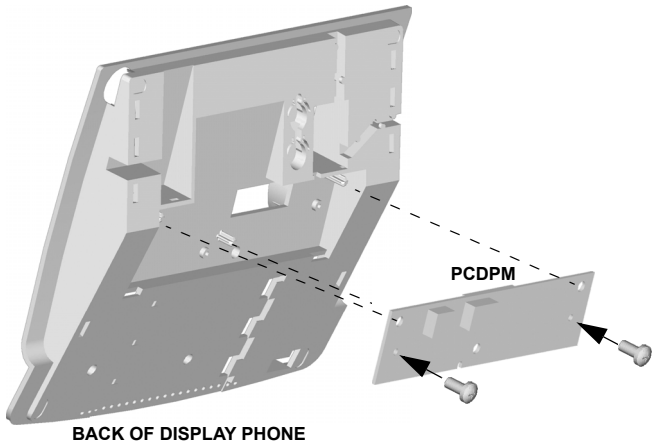
8. Carefully thread the cables connected to the PCDPM through the slots on the access panel. Re-attach the baseplate and reconnect the phone line cord.

13.22 Install the PC Data Port Module for a Professional Display or Associate Display Phone as outlined below.

1. Press up on the tabs and lift off the cover. Remove the cover and the display phone foot in the back of the display phone, as shown below.



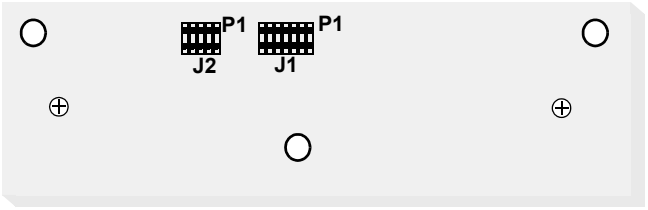
2. Align the PC Data Port Module over the connector inside of the display phone and the three pins on the outside of the display phone. Gently press the module into place until the connector is securely positioned into place, as shown on the next page.
3. Attach the module to the display phone using the two mounting screws.



4. Depending on the PCDPM's intended use, do one of the following:

   - **To connect to a PC/output device:**

     Attach one end of a PCDPM-to-DSS interface cable (part no. 813.1519) or PCDPM-to-RS232 interface cable (part no. 813.1631) to the RS-232 connector (J2) on the module, as shown below. (The other end of the cable is attached later to the 60-key DSS/BLF Console or PC/output device.) For complete RS232 cable pinout requirements, see page B-5.



     NOTE: Pin 1 of the cable (indicated by the stripe on the edge of the cable) must correspond to pin 1 on the RS-232 connector (indicated by the notch silk-screened on the module).

   - **To connect to an MDPM:**

     Plug the appropriate end of the PCDPM-to-MDPM interface cable (part no. 813.1566) into the MDPM connector (J1), shown above, on the module. (The other end of the cable is later attached to the MDPM after it is installed.)

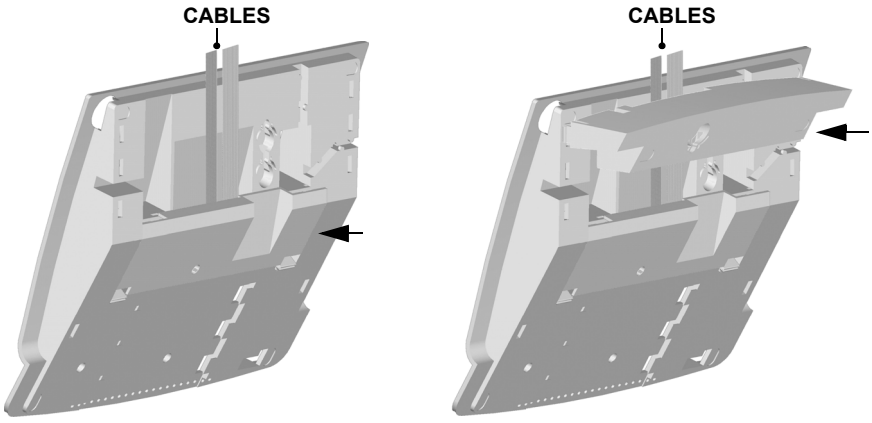
     NOTE: Pin 1 of the cable (indicated by the stripe on the edge of the cable) must correspond to pin 1 on the MDPM connector (indicated by the notch silk-screened on the module).

   - **To connect to a DSS/BLF Unit:**

     See page 3-205.

5. Carefully thread the cables connected to the PCDPM through the slots on the access panel, and re-attach the access panel to the phone. The cables should exit through the slot between the cover and the bottom of the display phone, as shown on the next page on the left.

6. Place the cables flat against the phone and carefully place the foot over the cables, as shown below on the right. For sample display phone PCDPM cable connections, see Figure 3-88 on page 3-201.



13.23 Install the PC Data Port Module for a Model 8520 or 8560 Phone as outlined below. For diagrams of the installation, see Figure 3-89 on page 3-202 and Figure 3-90 on page 3-203.

1. Unplug the line cord and headset from the modular jacks on the back of the phone.
2. Open the phone's base to its fully-extended position.
3. Gently slide the access panel off the phone to expose the PCDPM pin connector (J10) on the phone's control board.
4. Align the PCDPM's J3 connector with the board's J10 connector and plug the PCDPM into place on the phone's board (see page 3-202).
5. *If a DSS/BLF Unit (or a PC/output device) is attached to the PC Data Port Module,* attach one end of a PCDPM-to-DSS interface cable (or PCDPM-to-RS232 interface cable) to the 10-pin (J2) connector on the module, as shown in Figure 3-89. (The other end of the cable is later attached to the DSS/BLF Unit [or PC/output device] after it is installed.) For complete RS232 cable pinout requirements, see page 2-99.

   NOTE: If the correct end of the cable is not connected into J2, the attached device does not function properly. Make sure that the notched side of the cable's plastic connector is facing away from the PCDPM board and that the smooth side lies flat against it.

   If a Modem Data Port Module is attached to the PC Data Port Module, plug the appropriate end of the PCDPM-to-MDPM interface cable into the 14-pin (J1) MDPM connector on the PCDPM. (The other end of the cable is later attached to the MDPM after it is installed.)

   NOTE: If the correct end of the cable is not connected into J1, the attached device does not function properly. Make sure that the notched side of the cable's plastic connector faces away from the PCDPM board, and the smooth side lies flat against it.

6. Carefully thread the cables connected to the PCDPM through the slots on the access panel. Re-attach the access panel to the phone.
7. Secure the access panel to the PCDPM and the phone using the mounting screw supplied in the PCDPM kit.

8. *If a Modem Data Port Module is attached to the PC Data Port Module,* see the instructions on page 3-204.

   *If a Modem Data Port Module is **not** attached to the PC Data Port Module*, reconnect the line cord.

13.24 After the PCDPM hardware is installed, the telephone system database must be properly programmed for the PCDPM's intended purpose. Make sure the following areas in DB Programming are configured properly for the desired PCDPM application.

- **System Configuration**: In the Digital Keyset Card Configuration window, select the appropriate Port Type for the port assigned to the PCDPM.
  - — Select **Keyset** if you plan on using the PCDPM as a DSS/BLF Unit connection, Desktop Interface application, or for Off-Hook Voice Announce receive capability.
  - — Select **Keyset/MDPM** if you plan on attaching an MDPM to the PCDPM, and you still want to use the PCDPM for a DSS/BLF Unit connection, Desktop Interface application, or for Off-Hook Voice Announce receive capability.
  - — Select **Keyset/System Serial** if you do not plan on attaching an MDPM to the PCDPM, and you want to use the PCDPM for SMDR, Message Print, or a system serial port.
  - — Select **Keyset/MDPM/System Serial** if you plan on attaching an MDPM to the PCDPM, and you want to use the PCDPM for SMDR, Message Print, or a system serial port.
  - — Select **Keyset/MDPM** to use phantom extensions with or without an MDPM.

    NOTE: It is not necessary to install a cable phantom extension plug or attach an MDPM to the PCDPM when creating a phantom extension.

- **Premium Feature Configuration**: If using the PCDPM as an RS232 port for a CTI application that uses the Desktop OAI interface, make sure the Desktop Interface premium feature is enabled.
- **Individual Station Configuration**:
  - — *If attaching a DSS/BLF Unit to the PCDPM or if using the PCDPM as an RS232 port for a CTI application that uses the desktop OAI interface,* select either the DSS or Desktop Interface for the Attached Devices field in the Miscellaneous Port Information window.

    NOTE: The Attached Device Bit Rate field is automatically set to 2400 bps for the DSS selection and cannot be changed. The Bit Rate field is adjustable for the Desktop Interface selection, and, unless a slower speed is required, 4800 bps is recommended for the fastest performance.

  - — *If you want to allow the station with the attached PCDPM to have the ability to receive Off-Hook Voice Announce (OHVA) calls,* enable the OHVA Receive option in the Miscellaneous Flags window. (The system flag for OHVA must also be enabled.)
  - — *If attaching an MDPM to the PCDPM,* be sure to configure the MDPM as a single-line port.

- **System Serial Port Configuration**: If using the PCDPM as a system serial port (e.g., for SMDR or message print), program the Keyset Serial Port with the desired Bit Rate entry. Besides configuring the system serial port Bit Rate field, it may also be necessary to program the system to route the desired output to this port via the system serial port assignment.

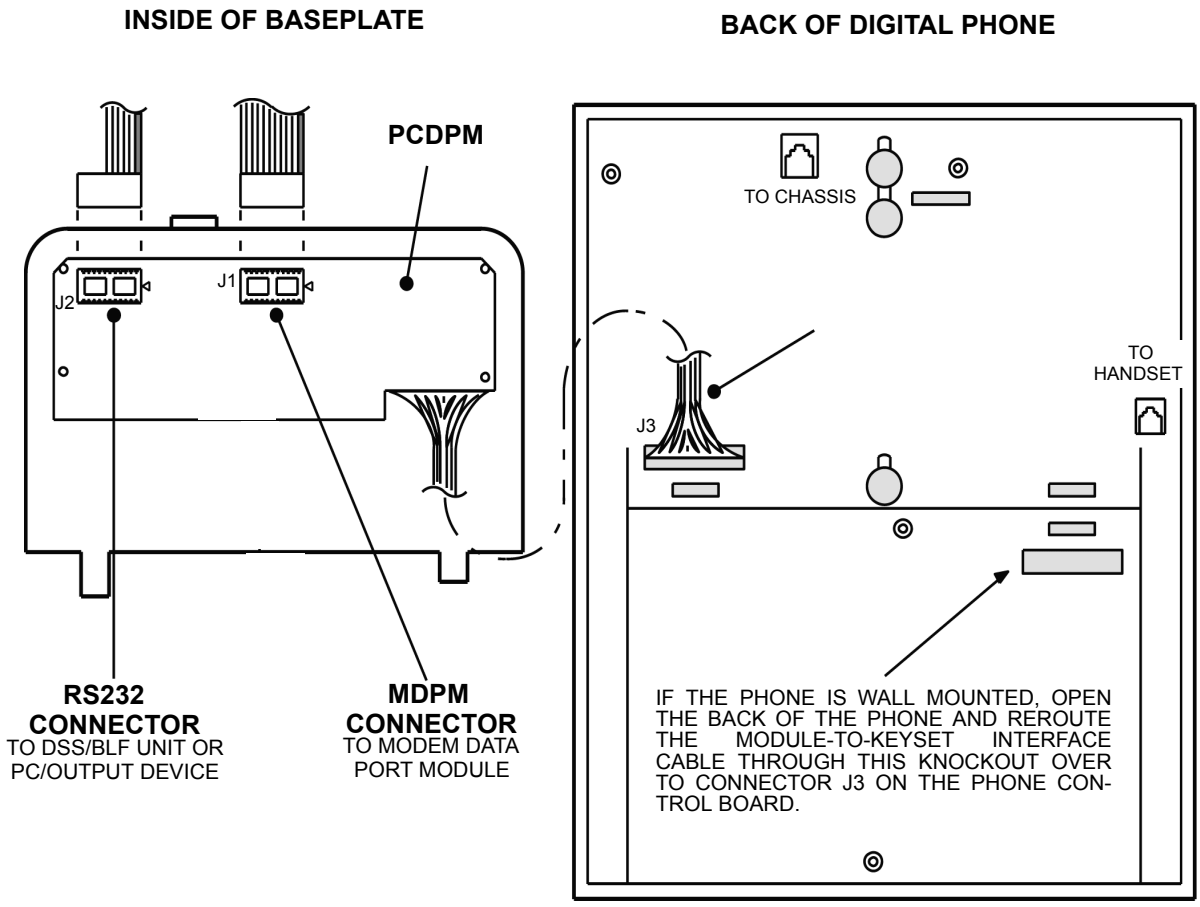
NOTE: The 4800 bps Bit Rate selection is generally recommended for optimal performance. However, depending on the attached device, a slower speed may be necessary. Also note that hardware handshaking is already checked and cannot be changed. By design, all system serial ports routed through a PCDPM always have hardware handshaking enabled and cannot use software handshaking.

CAUTION

Inter-Tel strongly recommends that you do not use a PCDPM system serial port for DB Programming. If flow control is not configured properly, data could be lost, and the system database could become corrupted.

13.25 If the PCDPM uses an RS232 connection (e.g., when using system serial port or desktop OAI applications), see the information on page 2-76 in SPECIFICATIONS to ensure a successful RS232 connection.

Figure 3-86. *Executive or Standard Display PCDPM Installation*



NOTE: Older PCDPMs (part no. 550.3014) require AC transformers when they are connected to DSS/BLF Units only or PC/output devices only. PCDPMs do not require AC transformers when they are used for off-hook voice announce only or have attached MDPMs. For a diagram of AC transformer requirements, see Figure 2-7 on page 2-78. Also, Some early PCDPM-to-MDPM cables may have the stripe on the wrong edge. If the MDPM doesn't respond properly, try reversing the connector on the PCDPM.

Figure 3-87. *Sample Executive/Standard Phone PCDPM Cable Connections*

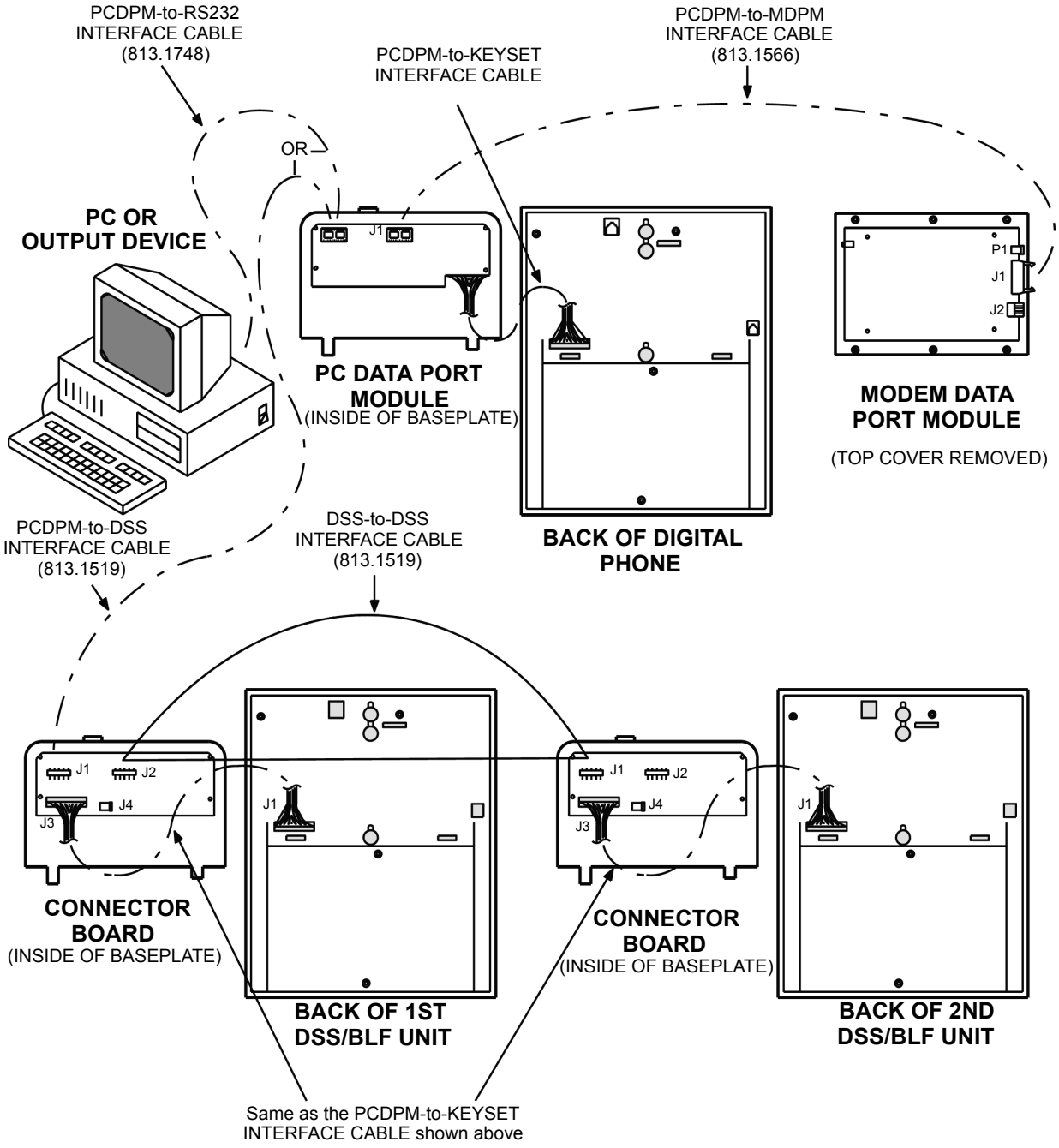


Figure 3-88. *Sample Professional/Associate Phone PCDPM Cable Connections*

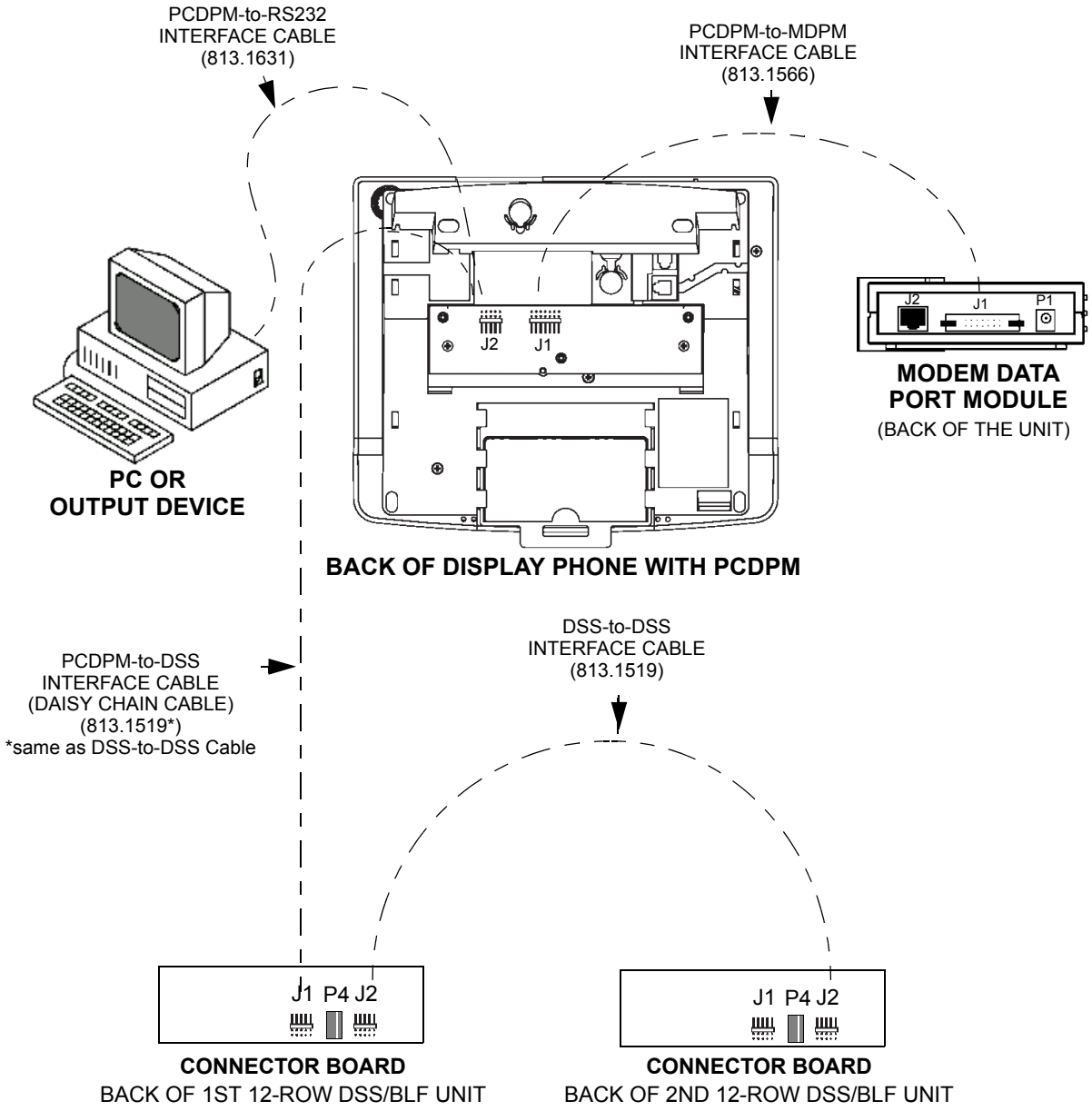


Figure 3-89. *Model 8520 or 8560 Phone PC Data Port Module (PCDPM) Installation*

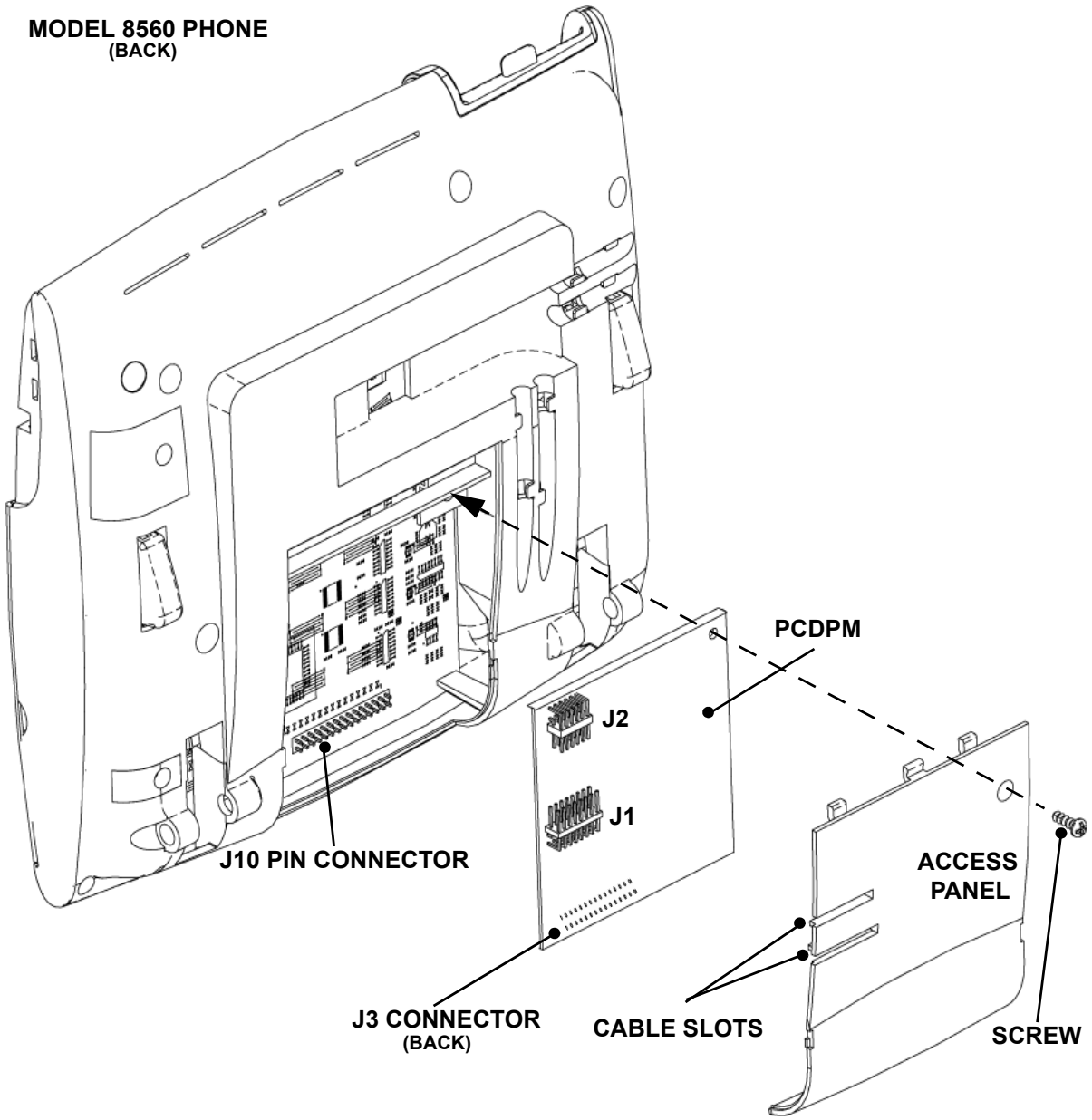
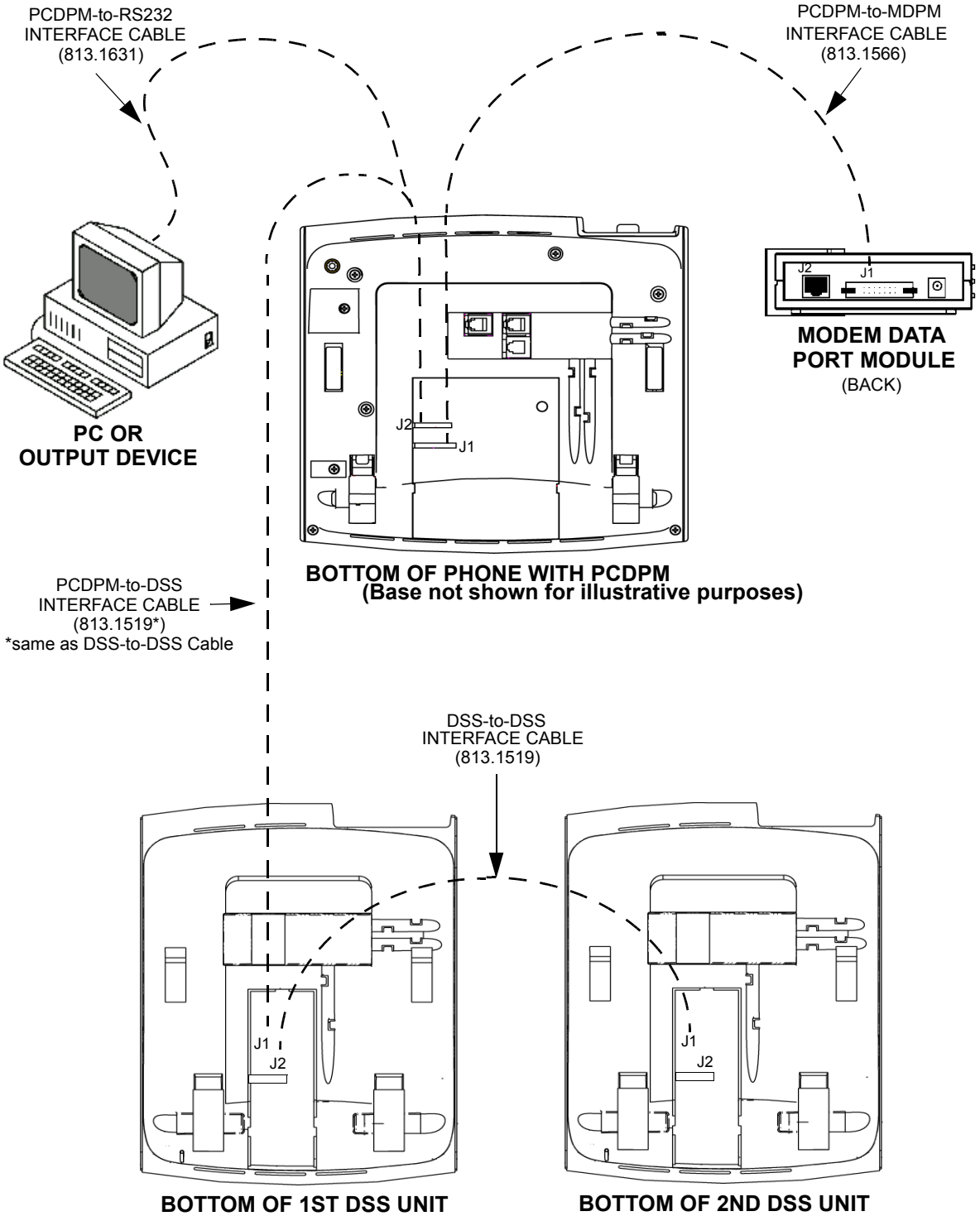


Figure 3-90. *Sample Model 8520 or 8560 Phone PCDPM Cable Connections*



Optional Modem Data Port Modules (MDPMs)

13.26 Digital phones equipped with PC Data Port Modules may be equipped with optional Modem Data Port Modules – part no. 550.3015 [PB2549-A in Europe]. The MDPM contains a four-conductor modular jack (line jack unit) that can be used to connect a data device (such as a personal computer with a direct-connect modem) or a single-line set to the phone.

NOTE: Each "equipped" MDPM installed counts as one single-line device.

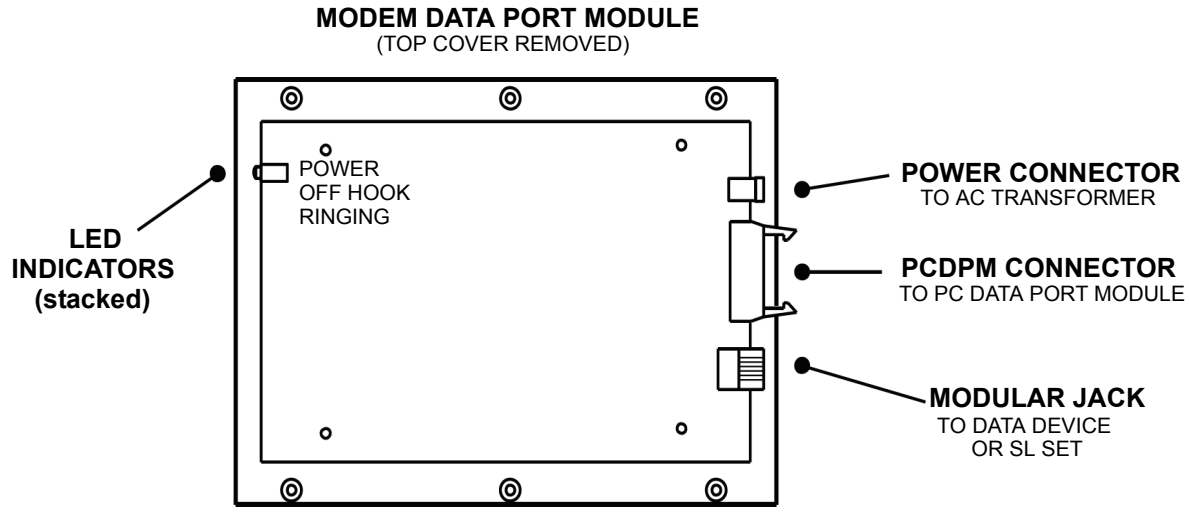
13.27 The data device can be used by the associated phone(s) to communicate with remote data equipment over voice channels being used for CO or intercom calls. The data device's modem must be externally powered (or capable of operating on 20mA of loop current) and have an RJ11 CO trunk interface.

13.28 The data device can be activated by the associated phone(s) or by an auto-dial modem. (For more details, see page 5-269 in FEATURES). The data device is disconnected whenever one of the following occurs: the modem attached to the phone disconnects from the call, the called modem disconnects from the call, or the data connection is transferred to the phone's primary voice path and the call is disconnected.

13.29 Install the MDPM and attach the data device as outlined below. For diagrams of the installation, see Figure 3-87 on page 3-200 and see Figure 3-91 below.

1. Ensure the phone line cord is unplugged from its modular jack (line jack unit) on the phone.
2. Plug the free end of the PCDPM-to-MDPM interface cable into connector J1 on the MDPM. (The other end of the cable was previously attached to the PCDPM; see page 3-191.)
3. Insert the modem line cord (which would normally be connected to a CO jack) into the modular jack (J2) on the MDPM.
4. Plug the main AC transformer unit into an available AC power source. *DO NOT* use the outlet for the equipment chassis.
5. Reconnect the phone line cord. The MDPM can be placed flat or standing on end.

Figure 3-91. *Digital Phone Modem Data Port Module (MDPM) Installation*



B. DSS/BLF UNIT INSTALLATION

NOTE: For information about installing the 48-key DSS/BLF Console for use with an Executone® IDS phone, refer to a version 7.0 or earlier *Eclipse Installation and Maintenance Manual*.

ALSO: Analog DSS/BLF Units cannot be installed on the European system.

13.30 Up to 20 digital Digital Direct Station Selection/Busy Lamp Field (DSS/BLF) Units can be installed on the system. A single phone can be equipped with up to four units.

13.31 There are three types of digital DSS/BLF Units available:

- **60-Key, 6-Row Model**: Used with Executive Display and Standard Display Phones.
- **60-Key, 12-Row Model**: Used with Professional Display and Associate Display Phones.
- **50-Key Model**: Used with Model 8520 and 8560 Phones.

13.32 Each unit provides one-button access to 50 or 60 extension numbers and/or other features/resources, while four units provide access to 200 or 240 extension numbers and/or other features/resources. Although the 60-key, 6-row model is designed for Executive Display or Standard Display Phones and the 50-key model is designed for Model 8520 or Model 8560 Phones, you can use either unit on these phones.

13.33 In addition, Model 8500, 8520, and 8560 Phones support a 16-key Mini-DSS.

13.34 Digital DSS/BLF Units are physically attached to the RS232 ports on phones equipped with PC Data Port Modules (PCDPMs). The units are powered by their own AC transformer and do not require a DKSC/DKSC-16 circuit that is separate from the phone's circuit. If desired, all 20 units can be connected to phones installed on the same Digital Keyset Card.

13.35 Unpack and inspect the DSS/BLF Units before installing them.

6-Row and 50-Key Models

13.36 Install the 6-Row or 50-Key DSS/BLF Unit as outlined below. For diagrams of the installation, see Figure 3-87 on page 3-200 and Figure 3-92 on page 3-207.

1. Remove the baseplate on the back of the DSS/BLF Unit.
2. On the connector board located in the baseplate, plug the free end of the PCDPM-to-DSS interface cable into the main RS232 connector (J1). (The other end of the cable was previously attached to the PC Data Port Module on the corresponding phone.)
3. Plug the AC transformer cable into the power connector (J4) on the baseplate connector board. (The transformer is later plugged into an AC power source.)

   NOTE: DSS/BLF Units always require their own separate AC transformers. For a diagram of AC transformer requirements, see Figure 2-7 on page 2-78.

4. To attach a second DSS/BLF Unit to the first unit:
   a. Plug one end of a DSS-to-DSS interface cable into the secondary RS232 connector (J2) on the baseplate connector board in the first unit.
   b. Remove the baseplate on the back of the second DSS/BLF Unit.

c. Plug the other end of the DSS-to-DSS interface cable into the main RS232 connector (J1) on the baseplate connector board in the second unit.

d. Plug the AC transformer cable into the power connector (J4) on the baseplate connector board in the second unit. (The transformer is later plugged into an AC power source.)

NOTE: If more than one DSS/BLF Unit is installed per phone, each individual unit must be equipped with its own AC transformer. In addition, the PC Data Port Module on the corresponding phone must be equipped with its own AC transformer.

5. To attach a third unit to the second unit, or to attach a fourth unit to third unit, follow the same procedure outlined in step 4.

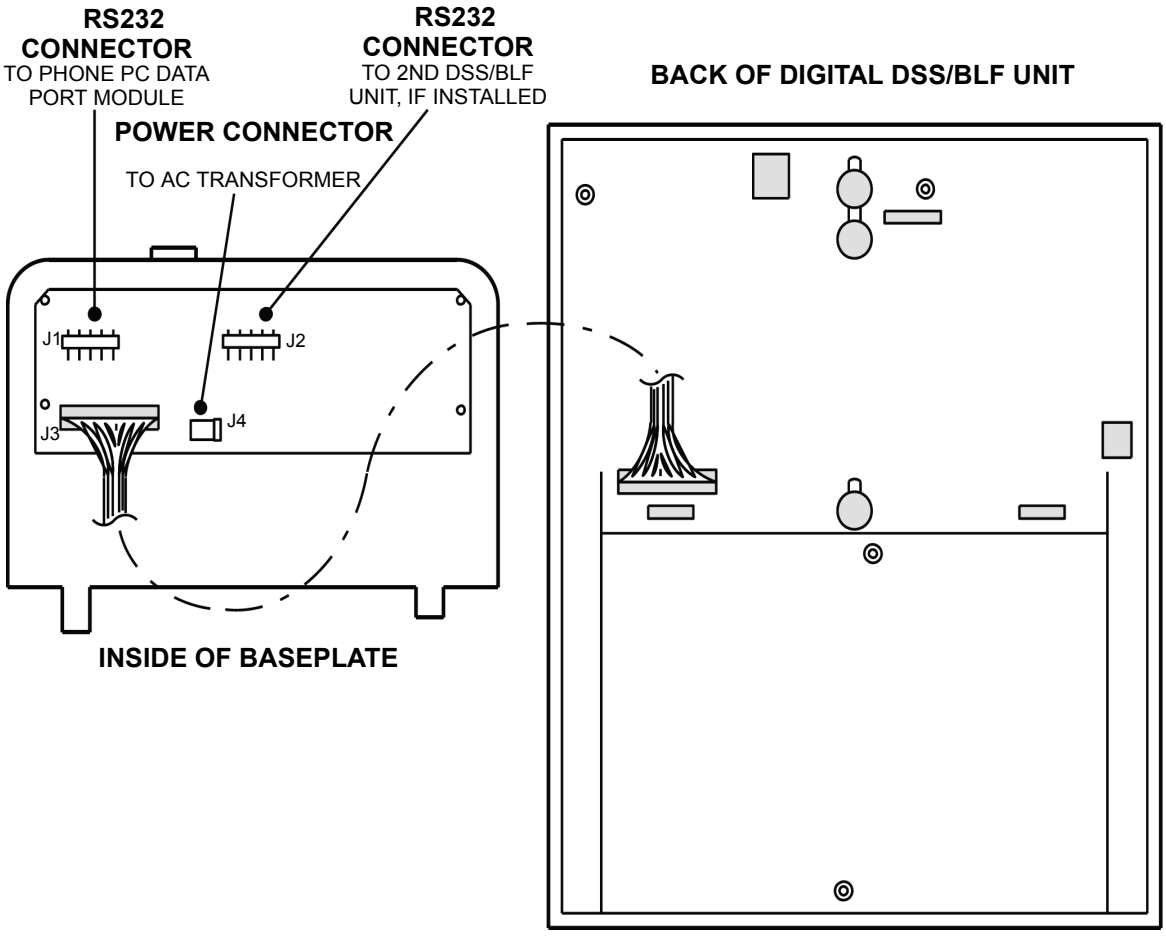
6. Perform the DSS/BLF Unit self-test:

a. While holding down the first two buttons in row one (extension numbers 1000 and 1001 in the default database), plug the main AC transformer unit into an available AC power source. (*DO NOT* use the outlet for the chassis.) All of the buttons should light up. If an LED does not light, unplug the AC transformer unit and plug it in while holding down the first two buttons in row one (ext. 1000 and 1001). If the test fails again, replace the DSS/BLF Unit.

b. Press each button individually (in any order) to remove the lit status. When finished, all buttons should be unlit. If an LED does not go out, unplug the AC transformer unit and repeat steps a and b. If the test fails again, replace the DSS/BLF Unit. After all of the buttons have been tested, it takes approximately five seconds for the DSS/BLF Unit to return to normal operation.

c. Being careful not to pinch any of the attached cables, replace the baseplate on the back of the DSS/BLF Unit.

Figure 3-92. *Back of 60-Key, 6-Row Digital DSS/BLF Unit*



12-Row Models

13.37 Install each 12-row DSS/BLF Unit as outlined below. For diagrams of the installation, see Figure 3-88 on page 3-201.

1. Attach one end of a PCDPM-to-DSS interface cable (part no. 813.1519) into connector J2 on the PCDPM.
2. Remove the baseplate on the back of the DSS/BLF Console.
3. On the connector board located in the baseplate, plug the free end of the PCDPM-to-DSS interface cable (part no. 813.1519) into the main RS232 connector (J1). (The other end of the cable was previously attached to the PC Data Port Module on the corresponding keyset.)
4. Plug the AC transformer cable into the power connector (J4) on the baseplate connector board. (The transformer is later plugged into an AC power source.)

   NOTE: DSS/BLF Units always require their own separate AC transformers. For a diagram of AC transformer requirements, see Figure 2-7 on page 2-78.

5. To attach a second 60-key DSS/BLF Console to the first unit:
   a. Plug one end of a DSS-to-DSS interface cable (part no. 813.1519) into the secondary RS232 connector (J2) on the baseplate connector board in the first unit.
   b. Remove the baseplate on the back of the second DSS/BLF Console.
   c. Plug the other end of the DSS-to-DSS interface cable into the main RS232 connector (J1) on the baseplate connector board in the second unit.
   d. Plug the AC transformer cable into the power connector (J4) on the baseplate connector board in the second unit. (The transformer is later plugged into an AC power source.)

      NOTE: If more than one 60-key DSS/BLF Console is installed per keyset, each individual unit must be equipped with its own AC transformer. In addition, the PC Data Port Module on the corresponding keyset must be equipped with its own AC transformer.

6. To attach a third unit to the second unit, or to attach a fourth unit to third unit, follow the same procedure outlined in step 4.
7. Perform the 60-key DSS/BLF Console self-test:
   a. While holding down the first two keys in row one (extension numbers 1000 and 1001 in the default database), plug the main AC transformer unit into an available AC power source. (*DO NOT* use the outlet for the chassis.) All of the keys should light up. If an LED does not light, unplug the AC transformer unit and plug it in while holding down the first two keys in row one (ext. 1000 and 1001). If the test fails again, replace the DSS/BLF Console.
   b. Press each key individually (in any order) to remove the lit status. When finished, all keys should be unlit. If an LED does not go out, unplug the AC transformer unit and repeat steps a and b. If the test fails again, replace the DSS/BLF Console. After all of the keys have been tested, it takes approximately five seconds for the DSS/BLF Console to return to normal operation.
   c. Being careful not to pinch any of the attached cables, replace the baseplate on the back of the DSS/BLF Console.

Mini-DSS

13.38 The Mini-DSS is an optional unit that provides 16 additional user-programmable buttons and connects directly to the Model 8660, 8560, or 8520 Phones. Because the Mini-DSS does not utilize any of the resources required by a DSS/BLF, a phone can have one Mini-DSS and up to four DSS/BLF Units.

NOTE: Version 7.0 or later software is required to use the Mini-DSS.

13.39 Mini-DSS Units are physically attached to a special cable pin connector on Model 8520 or Model 8560 Phones. Because the Mini-DSS Units draw power from the phone, external AC transformers are not required. In addition, separate circuits on the DKSC-16 or DKSC-16+ are not required.

13.40 Unpack and inspect the Mini-DSS Unit before installing it.

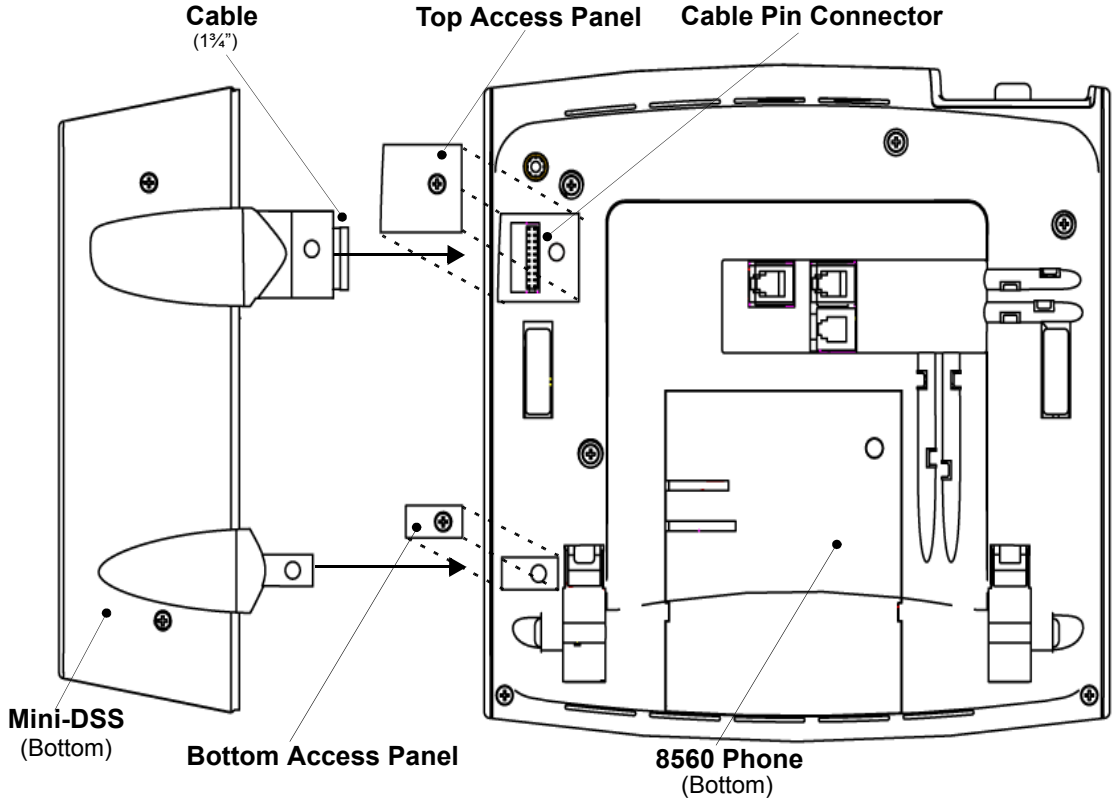
13.41 Install each Mini-DSS Unit as outlined below. For diagrams of the installation, see Figure 3-93 on page 3-210 and Figure 3-94 on page 3-210.

1. Lift and remove the original faceplate using the notch on the faceplate edge of the phone. (The original faceplate can be returned to Inter-Tel for recycling.)
2. Remove the original phone facemats and set them aside for later use.
3. Turn the phone over so that the Mini-DSS access panels located on the bottom of the phone are accessible. (The recess beneath the top access panel houses the pin connector for the Mini-DSS cable.)
4. Remove the screws from the access panels and set them aside for later use.
5. Remove the access panels from the phone. (The phone access panels can be returned to Inter-Tel for recycling.)
6. Turn the Mini-DSS over and pull a section of ribbon cable measuring 1¾ inches (4.5 cm) from the unit.

   NOTE: Avoid removing more cable than the recommended 1¾ inches (4.5 cm). The excess cable may be difficult to replace in the Mini-DSS.

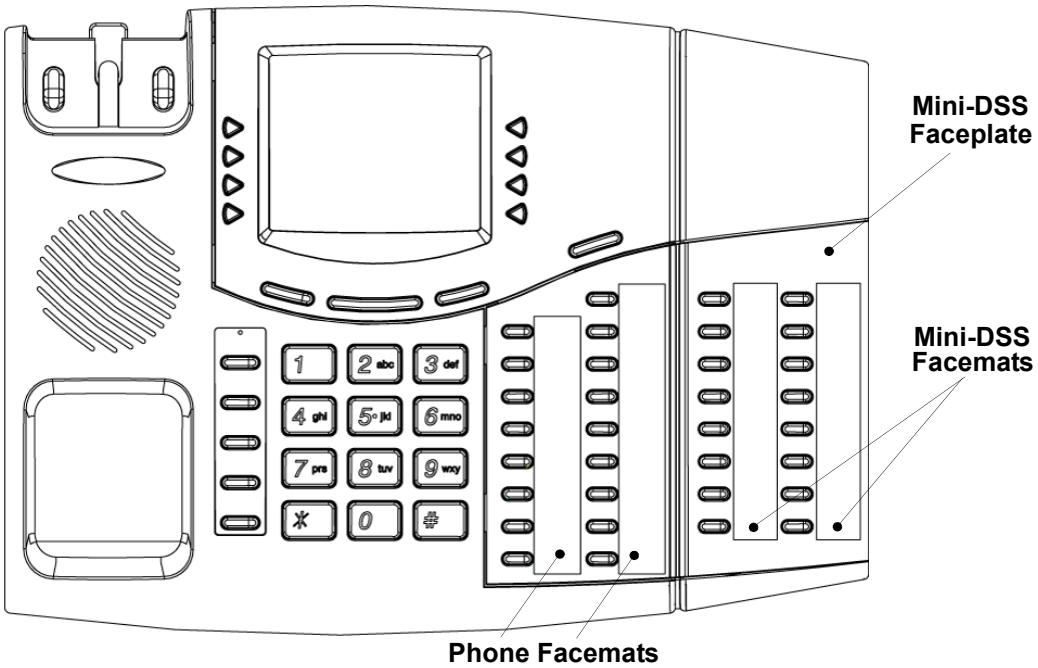
7. Plug the Mini-DSS cable into the pin connector on the phone.
8. Guide the “arms” of the Mini-DSS into the access panel slots (see Figure 3-93).
9. Secure the Mini-DSS unit to the phone by replacing and tightening the screws removed from the access panels in step 5.
10. Turn the phone (and attached Mini-DSS) over. Replace the original facemats on the phone and place the Mini-DSS facemats on the Mini-DSS unit.
11. Snap the Mini-DSS faceplate into place covering the facemats (see Figure 3-94).

Figure 3-93. *Mini-DSS Installation*



NOTE: For illustrative purposes, the base on the phone is not shown.

Figure 3-94. *Model 8560 with Attached Mini-DSS*



C. SINGLE-LINE SET AND PLAYBACK DEVICE INSTALLATION

Single-Line Sets

13.42 Single-line sets are connected to Single-Line Cards (SLCs), or they can be connected to Single-Line Adapters (SLAs), which are in turn connected to DKSC/DKSC-16 circuits. The single-line sets (including OPXs) must be equipped with AC ringers.

NOTE: DC-ringing single-line devices may not be connected to SLCs or SLAs.

13.43 SLAs do not support *visual* message waiting applications. If single-line sets with message waiting lamps are installed on SLAs, the lamps do not function. However, single-line sets connected to SLAs can be configured in DB Programming to receive *audible* message waiting indication tones.

13.44 Each SLC can be equipped with an optional Lamp Option Card (LOC) to provide visual message waiting capabilities for single-line sets equipped with message lamps. Single-line sets connected to SLCs are then configured in DB Programming to receive *visual* and/or *audible* message waiting indication tones.

Playback Devices

NOTE: For automated attendant features and for optimal playback device performance and functionality, use the Inter-Tel Voice Processing PC instead of SLC or SLA circuits as outlined below.

13.45 Playback devices can be used in place of single-line sets on SLC or SLA circuits. These devices answer an incoming call, play a recorded message, and automatically disconnect from the call. They are installed like single-line sets, using the same cabling and modular jack (line jack unit) assemblies. A playback device uses the extension number that is associated with its station circuit.

Single-Line Set and Playback Device Installation

13.46 To install a single-line set or a playback device:

1. *For SLAs only:*
   a. Plug the AC transformer cable into the power connector on the corresponding Single-Line Adapter (SLA).

      NOTE: SLAs always require their own separate AC transformers. For a diagram of AC transformer requirements, see Figure 2-7 on page 2-78.

   b. Plug the main AC transformer unit into an available AC power source. *DO NOT* use the outlet for the equipment chassis.
2. Before connecting a single-line set or playback device to the system, measure the voltage on the RED terminal of the modular jack (line jack unit) assembly with respect to the GREEN terminal. (The common probe of the voltmeter is placed on the GREEN terminal.) It must measure -48VDC (±2VDC). If +48 is measured, check the cabling for a reversed pair.

3. Mount the modular jack assembly on the wall at the station location.
4. Plug one end of the line cord into the modular jack (line jack unit) assembly and plug the other end into the jack on the single-line set or playback device.

Optional Handset Amplifiers

13.47 Single-line set users may want to have a handset amplifier installed. Typically, the amplifier is an external unit that is placed between the single-line set and the handset (see page 2-85 for specifications).

13.48 To install such an amplifier:

1. Unplug the coiled handset cord from the single-line set.
2. Plug the coiled handset cord into the amplifier jack labeled HANDSET.
3. Plug the amplifier line cord (coming from the jack labeled TELEPHONE) into the single-line set handset jack.
4. Plug the amplifier power supply cord into an AC outlet.
5. The handset volume can be increased or decreased, using the thumbwheel located on the amplifier. Verify that the amplifier is functioning correctly by placing a call and adjusting the volume from low to high.

14. VOICE PROCESSING PC

14.1 For integrated voice processing (voice mail, automated attendant, hunt group announcement and overflow, optional Fax-On-Demand, buffered SMDR, Record-A-Call, etc.), an optional Voice Processing PC with special applications software can be attached to the phone system. For more information, see SPECIFICATIONS, page 2-88, and VOICE PROCESSING FEATURES beginning on page 4-1.

NOTE: As of version 8.0, the OS/2 Voice Processing Unit is no longer supported.

CAUTION

Personal computer (PC) technology is continually evolving. As it does, the Inter-Tel Voice Processing PC continues to incorporate important design enhancements. Thus, depending on the actual version of equipment that is shipped, the following installation instructions could vary somewhat. If revised instructions are included with the equipment, refer to the new information instead.

14.2 The Voice Processing PC is required for the Unified Messaging feature and to support the increased voice mail capacities. Currently, the PC runs either Windows NT or Windows 2000. For details about the EM unit, see the *Enterprise Messaging Installation Manual* (document part no. 780.8006).

14.3 The Windows NT-based Voice Processing PC is equipped with the following items:

- Microsoft Windows NT Workstation version 4.0 operating system software (equipped with Service Pack 5)
- Pentium 133 MHz or AMD K6-2 or higher microprocessor (depending on the PC version)
- 32 MB or 64 MB RAM (depending on the PC version)
- 2.1 GB or larger hard disk drive

14.4 The Windows 2000-based Voice Processing PC is equipped with the following items:

- Microsoft Windows 2000 operating system software (equipped with Service Pack 4)
- AMD-K6®-2 300 MHz microprocessor (desktop) or Pentium III 850 MHz microprocessor (rackmount)
- 128 MB RAM
- 30 or 40 GB hard disk drive
- 235 Watt power supply

14.5 In addition, both Voice Processing PCs include the following:

- 3.5-inch (1.44MB) double-sided/high-density floppy disk drive
- CD-ROM
- Network Interface Card
- Standard VGA monitor, keyboard, and mouse

14.6 The Network Interface Card (NIC) is a specially designed network communications processor card that can transfer data packets between the Voice Processing PC and a Local Area Network (LAN).

14.7 The RJ45 connector on the Network Interface Card connects to the LAN via a customer-provided standard network interface cable.

14.8 If configuring an ATM-based system, you should place the Voice Processing Unit close to the first chassis that provides the first eight voice mail ports. The first chassis must also host the Options Card (OPC), which is used for the remaining 32 voice mail ports.

14.9 When connecting the Voice Processing Unit to the first chassis, you can use either an RS232 or TCP/IP connection. It is, however, recommended that you use TCP/IP because TCP/IP is faster. The voice mail should also be configured to match the end-user's Ethernet network.

A. HARDWARE INSTALLATION

14.10 Install the Voice Processing PC as outlined in the following steps. See Figure 3-95 on page 3-219 for sample diagrams of the assembled PC.

NOTE: Like the phone system itself, the Voice Processing PC *must* have its own isolated, dedicated, properly grounded AC circuit for proper operation.

1. Ensure that the Voice Processing PC's AC power switch is turned off and the AC power cord is unplugged.
2. *If the Audio Interface Card (AIC), Voice Processing Cards (VPCs), Video Card, and optional Fax Card are not already installed in the Voice Processing PC*, proceed with the installation as outlined in the following steps.

   If the AIC, VPCs, Video Card, and optional Fax Card are already installed in the Voice Processing PC, skip to step 9.
3. Remove the cover from the PC by removing the screws and sliding the cover off.

CAUTION

The cards and components are static sensitive. Handle the cards by the edges only and keep them in their protective bags until they are to be installed. Do not bend or touch the pins of the components or subject them to a static discharge. When working with the cards, use a properly terminated anti-static wrist strap and cover the work surface with the bag that protected the card during shipping. Any static discharge (no matter how small the charge) must be discharged from the body before touching the cards or components. *The warranty for this equipment does not cover damage caused by static or mishandling. Cards or components damaged in such a manner cannot be replaced.*

4. *If not already installed*, install the Audio Interface Card (AIC) as follows (see Figure 3-97 on page 3-221:
   a. Remove the bracket that covers the opening to the appropriate slot in the PC, and set the bracket and screw aside.

      NOTE: To facilitate connection to the Voice Processing Cards (VPCs), the AIC should be installed so that the VPCs can all be placed to one side of the card. (See Figure 3-95 on page 3-219 for a sample card slot arrangement.)

   b. Ensure that the AIC address selection DIP switches (S1) are set for address $3E0. DIP switches 1 and 2 must be in the ON (closed) position and switches 3-6 must be in the OFF (open) position.

c. Make sure that jumper straps are installed in the following locations on the AIC (see Figure 3-97 on page 3-221 for orientation):

Jumper Pins	Strap Location
J3	Over pins 1 and 2 (as shown on page 3-221)
J5	Over one pin only (or no strap)
J6	Over one pin only (or no strap)

d. Holding the AIC near the PC Motherboard, disconnect the two-pin reset switch cable from the PC Motherboard and attach it to connector J1 on the AIC. (The other end of the cable should remain attached to the reset switch on the PC.)

NOTE: It may be necessary to unbundle the reset switch cable to allow it to reach the AIC.

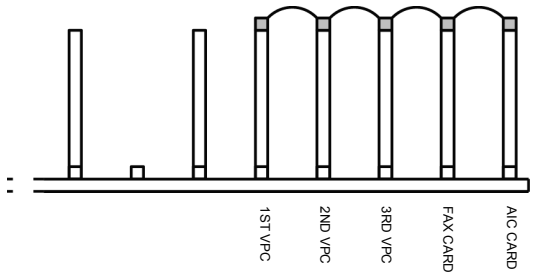
e. Attach one end of the two-pin AIC-to-reset switch cable (supplied with the PC) to connector J2 on the AIC. Attach the other end of the cable to the PC Motherboard connector that was previously vacated in step 4d.

NOTE: Be sure that pin 1 marked on each end of the cable matches pin 1 on the AIC and the PC Motherboard. (If connected incorrectly, the PC is reset continuously or does not boot up at all.)

f. Insert the AIC in the appropriate PC slot and attach it to the chassis using the screw previously removed. Make sure the card is securely attached to the PC Motherboard and does not touch any adjacent cards.

g. Attach one of the connectors on the end of the AIC-to-VPC interface cable to the 40-pin connector on the AIC, as shown below. (The other four connectors on the cable are attached to the VPCs and optional Fax Card as they are installed. Note that one end of the cable attaches to the AIC, and the other end of the cable attaches to the 1st VPC.)

NOTE: The colored edge of the AIC-to-VPC interface cable must align with pin 1 on the connector (see Figure 3-97 on page 3-221).

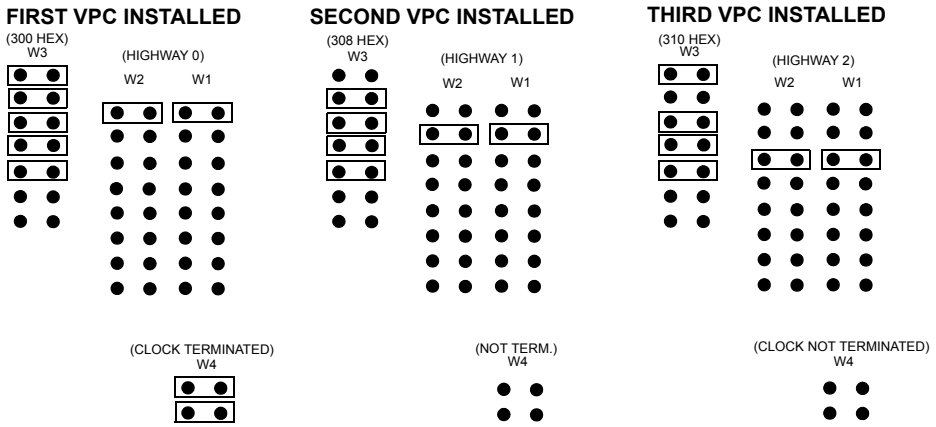


5. *If not already installed*, install each of the up to three Voice Processing Cards (VPCs) as follows (see Figure 3-98 on page 3-222):

a. Remove the bracket that covers the opening to the appropriate 16-bit (full-size) slot as shown in Figure 3-96 on page 3-220, and set the bracket and screw aside.

NOTE: The first VPC must be installed so that it can be connected to the far end of the AIC-to-VPC interface cable that was previously attached to the AIC.

b. Depending on which number VPC (1-3) is being installed, check to make sure that jumper straps are installed in the following locations on the VPC (see Figure 3-98 on page 3-222 for orientation).



c. Insert the VPC in the appropriate slot and attach it to the chassis using the screw previously removed. Make sure the card is securely attached to the PC Motherboard and does not touch any adjacent cards.

d. Attach the next available connector on the AIC-to-VPC interface cable to the 40-pin connector on the VPC. (One end of the cable was previously attached to the 40-pin connector on the AIC.)

NOTE: Be sure that the colored edge of the AIC-to-VPC interface cable aligns with pin 1 on the connector.

6. *If not already installed,* install the Video Card as follows:

   a. Remove the bracket that covers the opening to the appropriate 8-bit (half-size) slot as shown in Figure 3-96 on page 3-220, and set the bracket and screw aside.

   b. Insert the Video Card in the appropriate slot and attach it to the chassis using the screw removed in step 6a. Make sure the card is securely attached to the PC Motherboard and does not touch any adjacent cards.

7. *If desired and if not already installed,* install the optional Fax card. There are two different styles of fax cards. The settings for both fax cards are the same, but the location and labeling of the DIP switches and jumper settings have been rearranged. (See Figure 3-100 on page 3-224 for the older-style cards and see Figure 3-101 on page 3-225 for the newer-style fax cards.):

   a. Remove the bracket that covers the opening to the appropriate 16-bit (full-size) slot, and set the bracket and screw aside. (See Figure 3-96 on page 3-220 for a sample card slot arrangement.)

   b. Ensure that the IRQ selection jumper (J503 on 2-, 4-, and 8-port cards; J902 on 12-port cards; and J2502 on 16-port cards) is installed in location IRQ9.

c. Ensure that the eight address selection DIP switches (SW1 on 2-, 4-, 8-, and 12-port cards and SW2501 on 16-port cards) are set to 320:

   – 1-3 & 5-6 = ON for older cards (down); 4 & 7-8 = OFF for older cards (up).

   – 1-3 & 5-6 = CLOSED for newer cards (down); 4 & 7-8 = OPEN for newer cards (up).

d. Ensure the two MVIP clock termination DIP switches (SW2 on 2-, 4-, and 8-port cards; SW1001 on 12-port cards; and SW2601 on 16-port cards) are set in the OFF (up) position.

e. Insert the Fax Card in the appropriate slot and attach it to the chassis using the screw removed in sub-step a. Make sure the card is securely attached to the PC Motherboard and does not touch any adjacent cards.

f. Attach the next available connector on the AIC-to-VPC interface cable to the 40-pin connector on the Fax Card. (One end of the cable was previously attached to the 40-pin connector on the AIC.)

NOTE: Be sure that the colored edge of the AIC-to-VPC interface ribbon cable aligns with pin 1 on the connector.

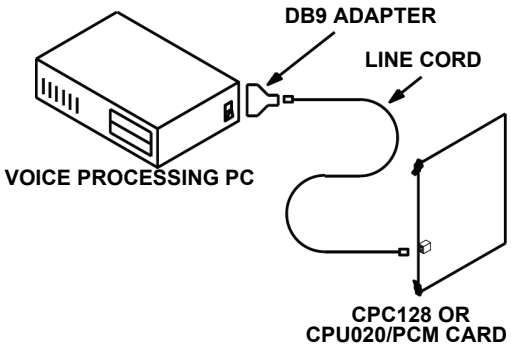
8. Replace the cover on the PC and re-install the screws.

9. To set up the RS232 interface between the phone system and the Voice Processing PC:

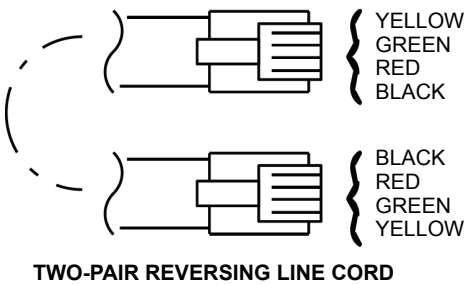
   a. Referring to the diagram below, attach the DB9 modular adapter (part no. 804.2420; supplied with the PC) to the DB9 (COM1) serial port connector on the PC.

   b. Plug one end of the mod-to-mod line cord – part no. 813.1592 [DMS67 in Europe]; supplied with the PC, into the DB9 modular adapter.

   c. Plug the other end of the line cord into an RS-232 port on the CPC or CPU Card. Do *not* use the Options Card.



NOTE: The RS232 interface consists of a standard, two-pair, reversing, mod-to-mod line cord with an attached 9-pin modular adapter. A reversing line cord can be identified by examining the modular plug connectors side-by-side as shown at the right.



TWO-PAIR REVERSING LINE CORD

10. To supply the first eight audio interface ports (e.g., dedicated system voice channels) to the Voice Processing PC, attach one end of the AIC-to-CPU/OPC interface cable (supplied with the PC) to the DB15 PCM connector on the CPC or CPU card. Attach the other end of the cable to the first DB15 PCM connector (J7) on the AIC (see Figure 3-97 on page 3-221).

NOTE: The right-angle connector end of the cable attaches to the CPC or CPU card. The straight-in connector end of the cable attaches to the AIC. Also, the slide latch on the end of the right-angle connector can be used to lock the cable to the card.

11. For an additional 4-24 (or 32) ports (voice channels), attach one end of a second AIC-to-CPU/OPC interface cable (supplied with 12-port or higher PCs) to the DB15 PCM connector on the Options Card. Attach the other end of the cable to the second DB15 PCM connector (J8) on the AIC (see Figure 3-97 on page 3-221). For an ATM-based system, the first chassis must host the OPC to use the remaining 32 ports.

NOTE: The Voice Processing Unit is required to support 40 ports.

ALSO: The Options Card must be configured to have a Voice Processing connection. See PROGRAMMING, page 6-117.

12. Use the slide latch on the end of the right-angle connector to lock the cable to the card.
13. Connect the power cable according to the PC manufacturer's instructions and turn on the Voice Processing PC's AC power switch.

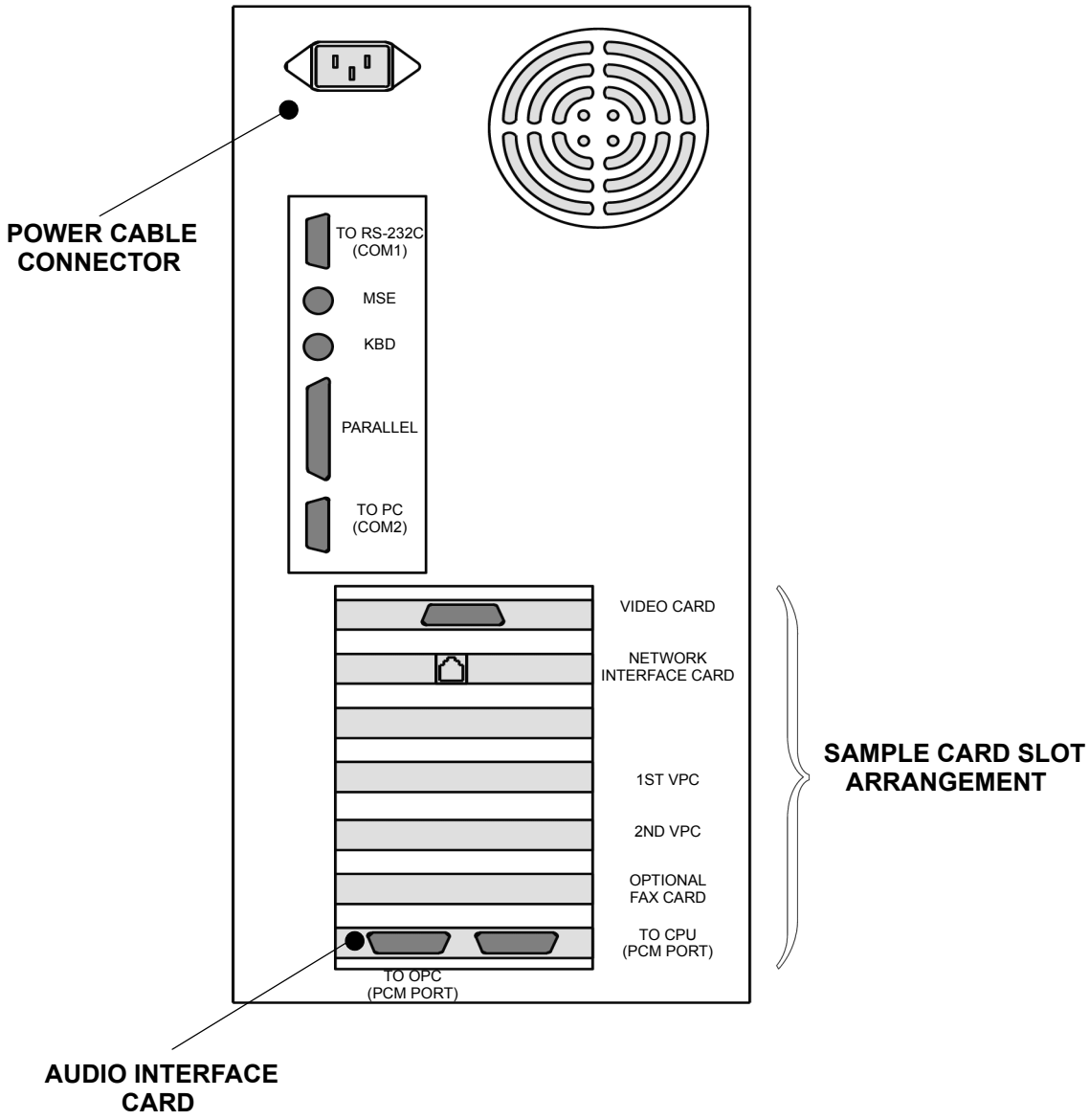
NOTE: For optimal performance, the Voice Processing PC is configured with turbo mode always enabled. Do not disable the turbo mode setting.

14.11 If you have any problems with Windows recognizing the Fax card, follow these troubleshooting steps:

- The IRQ should be set to 9 and the I/O address should be set to 320. Make sure that there are no IRQ or I/O conflicts.
- *For Windows NT-based VPUs:* In Control Panel, select **Services**. The *faxinit* service should be set to **automatic**. Then select Devices from the Control Panel. The bfax device should be set to **automatic** and started.

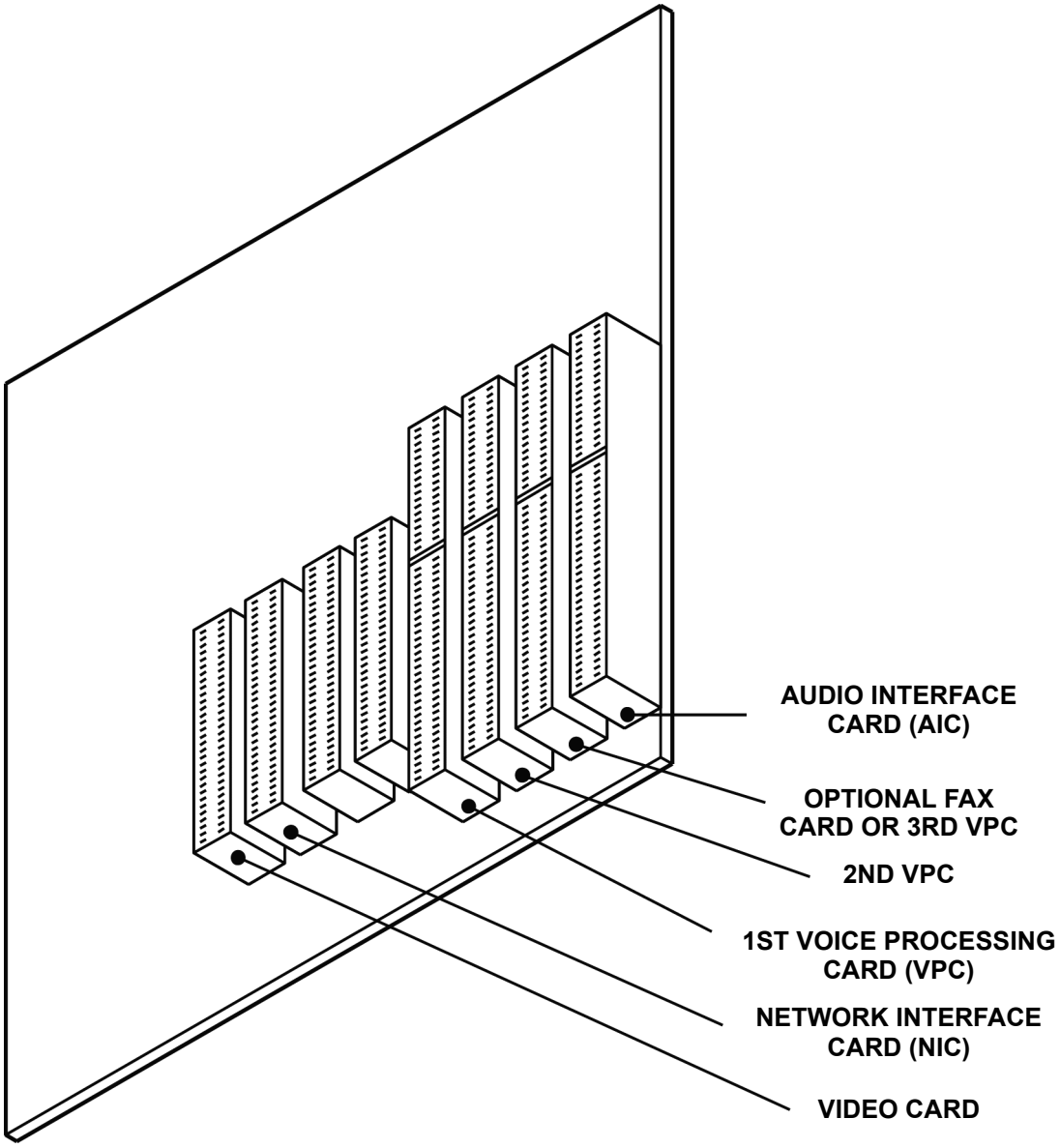
  For Windows 2000-based VPUs: In Control Panel, select **Administrative Tools** then **Services**. The *faxinit* service should be set to automatic. Then, under Administrative Tools, select **Computer Management**. Go to **Device Manager**, from the View menu on top, select **show hidden devices**. From there expand Non-plug and play drivers. The bfax device should be set to automatic and started.
- *If the driver is installed correctly,* after a system restart, there should be a faxinit.log file in the \winnt\bfax directory. It indicates how many fax ports were detected.
- *If the driver is not installed correctly,* uninstall the fax driver (from the \avdap\drivers\brooktrt\install directory type *install -r < uninstall.rsp*), remove and re-install the fax card, then reinstall the driver (*install < install.rsp)*. Restart the system; the system should be able to detect the fax card.
- If using the Unified Messaging feature, the Voice Processing PC must have a security key attached to the parallel port. See page 4-67 for more information on the security keys.

Figure 3-95. *Sample Back of Assembled Voice Processing PC*



NOTE: Depending on the model of PC being used, the sample card slot arrangement and connector locations shown above could vary somewhat.

Figure 3-96. *Sample PC Motherboard Card Slot Arrangement*



NOTE: Depending on the specific model of PC Motherboard being used, the sample card slot arrangement shown above could vary somewhat.

Figure 3-97. *Audio Interface Card (AIC)*

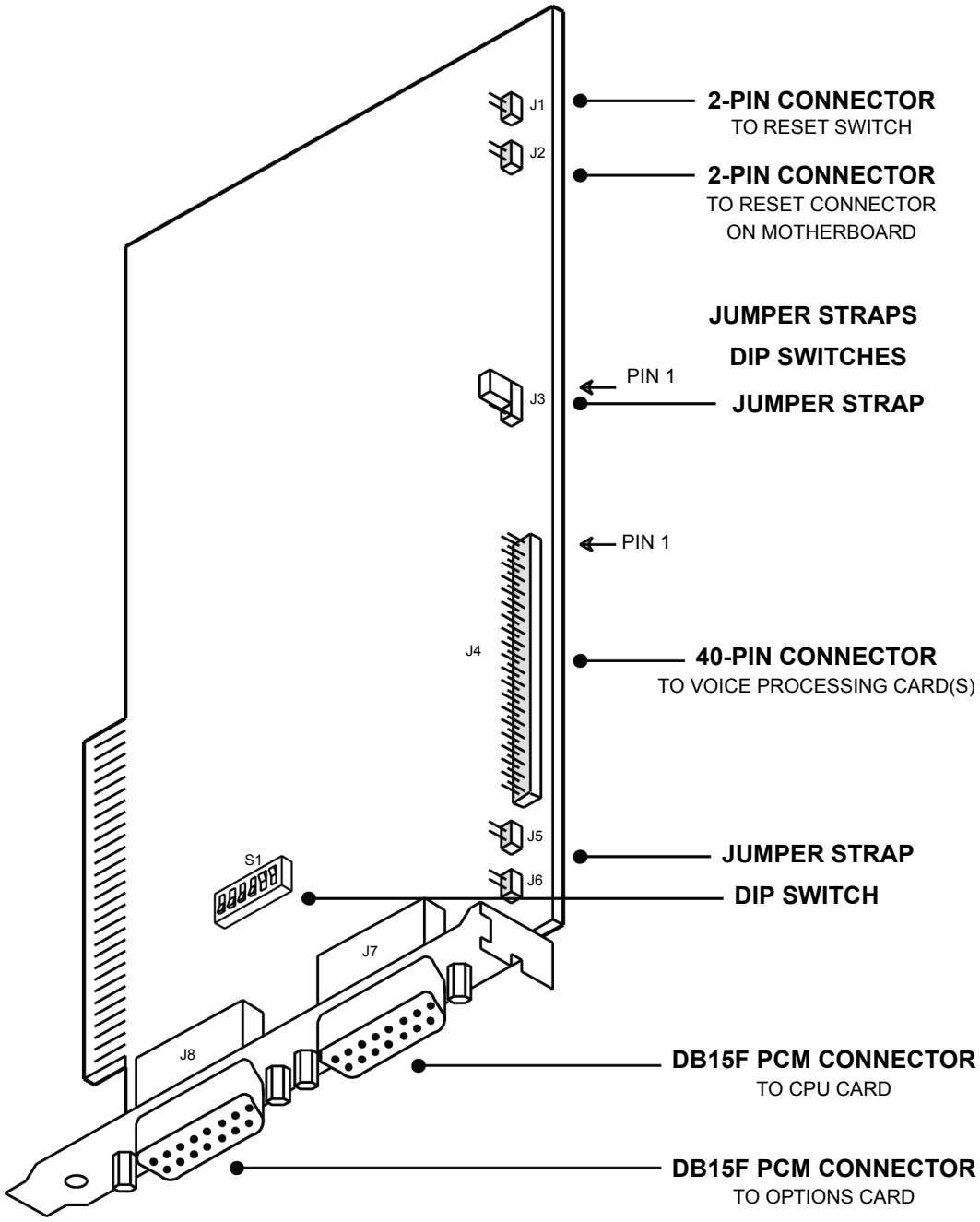


Figure 3-98. *Voice Processing Card (VPC)*

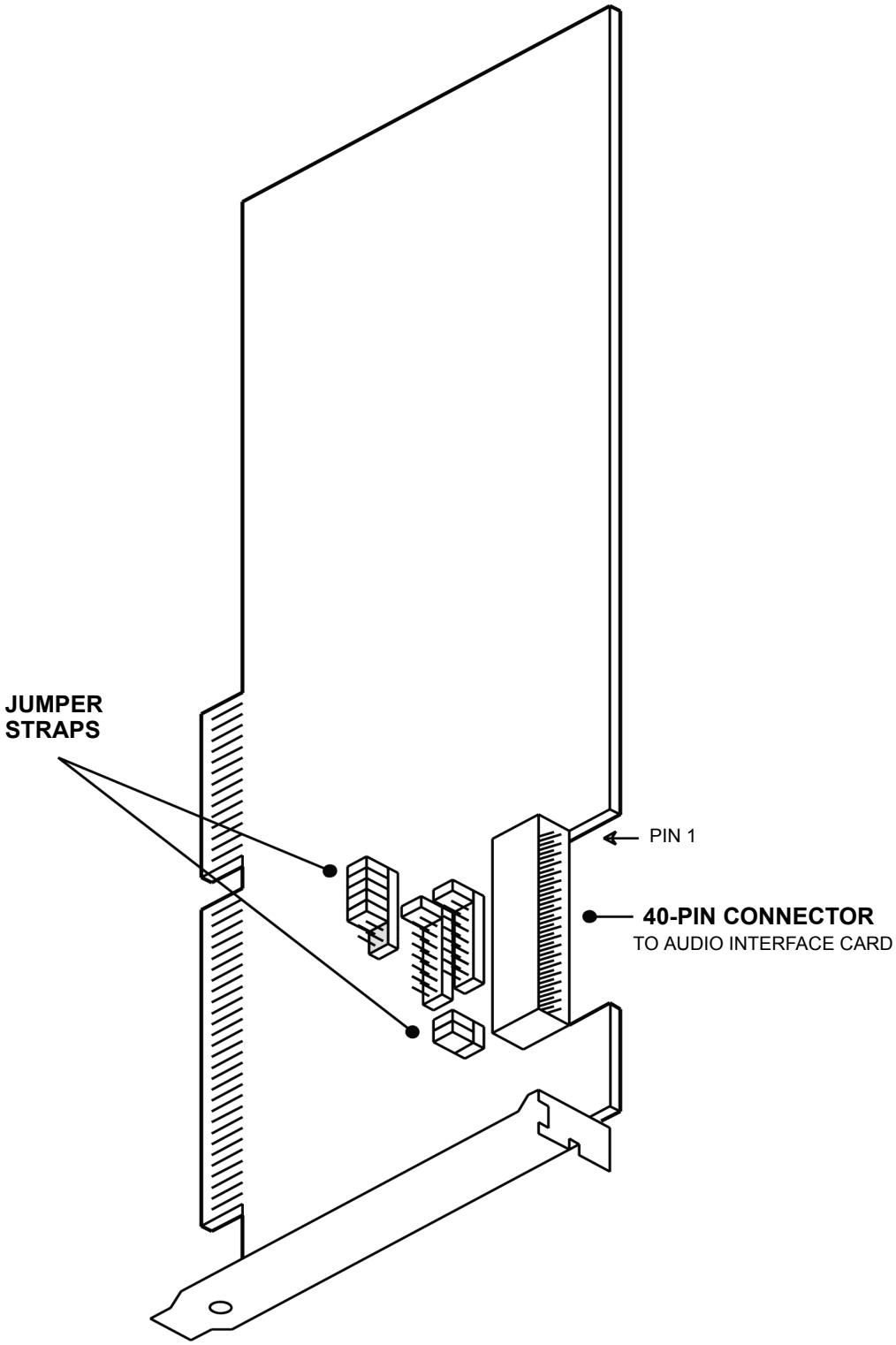


Figure 3-99. *Video Card*

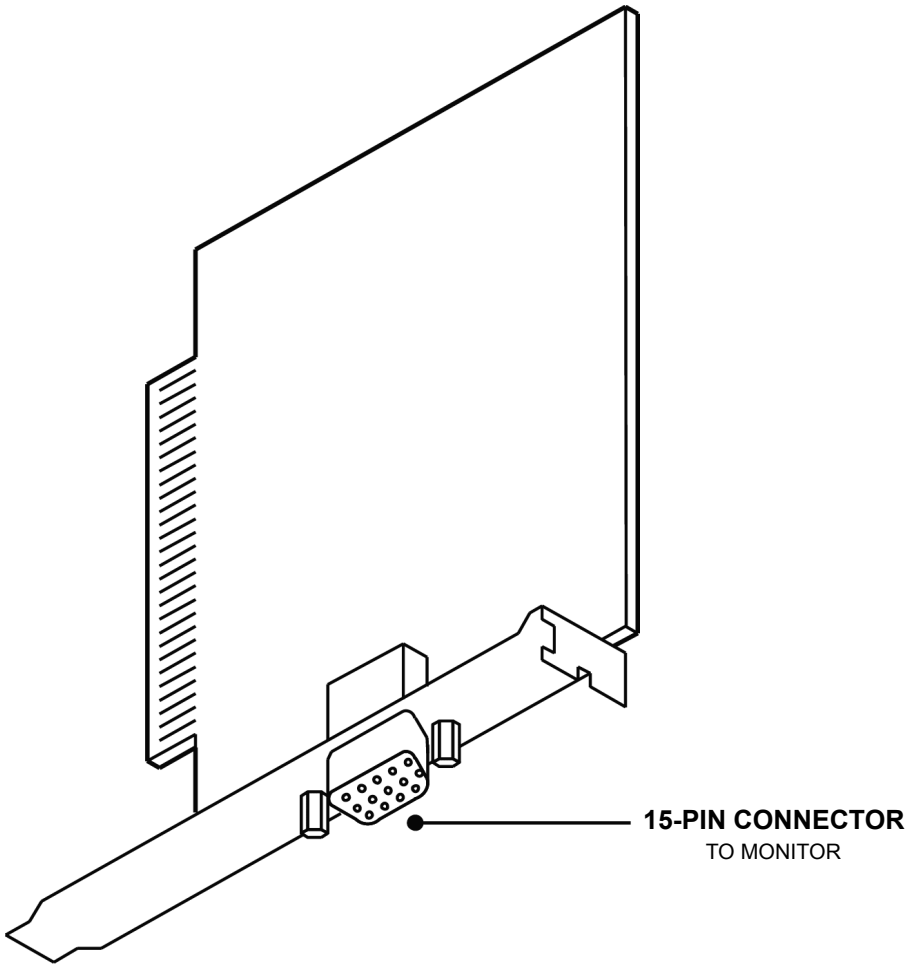


Figure 3-100. *Original Optional Fax Card*

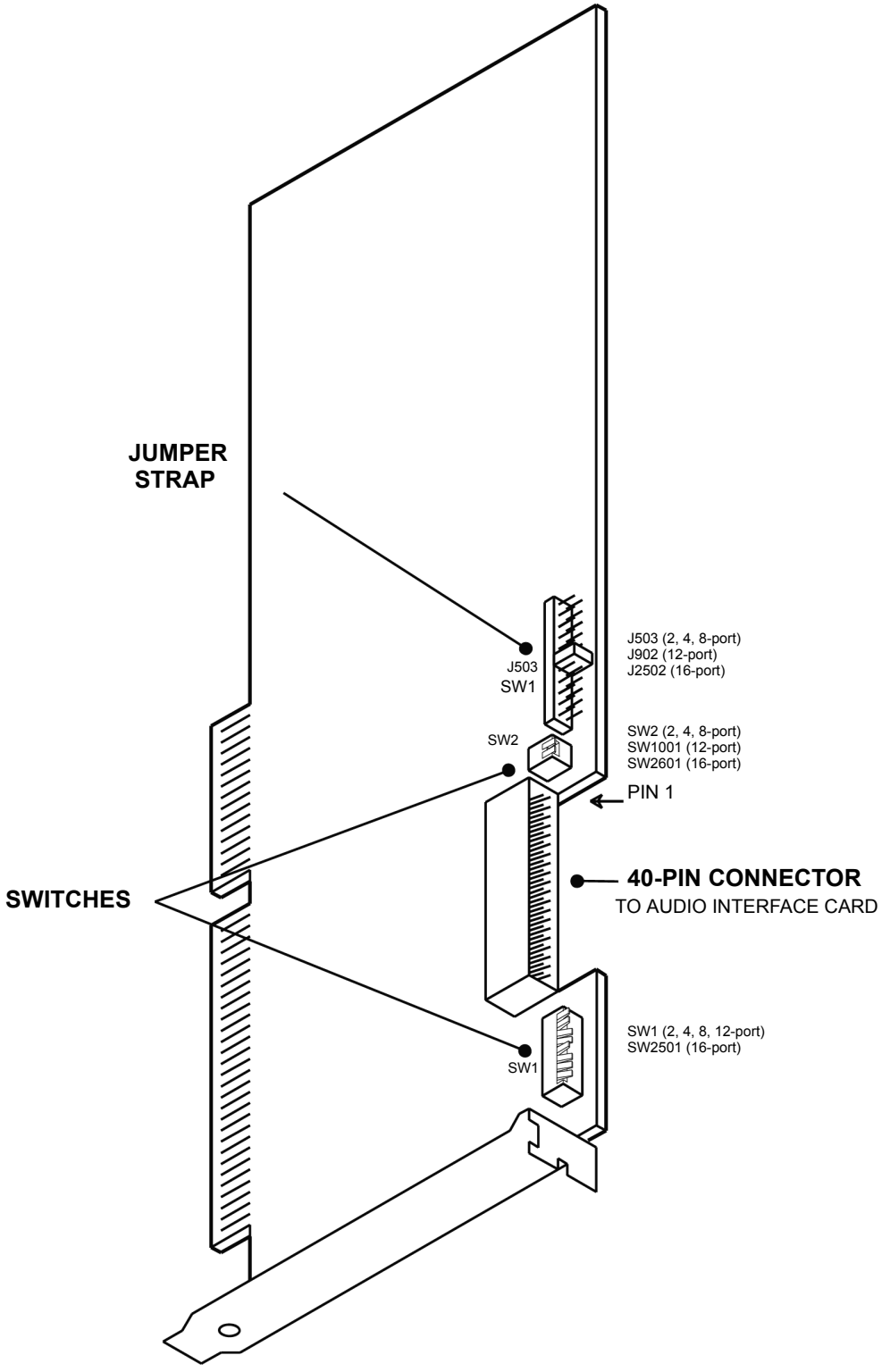
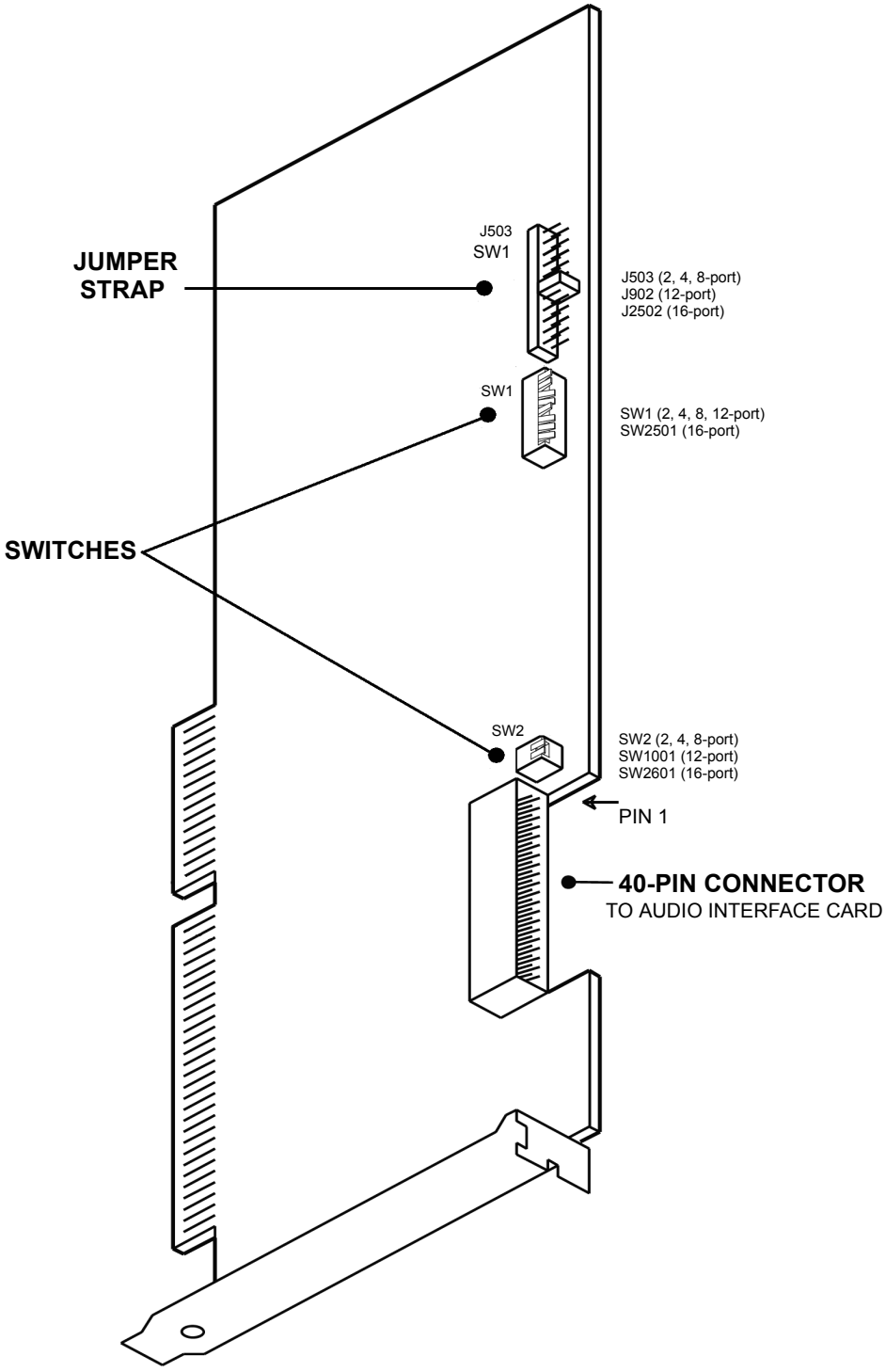


Figure 3-101. *New Optional Fax Card*



B. VOICE PROCESSING SOFTWARE

Software Upgrades

14.12 When upgrading from one version of Voice Processing software to a newer version, make sure you complete the following steps:

1. Log into the Voice Processing PC using the administrator's account.
2. Start the Avdapmon Utility. Enter a password, if necessary. See page 3-230 for more information about Avdapmon.
3. Stop the avdap service by entering a tilde (~) in the Avdapmon Utility window.

   NOTE: When Voice Processing is shut down, all deleted messages are permanently deleted and cannot be restored.

4. When the avdap service is stopped, close the Avdapmon window. (You can use Task Manager to make sure the avdap.exe process has been stopped.)
5. Install the Voice Processing software as usual. (See the next section for details.)
6. After the PC reboots, the avdap service should be running correctly. Login to verify that. If not, contact Inter-Tel Technical Support.

Voice Processing Software Installation

14.13 If necessary, use the following procedure to re-install the Windows version of Voice Processing applications software.

NOTE: The hard disk drive must be properly formatted and must have Windows NT Workstation version 4.0 with Service Pack 5 or Windows 2000 with Service Pack 4 installed. Also, if upgrading from a previous version of Voice Processing, first stop the Avdap service in Control Panel (see page 3-230 for details).

1. *If not already installed,* attach the monitor to the 15-pin Video card connector on the back of the Voice Processing PC. Then, plug in the monitor's AC power cord and turn ON the monitor's AC power switch.
2. *If not already installed,* attach the keyboard and mouse to their appropriate connectors on the back of the PC.
3. Turn ON the PC's AC power switch and allow the system to boot up.
4. Press **CTRL+ALT+DEL**, log on using an account that has administrator rights, and then press **OK** to continue.

   NOTE: In the default state, the Windows system administrator does not have a password. Inter-Tel recommends setting up a password.

5. Insert the Applications Software Disk in the PC's floppy disk drive or the Software CD-ROM in the CD drive.
6. Select **Start** and then **Run**.
7. In the Command Line text box, enter `A:setup` (or `B:setup` if the B drive is used). Then select **OK**.

NOTE: If necessary, you can select **CANCEL** from any setup window to exit without performing the installation. Also, during the file installation process, pressing **ESC** exits the setup program without completing the installation.

8. A Welcome screen appears. Select **Next** to begin the installation. If the software has been installed before, you are asked to stop the Avdap service.
9. For a new installation, a Select Destination Directory window asks you for the desired destination directory name. If the software has been installed before, the previous directory is the default value (default is C:\AVDAP or D:\AVDAP, depending on the factory configuration of your Voice Processing PC). To use the default drive and directory name, select **Next**. To select another drive and/or directory, select **Browse** and follow the displayed instructions.
10. If you are re-installing software, you are allowed to select which components you want to re-install.
11. The Ready To Install screen appears. Select **Next** to continue.
12. A Voice Board Driver screen appears. Select **Next** to continue.
13. A Board Information window appears. You need to know what voice boards are installed in your Voice Processing PC.
    a. To add a board, select **Add**. Then choose the appropriate model from the list. The model depends on number of ports. For example, use RDSP/4000 for four ports, RDSP/8000 for eight ports, RDSP/24000 for 24 ports, etc.
    b. Set the Address to **300** (default) for the first board in the PC. Use **308** for the second board and **310** for the third board.
    c. To change the Address, highlight the board and select **Edit**.
    d. To remove a board, highlight it and select **Remove**.
14. Select **Next** to continue.
15. An Interrupt Request (IRQ) window appears. Make sure the IRQ Value is set to **5** (default). Then select **Next** to continue.
16. A Shared Memory Address window appears. Make sure the shared memory address is set to **0xD0000** (default). Then select **Next** to continue.
17. An Advanced Options window appears. Select **Advanced Options**. In the Save Driver settings, make sure **No** is selected. Then select **Next** to continue.
18. In the next screen, select **Finish** to continue.
19. Another screen asks how many FAX ports the system has. Select the correct number and then **OK**.
20. A window appears stating that Voice Processing software has been successfully installed. Select **Finish** to continue.
21. A window appears informing you that you must restart the system to complete the installation. Select **OK**.
22. *To install the Japanese voice prompts (if desired),* remove the Applications Software Disk and insert the Japanese Prompts Disk 1 of 3. Select **Start** and then **Run**. Enter `a:pmtsetup` and press **ENTER**. Then follow the installation instructions displayed on the monitor.

NOTE: If Japanese language prompts are not installed, yet a user attempts to access them, English prompts are delivered instead.

Voice Processing Uninstall Program

14.14 The Voice Processing Windows software comes with an uninstall program. To access the uninstall program:

1. Select Start - Programs - Voice Processing - **Uninstall Voice Processing**.
2. A screen appears that says, "The uninstall program will restart the PC." Select **Yes** to continue, or select **No** to cancel the uninstallation.
3. If you select **Yes**, the uninstall program modifies the settings of several Voice Processing related programs and displays a screen that says, "Please run Uninstall again after the PC restart to complete the uninstallation." Select **OK** to continue.
4. The uninstall program terminates the AVDAP service and restarts the PC in a few seconds.
5. After the PC restarts, select Start - Programs - Voice Processing - **Uninstall Voice Processing** again.
6. The Select Uninstall Method screen appears. Choose **Automatic** and then select **Next**.
7. The uninstall program deletes all the Voice Processing files, services, and registry entries. The only exception is the avdap_root\avdap\db and avdap_root\avdap\drivers ("avdap_root" is the directory where you install the AVDAP files and the default is c:\avdap). Uninstall does not delete those files to prevent unwanted loss of system data. However, you can manually delete those if you do not need the Voice Processing data (such as custom recordings).
8. You can now reinstall Voice Processing software. If you did not delete the \avdap\db directory, the system retains the previous Voice Processing data.

NOTE: When downgrading Voice Processing software to an earlier version, you must first uninstall the newer version. If you do not uninstall the newer version before loading the older version, the Voice Processing audio may not operate properly.

C. WINDOWS INSTALLATION AND CONFIGURATION

14.15 The Voice Processing PC is shipped with Windows NT Workstation (version 4.0 with Service Pack 5) or Windows 2000 (with Service Pack 4) already installed and properly set up. If necessary, you can re-install Windows using the instructions outlined in Microsoft's installation manual, while observing the items listed below:

- Install TCP/IP protocol
- Install the appropriate Service Pack (4 for Windows 2000, and 5 for Windows NT); you can download these directly from Microsoft's Web site

NOTICE

Microsoft Updates. Inter-Tel Engineering routinely updates its Windows-based house and lab systems with critical updates from Microsoft as they become available. Should Inter-Tel become aware of a problem with any such update, Inter-Tel Engineering, in a timely manner, instructs the reseller channel to not install the offending critical update. The notification is e-mailed to all "Inter-Tel Technical Hot List" subscribers and posted under Tech Central on the *edGe*. For readers located in Europe, contact the Inter-Tel Europe office for technical information.

Inter-Tel encourages all customers to include their Windows-based products in their security umbrella and to update them as they would any other mission critical component on their system. Along this line, some Inter-Tel products are pre-configured to automatically download critical updates once a week, but not to automatically install them as that may require a reboot.

Because Inter-Tel is not given any advance notice regarding Microsoft critical updates, Engineering is unable to test them ahead of time. As with any non-Inter-Tel update, Inter-Tel cannot be responsible if a Microsoft critical update causes an Inter-Tel software application to fail. However, Inter-Tel responds to the issue in a timely manner.

For information about Inter-Tel compatibility with service packs and driver updates, search the Inter-Tel Knowledge Base or the Hot List archive on the Inter-Tel *edGe* Web site.

14.16 The following optional Windows software configuration changes can be made if necessary.

Voice Processing (Avdap) Service

14.17 The Voice Processing software is designed to start as a service using the default system service account. After started, it attempts to logon as a network user to gain network access, and, if necessary, it continues to try to log on about once a minute until it either succeeds or is terminated. However, these attempts do not delay the start of Voice Processing features, other than Unified Messaging. While the network connection is down (either while the service is awaiting logon or because of a later failure), Unified Messaging and activities, such as print report or save/restore, mapped to a network location fail. After the logon succeeds and while the network is up, activities involving the network connection succeeds.

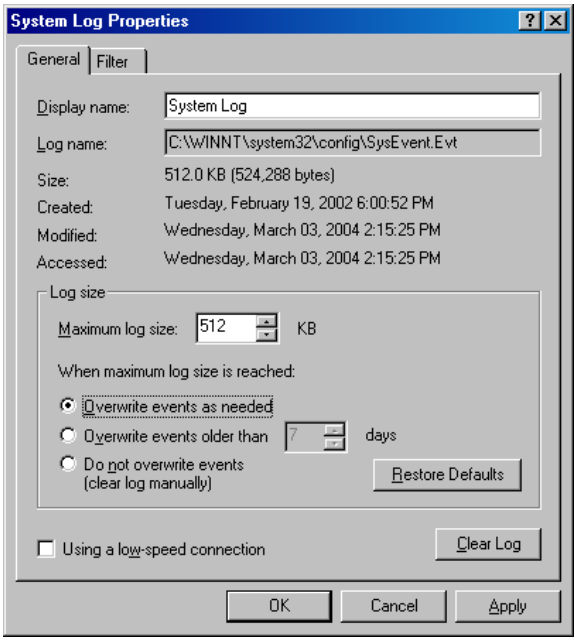
14.18 Voice Processing starts up automatically and does not require a user to log on to Windows in order to start it. However, the service must wait for "log on" to get access to network resources in the following situations:

- If the customer's E-mail system is configured to MAPI, the service needs to log on so that it can access the E-mail server.
- The B: drive is mapped to a network drive; the service needs to log on so that it can access the directory on the network drive.

14.19 To log on to the domain server, the service requires a valid user ID and password on the domain server. They are programmed through Voice Processing DB Programming (see page 6-351). The user ID and password can be obtained from the Windows system administrator. It is important that the system administrator set up the password so that it does not expire. If the password expires, the log on fails.

14.20 If Voice Processing is unable to log on to the network, a message is automatically entered into the Windows System Event Log. Because Voice Processing tries repeatedly to log in to restore a network connection, if the network is down for a long time (or Voice Processing is running in an environment without a network), many entries are entered into the Event Log. Eventually, the log fills up, and warning messages start appearing on the PC screen. The instructions below tell how to change the setting of the Event Log to handle this situation.

1. Select Start - Programs - Administrative Tools (Common) - **Event Viewer**.
2. Right-click System Log and select **Properties**.
3. In the General tab, under the "When maximum log size is reached" section, select **Overwrite events as needed**.



4. Select **OK** to save your change and exit.

14.21 The account that the Avdap service uses must be part of the local "Administrator" group on the Voice Processing PC and it must have an Advanced User Right set to **Log on as a batch job**. To do this, follow these steps:

1. Select Start - Programs - Administrative Tools (Common) - **User Manager**.
2. Select the Administrators group from the list in User Manager.
3. Select **Groups**.
4. Choose the Avdap service account from the Names list. Select **Add** to add this name.
5. Select **OK** to exit to the Local Group Properties window.
6. Select **User Rights** from the Policies drop-down menu.
7. Check the box in the lower left corner to **Show Advanced User Rights**.

8. Select **Log on as a batch job** from the Right drop-down list, then select **Add**.
9. Choose the Avdap account from the Names list and select **Add**. Then select **OK**.
10. Select **OK** to exit the User Rights window.
11. Exit the User Manager program.

14.22 There are two ways you can start the Avdap service manually:

- Open Services inside the Control Panel. Select **Avdap** and select **Start**.
- Open a command shell, then type `net start avdap`.

14.23 There are three ways you can shut down the Avdap service manually:

- From the avdapmon window (see the next section), type ~ (tilde).
- Open **Services** inside the Control Panel. Select **Avdap** and select **Stop**.
- Open a command shell, then type `net stop avdap`.

AvdapMon Utility

14.24 Because the Voice Processing software is running as a service, it does not have a console for displaying output. Inter-Tel provides a stand alone program called "AvdapMon" that can be used for viewing the Voice Processing software messages. The setup program automatically creates a shortcut to Avdapmon in the Voice Processing programming group in the Programs folder.

14.25 Avdapmon uses a TCP/IP socket to connect to the Voice Processing software. It can be run from the Voice Processing PC itself or any other PC running Windows NT, Windows 2000, or Windows 95, as long as there is a TCP/IP network connection from the PC running the AvdapMon utility to the Voice Processing PC. Note that the software supports only one active AvdapMon connection.

14.26 This program has the following parameters:

AVDAPMON <*IP number*> <*socket number*>

- The <*IP number*> is the IP number of the Voice Processing PC in "dotted notation" (e.g., 190.253.11.210).

> **NOTE:** The Voice Processing Unit requires a static IP address. If it obtains its IP address from the server, the AvdapMon utility does not load properly.

- The <*socket number*> is the socket number of the Avdap service. If not configured differently, the default socket number is 4444.

14.27 **For the Windows NT-based VPU running Avdap version 8.1 and earlier**: *If the IP address was assigned statically,* you do not need to modify the shortcut and follow the instructions above; *If the IP address was assigned dynamically,* you must specify an IP address in the shortcut. **For the Windows NT-based VPU running Avdap version 8.2 and later,** you must specify an IP address (static or dynamic) in the shortcut (see Table 3-27 on page 3-232). **For the Windows 2000-based VPUs,** you can specify 127.0.0.1 as the IP address in the shortcut or the specific IP address (see Table 3-28 on page 3-232).

Table 3-27. *VPU with Static IP*

Avdap Version	Windows NT VPU	Windows 2000 VPU
8.1 and Earlier	Do Not Modify	127.0.0.1
8.2 and Later	VPU's IP Address	127.0.0.1

Table 3-28. *VPU with Dynamic IP*

Avdap version	Windows NT VPU	Windows 2000 VPU
8.1 and Earlier	VPU's IP Address	127.0.0.1
8.2 and Later	VPU's IP Address	127.0.0.1

14.28 If the AvdapMon utility is run on the Voice Processing PC itself, there is a shortcut notation to specify the IP and port numbers. In this case, simply type **avdapmon**. (that is, *avdapmon* followed by a space, a period, a space, and a period again).

14.29 To protect the Voice Processing PC against unauthorized access through AvdapMon, a password can be required to use the AvdapMon utility. This password is set through DB Programming (see page 6-351).

Voice Processing Configuration

14.30 The Voice Processing software runs as a service (called "avdap") under the Windows operating system. It gets its startup parameters from the registry key HKEY_LOCAL_MACHINE\SOFTWARE\AVDAP\Parameters. Information such as the IP address AVDAP listens to, the size of log files, and the drive to map as B: Drive, can be specified there.

14.31 The Inter-Tel Voice Mail configuration utility provides a user-friendly graphical user interface to specify these parameters without accessing the registry directly. The fields that can be modified through the configuration utility include:

- IP Addresses and IP Port for AVDAP and CP
- Select or deselect IP for ASAI Connection
- Enable or disable log files
- The log files size and number of log files
- Serial port used by AVDAP for ASAI connection
- Enable or disable SMDR Polling on serial port
- Drive to map B: Drive to (map_b_drive for save/restore to/from network drive)
- Ignore the parameters in Avdap.opt file

14.32 The Configuration Utility uses various registry keys under HKEY_LOCAL_MACHINE\SOFTWARE\AVDAP to keep track of the values of the fields. When you perform a Voice Processing database save, it saves the new values to the keys. In addition, it manipulates the MULTI_SZ in HKEY_LOCAL_MACHINE\SOFTWARE\AVDAP\Parameters based on those values, so that Voice Processing obtains the appropriate values during startup.

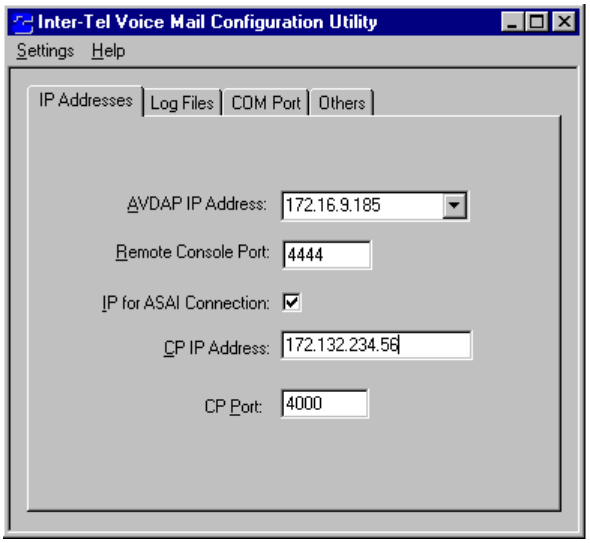
14.33 Use the Configuration Utility (avdapConfig.exe in the avdap directory) to program the necessary fields, as described in the following topics:

- IP Addresses Tab
- Log Files Tab
- COM Port Tab
- Others Tab

14.34 Each of these tabs are described in detail on the following pages.

14.35 IP Address Tab: The phone system normally uses a serial link (RS232 link) to establish the ASAI connection between Voice Mail and Call Processing. With the Call Processing Card (CPC) or Call Processing (CP) Server CPU, Voice Processing has the capability to establish the ASAI connection through a TCP/IP socket for voice mail.

14.36 This feature does not require any premium features. However, only the CPC and CP Server can use the socket connection. The CPC/CP Server and the Voice Mail must be able to communicate with each other across a LAN. It is also possible to connect the two directly together using a standard Ethernet cross-connect cable similar to the one provided with the CPC or CPU.

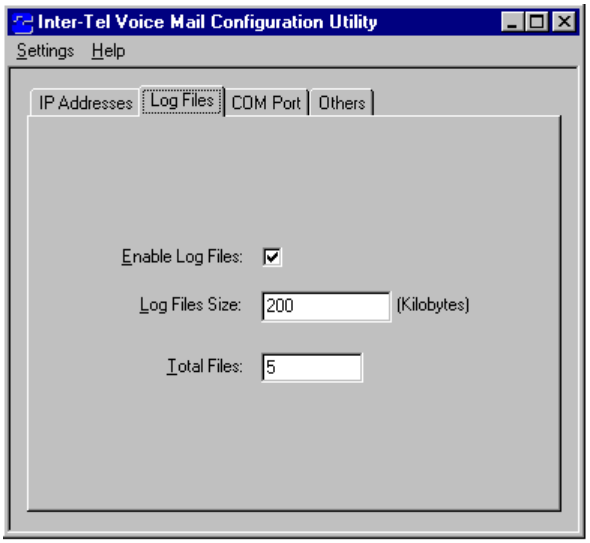


14.37 These fields are used by Voice Processing to communicate through TCP/IP sockets:

- **AVDAP IP Address**: Specifies the IP address that Voice Processing uses to communicate with the LAN. Voice Processing uses it to listen for remote console connection (e.g., AVDAPMON), CP ASAI connection, and TCP/IP Networking messages. This parameter is required, and has no default. Select from the drop-down list, which lists all the IP Addresses installed on the Voice Processing PC.
- **Remote Console Port**: Specifies the socket number that AVDAPMON uses to talk to Voice Processing. The default value is 4444.
- **IP for ASAI Connection**: Uses IP to connect to Call Processing, check this option, and program the following two fields below. If this option is not checked, Voice Processing uses the COM Port (specified in COM Port Tab) to communicate with Call Processing. At default, it is not enabled.

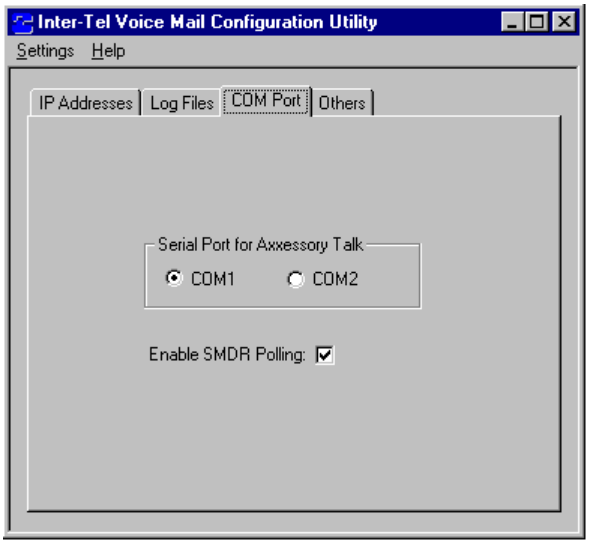
- **CP IP Address**: Specifies the IP Address of the Call Processing listening socket. Call Processing listens to this IP address for incoming ASAI message from Voice Processing. If "IP for ASAI Connection" is enabled, this field must be programmed so that Voice Processing knows which IP address to connect to Call Processing. If "IP for ASAI Connection" is disabled, this field is grayed out and not used. There is no default value for this field.
- **CP Port**: Specifies the socket number that Voice Processing and Call Processing use to communicate with each other. The default value is 4000. If "IP for ASAI Connection" is disabled, this field is grayed out.

14.38 Log Files Tab: Voice Processing saves a number of log files in the \avdap\db\msgs directory. The log files contain detailed information of the activities on the system. These fields determine the file sizes and number of log files in the system.



- **Enable Log Files**: Specifies whether to enable or disable log files. By default, it is enabled.
- **Log Files Sizes**: Specifies the size of the log files in kilobytes. The default size is 200 KB.
- **Total Files**: Specifies the number of the log files that are kept by the system. Default number is five files.

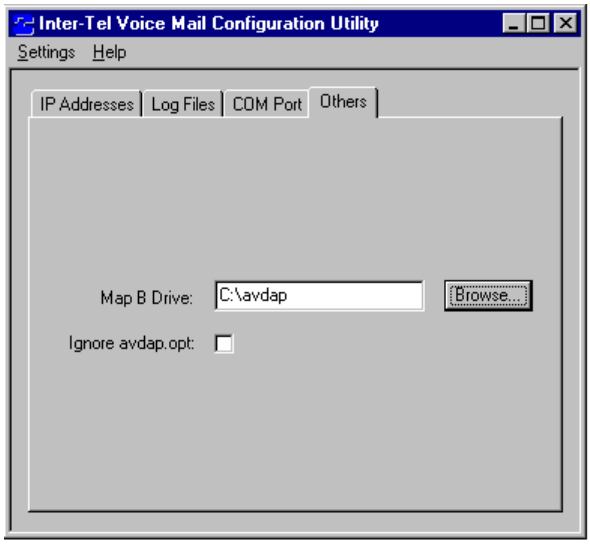
14.39 COM Port Tab: If "IP for ASAI Connection" in the IP Addresses Tab is not checked, Voice Processing uses the serial port to communicate to Call Processing. Voice Processing also uses the other serial port to poll for SMDR messages, if enabled.



14.40 These fields allow the user to specify the serial port used by Voice Processing and whether or not to enable SMDR polling.

- **Serial Port for Voice Processing**: Specifies COM1 or COM2 as the serial port to communicate to Call Processing. The default is COM1.
- **Enable SMDR Polling**: Specifies whether or not Voice Processing should perform SMDR polling on the other COM Port. The user can disable the SMDR polling if they want to use the other COM port for DB Programming. The default is enabled.

14.41 Others Tab: The Others tab has the following fields:



- **Map B Drive**: Specifies the drive (can be local or network) that Voice Processing maps as B: drive during startup. It is used to backup Voice Processing data. This field is optional. The default is empty.

- **Ignore avdap.opt**: Specifies whether or not Voice Processing should obtain additional startup parameters from the avdap.opt file. There are other parameters (such as print_asai_msgs) that cannot be specified through this utility. If you want to specify those values, you must do it in advap.opt, and choose not to ignore avdap.opt. At default, this option is not enabled.

NOTE: If there is a conflict between a value set in this utility and in avdap.opt (e.g., log files size), the value set in the configuration utility takes precedence and overwrites the value set in advap.opt.

D. UNIFIED MESSAGING

14.42 Unified Messaging versions 2.2 (VPU) and 2.3 (EM) have reached their end of sale and are no longer supported as of April 30, 2008.

14.43 Inter-Tel recommends that current UM v2.2 and v2.3 installations upgrade to UM Open Standards Edition (OSE). See below for more information.

E. UNIFIED MESSAGING OPEN STANDARDS EDITION (OSE)

14.44 Axxess v8.23x or later supports Unified Messaging OSE, a client-free Unified Messaging distribution for the Inter-Tel Enterprise Messaging voice messaging server. Unlike previous Unified Messaging products from Inter-Tel, users do not have to install a special Unified Messaging client application on their desktop computer to view, play, and manage voice mail messages. With Unified Messaging OSE, the e-mail client can be any Internet Message Access Protocol (IMAP) v4 compliant client that is configured to connect to the IMAP server located on Enterprise Messaging. To use the latest version of Unified Messaging OSE, v2.0, existing Axxess systems must upgrade to v11.0 or later, and to Enterprise Messaging v1.2 or later.

14.45 For Unified Messaging OSE v2.0 programming and feature information, refer to the latest version of the *Unified Messaging Open Standards Edition Administrator's Guide* (document part no. 835.3162). For end-user information and instructions for using features, refer to the *Unified Messaging Open Standards Edition User Guide* (part number 835.3164).

15. VOICE PROCESSING UNIT POWER SUPPLY REPLACEMENT

15.1 The Voice Processing PC is shipped with the power supply unit already installed. If the power supply fails, use the Voice Processing Power Supply Replacement Kit (828.1478). The kit includes a power supply module (550.0120) with a fan (part number 817.3135). You need a hex and/or #2 Phillips screwdriver (depending on the type of screws used on your Voice Processing PC). Replace the unit as follows (see Figure 3-95 on page 3-219):

1. Unpack the new power supply unit. (There may be four hex/Phillips-head screws included with the unit.)
2. Turn OFF the AC POWER switch on the front of the Voice Processing PC and wait at least five minutes for the power supply voltage to dissipate.
3. Disconnect the AC power cable and any other cables necessary to allow the Voice Processing PC to be moved. Note the placement of these cables for the purpose of reattachment following the replacement of the power supply.
4. Place the Voice Processing PC upright on a table and remove the screws securing the cover and side panel of the Voice Processing PC.
5. Slide the cover toward the rear of the Voice Processing PC until the cover is released from the chassis. Lift the cover off and set it aside.

6. Using a #2 Phillips screwdriver, remove the four screws securing the power supply unit to the rear of the Voice Processing PC chassis.
7. Slide the power supply unit an inch toward the front of the Voice Processing PC and lift the unit out.
8. Lower the new unit onto the power supply shelf in the Voice Processing PC and slide it back into place.
9. Replace the four screws removed in step 7 (or, if desired, use the four new screws that may have been included in this kit).
10. Verify that the 115V/230V selection switch on the power supply unit is set to the 115V setting.
11. Using the old power supply unit as a guide, route and reconnect all of the required cables and wires to their proper connectors in the Voice Processing PC.
12. Reconnect the AC power cable to the rear of the Voice Processing PC.
13. Ensure that the AC power source is properly grounded by performing one of the following tests:
    - Use a digital multimeter to measure the difference of potential between the Voice Processing PC chassis and the power neutral, and between the Voice Processing PC chassis and the third wire ground. If the voltage measurement exceeds 0.5V, noise may develop on the system. If this occurs, call an electrician.
    - Use a Megger to test the AC ground wire. (A Megger should be available through the local power company.) The reading should be 5 ohms or less.
14. Turn OFF the AC POWER switch.
15. Re-attach all cables to the back of the Voice Processing PC.
16. Slide the Voice Processing PC cover back into place, replace the screws, and turn the system AC POWER switch back ON.
17. To receive proper credit for the advance replacement, return the defective power supply unit to Inter-Tel.

16. VOICE PROCESSING TCP/IP NETWORK SETUP

16.1 The following steps describe the setup for TCP/IP networking on Voice Processing PCs.

A. SERVER REQUIREMENTS

16.2 The TCP/IP server requires the following:

- An IP address for the server
- A modem with speed greater than or equal to 28,800 connected to a serial port and set up for auto-answer
- A range of two or more IP addresses (e.g., 110.230.39.40 to 110.230.39.42 defines three IP addresses)

B. SERVER INSTALLATION

16.3 On the PC that serves as a voice mail server, create or select a conventional user account the voice mail client may use for the dial-in. You must have the privilege to modify this account to complete this installation. This requirement may prevent the use of an account from elsewhere in the domain. In the example below, account "voice_mail_networking" is used.

1. Install the Remote Access Server (RAS): Open **Network** in the Control Panel, then select **Services**. Then select **Add**.
2. Select **Remote Access Service** and select **OK**. You are asked to verify the location of the installation files, which is in the i386 directory on the Windows-based installation CD.
3. Next, you are asked if you want to install a modem. If you do, you'll need to specify the make and model of the modem as well as the port to which it is connected. (This information can also be set under the Modems icon in the Control Panel.) Configure the modem to receive calls (either exclusively or also permit dial out), then select **OK**.
4. Select **Network** in the Remote Access Setup dialog.
5. In the Network Configuration dialog, specify the level of encryption to use for account information exchange. Select either **Require encrypted authentication** or **Require Microsoft encrypted authentication**. Make sure that TCP/IP is selected and select **OK** to exit.
6. Select **Configure** for TCP/IP.
   a. Change the network access scope for remote TCP/IP clients to **This computer only**.
   b. Select the option to use a static address pool, and enter the range of IP addresses.
   c. Select **OK** for this dialog, then **OK** for the Network Configuration dialog.
7. If NetBEUI or IPX is selected in the Network Configuration dialog, dialog boxes with configuration options appear. Select **OK**.
8. If you are asked if you want to enable NetBIOS Broadcast Propagation, select the default answer **No**.
9. Select **Continue** in the Remote Access Setup dialog.
10. When you return to the Network dialog (the one with the Services tab), select **Close** and the configuration is reviewed and saved.
11. You are asked if you want to restart the computer. Select **Yes**.
12. Start **Remote Access Admin** (under Start - Programs - Administrative Tools).
13. Select **Permissions** under the Users menu.
14. Grant dial-in permission to the account that the client is to use for the dial-in, then exit the Admin program.
15. Open **Services** in the Control Panel, then select **Remote Access Server**.
16. Select **Startup** and make the Remote Access Server service's startup automatic.
17. Select **OK**, then select **Close** in the Services dialog.
18. Restart the computer. This should also start the Remote Access Server.
19. Open **Network** in the Control Panel, then select **Services**.
20. Select **Remote Access Service** and **Properties**.

C. CLIENT REQUIREMENTS

16.4 The TCP/IP server requires the following:

- An IP address for the client
- A modem with speed greater than or equal to 28,800 connected to a serial port

D. CLIENT INSTALLATION

1. Install the Remote Access Server (RAS).
2. Using DB Programming, store the information about the account on the server that this client uses when it dials in: server telephone number and server account (domain, username, and password).

E. PROGRAMMING

16.5 Within the Voice Processing software, to open a direct network connection to a remote address, you call the open_connection() procedure:

```
socket_status_t open_connection (void);
// Prepare a socket connection to a remote address.
remote_io = new communication_socket_t;
socket_status_t status = remote_io->initialize (ip_address, port);
if (status == SOCKET_STATUS_SUCCESS) {
    status = remote_io->open_connection ();
}
```

16.6 With this signature, a direct network connection is tried. Your socket is a direct network connection. The resulting socket can be used as-is in the Voice Processing application; no further code changes are necessary.

17. SMDR/ERROR MESSAGE OUTPUT DEVICE INSTALLATION

17.1 The output device(s) for the station message detail recording (SMDR) and error message recording features must meet the requirements and match the RS232 pinouts described in SPECIFICATIONS on page 2-100. The cables for the device(s) must not be longer than 50 feet (15 meters).

17.2 To connect an RS232 output device to the CPC, CPU card, Options Card(s), or PC Data Port Modules:

1. Match the bit rates on the output device and the RS232 port to which it is attached. See page 2-100 for the proper bit rate settings.
2. Turn ON the AC POWER to both the device and the system *before* connecting the interface cable to the circuit card or PCDPM. This prevents any electrical surges from being transmitted by the interface.
3. Carefully connect the interface cable from the device to the appropriate connector on the CPC, CPU128 card, CPU020/PCM card, Options Card, or PCDPM.
4. *If necessary*, using standard electrical tape, tape two ferrite split beads around the interface cable just below where the cable connects to the circuit card or PCDPM. (See Figure 3-8 on page 3-26 for an example of ferrite bead installation).

   NOTE: Ferrite beads may be necessary to help reduce electromagnetic interference (EMI) and radio frequency interference (RFI). The part number for a single split bead is 808.1195.

5. To set the parameters for the SMDR and error message outputs, see PROGRAMMING.

18. EXTERNAL MUSIC SOURCE INSTALLATION

18.1 If desired, external music sources may be connected to the system for background music and music-on-hold. The music jacks on the CPC, CPU cards, and Options Cards are the inputs for the optional external music sources (radio, tape player, etc.). The music source connected to the primary CPU can be used for music-on-hold and/or background music. However, the music sources connected to the secondary CPUs and Options Cards can only be used for music-on-hold.

18.2 The music sources on the OPC cards are intended to be used on a per trunk group basis so the end user can provide different recordings based on the trunk group.

18.3 For the music-on-hold feature, music, silence, ringback, or tick tones can be assigned on a station-by-station basis. For more information, see PROGRAMMING, page 6-198.

NOTE: There are often broadcast restrictions associated with copyrighted music. Check with the music's original distributor and/or the broadcast source for restrictions and usage concerning background music and music-on-hold. Inter-Tel is not responsible for any illegal or improper use of copyrighted music connected to the system. With European systems, contact PPL (Phonographic Performance Ltd.) for licenses.

18.4 If using a radio as a music source, place it 5 to 10 feet (1.5 to 3 meters) away from the equipment chassis to avoid RFI generated by the chassis. If the radio is placed more than 10 feet (3 meters) away from the chassis, use twisted-pair cable. For better reception, a radio with an external antenna is recommended.

18.5 To install the external music source:

1. Attach a 1/8-inch, two-conductor (mono), mini-phone plug to one end of a 5- to 10-foot (1.5- to 3.0- meter) length of shielded speaker cable (or twisted-pair cable) to run from the external music source to the CPC, CPU card, or OPC.

   NOTE: A stereo plug does not work properly in the MOH jack. If the stereo earphone output of an external music source is used, a stereo-mono adapter should be used between the stereo output jack and the monaural cable.

2. *EITHER,* connect the other end of the cable to the speaker output terminals of the music source.

   OR, if the music source has an earphone jack, attach the necessary connector to the other end of the cable, and plug it into the earphone jack on the music source.

   NOTE: If the earphone jack is "padded" so that the optimal volume level cannot be reached, the EITHER procedure (above) may be more effective.

3. Plug in the AC power cord for the music source. *DO NOT* use the outlet for the equipment chassis.
4. Turn on the AC power to the music source.
5. Insert the 1/8-inch mini-phone plug into the jack labeled MUSIC on the edge of the CPC, CPU card, or OPC.
6. From a phone:

   EITHER, select a trunk and dial the telephone number of one of the other trunks to call back into the system. Put the call on hold to hear the music.

   OR, place an intercom call to another station. Put the call on hold to hear the music.

7. While listening to music-on-hold, adjust the volume on the music source to an acceptable level.

19. EXTERNAL PAGING EQUIPMENT INSTALLATION

19.1 Optional external paging equipment can be connected to an available LGC or LSC circuit or to the speaker wires on a phone. In addition, the CPC or CPU card has its own dedicated external paging port – one in single-/dual-chassis systems, two in tri-/quad-chassis systems (see page 2-30 for more information). When connecting external paging equipment to an LGC or LSC circuit, use a paging amplifier that provides talk battery. When connecting external paging equipment to a phone, the CPC, or the CPU card, use a paging amplifier that *DOES NOT* provide talk battery.

CAUTION

Using a paging amplifier with talk battery on a phone may damage the phone or the card.

19.2 To install external paging equipment, follow these steps:

1. *If installing the paging equipment on an LGC or LSC circuit:* Cut a length of shielded speaker cable to run from the paging amplifier to the MDF backboard. Then, at the MDF, terminate one end of the cable onto tip and ring of an available LGC or LSC circuit.

   If installing the paging equipment on a phone speaker: Cut a length of shielded speaker cable to run from the paging amplifier to the phone. Then, at the phone, terminate one end of the cable onto the two wires leading from the phone's internal speaker.

   If installing the paging equipment on the CPC or CPU paging port: Cut a length of shielded speaker cable to run from the paging amplifier to the CPC or CPU card. Then attach an RCA-type phono plug to one end of the cable and insert the RCA-type phono plug into the external page jack on the CPC or CPU card.

2. Connect the other end of the cable to the amplifier high-impedance input according to the manufacturer's instructions.

3. Connect the paging speaker(s) to the amplifier using speaker cable.

4. Plug in the amplifier's AC power cord. *DO NOT* use the outlet for the equipment chassis.

5. Set the amplifier volume control to the lowest setting and turn on the amplifier.

6. *If installing the paging equipment on an LGC or LSC circuit:*

   a. Ensure that the LGC or LSC circuit is not toll restricted.

   b. From a station location, access the LGC or LSC circuit and adjust the amplifier to the desired level while placing the page.

   If installing the paging equipment on a phone speaker or the CPC or CPU paging port:

   a. From a station location, make a page by lifting the handset, pressing the **PAGE** button (or dialing 7 – the default Page feature code), and dialing a zone code that includes the external paging zone or dialing the extension number of the external paging port – default is 91000 [99000 in Europe] for the CPC, CPU 128, CPU 256, CPU 512 Master, or CPU ATM and 91001 [99001] for the CPU 512 Slave.

   b. Adjust the amplifier to the desired level while placing the page.

20. CT GATEWAY/TRANSCEIVER

20.1 System OAI Level 2 applications can be used in conjunction with the CT Gateway/ Transceiver.

- The CT Gateway provides a TCP/IP System OAI interface that supports a multi-node phone network. Each node in a phone network functions individually, as far as the System OAI streams are concerned. In other words, you cannot program a System OAI serial port on Node 1 that controls devices or receive events for other nodes. If you need to control devices on multiple nodes, you must install a CT Gateway. The CT Gateway receives the OAI streams from multiple nodes and consolidates them into a single stream for a System OAI application. The Gateway sends and receives all of its data via TCP/IP connections. (At this time, each application requires a separate Gateway.)

  NOTE: System version 9.1 or later requires CT Gateway version 4.30x or later.

- The Transceivers provide an interface between RS232 and TCP/IP. Because the CT Gateway only supports TCP/IP, and system serial ports and older System OAI applications communicate with RS232, a Transceiver is needed to translate the data protocol. Because Transceivers can be located on separate PCs, they allow the Gateway to communicate with multiple system nodes, regardless of the number of serial ports available on the Gateway PC and regardless of the distance between the nodes and the CT Gateway. Inter-Tel provides two different Transceivers: the Switch Transceiver and the Application Transceiver, part numbers 827.9000 and 827.9001 respectively.

  — The Switch Transceiver provides the interface between the system serial ports and TCP/IP.

  — The Application Transceiver provides the interface between System OAI application serial ports and TCP/IP.

NOTE: System versions 6.0 and later do not support System OAI Level 1.

20.2 For more information about the CT Gateway and its Transceivers, see the *CT Gateway Installation Manual*.

Voice Processing Features

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1. INTRODUCTION

NOTE: As of version 8.0, the OS/2 Voice Processing Unit is no longer supported.

A. VOICE PROCESSING APPLICATIONS

1.1 Voice Processing can be used for any of the following applications. Each application is described in detail in this chapter.

- **Automated Attendant**: The automated attendant is a programmable feature that can be used to provide automated call answering service. Calls can transfer, forward, or directly ring in to an automated attendant. When an automated attendant answers a call, it plays a recording that gives dialing instructions. After hearing the recording (or at any time while it is playing), the caller may then dial an extension or mailbox number.
- **Automated Attendant Recall Destination**: If a call that is transferred by the automated attendant is not answered before the Transfer Voice Processor timer expires, the call recalls the Automated Attendant Recall Destination. The Recall Destination announces that the station is unavailable and allows the caller to leave a message (if the station has an associated mailbox) or dial another extension.
- **Call Routing Announcement**: Call Routing Announcements can be used two ways:
  - — A Call Routing Announcement application can be used in place of a playback device. The playback device function is especially useful for programming hunt group announcement and overflow stations. When called, the Call Routing Announcement application plays a recording and then hangs up.
  - — The Call Routing Announcement application can use Digit Translation which allows the caller to press a single digit for access to a mailbox, a Fax-On-Demand function, or a station or hunt group that has an associated mailbox or extension ID. Digit translation can be programmed for each digit 0-9, #, and *, plus a Timeout that is used when the caller does not enter a digit. Each digit can lead to a "digit translation node" that has its own digit translation values. This layered Call Routing Announcement digit translation creates a "tree" of programmable digit translation nodes.

    NOTE: The phone system supports a Primary and Secondary Language which can each be programmed as American English, British English, Japanese, or Spanish (Mexico City Spanish Dialect for VPU and EVMC, American Spanish Dialect for EM). Digit Translations, therefore, can be programmed to use either language.

- **Message Notification/Retrieval**: Only one Message Notification/Retrieval application may be created and it *must* be created to allow Voice Mail message notification and quick mailbox access.
- **Record-A-Call**: This feature allows a station user to record an ongoing call in a mailbox message. When a station user enters the Record-A-Call feature code, the system places a call to the station's assigned Record-A-Call application. When the application answers, the system sets up a conference call with the station's Record-A-Call mailbox. At default, the mailbox plays a greeting to indicate that the recording is in progress.

- **STAR**: The Scheduled Time-Based Application Router (STAR) enhances the programmability of the Voice Mail application greetings. With STAR, applications can be programmed to play alternative greetings for holidays and weekends. A STAR application is a table of up to 20 entries that serves as a "routing table," which tells Voice Processing which application is used, based on day and time information in the table. (The applications are programmed to play the greetings, not the STAR application. The STAR routes the call to the right application.)
- **Voice Mail**: This application handles all calls that are directed to Voice Mail (other than through the Message Notification/Retrieval application) by subscribers and non-subscribers. Callers hear the main company greeting, followed by a menu of available options. Stations can forward or transfer calls directly to their mailbox using this application.
- **Directory Services**: Directory services provide callers with a list of mailboxes and extension IDs.
- **Station Message Detail Recording (SMDR) Information Storage**: SMDR information can be stored on the Voice Processing Unit's hard disk and then processed using call record sorting software, such as Inter-Tel® Inside Track® (not applicable to Europe).

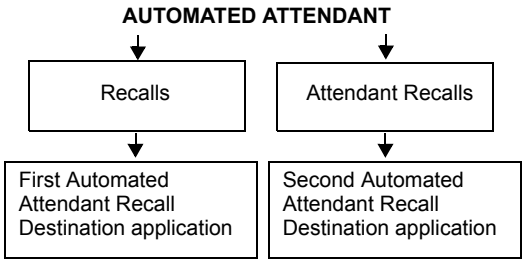
1.2 The applications are given extension numbers when they are created. Voice Processing application extension numbers are 2500-2649 at default, but may be reprogrammed.

B. TIME SLOT GROUPS

1.3 Each application is assigned to a "Time Slot Group" which determines the maximum number of voice channels that is used by the applications in that group. Voice channels are used for processing calls between the phone system and the Voice Processing applications.

1.4 The programmed voice channel limit for the time slot group may exceed the number of voice channels actually provided by the hardware, due to hardware limitations and/or heavy system traffic. For example, if an Automated Attendant application is assigned to a time slot group that has a programmed limit of five voice channels, it can normally support five simultaneous transfers of outside calls to extensions. However, if only four voice channels are available (because all other channels are in use by other applications or the hardware only supports four channels), a fifth call cannot be completed to that (or any other) application. When all voice channels are busy, intercom callers hear reorder tone and see a CALL CANNOT BE COMPLETED display; outside callers hear ringing but their calls are not answered.

1.5 Because multiple voice channels can be made available to the applications, callers are unlikely to encounter busy resources. In addition, the use of applications as recall destinations and attendants make it even less likely that a caller does not reach an available resource. For example, an Automated Attendant application can have one Automated Attendant Recall Destination with several voice channels as its recall destination and another Automated Attendant Recall Destination application (with several voice channels) as its attendant. Recalls from that Automated Attendant application would be very likely to reach an available voice channel at one of the Recall Destination applications.



C. MUSIC-ON-HOLD PARAMETERS

1.6 In previous system versions, the system determined the music source a caller heard when the call was waiting based on the trunk group on which the call resided. As of version 7.0, the system can be programmed to determine the music source a caller hears based on the device for which the caller is waiting. By default, the system determines the music source based on the trunk group on which the call resides.

1.7 The following fields are programmable for each voice processor application:

- **Audio for Calls Camped onto this Device**: Defines the audio that a caller who is camped-on to the device hears.
- **Audio for Calls Holding for this Device**: Defines the audio that a caller who is holding for the device hears.
- **Audio for Calls Ringing this Device**: Defines the audio that a caller who is ringing the device hears. The default is *Ringback*.

NOTE: If the trunk group's audio field, including Music-On-Hold, is set to Use Next Device's Audio Source, the system uses the station's programming. If the field is set to any other option, such as Board, the system uses the trunk group's audio source, overriding station programming.

1.8 When a caller on a node *with* the v7.0 or later functionality waits for a device on a node *without* this functionality, the caller hears the music source programmed for **the node trunk group** on the destination node. This is identical to the previous functionality.

1.9 When a caller on a node *without* the v7.0 or later functionality waits for a device on a node *with* this functionality, the caller hears the music source programmed for **the device** for which the caller waits. This is the new functionality.

D. COMMUNICATION BETWEEN THE SYSTEM AND THE VOICE PROCESSING UNIT

1.10 The software and hardware that control the Voice Processing applications is located in the Voice Processing PC. The phone system must be able to communicate with the Voice Processing PC for the applications to function. If the phone system is unable to communicate with the Voice Processing PC, the applications cannot be used and the following occurs:

- Direct ring-in calls to an application are sent to the primary attendant (if there is one).
- Intercom calls to applications receive reorder tone. Display phones show <*Application name*> IS UNPLUGGED.
- Administrator phone users see an Alarm 203 (Voice Processing: Communications Link Down) if the system detects a loss of communications between the phone system and the Voice Processing PC RS232 port.
- Calls do not recall to an application. Instead a recalling call remains at the station or hunt group and rings until it is answered or the Abandoned Call timer expires.
- If an application is used as a hunt group overflow or announcement station, calls are not sent to the application, but remain camped on to the hunt group.
- A transfer to an application camps on until the Recall timer expires. Then it follows the usual recall path.
- Calls cannot be forwarded or system forwarded to an application.
- SMDR information is be recorded in the buffer on the Voice Processing PC hard drive.
- Database (DB) programming for the Voice Processing features is not allowed.

1.11 Communication between the system and the Voice Processing Unit can be via RS232 connection to the equipment chassis or TCP/IP socket connection to the system CPC or CP Server. For an ATM-based system, the Voice Processing Unit should be placed close to the first chassis that provides the first eight voice mail ports. The first or second chassis must also host the Options Card (OPC) which is used for the remaining 32 voice mail ports.

2. VOICE PROCESSOR NETWORKING

NOTE: AMIS and other networking methods require specific premium features. Make sure the software license supports the required premium features.

2.1 The Voice Processing Unit network is programmed separately from the phone system network. There can be up to 99 nodes (one voice mail system per node) in the Voice Processing Unit network. There are new programming fields that provide networking support:

- **Node**: The Network Node table contains information about each Voice Mail node in the network. This table contains information that allows the system to identify nodes, contact nodes, and deliver messages to nodes. Each node has its own internal message queue (similar to a mailbox's message queues) and stores messages destined for other nodes until they are delivered.
- **Network Mailbox**: A network mailbox is a mailbox on the local Voice Processing Unit that accesses a mailbox on a remote node or the node itself. A Network Mailbox is very similar to an Extension ID. If you log on to a Network Mailbox, the only options available are the directory name, the password, and the transfer method. There are two types of network mailboxes, a "direct network mailbox" and a "node network mailbox."
  - A *direct network mailbox* is a network mailbox that accesses a mailbox on a remote node. To do this, the network mailbox is programmed to use a specific mailbox number.
  - A *node network mailbox* does not refer to a specific mailbox on a remote node, but instead, refers to a remote node in general. To do this, the mailbox is not programmed with a specific mailbox number. This tells Voice Processing to send the call to the remote node programmed. The directory name for this type of mailbox should be something that indicates the location of the node rather than an individual's name. For example, "Denver" or "Main Office."

2.2 Voice Processing has several different methods for networking Voice Mail systems together. Each node is programmed with a Network Type field that identifies how the local node communicates with the remote node. This field can be programmed to the following:

- **IC or CO AMIS**: Voice Processing sends messages via public network telephone lines or a private network using AMIS protocol. Voice Mail messages are sent by placing a call through the phone system to the destination Voice Processing Unit's phone system. An off-node Call Routing Announcement station, CO trunk group, or hunt group at the destination is programmed to receive the AMIS signals.
- **TCP/IP**: Voice Processing Unit sends messages using TCP/IP protocol. The originating and destination Voice Processing Units must have a TCP/IP service that provides a gateway to the IP network. Voice Mail messages are sent during a temporary TCP/IP session between the originating Voice Processing Unit and the destination Voice Processing Unit.

A. NETWORK OPERATION OVERVIEW

2.3 The Network Node table is used to define information about each node in the Voice Mail network. Before the local node can send messages to a remote node, there must be an entry in the Network Node table for the remote node.

2.4 To leave messages for mailboxes on remote nodes, you must have "network mailboxes" on the local node. If the local node only has network mailboxes associated with specific mailbox numbers, the local node can only leave messages for those mailboxes. If the local node has a network mailbox for a remote node (no mailbox number specified), the local node can leave messages for any mailbox on the remote node (as long as the Validate Off-Node Mailboxes flag is disabled).

Leaving Messages

2.5 Users can leave messages for any network mailbox as either a subscriber or a non-subscriber. Leaving messages for a *direct* network mailbox is just like leaving messages for a regular mailbox. After entering the network mailbox number, Voice Mail plays the mailbox directory name (if it is recorded, otherwise, the mailbox number), followed by the recording instructions, and the caller can leave a message as usual.

2.6 Users can also leave messages using a *node* network mailbox. Users who call the node network mailbox hear a prompt asking them to enter the desired mailbox on the remote Voice Processing Unit. After that step, it is the same as leaving a message for a regular mailbox. Node network mailboxes are useful when you do not know the direct network mailbox number and/or when the mailbox on the remote node does not have a direct network mailbox on the local node. (If the Voice Processing Unit attempts to deliver the message from a local mailbox user to the remote node and the mailbox number is invalid, the message is returned to the sender.)

2.7 If the Validate Off-Node Mailboxes flag is enabled, users can only leave messages for mailboxes on remote nodes that have a direct network mailbox on the local node. If there are no direct network mailboxes that refer to mailboxes on the remote node, the caller hears a prompt indicating that the message cannot be delivered to the remote node.

2.8 Unlike regular mailboxes, network mailboxes (both direct and node mailboxes) do not have their own message queues. When a message is delivered to a network mailbox, the Voice Processing Unit looks up the mailbox's node information, and stores the message locally. The messages are stored until they are transmitted to the remote node or returned to the sender.

Undeliverable Messages

2.9 The Voice Processing Unit delivers messages to another node by calling the System Number of the remote node. If the remote number is busy or there is no answer, the Voice Processing Unit continues attempting to contact the remote node until the remote node answers, or until it has reached the Maximum Network Call Attempts limit (see page 6-415). If the Voice Processing Unit has made consecutive, unsuccessful Maximum Network Call Attempts (due to busy or no answer), it stops attempting to contact the remote node and return any messages pending for the node. The messages are then considered "undeliverable." Undeliverable messages pending for a node are handled according to the Undeliverable Network Messages Destination field. The field can have a value of Delete, Sender, or System Administrator:

- When the field is set to Delete, all undeliverable messages are deleted.
- When the field is set to Sender, all undeliverable messages are returned to the sender's mailbox, if possible. If the sender is unknown, the messages are returned to the System Administrator's mailbox. If the System Administrator's mailbox does not exist, then the messages are deleted. When a subscriber listens to a returned message, he hears a prompt indicating that the message was undeliverable.
- When the field is set to System Administrator, all undeliverable messages are returned to the System Administrator's mailbox, if it exists. Otherwise the messages are deleted.

Examples

2.10 To help understand networking configurations, see the diagram on the next page. In the diagram:

- There are three nodes in the network, Chandler, Denver, and Houston.
- Chandler and Houston both have Voice Processing Unit, and Denver has a non-Inter-Tel Voice Mail that supports AMIS.
- The Voice Mail network depicted in the diagram does not have a universal numbering plan.
- Delivery schedules are not shown.

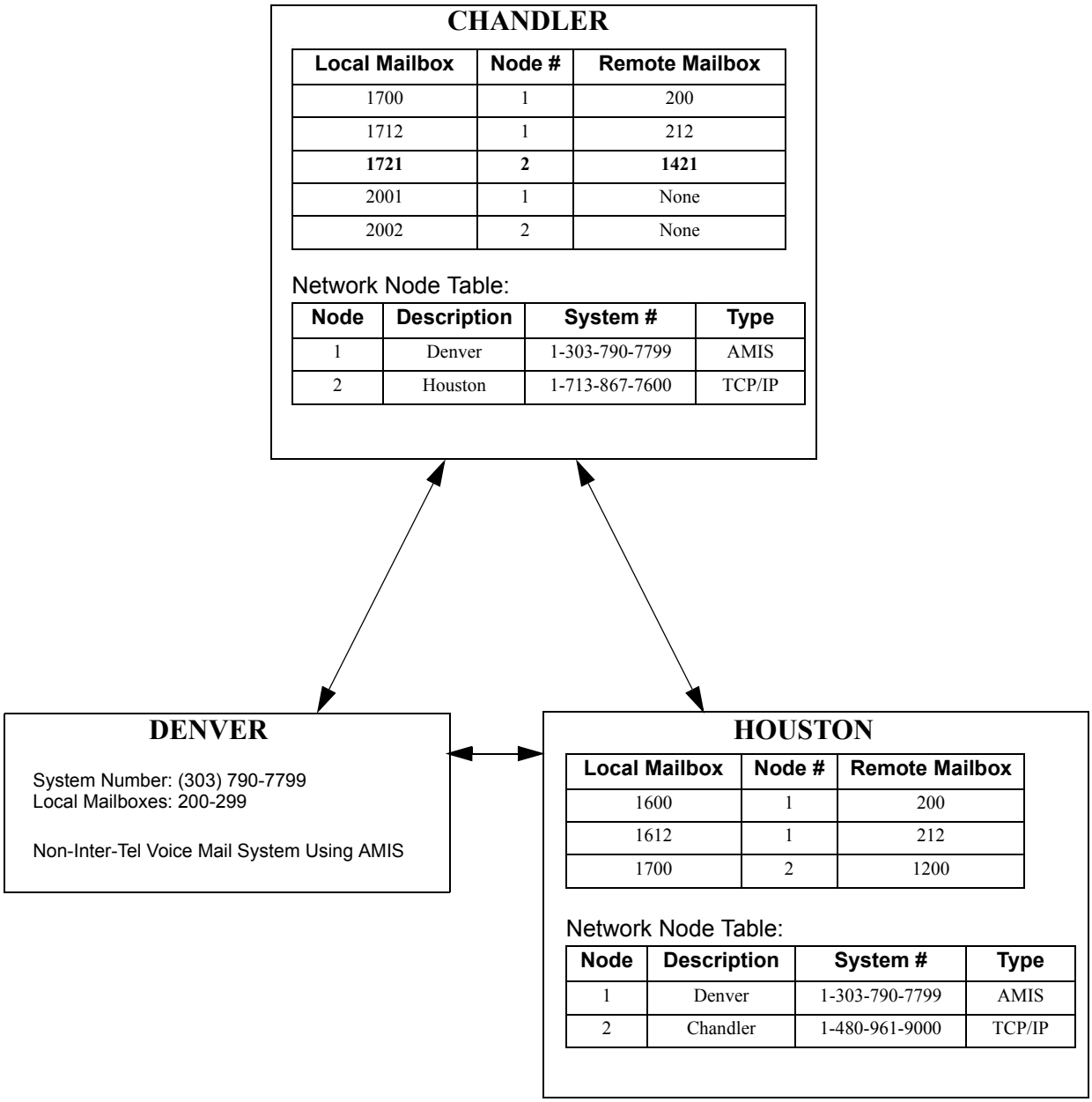
2.11 Look at mailbox 1721 in Chandler. This is a direct network mailbox that refers to mailbox 1421 in Houston. When someone delivers a message to mailbox 1721 in Chandler, the message is transmitted to Houston (via TCP?IP) and delivered to mailbox 1421. Note that if someone in Chandler delivers a message to mailbox 1421, it is delivered to Chandler's local mailbox 1421 and not to Houston's mailbox 1421.

2.12 The Chandler location also has two node network mailboxes, 2001 (for Denver) and 2002 (for Houston). These two mailboxes allow users in Chandler to deliver messages to any mailbox in Denver and Houston, respectively, if the Validate Off-Node Mailboxes flag is disabled. (If the flag is enabled, users in Chandler can only deliver messages to direct mailboxes 200 and 212 in Denver, and 1421 in Houston.)

2.13 Houston can only send messages to two mailboxes in Denver (200 and 212) and one mailbox in Chandler (1200). This is because Houston only has three direct network mailboxes and no node network mailboxes. If Houston had node network mailboxes for Denver and Chandler, users in Houston could send messages to any mailbox in Denver or Chandler (if the Validate Off-Node Mailboxes flag is disabled, as explained above).

2.14 Details about the configuration in Denver are not listed because this node is assumed to have a non-Inter-Tel Voice Mail that supports AMIS networking. The only information Chandler and Houston need to know about Denver are the mailbox numbers (if they want to set up direct network mailboxes) and Denver's system number.

Figure 4-1. *Example of Voice Processing Unit Network*



B. AMIS NETWORKING

2.15 Audio Messaging Interchange Specification (AMIS) is a vendor-independent protocol for networking Voice Mail systems. The Voice Processing Unit supports AMIS-Analog which uses DTMF control signaling and analog message transmission.

2.16 Voice Processing uses Call Routing Announcement applications (local or off-node) to detect and process incoming AMIS calls. The Call Routing Application's greeting must consist of only a 15-second pause (one long and one intermediate pause), because AMIS disconnects if it detects voice.

2.17 AMIS-Analog uses DTMF signaling to exchange control information between the originating system and destination system. Voice Mail messages are transmitted in analog form over the public telephone network. This means that the originating system uses its standard playback function to transmit, and the destination system uses its standard record function to receive. Because messages are transmitted in this manner, message transmission time is exactly the length of the message plus synchronization and control signaling overhead (which is typically less than one minute).

2.18 AMIS-Analog specifies that the destination system terminates the call if it detects a period of silence of ten seconds or more during message transmission. Therefore, messages being transmitted should not contain periods of silence that are longer than ten seconds. They should also not contain any DTMF tones or call progress tones that may cause the destination system to disconnect or stop recording prematurely.

AMIS Call Processing

2.19 An AMIS call is initiated by the originating system calling the destination system over the public telephone network. When the call is connected, the originating system and destination system perform a handshake to establish the AMIS session.

2.20 The originating system then identifies itself with its system number and AMIS protocol version. After acknowledgment by the destination system, the originating system specifies the type of message to be delivered, the length of the message (if available), the originator's mailbox number, and the recipient's mailbox number. The originating system then begins transmitting the previously recorded message by using its standard playback function. The destination system records the message and delivers it to the recipient's mailbox.

2.21 Outgoing AMIS calls are made by the Message Notification/Retrieval application. When a network mailbox that is set up for AMIS networking receives a message, it triggers the Voice Processing Unit to make an outgoing AMIS call to deliver the message.

2.22 The Voice Processing Unit begins by calling the System Number of the remote node. If the remote number is busy or there is no answer, the Voice Processing Unit continues attempting to contact the remote node until the remote node answers, or until it has made the Maximum Network Call Attempts. If the Voice Processing Unit has made consecutive, unsuccessful Maximum Network Call Attempts (due to busy or no answer), it stops attempting to contact the remote node and returns any messages pending for the node.

2.23 If the remote node answers the call, the Voice Processing Unit establishes the AMIS session with the remote node. After a successful handshake, the Voice Processing Unit starts the AMIS session. (If this is unsuccessful, it disconnects and tries again.) If the destination system successfully acknowledges the AMIS session, the message transmission process starts.

2.24 The Voice Processing Unit begins the message transfer process by transmitting the message type (new or reply), message length (if available), the originating mailbox, and the destination mailbox. If the remote system responds with an unsuccessful acknowledgment, the Voice Processing Unit returns the message. If the Voice Processing Unit has not reached the per-call message limit (nine messages per standard AMIS-Analog), and there are more messages, it continues with the next message. Otherwise, it ends the AMIS session and disconnects.

Incoming Calls

2.25 Call Routing Announcement applications are the only applications that can receive and process incoming AMIS calls. If any other application answers an incoming AMIS call, the application eventually times out. If a Call Routing Announcement application answers an incoming AMIS call, the application's Accept Network Calls flag must be enabled for it to process the AMIS call. If it is not enabled, the call times out.

2.26 After the Voice Processing Unit sets up the AMIS call, it performs the following tests to see if it can deliver the message:

- Checks to see if there is enough disk space to receive the message.
- Checks to see if the destination mailbox exists.
- Checks to see if the destination mailbox is full.

2.27 If any of the tests fail, the Voice Processing Unit sends the appropriate response to the originating system and then waits for the next message. If all the tests pass, it begins recording the message. The Voice Processing Unit continues recording the message until it detects DTMF tones or at least ten seconds of silence, or until the recording exceeds the maximum message length. If it detects ten seconds of silence or if the recording exceeds the time limit, it disconnects. If it detects DTMF tones during the recording, and the DTMF tones are not the End Message signal, then it ignores the tones and continues recording.

2.28 When the Voice Processing Unit receives the End Message frame, it delivers the message to the destination mailbox and waits for the next frame. If the next frame is the End Session frame, it disconnects. If the next frame is another Message Information frame, it verifies that it has not received the maximum number of messages during this call. If it has, it sends an error response and disconnects. If it has not, it begins receiving the next message.

AMIS-Analog Protocol Limitations

2.29 These limits are enforced when the Voice Processing Unit places calls to or receives calls from another Voice Processing Unit or a non-Inter-Tel Voice Mail system that supports AMIS-Analog.

- Maximum of nine messages per call.
- Maximum of one recipient per call. If the same message is to be delivered to multiple mailboxes, the message must actually be transmitted multiple times.
- Maximum message length is eight minutes.
- Certified delivery of messages is not supported.
- Priority message indication is not supported.
- Private message indication is not supported.

- Full duplex message flow is not supported. After the originating system has transferred its messages to the destination system, if the destination system has pending messages for the originating system, it cannot transmit them during the same call. The destination system must make a separate call to transfer its pending messages. In other words, only the system that originates the call can transfer messages during that call.
- The retransmission of data frames in the event of an interdigit time out is not supported.

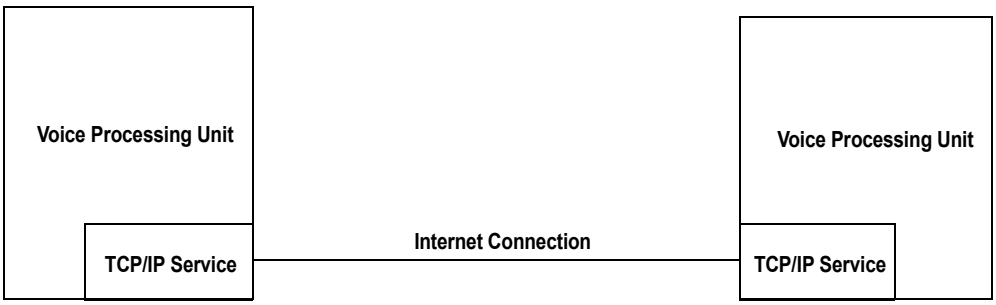
Software Configuration

2.30 The following list describes the programming requirements necessary for AMIS networking.

- **System\Premium Features**: The Voice Processor AMIS Networking software license is required.
- **System\Devices and Feature Codes\Stations**: There must be at least one off-node device for a station on the remote node programmed for AMIS networking. This off-node device is required before creating the network mailbox below.
- **Voice Processor\Devices\Applications**:
  - There must be a Message Notification/Retrieval Application programmed with outgoing access.
  - There must be a Call Routing Announcement application that has the Accept AMIS Calls flag enabled. The Call Routing Application's greeting must consist of only a 15-second pause, because AMIS disconnects if it detects voice.
- **Voice Processor\Devices\Nodes**: The node table must contain an entry that is programmed for AMIS networking.
- **Voice Processor\Devices\Mailboxes**: There must be at least one network mailbox on the local Voice Processing Unit that refers to a network node programmed for AMIS networking. For example, if entry 5 in the Network Node Table is programmed for AMIS networking, then at least one network mailbox must have node 5 in its Network Node Number field.
- **Voice Processor\Networking\AMIS**:
  - The IC or CO System Number field must be programmed to allow the Voice Processing Unit to identify itself when establishing an AMIS session and to allow the remote AMIS system to be able to return calls to the local Voice Processing Unit.
  - The Accept Unknown Network Calls flag must be enabled (or there must be a programmed node for each remote node from which the local Voice Processing Unit expects to receive AMIS calls). If the Accept Unknown Network Calls flag is enabled, the local Voice Processing Unit can accept calls from any Voice Mail system that supports AMIS networking. If this flag is disabled, the local Voice Processing Unit can only accept network calls from nodes that are programmed in the network node table.

C. TCP/IP NETWORKING

2.31 Messages are forwarded, one or more at a time, using the TCP/IP protocol. The TCP/IP service is added to the originating and destination Voice Processing PCs. A temporary TCP/IP session from the originating system to the destination system is established, and messages are forwarded. See the illustration below.

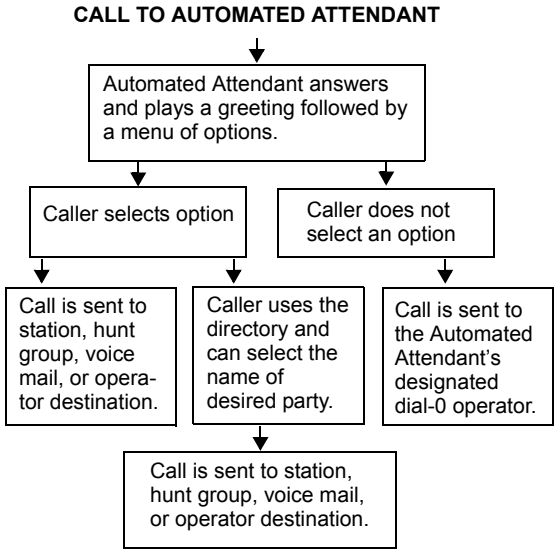


2.32 To set up a TCP/IP Network:

1. Set up the TCP/IP Service as described in the INSTALLATION section (page 3-229).
2. Make sure the Inter-Tel Messaging IP Networking premium feature is enabled.
3. In the Voice Processor\Devices\Nodes directory, right-click to create your nodes. A warning appears that states you must restart DB Programming. When you have created all of the nodes, select **Yes** and then **OK** to end the DB Programming session.
4. In the Session Manager, restart the DB Programming session.
5. In the Voice Processor\Devices\Nodes directory, you should see the newly created Voice Processing Node. Program the description.
6. Select **Network Type** and scroll to TCPIP.
7. Select **System Number** next to the description you just programmed. This system number, for this method (TCP/IP), is the IP number for the Remote Node Voice Processing Unit. Remember, this is TCP/IP Voice Mail networking, and both Voice Processing PCs are on a TCP/IP Network. If you do not know the IP number, you must get it.
8. Create off-node associated and/or non-associated mailboxes in Devices\Mailboxes\Remote Node.

3. AUTOMATED ATTENDANT

3.1 The automated attendant is a programmable feature that can be used to provide an automated call answering service. Calls can transfer, forward, or directly ring in to an automated attendant. A Voice Processing Unit is required for the Automated Attendant feature on the Inter-Tel system. Calls to the automated attendant application are processed as follows.



3.2 When an automated attendant answers a call, it plays a recording that gives dialing instructions. During or after the recording, the caller may then directly dial a station extension number, Voice Mail access number (if there is no associated mailbox), or hunt group pilot number. Or, the caller may use the directory to look up the desired extension.

3.3 In a network setting, a trunk on another node can ring in to a Voice Processing application. (See page 5-87 in FEATURES and page 6-237 in PROGRAMMING for trunk group information.)

3.4 When the automated attendant answers a call, the caller hears the company greeting, followed by instructions and the list of available options. The caller then has the following options:

- **Dial a station extension number**: If an extension number is dialed, the call is transferred to the selected station. If ringback tones are enabled, the caller hears ringing while the call is being transferred. If ringback is not enabled, the caller hears music (see PROGRAMMING, page 6-237). If the called station is forwarded, the call follows the programmed forward.
- **Dial a hunt group number**: When a hunt group number is dialed, the call is transferred to the selected hunt group. The call rings or circulates according to how the hunt group is programmed (linear or distributed). If ringback tones are enabled, the caller hears ringing while the call is being transferred.
- **Dial the Voice Mail application's extension number**: The caller can reach the Voice Mail main greeting by dialing the application's extension number (access number) assigned to the Voice Mail feature. The caller can then leave a message as a non-subscriber or access any of the Voice Mail subscriber features. (See page 4-36 for Voice Mail information.)

- **Use the directory**: If the caller does not know the extension or mailbox number of the desired party, the caller can spell the name using the keypad buttons and "look up" the number in the directory. (This option can be disabled in the database. Or, if there are no names recorded for the individual mailboxes or for the system's extension IDs, this option is not provided.) Directory names can be sorted by first or last name. (See page 4-62 for information about using the directory.)
- **Dial the operator access destination**: If the caller needs further assistance, dialing 0 accesses the Voice Processing Unit's programmed operator destination. Or, if the caller is on a rotary telephone and cannot enter a digit, the call is automatically transferred to the operator destination. (The operator access destination is programmed in the database. There can be separate destinations for day and night modes.)

NOTE: Due to the natural characteristics of the trunk, the volume level of DTMF tones transmitted over the trunk may be substantially reduced before reaching the phone system and Voice Processing Unit. This natural degradation in tone volume may adversely affect the reliability of the Automated Attendant feature. Other factors which can affect automated attendant performance are trunk noise and the quality and strength of the DTMF tones generated by the off-premises phone itself.

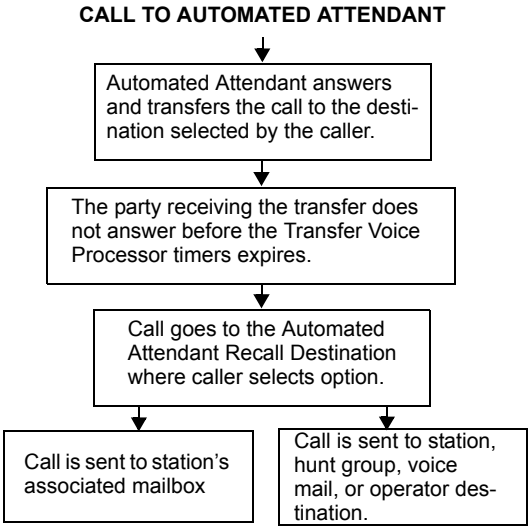
A. AUTOMATED ATTENDANT APPLICATIONS

3.5 There are a number of different uses for this feature. Below are two examples:

- Direct ring-in calls to a busy attendant could be forwarded to an automated attendant (using the Call Forward If Busy feature or system forwarding).
- Calls could ring in directly to an automated attendant application's extension number when the system is in day and/or night mode.

B. AUTOMATED ATTENDANT RECALL DESTINATION

3.6 When a station receives a call that has been routed through the automated attendant, the call is handled as a transferred call, and the display shows TFR FROM *(description)*. If the call is not answered before the Transfer Voice Processor timer expires, the call recalls the automated attendant's recall destination. The recall destination is usually the Automated Attendant Recall Destination application which announces that the called station is unavailable and allows the caller to choose to leave a message (if the station has an associated mailbox) or dial another extension number.



3.7 If the Recall Destination fails to answer a call, it is automatically sent to the recall destination's programmed attendant. If the call is not answered there, it is disconnected after the Abandoned Call timer expires.

3.8 If an invalid number is dialed, the caller is prompted to enter another number. If a caller fails to make an entry before the Inactivity Alarm timer expires, the caller is prompted again to make an entry.

3.9 The caller cannot access trunks or enter feature codes through the Automated Attendant application. Trunk access codes and feature codes are considered invalid numbers.

C. AUTOMATED ATTENDANT CUSTOM AUDIOTEX RECORDINGS

3.10 Custom Audiotex recordings are made using the Voice Mail System Administrator's mailbox as described on page 4-58. Each recording is associated with a recording number and assigned to the application(s) in DB Programming or using the System Administrator's mailbox. (See page 4-60 in this section and page 6-353 in PROGRAMMING.)

D. CALL SCREENING

3.11 Calls transferred from the Automated Attendant or a Call Routing Announcement application can be screened, announced, or unannounced.

3.12 When a call is received by an automated attendant or Call Routing Announcement application, and the caller enters an extension number, the station's programmed Transfer Method determines how the call is transferred. The Transfer Method flags can be programmed in the database. If allowed in mailbox programming, they can also be programmed by the mailbox user. The available Transfer Methods are as follows:

- **Announce Only**: The caller is asked to state his or her name. Then the call is transferred to the associated extension number. When the station user answers the transfer, the application plays the caller's name and completes the transfer.
- **Screened**: The caller is asked to state his or her name. Then the call is transferred to the associated extension number. When the station user answers the transfer, the application plays the caller's name. The station user has the options of replaying the name, sending the call to Voice Mail (if the extension has a mailbox), transferring the call to another extension, accepting the call, or rejecting the call.
- **Unannounced**: The call is transferred to the associated extension number after the Voice Processing Unit checks the station to determine its status (busy, available, ringing, etc.). This is the default method.

3.13 In a network setting, the Voice Processing Unit can provide call screening for a destination extension on another node. However, the node where the Voice Processing Unit is connected must have an off-node device programmed for the destination extension and access to the remote node.

3.14 See page 5-211 for instructions on receiving announced or screened calls. See page 4-54 for subscriber mailbox programming.

E. EXTENSION ID

3.15 Extension IDs are used in conjunction with transferring calls through the Automated Attendant or using the Extension Directory. The extension ID allows callers to be transferred to stations and applications that do not have mailboxes. It also allows the station or application to have a recorded name in the directory. This feature is set up and initialized using DB Programming and Voice Mail features.

3.16 If an Extension ID has been created in DB Programming for a station extension number or a Call Processing Card (CPC) modem, either the principal owner of the extension or the Voice Mail System Administrator must set up (initialize) the ID with a new password and record a name for use in the Extension Directory. At default, the extension ID password is the same as the extension number.

3.17 If the Extension ID has not been "initialized," calls can still be transferred to the associated extension number. However, they cannot be accessed from the directory. If the Extension ID's user name has not been recorded, it cannot be heard when callers access the Extension Directory. The name must be recorded in order to fully initialize the Extension ID. See page 4-45 for more information on initializing Extension IDs.

NOTE: To provide system security, ***all*** extension IDs should have a password. To make the passwords difficult to guess, they should not match the station's extension number or consist of one digit repeated several times. The default password should be changed the first time the user logs in.

3.18 After a password has been set up and the name recorded, the extension owner may access Extension ID Options which allows the associated directory name and password to be modified.

3.19 If an Extension ID has not been created for a station, callers using the automated attendant cannot be transferred to that destination. Instead, these callers receive a system recording notifying them of an invalid entry and are routed back to the automated attendant's main menu.

3.20 Individual extension IDs can be programmed as "Private" in DB Programming. A Private number is included in the directory and can be dialed if the caller knows the number, but the extension number information is not available to the caller; only the name is played.

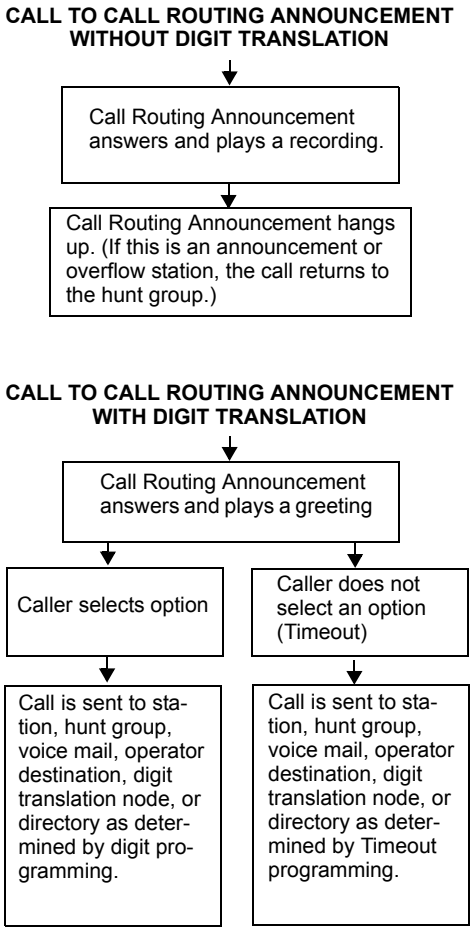
3.21 Extension IDs can also be programmed as "Unlisted." That means that the number is not included in the directory but can be dialed if the caller knows the extension number.

3.22 Calls transferred from the Automated Attendant or a Call Routing Announcement application to stations with extension IDs can be screened, announced, or unannounced. Programming flags determine the methods used for transferring calls to stations with extensions IDs. See page 4-19 for details.

3.23 In a network setting, the Voice Processing Unit cannot create extension IDs for off-node stations included in wildcard ranges. Each associated station must have its own off-node device entry. (See Appendix A — Networking for more information.)

4. CALL ROUTING ANNOUNCEMENT

4.1 The Call Routing Announcement application can be used as a simple playback device that plays a message and then hangs up to disconnect the call. This is especially useful for programming hunt group announcement and overflow stations. Or, the Call Routing Announcement application can use digit translation which allows the caller to press a single digit for access to a station, hunt group, or mailbox.



NOTE: The phone system supports a Primary and Secondary Language which can each be programmed as American English, British English, Japanese, or Spanish (Mexico City Spanish Dialect for VPU and EVMC, American Spanish Dialect for EM). Digit Translations, therefore, can be programmed to use either language.

4.2 When a Call Routing Announcement application is used as a hunt group announcement or overflow station, calls to the application automatically stop circulating through the hunt group if the caller selects a valid digit translation option. This allows the application to send the call to other stations without the call being "pulled back" into the hunt group when a hunt group station becomes available and answers the call. However, if the caller does not dial a valid digit translation option, the call is pulled back if a hunt group member answers.

4.3 Here are some tips for designing an effective Call Routing Announcement application:

- Design with the caller in mind, not just the information you want to include.
- Keep menus as simple as possible, with four or fewer options per menu.
- Number options sequentially and do not skip numbers. List "transfer to operator" last.
- Use consistent digits for options, such as 1 for Yes, 2 for No, and 0 for the operator.
- State the option before the digit. For example, say, "For account information, press 1," instead of, "Press 1 for account information."
- Draw a map of your arrangement to avoid "dead ends" or endless loops.
- Take advantage of Caller ID [CLIP] and DNIS to route calls to suitable menus.
- Keep recordings short (under 60 seconds) and do not use jargon.
- Give the most frequently requested information in the first ten seconds, without requiring the caller to press a digit.
- Make sure the recordings are clear and the voice is consistent from prompt to prompt. Avoid heavy regional accents.
- Do not repeat the main greeting on any other level.
- Make seasonal changes when necessary, but keep menu options the same (callers get used to them).
- Include an option for overriding the Primary Language. For example, say, "For English, press 1. Para Español, empuje 2."
- Test your application any time you make a change. Listen to your prompts periodically.

4.4 **To program a Call Routing Application**:

1. Create the Call Routing Application and enter its description and user name.
2. Change the day and night greetings to custom recordings. Remove the default recordings and assign new recording numbers. (Write down the recording numbers and their assignments. You record them later.)
3. If using Digit Translation, create the digits and nodes you are using.
   a. Give them appropriate descriptions and usernames (for example: Sales 1, Service 2, Repair 3, etc.)
   b. Go back to the Applications Programming screen and expand the Call Routing application that you have just created.
   c. Program each digit or node individually and set the greetings as desired. Repeat this step for all nodes and digits.
4. Use the System Administrator's mailbox to record your custom greetings.
5. Test the application and all of its nodes by calling the application and checking each digit and node.

A. CALL ROUTING CUSTOM AUDIOTEX RECORDINGS

4.5 When accessed, the Call Routing Announcement application plays a custom Audiotex recording and then hangs up.

4.6 Call routing announcements are recorded using the Voice Mail System Administrator's mailbox as described on page 4-58. Each announcement is assigned to use a "custom greeting" number in DB Programming. (See page 6-353 in PROGRAMMING.)

4.7 A Call Routing Announcement application message can be programmed to include the caller's queue position and/or estimated wait time. The queue position announcement tells the caller how many calls are ahead of his or her call. This includes calls being served and waiting calls. See page 5-35 for details.

4.8 Recordings for fax documents should include all dialing instructions. The recording should state if documents can be selected by number and/or list all options. If fax delivery times are set to specific days or times, the recording should also include this information.

B. CALL SCREENING

4.9 Calls transferred from the Automated Attendant or a Call Routing Announcement application can be screened, announced, or unannounced. Separate programming flags determine the methods used for transferring calls to stations with mailboxes, stations with extension IDs, and extensions without mailboxes or IDs. See page 4-19 for details.

C. DIGIT TRANSLATION

4.10 Digit translation allows callers to dial a single digit to access a designated extension number, mailbox, or hunt group. Up to 12 digit translation storage locations (#, *, 0-9) are available for each Call Routing Announcement application (see PROGRAMMING page 6-371).

4.11 To use digit translation, the Call Routing Announcement application must have special programming. Digit translation is programmed by assigning specific "Actions" for each digit and determining what happens if the caller does not enter a digit (Timeout). The Timeout period is determined by the length of the pause at the end of the greeting. The Actions that can be selected are as follows:

NOTE: The Fax actions require the Fax-On-Demand feature. For Fax-On-Demand setup information, see page 4-29.

- **Transfer To Operator**: Transfers the call to the Dial-0 Destination that is programmed in the Voice Processing database.
- **Company Directory - First Name**: Sends the caller to the directory prompt that asks the caller to enter the first name of the desired party.
- **Company Directory - Last Name**: Sends the caller to the directory prompt that asks the caller to enter the last name of the desired party.
- **Subscriber Access**: Sends the caller to the mailbox prompt that requests a mailbox number.
- **Invalid**: (Not used for the Timeout.) The digit is not used. Callers who press this digit hear a recording that tells them that it is invalid.

- **Select Fax Document**: (Not used for Timeout and not available if there are no fax documents in the system.) Selects a specific document for faxing. Callers should be prompted to select this digit with a recording that tells them the name of the document. For example, "For a price list, press 2." This Action must be accompanied by an End Fax Selection Action and should also be accompanied by a Cancel Fax Selection Action.
- **Select Fax By Document Number**: (Not available for Timeout or digits * and #.) Is similar to the Transfer to Collected Extension Action described below. It allows the caller to enter the exact four-digit number(s) of the desired document(s). Assign this Action to the digit(s) that correspond to the first digits of the document numbers. For example, if you have documents numbered 0001-1000, digits 0 and 1 should have a Select Fax By Document Number Action. Callers can then dial four-digit document numbers that begin with 0 or 1. This Action must be accompanied by an End Fax Selection Action and should also be accompanied by a Cancel Fax Selection Action.

NOTE: When document numbers begin with 0, the caller can select documents by entering the number without 0 and pressing #. For example, the caller can enter 99# for document number 0099.

- **End Fax Selections**: (Usually assigned to the digit #.) Used for signaling the system that the caller has entered all desired fax document selections (either through a Select Fax Document By Number Action or a Select Fax Document Action). When the caller presses this digit, the system checks to see if any documents have been requested. If not, it issues a warning to the caller and returns to the greeting. If documents were selected, the system requests fax dialing information from the caller.
- **Cancel Fax Selections**: (Usually assigned to the digit *.) Used for canceling fax document selections that a caller has entered. If the caller has not made any entries, the system notifies the caller that no selections were made and returns to the greeting.
- **Transfer To (Digit Translation) Node**: (Not available for Timeout and not available if there are no programmed digit translation nodes.) Sends the call to a digit translation node that allows access to further digit translation options as described in paragraph 4.14.
- **Transfer To Extension**: Sends the call to the designated extension (station, hunt group, trunk, trunk group, modem, or application).
- **Transfer To Mailbox**: (This does not appear if there are no programmed mailboxes.) Sends the call to the designated mailbox.
- **Transfer To Collected Extension**: (Cannot be used for Timeout, #, *, or 0.) Allows callers to dial extension numbers of stations and hunt groups that have a mailbox or extension ID. Assign this Action to digits that correspond to the first digits of extension numbers. For example, if digit 1 is "Transfer To Collected Extension," callers can dial extension numbers that begin with 1. However, if digit 1 is "Transfer To Extension 2000," a caller attempting to dial a station extension number that begins with 1 is instead transferred to 2000.
- **Hang Up**: (Used for Timeout only.) Disconnects from the call if the user does not enter a digit.

4.12 After the application is created and programmed, the Voice Mail System Administrator's mailbox is used to make a custom Audiotex recording that explains the options to callers, as described on page 4-60. The custom Audiotex recording is then assigned to the Call Routing Announcement application. A caller who listens to the Call Routing Announcement recording can dial the single-digit location number to reach the designated destination. For example, if the hunt group extension for a customer support group is programmed for digit 3, the recording should say something like, "For customer support, press 3."

4.13 The system provides a choice between Primary and Secondary Language prompts and displays. The available languages are American English, British English, Spanish, and Japanese. Note that Spanish voice prompts use Mexico City Spanish Dialect for VPU and EVMC, and American Spanish Dialect for EM. The system selects the language to use for each call, as determined by the trunk, station, and Voice Processing programming. When using a Call Routing Announcement application with digit translation, the individual Voice Processing applications assigned to the digits can be programmed to override the device language and provide prompts in one language only. Using this method, you can program a Call Routing Announcement tree that offers callers a choice between languages (Primary and Secondary). See page 4-27 for more information.

D. DIGIT TRANSLATION NODES

4.14 A "digit translation node" is a digit translation destination that allows further digit translation options. Unlike extension and mailbox destinations, a digit translation node does not move the call to a specific location. Instead, it offers additional dialing options. A digit translation node can use any of the Actions, including transfers to other digit translation nodes.

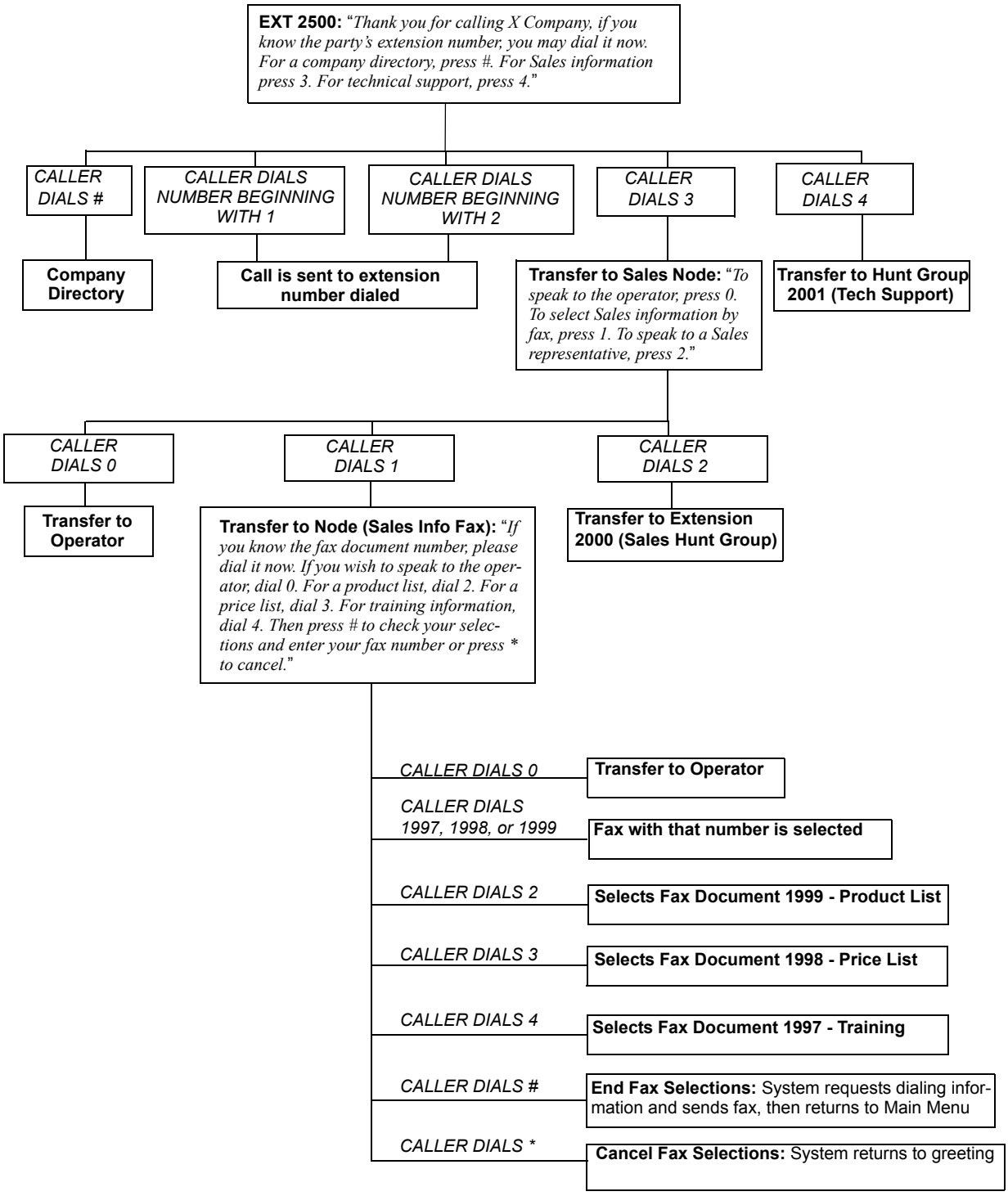
4.15 The use of digit translation nodes allows the programmer to create a Call Routing Announcement with several layers to form a digit translation "tree." The tree can be made up of digits with destinations and/or digit translation nodes with additional translations. There can be up to 200 digit translation nodes in the system and up to 20 per Call Routing Announcement application. An example of a Call Routing Announcement application with a multi-layered tree is shown on the next page.

4.16 After a digit translation node is created, it can be assigned to more than one Call Routing Announcement application. This allows entire digit translation node hierarchies to be shared or moved without reprogramming. A digit translation node can only be deleted if it is not associated with a Call Routing Announcement application.

4.17 Digit translation nodes can be programmed to use either the Primary or Secondary Language.

4.18 See "Applications" on page 6-356 for information on creating the Call Routing Announcement applications and programming digit translation nodes.

Figure 4-2. *Sample Call Routing Announcement Tree*



5. MULTILINGUAL CAPABILITY

NOTE: This feature requires the Multilingual Support software license. This prevents unintentional language changes by users in an all-English system.

5.1 The system provides a choice between American English, British English, Spanish, and Japanese prompts and displays. Note that Spanish voice prompts use Mexico City Spanish Dialect for VPU and EVMC, and American Spanish Dialect for EM. The system selects the language to use for each call, as determined by the trunk, station, and Voice Processing programming.

NOTE: If Secondary Language language prompts are not installed, yet a user attempts to access them, Primary Language prompts are delivered instead. Also, only Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones can be used for viewing Japanese prompts. Analog phones do not support Japanese displays.

5.2 When a Voice Processing application receives a call from a station or trunk, the system tells the application which language is programmed for that device. For example:

- If a trunk programmed for the Primary Language rings in to an application, the Voice Processing Unit plays the Primary Language prompts.
- If a station programmed for the Secondary Language calls Voice Mail, the station user hears Secondary Language prompts.
- If a station programmed for the Primary Language receives a call on a trunk that is programmed for the Secondary Language, and then forwards the call to Voice Mail, the caller hears Secondary Language voice prompts.

5.3 When using a Call Routing Announcement application with digit translation, the individual Voice Processing applications assigned to the digits can be programmed to override the device language and provide prompts in one language only. Using this method, you can program a Call Routing Announcement tree that offers callers a choice between languages. For example, in a system that uses English and Japanese:

- The Call Routing Announcement application could have a greeting that says, “Thank you for calling. For English prompts, press 1. NIHONGO WA, 2 WO OSHITE KUDASAI (for Japanese, press 2).”
- The digit translation for digit 1 would lead to a digit translation node that overrides the calling device’s programming and uses only English prompts.
- The digit translation for digit 2 would lead to a digit translation node that overrides the calling device’s programming and uses only Japanese prompts.

5.4 In the example above, the digit translations could be digit translation nodes that lead to various other English-only or Japanese-only applications. Or, the first level can give more choices, such as English Voice Mail or automated attendant and Japanese Voice Mail and automated attendant. Either way, the individual applications or digit translation nodes can be programmed to play only one language or they can use the device’s language, as needed.

5.5 Multi-Language Feature on Network Operation

5.6 Nodes do not have to match the languages because the Voice Mail system installs any existing language prompt, not just the languages designated as Primary or Secondary. Therefore, for calls across network nodes, if the language does not match between the two nodes, the called Voice Mail system searches for the designated language in its system in the following order: Primary language, Secondary language, American English, British English, Japanese, and Spanish (Mexico City Spanish Dialect for VPU and EVMC, American Spanish Dialect for EM) and uses the designated language. If the designated language does not exist in the Voice Mail system, the system uses the default Primary language instead.

5.7 For more information about multilingual capabilities, see page 5-160 in SYSTEM FEATURES.

6. VOICE PROCESSING FAX-ON-DEMAND

6.1 The Fax-On-Demand feature provides fax services to callers. It is a specially programmed Call Routing Announcement application that uses digit translation to allow callers to select the documents they want to have faxed to them.

6.2 With Fax-On-Demand, callers can use a DTMF telephone to request one or more documents from the company's fax library. (This library of fax documents is also used in the Unified Messaging faxing feature.) When the request is completed, the Voice Processing Unit places a call to the caller's fax machine to deliver the requested documents.

A. HOW FAX-ON-DEMAND WORKS

6.3 The Fax-On-Demand process is shown in the following steps:

1. The caller reaches the Call Routing Announcement application either through direct ring-in or transfer.
2. After listening to the introductory recording, the caller selects a document by entering a document number or using digit translation, depending on digit translation node programming. If the document is selected by number, the system asks the caller to verify the entry by pressing #. If the number is invalid, the system alerts the caller and returns to the introductory recording.
3. The system checks the availability of the document and one of the following occurs:
   - *If the document is available,* the system tells the caller how many documents have been selected and how many more documents can be requested (as determined by the Maximum Selections Allowed value in the database). It then returns to the recording to allow the caller to select more documents or end the selection process.
   - *If the document has been deleted or is being updated,* the caller is told that the document is unavailable and returns to the recording.
4. If desired, the caller selects additional documents, and the system checks for their availability as described above. If the maximum number of allowed documents has been reached, the system informs the caller and gives the option of returning to the greeting or ending the selection process. If a caller selects the same document more than once, the system informs the caller of the error and allows another selection.
5. If desired, the caller can cancel the document selections at this point by pressing the Cancel Fax Selections digit (usually *). However, once the next steps are completed, the faxes are delivered.
6. When the caller ends the selection process by pressing the End Fax Selections digit (usually #) or reaches the allowed maximum number of documents, the system requests fax dialing information from the caller.
7. The caller enters and verifies his or her extension number (to be printed on the cover sheet) and the caller's fax machine telephone number.

8. The system checks the entered number and one of the following occurs:
   - *If the number is valid,* the system tells the caller when the document are delivered (this depends on the programmed fax delivery times in the Voice Processing database) and returns to the recording.
   - *If the number cannot be dialed by the Message Notification/Retrieval application due to toll restrictions,* the system tells the caller that the number cannot be dialed and asks for a new number.
   - *If the caller enters an invalid number,* the system tells the caller the number is invalid and asks for a new number.
9. The caller hangs up or returns to step 4. to select more documents. (If more documents are selected and canceled, the documents that were selected and accepted previously are still sent. Each time the caller accepts the documents and enters the dialing information, the documents are prepared for transmittal. They cannot be canceled after that point.)
10. The Message Notification/Retrieval application places a call to the caller's fax machine as soon as a fax port is available, unless one of the following occurs:
    - *If the fax was requested outside of the programmed fax delivery times,* the system waits until the Fax Delivery Start Time on an allowed day of the week before it attempts to send the document. The Fax Delivery Start and Stop times determine when faxes can be sent. Start and stop times can be set to any time period, up to 24 hours. The programmer can also determine which days of the week faxes are sent.
    - *If other calls are going out and the programmed Maximum Number of Outgoing Calls has been met,* the system waits for an available outgoing call to send the fax.
    - *If a fax port or trunk is not available or if a requested document is not available,* the system delays delivery until the resource or document is available.
    - *If a requested document has been deleted,* the system completes the delivery, but indicates on the cover sheet that the document is no longer available.
    - *If a fax delivery fails after one or more pages have been transmitted,* the system remembers which pages were not sent and tries again to send only those pages. For example, if the transmission failed in the middle of page 3 of 5, the system would attempt to send only pages 3, 4, and 5 on the retry. The cover sheet on the retry says, "This fax is the continuation of..." and the description for the fax includes "continued" in the title.

6.4 Documents are stored in the fax library using the System Administrator's mailbox. To do so, a fax machine places a call to the System Administrator's mailbox and sends the document. (See page 4-61 for instructions.)

6.5 Fax-On-Demand transmits a cover sheet with each faxed document that includes the recipient's extension number, your company name (as programmed in the database), the date, the number of pages to be sent, and a list of included documents. If you wan, you can store a company logo to be used on the cover sheet. The logo can be up to 5.5 inches tall; if it is larger, the system automatically cuts it off at the 5.5 inch mark. The logo document is stored, like other fax documents, using the System Administrator's mailbox.

6.6 If you have several fax documents, you may want to create and store a "catalog" document that callers can request before continuing their selections. For example, the catalog can include the document number, title, description of the contents, number of pages, and last revision date for each document.

6.7 Fax documents can be saved and restored using the Voice Processing Save And Restore database program. The system Save And Restore operation has no effect on fax documents. However, if the system database is defaulted, all fax documents are deleted.

6.8 The digit translation node used for fax document selection should have a greeting that explains how to select documents. For example, it can say something like, "Welcome to Fax-On-Demand. For a product list, press 1. For a price list, press 2. When you have made your selections, press # to continue. Or, to cancel your selections, press *." Or, you can use a series of digit translation nodes that break the document selection into categories and/or allow callers to dial individual document numbers. If fax delivery times are set to specific days or times, the introductory recording should also include this information.

6.9 Note that the Fax-On-Demand feature with a U.S. system uses User Group 1 and the Home Area Code for toll restriction information. Ensure that these are programmed correctly for outgoing fax delivery calls. Also ensure that the Message Notification/Retrieval application has the appropriate trunk access for placing outside calls.

B. FAX-ON-DEMAND SETUP

6.10 To set up Fax-On-Demand on an Inter-Tel system:

1. **Fax card installation**: Make sure the Fax card is installed in the Voice Processing PC, the DIP switches are set correctly, and the card is connected to the AIC-to-VP ribbon cable.
2. **Program Fax-On-Demand Library Size**: In Fax-On-Demand Programming, set the Maximum Fax-On-Demand Library Size to an appropriate number. (You adjust the other settings later.)
3. **Import documents**: Using the System Administrator's mailbox and a fax machine, import the documents that are used by the Fax-On-Demand feature. (See page 4-61 for instructions.) The first document should contain your logo. The second document should be a list of available documents that callers can use as an index. Make note of the numbers assigned to each of the documents.
4. **Program the document details**: In Fax-On-Demand Programming, view the list of fax documents you have just created. Give each document a short description and a detailed description. If you need to renumber them to match the index document, do it now.
5. **Configure the Fax-On-Demand feature**: In Fax-On-Demand Programming, set the Allow International Calls flag, Start/Stop Time, days of the week, Outgoing Access, Format - Local Fax ID, Format - Company Name, and Format - Logo Document number to the proper settings.
6. **Set the miscellaneous information**: In Fax-On-Demand Programming, set the timers to the appropriate settings for your installation.
7. **Program the Message Notification/Retrieval application**: If you have not already done so, create a Message Notification/Retrieval application. Then, in Trunk Group Programming, ensure it has access to an outgoing trunk group. If desired, you can set toll restrictions for it in Voice Processor - Devices - Applications - Class of Services.

8. **Program the Call Routing Announcement application**: In Voice Processor\Devices\Applications, create a Call Routing Announcement application that allows access to the Fax-On-Demand feature.

   a. Program a description and username.

   b. Program the digit translation. For example:

      # = End Fax Selection
      * = Cancel Fax Selection
      0 = Select Fax by Document Number
      1 = Select Fax Document (first document)
      2 = Select Fax Document (second document)
      3-9 = Select other fax documents
      Timeout = Transfer to Operator

   c. Use the same greeting for day and night modes. Assign the custom recording number to the greeting that you record later. The greeting includes instructions for using the Fax-On-Demand feature and a pause to allow the caller to respond, and it may include a list of available documents.

9. Using the System Administrator's mailbox, make the custom Audiotex recording assigned above. (See page 4-58 for instructions.)

10. Call in to the application and fully test it.

7. RECORD-A-CALL

NOTE: This feature requires the Record-A-Call software license.

7.1 If the Voice Processing Unit has a Record-A-Call application, the users can enter a feature code to record ongoing calls in their designated Record-A-Call mailboxes. Users can retrieve the recorded messages later, just as they would any other mailbox message.

7.2 The Record-A-Call mailbox records the call as a Voice Mail message. All parties are included in the recording. If desired, the Record-A-Call mailbox can be programmed to play a message announcing that the Record-A-Call feature is in progress. There can be separate messages for day and night modes.

7.3 A Record-A-Call tone can be programmed to alert callers at the beginning of the recording. Also, it can be programmed to beep periodically throughout the recording. Record-A-Call tone is enabled at default.

7.4 There are two ways a station can be programmed to use this feature:

- The station can use its personal mailbox, or any other mailbox, as the assigned Record-A-Call mailbox. No other mailbox can be selected. (This is the default programming for all stations.)
- The station can be programmed with a default mailbox but with the option of selecting a different mailbox. If the user chooses to not enter a mailbox number, the system automatically selects the default mailbox.

7.5 When the requesting station user hangs up, all parties are automatically disconnected. If all parties on the call hang up, except the requesting station user, an intercom call remains connected between the requesting station and the Record-A-Call application. This allows the user to make additional comments before ending the recording.

7.6 When the station user turns off Record-A-Call or ends the call being recorded, the Voice Processing Unit delivers the message to the mailbox. The station associated with the mailbox receives message waiting indications as usual. See page 4-46 for message retrieval instructions.

7.7 The programmer can set a maximum length for Record-A-Call messages. The Record-A-Call Maximum Message Length timer can be set at 0-600 minutes. A 0 setting allows messages of any length, limited only by the available disk space. The default is 30 minutes

7.8 In a network setting, the Record-A-Call destination does not have to be on the same node as the station, but the Voice Processing Unit must have a mailbox for that station.

7.9 For more information on using the Record-A-Call feature, see page 5-218.

8. SCHEDULED TIME-BASED APPLICATION ROUTER (STAR)

8.1 Scheduled Time-Based Application Routing (STAR) allows you to have applications with alternate greetings and/or different programming set up for holidays, weekends, and other scheduled events.

8.2 A STAR application is basically a "routing table" for Voice Processing applications. When a direct ring-in call (from a trunk group or call routing table) rings in to a STAR application, it sends the call to another Voice Processing application, according to its programmed schedule. The caller is not aware of this transfer, but hears the programmed greeting for the destination application. (The STAR application itself does not play a greeting.)

8.3 STAR can be used with any type of Voice Processing application except Auto Attendant Recall and Record-A-Call. You can even send calls from one STAR application to another, thereby "chaining" the applications to increase the number of available schedules.

8.4 A STAR application can contain up to 20 scheduling entries with the following fields:

- **Application**: This is the application (call routing announcement, automated attendant, another STAR application, etc.) that is used when the scheduling information applies to the incoming call.
- **Start/Stop Date**: If the schedule is going to be active on a single day or for a period of days, enable the Specific Date flag and enter Start and Stop Dates. (To have the schedule active on only one day, the Start and Stop Dates can be the same day.)
- **Days of the Week**: If the Specific Dates flag is disabled, the schedule can be used on specific days of the week.
- **Specific Times**: The schedule can be set to be active for a specific period of time on selected days, if you enable the Specific Times flag and enter a Start and Stop time.
- **Day/Night Mode**: The schedule can be set to be active in day or night mode, if the Specific Time flag is disabled.

8.5 The chart on the next page shows several samples of STAR entries. Note that there are two entries for Labor Day and Memorial Day [Summer Bank Holiday and Spring Bank Holiday in Europe], and that Christmas Day and Christmas Holiday overlap. The first entry in the table is checked first, and so on, until a match is found. So it is important that you program the applications in the correct order. For example:

- The Labor Day (Summer Bank Holiday) entry, with a specific time, comes before the Day/Night entry so that a different message is played from 10:00 to 2:00 only.
- The early time intervals come before the later intervals on Memorial Day (Spring Bank Holiday).
- Dec 25 comes before the range of dates Dec 24- Jan 4 so that the special holiday message is played on that day only.

8.6 After a match is found, the application corresponding to the matched entry is invoked. If no match is found, the application programmed in the "Default Application" field is invoked.

8.7 When programming the days and times, be careful not to overlap. For example, if you place a date entry for Dec. 25 after a day of the week entry for Monday, and Christmas falls on a Monday, the "Monday" application is selected instead of the "Dec. 25" application.

Table 4-1. *STAR Programming Examples*

Entry	Description (U.S./ Europe)	Date (U.S./Europe)	Day	Time Interval	Day/Night	Application
1	Labor Day/ Summer Bank Holiday	Sep. 1/ Aug. 28		10:00am-2:00pm		2600
2	Labor Day/ Summer Bank Holiday	Sep. 1 /Aug. 28			Both	2601
3	Memorial Day/ Spring Bank Holiday	May 30/May 29		8:00am-5:00pm		2605
4	Memorial Day/ Spring Bank Holiday	May 30/May 29		5:00pm-8:00am		2606
5	Christmas Day	Dec. 25			Both	2607
6	Christmas Holiday	Dec. 24 - Jan. 2			Both	2608
7	Weekends		Sat, Sun		Both	2612
8	Lunch Break		M - F	11:00am- 1:00pm		2613
9	Night Shift		M - F		Night	2614
10	Day Shift		M - F		Day	2615

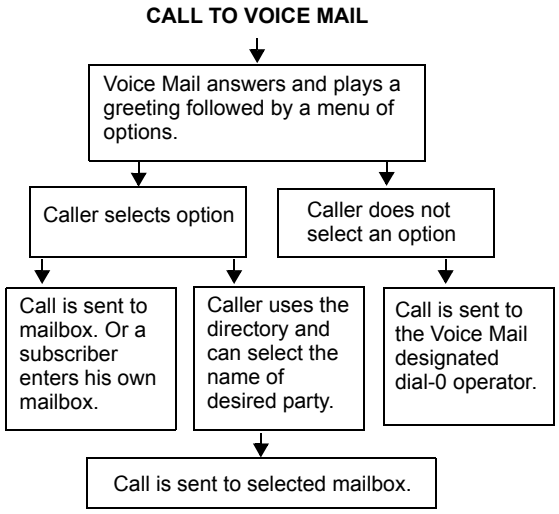
9. VOICE MAIL

9.1 The Voice Mail feature allows callers to send and receive recorded messages from any station or DTMF telephone. (Dial-pulse signaling is not recognized by Voice Processing. Callers using dial-pulse phones are routed to the Voice Mail application's Dial-0 Destination for assistance.)

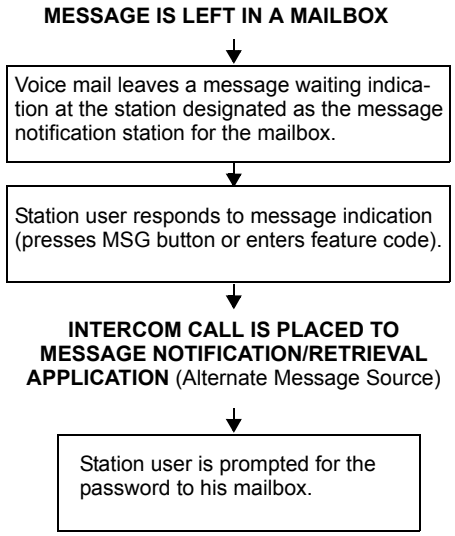
A. VOICE MAIL AND MESSAGE NOTIFICATION/RETRIEVAL APPLICATIONS

9.2 Two types of applications are combined to provide the Voice Mail feature: Voice Mail and Message Notification/Retrieval.

- **Voice Mail**: This application handles all calls that are directed to Voice Mail (other than the Message Notification/Retrieval application) placed by subscribers and non-subscribers. Callers hear the main company greeting, followed by a menu of available options. Stations can forward or transfer calls directly to their mailbox using this application's extension number. This application can also be the message center for the subscribers' stations.



- **Message Notification/Retrieval**: Only one Message Notification/Retrieval application may be created for the system. When station users respond to message indications left by Voice Mail, they reach the Message notification/Retrieval application and are prompted only for their password.



9.3 Station users and outside callers gain access to the Voice Mail application by:

- Dialing the telephone number or extension number assigned to the Voice Mail application
- Being transferred to the Voice Mail by the automated attendant
- Being forwarded to the Voice Mail when a called station is unavailable (via the Call Forwarding or System Forwarding features)
- Responding to message waiting indications

9.4 Callers hear the Voice Mail company greeting and recorded instructions that tell them what to do next. Users simply listen to the prompts and press the keypad button that corresponds to the desired choice. If the user does not respond immediately, a second set of prompts is played.

9.5 Most prompts are interruptible, and users can press the desired button at any time during the prompt. The prompt then stops, and the system acts on the requested choice.

B. MAILBOXES

9.6 A mailbox is a storage location on the Voice Processing PC hard disk that stores all messages that have been directed to it. (The hard disk also stores prompts, greetings, and special programming.) Each "Subscriber" (member of the voice message system) is assigned a unique mailbox number. At system default, the passwords are the same as the mailbox numbers. (For example, the default password for mailbox 1001 is "1001.")

9.7 There are three types of mailboxes on the Voice Mail system: Standard, Receive-Only and System Administrator. There can also be Extension IDs which are stations that do not have mailboxes, but that are included in the recorded directory.

- The Standard mailbox owner receives all of the available subscriber options as authorized through DB Programming. Standard mailbox features include:
  - — Listening to new messages
  - — Recording and sending messages
  - — Deleting and restoring messages
  - — Saving messages
  - — Accessing personal options (greeting, password, etc.)
  - — Accessing remote messaging (if authorized through DB Programming)
- A Receive-Only mailbox type may only listen to new messages, save, delete, or recover them and access personal options. This type of mailbox is restricted from recording and sending messages and has no access to remote messaging.
- There can be only one Voice Mail System Administrator. The Voice Mail System Administrator has all of the options a Standard mailbox subscriber has, plus the following features:
  - — Record and send broadcast messages to all subscribers of the Voice Mail system
  - — Provide mailbox and group list maintenance
  - — Record and select **Custom Audiotex Recordings**
- Extension IDs are used in conjunction with transferring calls through the Automated Attendant or using the Extension Directory. The extension ID allows callers to be transferred to station extensions which do not have mailboxes and allows the station user to record a directory name and set up a password.

9.8 Calls transferred from the Automated Attendant or a Call Routing Announcement application to a station with a mailbox can be screened, announced, or unannounced. Separate programming flags determine the methods used for transferring calls to stations with mailboxes. See page 4-19 for details.

9.9 During DB Programming, each mailbox is assigned a dial-0 operator destination. When a caller presses **0** while listening to the mailbox's personal greeting or recording instructions, or while recording a non-subscriber message, the operator destination is called. The operator can be one of the following types:

- **Mailbox**: If the operator is a mailbox, the caller hears the operator destination's personal greeting after pressing **0**.
- **Station extension number**: If the operator destination is an extension number, the caller hears the Automated Attendant transfer prompt while the call is placed to the extension number ("*Please hold while I transfer your call to...*"), unless it has been disabled at the mailbox. If the station user does not answer, the callers are asked if they want to leave a message, if the operator's extension has an associated mailbox. If the caller chooses not to leave a message, the call returns to the main menu of the application being used (Call Routing Announcement, Voice Mail, or Auto Attendant).
- **Application extension number**: If the operator destination is an application's extension number, the call is transferred to that application's main menu.
- **Operator**: If the operator destination is set to "Operator," the caller is transferred to the operator destination programmed in the Voice Processing database.
- **None**: If the operator destination is set to "None," the **0** is ignored.

C. GROUP LISTS

9.10 Group lists are programmed in the database (see page 6-375). They are lists of mailboxes that can be used by any Subscriber for sending messages to several mailboxes simultaneously. The information that is programmed for group lists include the following:

- Group list description
- Group list number
- Mailboxes included in the list

9.11 There can be up to 1000 group lists in the system and up to 1500 entries per group list.

9.12 Group Lists are not included in the Automated Attendant or Voice Mail directories.

9.13 Instructions for using group lists are given on page 4-48.

D. MESSAGE NOTIFICATION TO STATIONS

9.14 Each mailbox has a programmed Message Notification Station. This is usually the extension number that matches the mailbox number (associated mailbox). However, a different number can be used for the mailbox's message notification station (non-associated mailbox).

- **Associated**: When the extension number, message notification number, and mailbox number for a station all match, that station user can forward calls to Voice Mail and have the call go directly to the associated mailbox's personal greeting. (Even if other mailboxes use that extension number as their message notification extension.)
- **Non-Associated**: The extension number assigned as the Message Notification Station for a non-associated station is an extension number that does not match the mailbox number. (For example, a hunt group pilot number can have a mailbox, but the message notification must be sent to a specific station so that a message lamp can be lit or message notification signal can be sent.) If a station user's extension number does not match a mailbox number, and the station user forwards calls to Voice Mail, the caller hears the main Voice Mail greeting and must enter a mailbox number.

NOTE: If there are non-associated mailboxes in the Voice Processing Unit, the system database must have the Validate Voice Mailbox flag disabled. When enabled, this flag prevents users from dialing mailbox numbers that do not match valid extension numbers.

9.15 When a mailbox receives a message, the message notification station is signaled. If it is a phone, the MSG button is lit, and the display shows that a message has been received. A single-line set receives message waiting signals, if enabled system-wide.

E. CASCADING REMOTE MESSAGE NOTIFICATION

9.16 Remote Messaging is a subscriber feature that is enabled through DB Programming. Subscribers may program a series of specific telephone numbers (a "cascade") for the Voice Mail system to call when new messages are received by their mailboxes. See page 4-43 for a sample notification cascade setup. See page 4-51 for subscriber instructions on setting up remote messaging.

Primary and Alternate Notification

9.17 Each mailbox can have a Primary and/or an Alternate notification cascade of numbers. The notification numbers are programmed to be accessed during specific days and times.

9.18 The Voice Mail system monitors subscriber mailboxes continuously. During monitoring, if new messages have been received, the system checks to see if Primary Notification is turned on. If it is, the system checks the day and time programming. If the current time is within the programmed notification day and time, the Voice Mail system makes the notification call(s) to the numbers in the Primary Notification cascade. If Primary Notification is turned off or if the day or time does not match, the Voice Mail system immediately makes the same check for Alternate Notification and possibly places calls to the numbers in the Alternate Notification cascade.

9.19 When the Voice Processing Unit makes a notification call, it moves through the appropriate notification cascade level by level as determined by the notification and retry programming described in the following sections. When the system reaches the last level in the cascade, it returns to level 1 and begins again, if necessary.

9.20 If the system is not able to place the call within the parameters of the programmed notification day and time (due to busy facilities or invalid programming), the mailbox receives a message stating that notification could not be completed (the mailbox user hears the message next time the user accesses the mailbox.)

9.21 The number of calls that the Voice Processing Unit can process simultaneously is determined by the Maximum Number of Outgoing Calls flag. It defaults to 2, but can be programmed to use all of the enabled Voice Processing Unit voice channels.

NOTE: The number of calls that the Voice Processing Unit can process simultaneously is also limited by the available time slots (maximum channel allocations) for the application (see page 2-17 for details). An outgoing message notification call that exceeds the number of channels allocated to the Message Notification application results in system message as described in paragraph 9.20.

All Messages, Priority Only, and Each New Message Flags

9.22 When the mailbox is being programmed for remote notification, message notification can be set to place the notification call when any message is received or only when priority messages are received. (This defaults to "All Messages.") Another mailbox flag, called "Call For Each New Message," determines when the Voice Processing Unit attempts to make the notification call when more than one message is waiting to be heard:

- If the Call For Each New Message flag is *enabled,* the Voice Processing Unit attempts *immediate* notification each time a new message (or priority message, depending on the option selected) is received, regardless of the status of the applicable retry timer. If a retry is in progress, it places the call to the number on the cascade table that would have received the retry call, as described below. (For example, if the Voice Processing Unit was on level 3 of a cascade, attempting retries, it does not start over again at level 1. It places the call to level 3.)
- If the Call For Each New Message flag is *disabled,* new messages do not generate a remote notification attempt until the Subscriber has logged on and listened to the waiting message(s). (However, retry calls are attempted if the applicable retry timer expires, as described below.)

Retry Call Attempts

9.23 The mailbox can be programmed to retry notification calls when it encounters a busy trunk or trunk group. Each cascade level can be programmed with a Number of Call Attempts value of 1-1000 calls.

9.24 If the mailbox user accesses the mailbox between the time the message is received and when remote notification is successful, the system stops attempting remote notification. It is assumed that the mailbox user listened to the message when the mailbox was accessed. However, the MSG button at the station remains lit if there are any messages that have not been heard.

9.25 Each level also has a Number Called Busy timer, and a Pager Notification Retry timer or Personal Number No Answer timer. These timers determine how long the Voice Processing Unit waits before making the next notification attempt when messages are waiting to be heard (unless it is overridden by the Each New Message flag). These timers can be set to a value of 0-255 minutes. If set to 0, it retries the cascade level immediately, for as many attempts as allowed by the Number of Call Attempts field. If the message(s) have not been picked up by then, the system moves to the next cascade level. The default setting for the Pager Notification Retry timer is 20 minutes. The Number Called Busy and Personal Number No Answer timers default to 5 minutes.

NOTE: The 0-minute timer value should be used cautiously. If the system is forced to make several calls in quick succession, it impairs the Voice Processing Unit's ability to place other outgoing calls. It should be used primarily with cascades in which all of the members want to receive notification at approximately the same time. In this case, all levels except the last could be set to 0-minute retries with 1 allowed call attempt. However, the last level in the cascade should be programmed with a longer Pager Notification Retry timer so that the system does not continuously cycle through the cascade levels. (Continuous cycling is also prevented somewhat by a per-mailbox 15-call limit. Due to FCC regulations, the Voice Processing Unit can allow a mailbox to make only 15 calls in rapid succession without a ten minute pause.)

Notification Numbers

9.26 Each cascade level has a programmed notification number that can be an extension number or an outside telephone number.

9.27 When an outside telephone number is used, the system accesses an outgoing trunk using the code programmed for that level that contains a pre-programmed trunk access code and followed by an outgoing access termination feature code, if needed.

9.28 Notification calls to outside telephone numbers can be sent to a pager or to a number where a person answers (personal number).

- If the call goes to a pager, the system dials the programmed Pager Dial String, then disconnects the call. The dial string should include any digits that the paging company requires when the call is answered, the pager LCD number, and the pager termination code, if needed. Also, if desired, the pager string can be programmed to show the number of the mailbox that placed the call and/or the number of waiting messages in the mailbox.
- If the call goes to a personal number, the system plays a prompt announcing that the call is from Voice Mail, play the directory name or mailbox number that originated the call, and plays user instructions. The listener can then enter the mailbox number's password to gain access to the mailbox and hear the message. The listener has three chances to enter a correct password. If the system receives an incorrect password three times, the Voice Mail system disconnects the call immediately, and the attempt is considered unsuccessful.

Sample Pager Programming

9.29 The following is a sample programming sequence for enabling pager notification. In the example, the pager number is 555-2500, trunk group 1 is used for notification, the display should show 961-9000 when a message is left in mailbox 1001, and the paging company requires a # after the digits.

1. In Voice Processor Programming, create a Message Notification/Retrieval application.
2. If desired, set toll restriction for the Message Notification/Retrieval application in Voice Processor\Devices\Applications\Class of Services.
3. Ensure the Message Notification/Retrieval application has outgoing access for trunk group 92001.
4. In Mailbox Programming for mailbox 1001, enable Remote Notification and program the following for the primary and/or alternate message notification:
   a. **Notification Category**: Determine whether all messages or only priority messages cause message notification.
   b. **Call For Each New Message flag**: Determine if you want to call the pager every time a new message arrives, even while there is an unheard message waiting.
   c. **Start Time, Stop Time, and Day of Week**: Set the parameters for when you want the pages to be sent.
   d. **Cascade Table**: Program the cascade table(s) you need. In the example, the table would have:
      – Notification Type: Pager
      – Notification: Enabled
      – Notification Destination Type: CO

- Notification Number: 555-2500
- Outgoing Access: 92001 (Trunk Group 1)
- Pager Dial String: 961-9000#
- Timers: Can be adjusted to suit the pager user's requirements.

Sample Notification Cascade

9.30 The example on the next page shows how each level in a Primary and Alternate cascade is programmed to meet a customers needs. In the example, the pager dial string (9619000*MN#) includes the Voice Processing Unit's telephone number (961-9000), an asterisk to separate digits, the mailbox number (M), the number of messages in the mailbox (N), and a pager termination digit (#).

Table 4-2. *Sample Notification Cascade*

Customer Waits:	Cascade Level	Program these Parameters
During weekly business hours (8-5), he wants to be notified for all messages, each time a message is received...	Primary Notification	Notification Category: All Messages Call For Each New Message: Enabled Start Time/Stop Time: 8:00 AM/5:00 PM Days of Week: Monday-Friday
First, call his desk (extension 1234) for up to 3 attempts, 1 minute apart...	Primary – Level 1	Enable Notification: Yes Notification Destination Type: IC Notification Number: 1234 Notification Type: Personal Number of Call Attempts: 3 calls Number Called Busy: 1 minute Personal Number No Answer: 1 minute
If he doesn't answer at his desk, he wants Voice Mail to call his car phone (555-1000) up to 5 times, every three minutes...	Primary – Level 2	Enable Notification: Yes Notification Destination Type: CO Notification Number: 5551000 Notification Type: Personal Outgoing Access: 92001 (trunk group 1) Outgoing Access Termination: # Number of Call Attempts: 5 calls Number Called Busy: 3 minutes Personal Number No Answer: 3 minutes
If he doesn't answer at his car phone, he wants his pager (555-2000) to be called...	Primary – Level 3	Enable Notification: Yes Notification Destination Type: CO Notification Number: 5552000 Notification Type: Pager Pager Dial String: 9619000*MN# Outgoing Access: 92001 Outgoing Access Termination: # Number of Call Attempts: 1 call Number Called Busy: 0 minutes Pager Notification Retry: 0 minutes

Table 4-2. *Sample Notification Cascade (Continued)*

Customer Waits:	Cascade Level	Program these Parameters
Immediately after it calls his pager, he wants it to call his assistant's pager (555-3000) right away.	Primary – Level 4	Enable Notification: Yes Notification Destination Type: CO Notification Number: 5553000 Notification Type: Pager Pager Dial String: 9619000*MN# Outgoing Access: 92001 Outgoing Access Termination: # Number of Call Attempts: 1 call Number Called Busy: 255 minutes Pager Notification Retry: 255 minutes
After business hours and during the week, he wants to be notified for only priority messages, each time a priority message is received...	Alternate Notification	Notification Category: Priority Only Call For Each New Message: Enabled Start Time/Stop Time: 5:00 PM/8:00 AM Days of Week: Monday-Friday
He wants his pager called (555-2000) for all priority messages.	Alternate – Level 1	Enable Notification: Yes Notification Destination Type: CO Notification Number: 5552000 Notification Type: Pager Pager Dial String: 9619000*MN# Outgoing Access: 92001 Outgoing Access Termination: # Number of Call Attempts: 1 call Number Called Busy: 0 minutes Pager Notification Retry: 0 minutes

F. SUBSCRIBER USE OF VOICE MAIL

9.31 Voice Mail subscriber features include the following:

- Listen to new messages, if any
- Send a message
- Listen to saved messages, if any
- Set personal options
- Set up remote messaging, if enabled

9.32 When a subscriber enters the mailbox, the Voice Mail application may play one of the following messages:

- **Mailbox almost full or full**: If a mailbox is full, no new messages can be received at that mailbox until the subscriber deletes waiting or saved messages.
- **Message count**: A prompt tells the subscriber how many messages are waiting to be heard, if any, and how many are priority messages.
- **Remote messaging**: A prompt indicates whether primary or alternate remote messaging is selected when remote messaging is enabled. It also alerts the subscriber if a programming error has been detected and/or if the system encountered busy resources when attempting to place a remote messaging call.

Access to a Mailbox

9.33 To access an individual mailbox:

1. Dial the Voice Mail application access number. You hear the main menu.
2. During or after the greeting, press * to identify yourself as a subscriber.
3. Enter your mailbox number and personal password.

Initializing a Subscriber Mailbox or Extension ID

9.34 The first time a subscriber uses Voice Mail, the user must customize the mailbox. System prompts instruct the subscriber to:

- Change the default password number (same as the mailbox number) to a personal password.

> **NOTE:** To provide system security, ***all*** mailboxes should have a password. To make the passwords difficult to guess, they should not match the mailbox number or consist of one digit repeated several times. The default password should be changed the first time the user logs in. This is especially important in the System Administrator's mailbox which allows programming access to the Voice Processing Unit.

- Record a name to identify the mailbox owner in the company directory. (Directory names can be up to 1 minute in length.)
- Listen to the Voice Mail application introduction. (You can skip this introduction by pressing #, if desired.)

9.35 To initialize a mailbox or extension ID:

1. Dial the Voice Mail extension number. (You hear the main menu.)
2. Press * to identify yourself as a subscriber.
3. Enter your mailbox number and default password. (Your default password is your mailbox number.)
4. *If you want a password,* enter a new password using digits 0-9 (up to 12 digits). Then press #. Voice Mail plays back your password.

   If you do not want to use a password, just press #.
5. Press # to accept the entry. (Or, press 3 if you want to re-enter your password.) The system prompts you to record your directory name.
6. After the tone, record your first and last name. When finished, press #.
7. Press # again to accept the name. (Or, press 1 to replay the name you just recorded, press 2 to add to your name, or press 3 to erase and re-record it.)
8. The system then plays a prompt that introduces you to the basic Voice Mail features. (You can skip this introduction by pressing #, if desired.)

9.36 After your mailbox is initialized, you may use any of the mailbox features provided. To record a personal greeting, or to change your password or directory name, see paragraph 9.45 on page 4-51.

Listening to Messages

9.37 There are three types of Voice Mail messages:

- **New Messages** are messages that you have not yet heard, including Broadcast Messages from the Voice Mail System Administrator and system-sent messages. The order in which the messages are played depends on your message retrieval settings (see page 4-51), except that messages marked "priority" are played first. Some special messages that may be played include the following:
  - *Message receipt:* If you send a certified message, a system message is played when the recipient has listened to the message.
  - *Caller Information:* If a call was placed to your mailbox on a trunk that supports the Caller ID [CLIP] feature, and the caller hung up without leaving a message, a system message is played informing you that the calling party did not leave a message. It also includes, within the corresponding message envelope, the number of the calling party as the message source.
  - *Security violation:* The Voice Mail system has an extra security measure built in which plays a message, when you log into your mailbox, that tells you if three invalid attempts have been made to access your mailbox within a single call. This announcement message is sent to your mailbox each time this situation occurs. It is received as a new message and is accompanied by a message envelope.
- **Saved Messages** are messages that you previously listened to and saved (or deleted and restored). You can play them back, as necessary.
- **Undeliverable Messages** are messages that you left for a mailbox on another node that could not be delivered. They are returned to your mailbox as "undeliverable."

9.38 The Return Call feature allows Voice Mail users to return a call to the person who left a message, as long as the call was received from an extension number or an outside number with Caller ID [CLIP]. It is part of the "Reply" option described in the procedure below.

9.39 Whenever you access Voice Mail as a subscriber, you are told how many new and saved messages you have. From the Main Menu:

1. *EITHER,* Press 1 to listen to your new messages.

   OR, Press 3 to listen to your saved messages.

2. The system plays each message in the queue selected. While you are listening to a message, you can use the following options:

   - Press # to skip to the end of the envelope or recording.
   - Press 1 to back up.
   - Press 2 to pause. (Then press any button to continue.)
   - Press 3 to skip ahead.
   - Press 4 to lower the volume.
   - Press 5 to play the message envelope again.
   - Press 6 to raise the volume.
   - Press 7 to save the new message in your mailbox.
   - Press 9 to delete the message from your mailbox.

3. When the message has finished playing, you have the following options:

   - Press 1 to replay the message from the beginning.
   - Press 2 to reply to the message. You can then do one of the following:
     - Press 1 to leave a Voice Mail message for the caller. *If the caller had a mailbox number,* it asks you to verify the destination by pressing #. *If the number was not associated with a mailbox,* it asks you to enter a mailbox number. *If the message was from an outside caller,* you cannot leave a Voice Mail message.
     - Press 2 to make a return call. Your call is transferred automatically to the caller's extension or telephone number, if available. *If the extension number is not available,* you have the option of leaving Voice Mail. *If the telephone number is not available,* you cannot reply to the message.
   - Press 3 to forward a copy of the message to another subscriber. If you want to include introductory comments, press 1 and record an introduction. If not, press # to forward the message without an introduction.
   - Press 4 to listen to the previous message.
   - Press 5 to play the introductory message "envelope" again.
   - Press 6 to listen to the next message.
   - Press 7 to save the new message in your mailbox.
   - Press 9 to delete the message from your mailbox.

Recovering Deleted Messages

9.40 This feature allows you to "undo" the Delete Message operation and restore previously deleted messages. The messages are restored to your saved-message queue. You have a set amount of time to recover messages, depending on DB Programming (up to 24 hours). After that, Voice Mail erases them.

NOTE: When Voice Processing is shut down, all deleted messages are permanently deleted and cannot be restored.

9.41 To recover deleted messages:

1. Dial the Voice Mail application access number. You hear the main menu.
2. During or after the greeting, press * to identify yourself as a subscriber.
3. Enter your mailbox number and password.
4. When you hear the Main Menu list of options, press 5 to select Message Options. Some phone users also see MESSAGE OPTIONS on their display menu.
5. The next prompt gives you the option of recovering deleted messages. Press 2 to continue.
6. Your options are:
   - Press 1 to listen to your deleted messages and choose which ones to delete or recover. After each message you can:
     - Press 1 to replay the message.
     - Press 2 to reply to the message.
     - Press 3 to forward the message to another mailbox.
     - Press 4 to listen to the previous message.
     - Press 5 to replay the message envelope.
     - Press 6 to listen to the next message.
     - Press 7 to recover the message.
     - Press 9 to purge the message from your mailbox.
   - Press 2 to recover all of your deleted messages and store them as saved messages.
   - Press 3 to purge all of your deleted messages so that they are erased from voice mail.

Sending a Message

9.42 As a subscriber, you can record and send voice messages to other users from the subscriber Main Menu. Follow these steps:

1. Dial the Voice Mail application access number. (You hear the main menu.)
2. During or after the greeting, press * to identify yourself as a subscriber.
3. Enter your mailbox number and personal password.
4. When you hear the Subscriber Menu, press 2 to send a message.

5. Dial the desired mailbox number or group list number. *If you do not know the mailbox number,* you can press # to access the company directory to locate and select the mailbox. See page 4-62 for directory instructions. Group lists are not included in the directory.

   If the mailbox is full, a prompt notifies you that the mailbox cannot receive messages. However, it allows you to leave a message at another mailbox. Leave your message with another subscriber or try again later.

6. After hearing the subscriber's name, press # to accept it.

7. The subscriber's mailbox message is played (To skip this message, press #.) When you hear the tone, record your message. When finished, hang up or press # and go to the next step.

   NOTE: If the system detects silence for longer than 5 seconds, it prompts you to speak up. The recording, if any, remains unchanged.

   - Press 2 to pause while recording. To continue, press any button. (The pause length is determined by the Pause Voice Mail timer.)
   - Press 3 to erase your message.

   If you exceed the maximum allotted time for message length, a prompt notifies you and allows you to re-record the message or send the message as it is.

   If Voice Mail is full, a message notifies you that it cannot accept your message. Hang up and try again later.

8. When you have completed your message, you have the following options:

   - *P*ress 1 to hear your message.
   - Press 2 to add to your message.
   - Press 3 to erase and re-record your message.
   - Press 9 to use the special delivery option. Then go to the next step.
   - Press # to send your message. (If you want to mark the message certified, private, and/or priority, do so before completing this step.) If you want to send another message, return to step 5.

9. You can do one or more of the following if you selected "special delivery options" above:

   - Press 1 to mark the message "private." (This prevents the recipient from forwarding it to other subscribers.)
   - Press 2 to mark the message "certified." (When a certified message is heard by the recipient, you receive a receipt notice.)
   - Press 3 to mark the message "priority." (This places your message ahead of all other waiting messages in the receiving mailbox.)
   - Press * to cancel delivery options.

10. After the message has been sent with special delivery options, you may choose to do either of the following:
    - Press # and enter another mailbox number to send the message to another destination.
    - Press * to exit.

Canceling Unheard Messages

9.43 The Cancel Unheard Messages feature allows you to cancel voice messages you have sent to other subscribers, provided the messages have not yet been heard by their recipients and they were not sent to a group list.

9.44 To cancel unheard messages:

1. Dial the Voice Mail application access number. You hear the main menu.
2. During or after the greeting, press * to identify yourself as a subscriber.
3. Enter your mailbox number and password.
4. When you hear the Main Menu list of options, press 5 to select Message Options. Some phone users also see MESSAGE OPTIONS on their display menu.
5. The next prompt gives you the option of canceling unheard messages. Press 1 to continue.
6. When prompted, enter the number of the mailbox that has the unheard message. Then you are asked to confirm the mailbox you selected by pressing #.
7. It tells you how many messages you have waiting at the mailbox and plays them for you. After each message, you can:
    - Press 1 to replay the message.
    - Press 2 to add to the message.
    - Press 3 to forward the message to another mailbox.
    - Press 4 to listen to the previous message.
    - Press 5 to replay the message envelope.
    - Press 6 to listen to the next message.
    - Press 7 to save the message in your mailbox and remove it from theirs.
    - Press 9 to delete the message.

Personal Options

9.45 Personal Options allow you to customize your mailbox. Each subscriber's mailbox has the following personal options:

- **Greetings**: Your primary or alternate greeting is played to callers when they reach your mailbox. You can change your primary greeting at any time to reflect a change in status and/or record and enable an alternate greeting. If you elect not to record a personal greeting, the default system greeting is used.

  NOTE: There is a database option called "Play Recording Instructions" that affects mailbox greetings. If this option is enabled, the system prompt that tells the caller to leave a message after the beep plays after the primary or alternate greeting. If disabled, the beep occurs as soon as the primary or alternate greeting ends. In the default state, this option is enabled. For programming information, see page 6-391 in PROGRAMMING.

- **Directory Name**: Your recorded name is used to identify you in the directory and to verify your mailbox number when messages are addressed to you by non-subscribers and other subscribers. Although your name is recorded when you first set up your mailbox, you can change it at any time.
- **Password**: Your password is used to prevent unauthorized access to your mailbox. Although you create a personal password when you first set up your mailbox, you can change it at any time.

  NOTE: To provide system security, all mailboxes should have a password. To make the passwords difficult to guess, they should not match the mailbox number or consist of one digit repeated several times. The default password should be changed the first time the user logs in. This is especially important in the System Administrator's mailbox, which allows programming access to the Voice Processing Unit.

- **Message Envelope Contents**: Each message is preceded by an "envelope" message that can include the time and date the message was left, the source of the message, and the length of the message. You can choose the contents of the envelope or disable the envelope completely. (Even if the envelope is disabled, the subscriber can still enter the code to play the envelope when listening to the message.)
- **Remote Messaging**: Remote messaging is described in detail on page 4-40.
- **Fax Destination Number**: If your station is programmed to receive incoming faxes, you can program the extension number of the fax machine to receive the faxes that are sent to your mailbox.
- **Message Retrieval Order**: This option allows you to determine how to retrieve your saved or new messages based on the date and time they were received. You can listen to the earliest-received messages first (First In/First Out) or the latest-received messages first (Last In/First Out).
- **Transfer Method**: When a call is received by an Automated Attendant or Call Routing Announcement application and the caller enters an extension number, the programmed Transfer Method determines how the call is transferred. See page 5-211 for a list of Transfer Methods and information about receiving screened and announced calls.

9.46 To program personal options:

1. Dial the Voice Mail application access number. You hear the main menu.
2. During or after the greeting, press ✱ to identify yourself as a subscriber.
3. Enter your mailbox number and password.
4. When you hear the Main Menu list of options, press 4 to select the Personal Options Menu. You may then do any or all of the following steps.
5. **To change your personal greeting,** press 1. You have the following options:
   a. Press 1 to record and/or enable your primary personal greeting or 2 to record and/or enable your alternate greeting. The system plays your current greeting, if one exists. To replay the greeting, you can press 1. Press 3 to erase and re-record your greeting as follows:
      1.) Record your greeting after the tone, then press #.
      2.) Press # again to accept your greeting. (Or, you can press 1 to replay your greeting, 2 to add to your greeting, or 3 to erase and re-record your greeting.)
   b. Press 3 to enable the system default mailbox greeting. The system greeting is automatically enabled.
   c. Press ✱ to exit without changing your greeting.
6. **To change your recorded directory name,** press 2.
   a. After the tone, record your first and last name, then press #.
   b. Press # again to accept your recorded name. (Or, press 1 to replay your recorded name, 2 to add to your name, or press 3 to erase and re-record your name.)
7. **To change your password,** press 3.
   a. Choose and enter a new password using digits 0-9 (up to 12 digits). *If you do not want to use a password,* skip this step.
   b. Press #. Your password is played back.
   c. Press # to accept the password as entered. (Or, press 3 if you want to re-enter your password.)
8. **To change your envelope contents,** press 4. A prompt plays to tell you the current status of your envelope contents.
   a. Press 1 to enable or disable the time/date option. A prompt plays to tell you the current status of the time/date option.
   b. Press 2 to enable or disable the message source option. A prompt plays to tell you the current status of the message source option.
   c. Press 3 to enable or disable the message length option. A prompt plays to tell you the current status of the message length option.
   d. Press 4 to enable all options. A prompt tells you that all options are enabled.
   e. Press 5 to disable all options. A prompt tells you that all options are disabled.
   f. Press ✱ or # to return to the Personal Options Menu.

9. **To set up Remote Messaging,** press 5.

NOTE: In the following procedure, some phone users can use the menu buttons next to their displays instead of pressing the digits on the keypad.

a. Press 1 to set up a Primary cascade, or press 2 to set up an Alternate cascade. A prompt tells you what the cascade's current status is. (If desired, press * to return to the Personal Options Menu.) Then perform one of the following steps.

b. **To program a cascade level:** Press 1 then program the following:

1.) Enter the number of the level you want to program (1-9). A prompt tells you the current status of the level.

2.) You have the following options:

Press 1 to enable or disable this level. (You cannot enable the level until a notification number is programmed.)

Press 2 to set up or change the personal telephone number. If it is an extension number, press 1. Or, if it is an outside number, press 2. Then enter the number. (If your phone has menu keys, you can enter the special characters for pause, mailbox, and new message count by pressing their menu buttons.) A prompt plays back the number for your verification. If you entered an invalid or toll-restricted number, you hear a warning and must enter the number again.

Press 3 to set up or change the pager telephone number. Then enter the number. (If your phone has menu keys, you can enter the special characters for pause, mailbox, and new message count by pressing their menu buttons.) A prompt plays back the number for your verification. If you entered an invalid or toll-restricted number, you hear a warning and must enter the number again.

c. **To set up time of day for notification:** Press 2. Then, when prompted, enter the time at which you want to have the message notification start and stop. Enter the times with two digits for the hour and two digits for the minutes. If entering the time in 12-hour format, you are prompted to press 1 for AM or press 2 for PM.

NOTE: If you want to have 24-hour notification, program the starting and ending times to be the same.

d. **To set up the days of the week for notification:** Press 3. Then press 1 for messaging Monday-Friday, 2 for all days, or 3 to select individual days. If you selected individual days, you are prompted to press digits 1-7 which correspond to the days Sunday through Saturday. Your selection is played back.

e. **To select all or priority-only message notification:** Press 4. Then press 1 for all messages or 2 for priority messages only.

f. The options prompt returns. Either select another option, as described above, or press # to save the settings and exit.

10. **To change your Fax Destination Number,** press 9 for More Options and then press 1.

    a. Enter the number of your fax destination, followed by #.

    b. To accept press #. Or, to erase the number and re-enter it press 3.

11. **To change your Message Retrieval method,** press 9 for More Options and then press 2 or press the **MSG SEARCH ORDER** menu button. This allows you to determine how to retrieve your new or saved messages based on the date and time they were received. Press 1 to change the search order for new messages, or press 2 to change the search order for saved messages. Your options are:

    - Press 1 or the **1ST IN/1ST OUT** menu button to retrieve the earliest-received messages first.
    - Press 2 or the **LAST IN/1ST OUT** menu button to retrieve the latest-received messages first.

12. **To change your Transfer Method,** press 9 for More Options and then press 5. The system tells you what your current Transfer Method is. You can then select the desired Transfer Method, as follows:

    a. To select unannounced transfers, press 1 or the **UNANNOUNCED** menu button.

    b. To select screened transfers, press 2 or the **SCREENED** menu button.

    c. To select announce only transfer, press 3 or the **ANNOUNCE ONLY** menu button.

G. NON-SUBSCRIBER USE OF VOICE MAIL

9.47 Callers who do not have a subscriber mailbox can leave messages for subscribers. Also, Subscribers can use this method for leaving messages without accessing their mailboxes. If a non-subscriber message is left using a station with an associated mailbox, the receiving mailbox Subscriber can reply to the message, if desired, and the reply is sent to the caller's associated mailbox.

9.48 Non-subscribers can gain access to the Voice Mail system through the automated attendant (by dialing the Voice Mail application extension number), through a call transfer, by placing an intercom call to the Voice Mail application extension number, or by being forwarded to Voice Mail.

Leaving a Message

9.49 To leave a message as a non-subscriber:

1. *If you hear the Voice Mail greeting or menu,* dial the desired mailbox number. (If you do not know the number, you can press **#** to access the mailbox directory to locate and select the mailbox. See page 4-62 for directory instructions.)

   If you hear a subscriber's personal mailbox greeting, skip this step.

2. *If the mailbox is full,* a prompt notifies you that it cannot receive messages. However, it allows you to leave a message at another mailbox. Leave your message with another subscriber or try again later.

3. After hearing the subscriber's greeting and a tone, record your message. You have the following options while recording your message:

   To pause while recording: Press **2**. To continue, press any button except **0**.

   To erase and re-record your message: Press **3**.

   To call the mailbox user's attendant: Press **0**.

   To cancel the recording and return to the main menu: Press *****.

   If you exceed the maximum allotted time for message length, a prompt notifies you and allows you to re-record the message or send the message as it is.

   If Voice Mail is full, a message notifies you that it cannot accept your message. Hang up and try again later.

   NOTE: If the system detects silence for longer than 5 seconds, it prompts you to speak up. The recording, if any, remains unchanged.

4. When you have completed your message, you have the following options:

   To hear your message: Press **1**.

   To add to your message: Press **2** and continue your message as described in step 3.

   To erase and re-record your message: Press **3**.

   To use the special delivery options: Press **9**. Go to the next step.

   To send your message: Press **#**. (If you want to mark the message certified, private, and/or priority, do so before completing this step.) If you want to send another message, return to step 1.

5. You can do one or more of the following if you selected "special delivery options" above:
   - Press 1 to mark the message "private." (This prevents the recipient from forwarding it to other subscribers.)
   - Press 3 to mark the message "priority." (This places your message ahead of all other waiting messages in the receiving mailbox.)
   - Press * to cancel delivery options.
6. After the message has been sent with special delivery options, you may choose to do either of the following:
   - Press # and enter another mailbox number to send the message to another destination.
   - Press * to exit.

H. VOICE MAIL SYSTEM ADMINISTRATOR FEATURES

9.50 The Voice Mail System Administrator can use special features that are not provided to Subscribers. The System Administrator mailbox has all standard subscriber features plus the ability to do the following:

- Record a broadcast message
- Perform mailbox and group list maintenance
- Create and select custom Audiotex recordings (Voice Mail company greetings, auto attendant recordings, call routing announcements, and hunt group overflow and announcement station recordings)
- Import fax documents
- Customize Voice Mail prompts

9.51 To enter the System Administrator's mailbox, the System Administrator dials the Voice Mail access number, presses * to access the main menu, enters the System Administrator's mailbox number, and then presses 9 to reach the System Administrator Menu. (There is not a prompt for entering the 9.)

Broadcast Messages

9.52 The System Administrator has the ability to make a single recorded message and send it to all Standard and Receive-Only subscriber mailboxes. (Extension IDs do not receive Broadcast Messages.)

9.53 To record a broadcast message:

1. Dial the Voice Mail application access number. You hear the main menu.
2. During or after the greeting, press * to identify yourself as a subscriber.
3. Enter the System Administrator mailbox number and password.
4. Press 9 to access the System Administrator's Menu.
5. When you hear the System Administrator Menu list of options, press 1 to send a broadcast message.
6. Record your message.
   - *To pause while recording:* Press 2. To continue, press any button except 0.
   - *To erase and re-record your message:* Press 3.

If you exceed the maximum allotted time for message length, a prompt notifies you and allows you to re-record the message or send it.

If Voice Mail is full, a message notifies you that it cannot accept your message. Hang up and try again later.

7. When you have completed your message, you have the following options:
   - **If you are satisfied with your message**: Hang up. OR press # for more options.
   - **To hear your message:** Press 1.
   - **To add to your message:** Press 2 and continue your message as described in step 6.
   - **To erase and re-record your message:** Press 3.

Mailbox/Group List Maintenance

9.54 Using the System Administrator's mailbox, the administrator may provide basic Voice Mail mailbox and group list system maintenance. This maintenance includes the following:

- Record the name of a standard subscriber mailbox, extension ID, or group list
- Record the mailbox greeting and set personal options for a subscriber mailbox
- Set the password for a subscriber mailbox or extension ID

NOTE: To provide system security, all mailboxes and extension IDs should have a password. To make the passwords difficult to guess, they should not match the mailbox number or consist of one digit repeated several times. This is especially important in the System Administrator's mailbox which allows programming access to the other mailboxes.

9.55 The System Administrator does not need to know the Subscriber's password to perform maintenance on a subscriber mailbox.

9.56 To perform mailbox/group list maintenance:

1. Dial the Voice Mail application access number. You hear the main menu.
2. During or after the greeting, press * to identify yourself as a subscriber.
3. Enter the System Administrator mailbox number and password.
4. Press 9 to access the System Administrator's Menu.
5. When you hear the System Administrator Menu list of options, press 2 to access the Mailbox/Group List Maintenance option.
6. Dial the mailbox, extension ID, or group list number to be accessed.

   If programming a mailbox or extension ID, you hear the Personal Options menu. Follow the instructions on page 4-51 to program the personal options (name, password, message envelope, or greeting) for the selected mailbox.

   If programming a group list, the current name is played. You are prompted to enter a name. After the tone, record the name for the selected list. When you are finished, press #. Then press # again to accept the name. (Or, press 1 to replay the name you just recorded, or 3 to erase and re-record it.)

Custom Audiotex Recordings

9.57 The Voice Mail System Administrator can create and select up to 500 custom Audiotex recordings (Voice Mail company greetings, auto attendant greetings, call routing announcements, and hunt group announcement/overflow recordings). Each recording has an assigned number. Numbers are then assigned to specific applications using the System Administrator's mailbox or through DB Programming.

9.58 Any recording number (001-500) can be assigned to any Voice Processing application. In the default state, the Inter-Tel Identification Prompt is played before the mailbox greeting whenever outside callers access or are transferred/forwarded to a voice mailbox (see page 6-347 for more information). If disabled, Recording 001 becomes the system-provided Voice Mail main greeting. Recording 002 is always the default Automated Attendant main greeting, regardless of the Identification Prompt setting. If necessary, the default recordings can be changed. Before making any changes, however, back up the existing recordings using the Voice Processor save feature (see page 6-71). A single recording may be used for multiple applications.

9.59 After making the required recordings, the System Administrator must determine which ones play during the telephone system's day and night modes of operation. This can be done from either the System Administrator's mailbox or through DB Programming.

9.60 The System Administrator must enter a valid three digit recording number (or a single digit followed by # for recordings 1-9) before a custom Audiotex recording can be made. If a valid number has been entered and a previous recording exists, Voice Processing plays it. (This includes the two standard default greetings.) When no recording is associated with a recording number, the System Administrator is prompted to create a recording.

9.61 After the recording is installed, it replaces any previous recording. The only recordings that can be retrieved are system default recordings 001 and 002. They can be retrieved by defaulting the Voice Processing database.

9.62 The maximum recording time for each custom Audiotex recording is determined in DB Programming when the maximum greeting lengths are established for all subscribers.

9.63 In DB Programming, several greetings can be linked together for one application and/or use the special UCD/ACD hunt group characters that tell the caller how many calls are ahead and how long the wait will be. See page 6-364 in PROGRAMMING for information on this option. If you use the System Administrator's mailbox to record or assign a recording to an application with a list of several recordings, your input changes only the first entry in the list. You cannot change the other entries using the System Administrator's mailbox.

9.64 To record custom Audiotex recordings:

1. Dial the Voice Mail application access number. You hear the main menu.
2. During or after the greeting, press * to identify yourself as a subscriber.
3. Enter the System Administrator mailbox number and password.
4. Press 9 to access the System Administrator's Menu.
5. When you hear the System Administrator Menu, press 3 to select the Custom Recordings option.
6. The next prompt asks whether you want to make a custom recording or select a custom recording. Press 1 (make a recording).

7. When prompted, enter the desired recording number (001-500). If a recording exists, it is played. If not, go to the next step. After the recording has played, you have the following options:
   - **To hear the recording again:** Press 1.
   - **If you want to leave the recording unchanged**: Press #.
   - **If you want to re-record the recording**: Press 3 and continue to the next step.
8. When prompted, record the greeting or message.
   - **To pause while recording:** Press 2. To continue, press any button.

   If you exceed the maximum allotted time for recording length, a prompt notifies you and allows you to re-record or save it.

   If the Voice Processing PC disk is full, a prompt notifies you that it cannot accept your recording. Hang up and try again later.
9. When you have completed your recording, you have the following options:
   - **If you are satisfied with your recording**: Hang up. Or, press # for more options.
   - **To hear your recording:** Press 1.
   - **To add to your recording:** Press 2 and continue your recording as described in steps 3-8.
   - **To erase and re-record your recording:** Press 3.

9.65 To assign custom Audiotex recordings to specific application extension numbers:

1. Dial the Voice Mail access number. You hear the main menu.
2. During or after the greeting, press * to identify yourself as a subscriber.
3. Enter the System Administrator mailbox number and password.
4. Press 9 to access the System Administrator's Menu.
5. When you hear the System Administrator Menu list of options, press 3 to select the Custom Recordings options.
6. The next prompt asks if you want to make a custom recording or select a custom recording. Press 2 (select a recording).
7. When prompted, enter the extension number of the application to be programmed. (For example, if users dial 299 to reach Voice Mail, enter 299 to program the Voice Mail greeting.) Or, press * to exit to the System Administrator menu.
8. When prompted, select whether the recording is played during day mode (by pressing 1) or during night mode (by pressing 2).
9. When prompted, enter the desired recording number (001-500). If a recording exists, it is played. (If not, you hear a warning that the recording has not yet been made.)
10. If it is the correct recording, press #. Return to step 7.

Customized Voice Processing Prompts

9.66 All of the pre-recorded prompts in Voice Processing can be customized, using the Voice Mail System Administrator's mailbox.

9.67 If the Multilingual feature is enabled, and the Secondary Language prompts have been loaded, the System Administrator can change either the Primary or Secondary Language prompts. When the System Administrator's phone is in Primary Language mode, the Primary Language prompts is changed when the following procedure is performed. If the System Administrator's phone is in Secondary Language mode, the Secondary prompts is changed.

9.68 A list of all pre-recorded prompt numbers (001-944) and their text is provided in Table 4-4, beginning on page 4-72. The Secondary Language prompts have the equivalent text in the same prompt numbers. If you change any of the prompts, you should create a chart like the one in the table that shows the new text.

NOTE: If you change the text of the prompts, be sure you keep the content similar to the default text. Otherwise, the prompts do not make sense to the listeners when played by Voice Processing.

9.69 To record custom system prompts (or restore system prompts):

1. Dial the Voice Mail application access number. You hear the main menu.
2. During or after the greeting, press ✱ to identify yourself as a subscriber.
3. Enter the System Administrator mailbox number and password.
4. Press 9 to access the System Administrator's Menu.
5. When you hear the System Administrator Menu, press 6 to select the System Prompt Replacement option.
6. Enter the desired prompt number (001-944) followed by # or the **ACCEPT** menu button. If the number is valid, the current system and/or custom prompt is played. (*If the number is not valid,* you hear an error message and must choose another prompt number.)
7. Do one of the following:
   - **To hear the prompt again:** Press 1.
   - **If you want to replace the current system or custom prompt with a new custom prompt**: Press 3 and continue to the next step.
   - **To delete a custom prompt and reinstate the system prompt:** Press 9.
   - **If you want to leave the prompt unchanged and return to the previous menu**: Press ✱.
8. Record the new prompt. *(To pause while recording the prompt,* press 2. To continue, press any button.)

   If the Voice Processing PC disk is full, a prompt notifies you that it cannot accept your prompt. Hang up and try again later.

9. When you have completed recording your prompt, you have the following options:
   - **If you are satisfied with your prompt**: Hang up.
   - **To hear your prompt:** Press 1.
   - **To add to your prompt:** Press 2 and continue as described in steps 3-8.
   - **To erase and re-record your prompt:** Press 3.

Importing Fax Documents

9.70 If you have a fax card installed on your Voice Processing Unit, you can use the Voice Mail System Administrator's mailbox to import fax documents. Imported documents can then be requested by callers who use the Call Routing Announcement application that is programmed to provide the Fax-On-Demand service.

9.71 To import a fax document:

1. Using a fax machine, dial the Voice Mail application access number. You hear the main menu.
2. During or after the greeting, press * to identify yourself as a subscriber.
3. Enter the System Administrator mailbox number and password.
4. Press 9 to access the System Administrator's Menu.
5. When you hear the System Administrator Menu list of options, press 4 to select the Import Fax option. (Note that, if the Fax-On-Demand feature is not enabled, the system does not accept this digit). The system may inform you of one of the following, if not, continue to the next step.

   All system fax ports are busy: Hang up and try again later.

   The system already contains the maximum number of fax documents allowed: Until one or more fax documents are deleted from the system, there is no room for additional documents.

   The system does not have enough disk space to import any more documents: Until one or more fax documents are deleted from the system, there is no room for additional documents.

   The system currently is at or above the Maximum Fax Library Size: Until one or more fax documents are deleted from the system, there is no room for additional documents.
6. When prompted, enter the fax document number. (Or, to exit to the System Administrator's Menu, press *.) One of the following occurs when you enter the number:

   If the document number does not already exist, the system asks you to verify the number. Press # if the number is correct and go to the next step. Or, press 3 if you want to re-renter the number.

   If the document number already exists, the system asks if you want to replace it. Press # if you want to replace the document and go to the next step. Or, press 3 if you want to re-renter the number.

   If the document is currently being sent, updated, or deleted, that document number cannot be used right now. You are prompted to enter a new number. Repeat step 6.

   If you did not enter a valid number, you are prompted to enter a new number. Repeat step 6.

7. When prompted, press **START** on your fax machine. One of the following may occur:

   If the system cannot communicate with your fax machine, check that it is working properly and press # to try again.

   If the system runs out of disk space while importing the document or if the document causes the system to exceed the Maximum Fax Library Size, you cannot import this document until one or more fax documents have been deleted from the system to create disk space. Press # to return to step 6.

8. When the fax transmission is complete, the system tells you that the document was received. Press # to continue importing documents, or press * to exit.

10. DIRECTORIES

10.1 There are two types of Voice Processing directories that can be enabled (or disabled) in the system: Mailbox and Extension.

- **The Voice Mailbox Directory** is a list of mailbox subscribers, their recorded names, and mailbox numbers. (See page 6-354 for programming information.)
- **The Automated Attendant Directory,** provided to all Auto Attendant callers, is a list of all mailbox subscribers and extension ID owners and their recorded names. (See page 6-355 for programming information.)

10.2 The directories can be selected in the following ways:

- **From the Voice Mail or Automated Attendant main menu**: Press the # button.
- **Using a Call Routing Announcement application**: Press the Digit Translation digit designated for directory access. (See page 4-21 for a detailed explanation of digit translation.)
- **As a Voice Mail Subscriber**: Subscribers can access the directory by pressing # whenever they are prompted to enter a mailbox number.

10.3 If a directory is disabled or empty because no names are recorded for any of the system's mailboxes and extension IDs, callers do not hear the prompt that allows access to the directory. However, if the caller does press # or a Directory menu button, the user is instructed that the selection is invalid and returned to the initial instructions.

10.4 Recorded names are added to the directories when the owner of the mailbox or extension ID has initialized the name. If the mailbox or ID is not initialized, the directory includes only the mailbox number or extension ID number. (Group Lists are not included in either type of directory.)

10.5 The caller uses the keypad buttons to enter the name. The application then plays the closest matching directory name that corresponds to the digits entered by a caller. After the name has been played, the system returns a menu of options, including the following:

- Listening to the previous or next name in the directory
- Listening to additional information (not available if the mailbox or extension ID number has been classified in the database as a "Private" number)
- Spelling a new name
- Toggling from a last to first name search mode
- Accepting the name

A. LOCATING A NAME

Entering a Name

10.6 There are two methods used for entering a name: Quick Spell and Exact Spell. Outside callers and station users with non-display phones and single-line sets use Quick Spell. Display phone stations use the Exact Spell method.

- Quick Spell callers press a single digit (or button) from their digital telephone dial pad for each letter or character entered. For example, keypad button 2 shows ABC, button 3 shows DEF, etc. To enter JONES, you would press 5 6 6 3 7. Some characters are not shown on the buttons: for "Q" press 7, for "Z" press 9, for punctuation marks press the 1 button.
- Exact Spell callers press the keypad buttons to enter the name. The number of times a button is pressed determines which character is entered, as shown on the chart below. When adjoining characters are located under the same button, press FWD [] once to advance to the next character. For example, 5666 FWD [] 66337777 would enter JONES.

Table 4-3. *Number of Times Button is Pressed*

BTN	Number of Times Button is Pressed				
	1	2	3	4	5
1	-	&	(	)	1
2	A	B	C	'	2
3	D	E	F	!	3
4	G	H	I	*	4
5	J	K	L	# or /*	5
6	M	N	O	Ñ or #*	6
7	P	Q	R	S	7
8	T	U	V	?	8
9	W	X	Y	Z	9
0	@	:	.	,	0

*The character available depends on the software version.

10.7 After the digits have been entered, the caller presses # to begin the search. The Automated Attendant application plays the name that most closely matches the digit(s) that were entered. If # is pressed without entering any digits, the caller hears the first name in the directory.

10.8 If a user presses 0 at any time while spelling a name, the system plays a helpful prompt instructing the caller how to enter a name. (Display phone users receive Exact Spell instructions; all other users hear Quick Spell instructions.)

Changing the First/Last Name Search

10.9 Each directory can be programmed to be organized by last name or by first name. Callers accessing a directory receive system voice prompts that ask them to enter the first or last name of the person for whom they want to leave a message.

10.10 If a phone with menu buttons is used to access a directory, the caller may switch back and forth from looking up a last name, to a first name simply by pressing the corresponding menu button. Callers using Standard Display, Associate Display, or Model 8520 Phones or single-line sets can press **5** to toggle between last and first name.

Listening to the Next/Previous Name

10.11 After a name has been played, the caller can listen to the previous name in the directory by pressing **1**. To listen to the next name, the caller can press **3**.

10.12 The directory lists are circular. That is, when the end of the list is reached, the next name played is the first name in the directory. Or, if the caller scrolls to the beginning of the list, the "previous" name played is the last name in the directory.

B. ACCEPTING A NAME

10.13 When the caller is using the Voice Mail feature and accepts a name, the caller is transferred to the corresponding mailbox. Then the caller hears either the subscriber's recorded greeting, recorded directory name, or, if no recording has been made, the "mailbox number XXX is not available" prompt. The caller can then record a message.

10.14 If the caller is using the Automated Attendant feature and accepts the name, the caller is transferred to the selected destination (station or mailbox) if it is available. If a station extension is dialed and the destination is not available, the caller is sent to the associated mailbox, if one exists.

C. REQUESTING ADDITIONAL INFORMATION

10.15 After a name is selected from the directory, the caller has the option of requesting additional information. When requested, the mailbox number, extension number, and name are played (if they have not been disallowed).

10.16 Individual mailboxes and extension IDs can be programmed as "Private" in DB Programming. A Private number is included in the directory and can be dialed by a caller who knows the number, but the associated mailbox and extension number information are not available to the caller; only the name is played.

10.17 Mailboxes and extension IDs can also be programmed as "Unlisted." That means that the number is not included in the directory but can be dialed if the caller knows the extension number.

D. USING THE DIRECTORIES

10.18 To use a directory:

1. Dial the Voice Mail access number. You hear the recorded system greeting.
2. Press # to access the Directory Services application.
3. The buttons on your telephone are labeled with letters. Use Quick Spell (non-display phone or single-line) or Exact Spell (display phone) to enter the desired name, as described on page 4-63. Then press #. (If you want to exit from the directory without selecting a name, press * or hang up.)
4. The Voice Mail application plays the selected name. Do one of the following:
   - **To accept the name:** Press #.
   - **To hear the previous name in the directory:** Press 1.
   - **To hear additional information for the selected name (if allowed):** Press 2.
   - **To hear the next name in the directory:** Press 3.
   - **To spell a different name:** Press 4 and repeat step 3.
   - **To switch the first/last name sort order:** Press 5.

11. SMDR INFORMATION STORAGE AND RETRIEVAL

NOTE: The SMDR Information Storage and Retrieval feature is not applicable to Europe.

11.1 The Voice Processing PC can be used to capture Station Message Detail Recording (SMDR) information. The SMDR information is stored in a file on the Voice Processing PC's hard disk for later retrieval.

11.2 Every time a call is completed, the system SMDR feature records the call, formats it according to SMDR programming (see page 5-308), and sends the record to the SMDR output device, if enabled. The formatted record is also sent to the Voice Processing PC if SMDR buffering is enabled.

11.3 The Voice Processing PC adds each call record to the SMDR buffer file as it is received. The amount of disk space allocated for SMDR storage is determined in DB Programming. The default value is zero megabytes of memory. If the call records use disk space exceeding 80% of the allotment, the system generates an alarm and displays it at the primary attendant station and all telephone system administrator stations. Another alarm message is generated if the allotted space is 100% filled. If the maximum buffer capacity is exceeded, the oldest call record(s) are discarded as new ones arrive.

NOTE: Because each SMDR call record takes up approximately 81 bytes on the Voice Processing PC, nearly 13,000 calls can be stored in each megabyte of hard disk space.

11.4 The SMDR records, which are stored in ASCII/binary format, are buffered until they are cleared by the programmer. If desired, the programmer may save and/or restore SMDR data (along with the system database) using the Voice Processing Save and Restore feature.

11.5 If desired, the SMDR records can be retrieved and processed using Inter-Tel Inside Track, a separate call accounting software package, to create customized reports. In a network setting, the Voice Processing Unit buffers SMDR records that are sent from other nodes in the network.

12. UNIFIED MESSAGING

12.1 Unified Messaging versions 2.2 (VPU) and 2.3 (EM) have reached their end of sale and are no longer supported as of April 30, 2008. Inter-Tel recommends that current UM v2.2 and v2.3 installations upgrade to UM Open Standards Edition (OSE). See below for more information.

13. UNIFIED MESSAGING OPEN STANDARDS EDITION (OSE)

13.1 Axxess v8.23x or later supports Unified Messaging OSE, a client-free Unified Messaging distribution for the Inter-Tel Enterprise Messaging voice messaging server. Unlike previous Unified Messaging products from Inter-Tel, users do not have to install a special Unified Messaging client application on their desktop computer to view, play, and manage voice mail messages. With Unified Messaging OSE, the e-mail client can be any Internet Message Access Protocol (IMAP) v4 compliant client that is configured to connect to the IMAP server located on Enterprise Messaging. To use the latest version of Unified Messaging OSE, v2.0, existing Axxess systems must upgrade to v10.2 or later, and to Enterprise Messaging v1.2 or later.

13.2 For Unified Messaging OSE v2.0 programming and feature information, refer to the latest version of the *Unified Messaging Open Standards Edition Administrator's Guide* (document part no. 835.3162). For end-user information and instructions for using features, refer to the *Unified Messaging Open Standards Edition User Guide* (part number 835.3164).

14. AUTOMATIC FAX DETECTION

14.1 With Automatic Fax Detection, Call Routing Announcement applications and mailboxes can be programmed to automatically route incoming fax calls to a specified extension or to an e-mail address.

14.2 If the feature is enabled, the mailbox or Call Routing Announcement Application automatically listens for fax tone.

- A mailbox listens for fax tones during the mailbox greeting and while a message is being recorded. If the mailbox detects fax tones, the call is transferred to the specified extension or e-mail address. If the tones are detected after a recording has started, the call is disconnected.
- Call Routing Applications can detect fax tones during the greeting and up to timeout. The fax tone detection also is disabled if the caller performs an action that removes them from the Call Routing Announcement (transferring to an extension, transferring to a mailbox, etc.).

14.3 If the fax card is busy when an incoming fax call is received, the call is disconnected.

A. UNIFIED MESSAGING REQUIRED FOR FAX E-MAIL

14.4 To send faxes to an e-mail address using the Fax-On-Demand feature, the Unified Messaging feature, described on page 4-67, must also be enabled. Note that programming a mailbox Fax E-mail Address counts as one user, as described in the *Security Key* section on page 2-92.

B. FAX CARD REQUIRED

14.5 To send and receive faxes, the Voice Processing PC must have a fax card installed. This can be the same fax card that supports the Fax-On-Demand feature. The software supports up to eight fax ports. See page 6-401 for more information on allocating fax ports for the Automatic Fax Detection and Fax-On-Demand features.

C. PROGRAMMING CALL ROUTING ANNOUNCEMENTS AND MAILBOXES

14.6 Both Call Routing Announcements and voice mailboxes now have two Fax Detection fields.

- **Fax Destination**: Specifies the extension of the fax machine that receives incoming faxes.
- **Fax E-Mail Address**: Specifies the E-mail Address of the account that receives incoming faxes. The fax is converted to a TIFF file and sent to the e-mail address as an attached file. The address can be up to 127 characters. To view a message, use any TIFF file viewer (such as Imaging for Windows).

14.7 If only the Fax Destination field is programmed, the Voice Processing Unit performs a blind transfer to send the fax to the programmed extension, allowing the fax machine to process the incoming fax.

14.8 If the Fax E-mail Address field is programmed, the disk is not full, and the Voice Processing Unit has an available fax port to receive the incoming fax, the Voice Processing Unit receives the fax and stores the image in TIFF Class F format. If the Voice Processing Unit cannot receive the fax because the disk is full or the fax ports are busy, the fax is transferred to the Fax Destination extension. If the Fax Destination is not programmed, the Voice Processing Unit disconnects the call (because it cannot process the incoming fax).

14.9 If the disk becomes full while the Voice Processing Unit is receiving the fax, the Voice Processing Unit disconnects the call and deletes the portion of the fax received to that point. If the Voice Processing Unit receives the fax successfully, it disconnects the call and places the received fax in a queue to be delivered to e-mail. The Voice Processing Unit processes the fax queue and delivers the fax images to the appropriate destination e-mail accounts in the order they were received.

D. FAX E-MAIL MESSAGE FORMAT

14.10 The e-mail message sent by the Voice Mail has the following components:

- **To**: Contains the recipient's e-mail address.
- **From**: Contains the e-mail address of the Voice Processing Unit. If an E-mail Real Name has been programmed for the Voice Processing Unit, it is also shown on this line.
- **Subject**: Contains the line "Fax received from <*source*>." The source is the Call Routing Application number or the mailbox number from where the fax is received, for example: CRA 2505 or MB 1000.
- **Received On**: Contains the date and time that fax was received.
- **FAX.TIF file attachment**: Appears as a file with the name FAX.TIF. To view the fax, the user must display the TIF file using graphics software.

15. ENTERPRISE MESSAGING ENHANCEMENTS

The following section describes new features that are to be introduced in EM v2.1. These new features require a v11.0 or later system:

- Option Swap: 7 and 9 Keys (see below)
- VPIM Networking (see page 4-71)

A. OPTION SWAP: 7 AND 9 KEYS

15.1 EM provides a swap feature whereby the 7 and 9 digit operations in Message Options can be reversed. By default, the 7 digit is used for SAVE and the 9 digit is used for DELETE. The swap feature allows the 7 digit to be used for DELETE and the 9 digit to be used for SAVE, just like the functionality operated by the cell phone voice mail systems.

15.2 This helps users who are familiar with a cell phone functionality from accidentally deleting messages by pressing 9, intending to save messages. This feature can be programmed for individual mailboxes by the subscribers (see below) or for all mailboxes in DB Programming (see page 6-347).

15.3 To use the 7 and 9 key swap feature at the endpoint display:

1. On the endpoint display, locate the these changes in the **Additional Options** menu under the **Personal Options – More** selection.
2. Do either of the following:
   - Select **Save Del** to display the current settings.
   - Select **Save Del** again to toggle the settings for the 7 and 9 digits. When you toggle the settings, the display shows the new settings for a few seconds before returning to the previous menu.

15.4 The new voice prompts 947 through 953 have been added for this feature (see page 4-98).

B. VPIM NETWORKING

15.5 Voice Profile for Internet Mail (VPIM) allows Enterprise Messaging to communicate with other voice processing systems. VPIM networking requires the Inter-Tel Messaging IP Networking premium feature.

NOTE: To receive and send e-mail across the Internet to a remote site, Enterprise Messaging needs access outside its local network. Information needs to come into port 25 on the Inter-Tel Axxess. This may require changes to any firewall between the Inter-Tel Axxess and the Internet.

15.6 You must configure the following areas to enable VPIM:

- VPIM nodes (see page 6-395)
- System-level VPIM settings
  - — VPIM Postmaster (see page 6-345)
  - — VPIM Home Domain (see page 6-346)

16. DEFAULT VOICE PROMPTS

16.1 The following table shows all of the American English and British English default system voice mail prompts for ASR- and non ASR-enabled voice mail systems.

NOTE: Prompts designated by an asterisk (*) cannot be re-recorded.

Table 4-4: *Voice Prompts: Non-ASR and ASR*

Prompt ID	Content (Non-ASR)	Content (ASR)
001	To leave a message, please enter a mailbox number.	To leave a message, please say or enter a mailbox number.
002	For the directory, press pound (or hash in European systems).	For the directory, say “Directory” or press pound (or hash).
003	To reach an operator, press zero.	To reach an operator, say “Operator” or press zero.
004	I’m sorry...	N/A
005	...is not available.	N/A
006	The number entered is not valid.	N/A
007	Using the keys on your telephone, please spell the name of the person, then press pound (or hash). For the letter Q, press seven. For the letter Z, press nine. To advance to the next character, please wait for two seconds or press another key. To exit the directory, press star.	Please say the name of the person you are trying to reach. Otherwise, using the keys on your telephone, please spell the last name of the person, then press pound (or hash). For the letter Q, press seven. For the letter Z, press nine. To advance to the next character, please wait for two seconds or press another key. To exit the directory, press star.
008	To exit the directory, press star.	To exit the directory, say “Cancel” or press star.
009	Mailbox number...	N/A
010	If this is correct, press pound (or hash).	If this is correct, say “Yes” or press pound (or hash).
011	For the previous name in the directory, press one.	For the previous name, say “Previous” or press one.
012	To hear additional information, press two.	To hear additional information, say “More” or press two.
013	To hear the next directory name, press three.	To hear the next name, say “Next” or press three
014	To spell a new name, press four.	To spell a new name, say “New” or press four.
015	Extension number...	N/A
016	Not used	Not used
017	...is not a valid choice.	N/A
018	Please try again.	N/A
019	After the tone, please record your message. When finished, you may hang up to deliver the message or press pound (or hash) for more options.	N/A
020	Message delivered.	N/A
021	To replay, press one.	To replay, say “Replay” or press one.

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
022	To erase and re-record, press three.	To erase and re-record, say “Erase” or press three.
023	To append this recording, press two.	To append this recording, say “Append” or press two.
024	To accept, press pound (or hash).	To accept, say “Accept” or press pound (or hash).
025	Please record after the tone. To end, press pound (or hash).	N/A
026	Recording erased.	N/A
027	Recording stopped.	N/A
028	Please begin speaking after the tone.	N/A
029	Message play stopped. To continue, press pound (or hash).	Message play stopped. To continue, say “Resume” or press pound (or hash).
030	Enter the destination mailbox or group list number.	Say or enter the destination mailbox or group list number.
031	Logon canceled.	N/A
032	Enter your password. Then press pound (or hash).	Say or enter your password.
033	There have been three invalid attempts to log onto your mailbox.	N/A
034	Invalid password.	N/A
035	Thank you for calling. Good-bye.	N/A
036	To access mailbox features, you must change your system-assigned password.	N/A
037	You have not finished setting up your mail box.	N/A
038	Subscriber access denied.	N/A
039	Because this is the first time you are entering your mailbox, you must record your name for the company directory.	N/A
040	Since you are entering your mailbox for the first time, you should be aware of a few standard features. To cancel an option you have accessed and return to the previous menu press star. To accept your recording or dialing sequence, or to advance to your next message press pound (or hash). For help, you may press zero at any time you are at a menu and the system replays the current menu and or gives you additional information. Certain mailbox features such as remote messaging are available only if authorized by your System Administrator. Your mailbox is now fully set up.	Since you are entering your mailbox for the first time, you should be aware of a few standard features. To cancel an option you have accessed and return to the previous menu, say “Cancel” or press star. To accept your recording or dialing sequence, or to advance to your next message, say “Accept” or press pound (or hash). For help, say “Help” or press zero to replay the current menu and/or give you additional information. Certain mailbox features such as remote messaging are available only if authorized by your System Administrator. Your mailbox is now fully set up.
041	Your password may be up to 12 digits.	N/A
042	Enter your new password, then press pound (or hash). If you don’t want a password, press pound (or hash) now.	Say or enter your new password. If you don’t want a password, say “Skip” or press pound (or hash).
043	Your new password is…	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
044	The number entered is too long.	N/A
045	To accept this entry, press pound (or hash).	To accept this entry, say “Accept” or press pound (or hash).
046	To erase and re-enter, press three.	To erase and re-enter, say “Re-enter” or press three.
047	Password erased.	N/A
048	Password saved.	N/A
049	Your mailbox is full and cannot receive any new messages. Please delete old messages.	N/A
050 - 051	Not used	Not used
052	Your mailbox is almost full. Please delete any messages you no longer need so new messages may arrive.	N/A
053	You have...	N/A
054	You have one new message.	N/A
055	...new messages.	N/A
056	...of which is priority.	N/A
057	You have no new messages.	N/A
058	You have one saved message.	N/A
059	...saved messages.	N/A
060	Remote messaging is currently enabled.	N/A
061	To listen to new messages, press one.	To listen to new messages, say “New” or press one.
062	To record a message, press two.	To record a message, say “Record” or press two.
063	To listen to saved messages, press three.	To listen to saved messages, say “Saved” or press three.
064	For personal options, press four.	For personal options, say “Personal” or press four.
065	To setup remote messaging, press five.	To setup remote messaging, say “Remote” or press five.
066	For your primary setup, press one.	For your primary setup, say “Primary” or press one.
067	For your alternate setup, press two.	For your alternate setup, say “Alternate” or press two.
068	If this is the correct destination, press pound (or hash).	If this is the correct destination, say “Yes” or press pound (or hash).
069	To send this message now, press pound (or hash).	To send this message now, say “Send” or press pound (or hash).
070	For special delivery options, press nine.	For special delivery options, say “Options” or press nine.
071	To return to the previous menu, press star.	To return to the previous menu, say “Cancel” or press star.
072	Message canceled.	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
073	To mark this message private, press one.	To mark this message private, say “Private” or press one.
074	To mark this message for certified delivery, press two.	To mark this message for certified delivery, say “Certified” or press two.
075	To mark this message priority, press three.	To mark this message priority, say “Priority” or press three.
076	Message marked private.	N/A
077	Message marked certified.	N/A
078	Message marked priority.	N/A
079	To send this message to additional mail boxes, press pound (or hash).	To send this message to additional mail boxes, say “Yes” or press pound (or hash).
080	To rewind your message or recording, press one.	To rewind your message or recording, say “Backup” or press one.
081	To pause at any time, press two.	To pause at any time, say “Pause” or press two.
082	To fast forward your message or recording, press three.	To fast forward your message or recording say “Forward” or press three.
083	To lower the volume, press four.	To lower the volume, say “Lower” or press four.
084	To raise the volume, press six.	To raise the volume, say “Higher” or press six.
085	To save this message, press seven.	To save this message, say “Save” or press seven.
086	To delete this message, press nine.	To delete this message, say “Delete Message” or press nine.
087	To skip to the end of your message or recording, press pound (or hash).	To skip to the end of your message or recording, say “Skip” or press pound (or hash).
088	To return to the main menu, press star.	To return to the main menu, say “Cancel” or press star.
089	To reply to this message, press two.	To reply to this message, say “Reply” or press two.
090	To forward a copy of this message to another destination, press three.	To forward a copy of this message to another destination, say “Forward” or press three.
091	To replay the previous message, press four.	To replay the previous message, say “Previous” or press four.
092	To replay the message envelope, press five.	To replay the message envelope, say “Envelope” or press five.
093	To listen to your next message, press six.	To listen to your next message, say “Next” or press six.
094	There are no previous messages.	N/A
095	Message saved.	N/A
096	Message deleted.	N/A
097	There are no further messages.	N/A
098	Message received...	N/A
099	...at...	N/A
100	...from...	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
101	...totaling...	N/A
102	...minutes...	N/A
103	...minute...	N/A
104*	...seconds...	N/A
105	...second...	N/A
106	Message received from an outside caller.	N/A
107	To record an introduction with a copy of this message, press one.	To record an introduction with a copy of this message, say “Record” or press one.
108	To change your personal greeting, press one.	To change your personal greeting, say “Greeting” or press one.
109	To change your name in the directory, press two.	To change your name in the directory, say “Name” or press two.
110	To change your password, press three.	To change your password, say “Password” or press three.
111	To change your message envelope, press four.	To change your message envelope, say “Envelope” or press four.
112	To replay the message, press one.	To replay the message, say “Replay” or press one.
113	To select your primary greeting, press one.	To select your primary greeting, say “Primary” or press one.
114	To select your alternate greeting, press two.	To select your alternate greeting, say “Alternate” or press two.
115	To select the system greeting, press three.	To select the system greeting, say “System” or press three.
116	After the tone, please record your greeting, then press pound (or hash).	N/A
117	Greeting saved.	N/A
118	After the tone, please record your name, then press pound (or hash).	N/A
119	Name saved.	N/A
120	Your message envelope is currently set to return...	N/A
121	...all information.	N/A
122	...time and date...	N/A
123	...message source...	N/A
124*	...and...	N/A
125	...message length...	N/A
126	To turn on all envelope options, press four.	To turn on all envelope options, say “All Options” or press four.
127	For message source, press two.	For message source, say “Source” or press two.
128	For message length, press three.	For message length, say “Length” or press three.
129	This option is now off.	N/A
130	This option is now on.	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
131	For time and date, press one.	For time and date, say “Time, Date” or press one.
132	Your message envelope will play all options.	N/A
133	...is turned on to call you at...	N/A
134*	AM	N/A
135*	PM	N/A
136	...Extension ID number...	N/A
137	...Every day.	N/A
138	...Monday through Friday.	N/A
139*	Sunday	N/A
140*	Monday	N/A
141*	Tuesday	N/A
142*	Wednesday	N/A
143*	Thursday	N/A
144*	Friday	N/A
145*	Saturday	N/A
146	To turn this number on or off, press one.	To turn this number on or off, say “Change” or press one.
147	To change the phone number, press two.	To change the phone number, say “Number” or press two.
148	To change times of day, press two.	To change times of day, say “Time” or press two.
149	To change days of the week, press three.	To change days of the week, say “Days” or press three.
150	For pager notification, press one.	For pager notification, say “Pager” or press one.
151	For personal phone notification, press two.	For personal phone notification, say “Personal” or press two.
152	For notification of all new messages, press one.	For notification of all new messages, say “All Messages” or press one.
153	For priority-only, press two.	For priority-only, say “Priority” or press two.
154	Please enter the destination telephone number and wait for further instructions.	Please say or enter the destination telephone number and wait for further instructions.
155	The new number is...	N/A
156	Please enter the start time.	Please say or enter the start time.
157	Please enter the stop time.	Please say or enter the stop time.
158	I’m sorry. You must program a telephone number first.	N/A
159	Invalid time.	N/A
160	For AM, press one.	For AM, say “AM” or press one.
161	Please select the days of the week on which you wish to be notified.	N/A
162	Not used	Not used
163	Please enter your mailbox number.	Please say or enter your mailbox number.

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
164	Today	N/A
165	Yesterday	N/A
166*	January	N/A
167*	February	N/A
168*	March	N/A
169*	April	N/A
170*	May	N/A
171*	June	N/A
172*	July	N/A
173*	August	N/A
174*	September	N/A
175*	October	N/A
176*	November	N/A
177*	December	N/A
178*	First	N/A
179*	Second	N/A
180*	Third	N/A
181*	Fourth	N/A
182*	Fifth	N/A
183*	Sixth	N/A
184*	Seventh	N/A
185*	Eighth	N/A
186*	Ninth	N/A
187*	Tenth	N/A
188*	Eleventh	N/A
189*	Twelfth	N/A
190*	Thirteenth	N/A
191*	Fourteenth	N/A
192*	Fifteenth	N/A
193*	Sixteenth	N/A
194*	Seventeenth	N/A
195*	Eighteenth	N/A
196*	Nineteenth	N/A
197*	Twentieth	N/A
198*	Twenty-first	N/A
199*	Twenty-second	N/A
200*	Twenty-third	N/A
201*	Twenty-fourth	N/A
202*	Twenty-fifth	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
203*	Twenty-sixth	N/A
204*	Twenty-seventh	N/A
205*	Twenty-eighth	N/A
206*	Twenty-ninth	N/A
207*	Thirtieth	N/A
208*	Thirty-first	N/A
209	...the Voice Mail system.	N/A
210	The system is set to notify you between...	N/A
211	...is turned off.	N/A
212	...is turned on to call your pager at...	N/A
213	...with all new messages.	N/A
214	...with all priority messages.	N/A
215	...starting at...	N/A
216	...ending at...	N/A
217	For mailbox and group list maintenance, press two.	For mailbox and group list maintenance, say “Mailbox Maintenance” or press two.
218	To send a message to all subscribers, press one.	To send a message to all subscribers, say “Broadcast Message” or press one.
219	To make a custom recording, press one.	To make a custom recording, say “Custom” or press one.
220	To select a custom recording, press two.	To select a custom recording, say “Select” or press two.
221	Not used	Not used
222	Please enter the recording number.	N/A
223	After the tone, please record. Then press pound (or hash).	N/A
224	Recording saved.	N/A
225	To continue recording, press pound (or hash).	To continue recording, say “Continue” or press pound (or hash).
226	Not used	Not used
227	You have no saved messages.	N/A
228	To spell the first name, press five.	To search by first name, say “First” or press five.
229	To spell the last name, press five.	To search by last name, say “Last” or press five.
230	Using the keys on your telephone, please spell the first name of the person. Then press pound (or hash). For the letter Q, press seven. For the letter Z, press nine.	Please say the first name of the person you are trying to reach. Otherwise, using the keys on your telephone, please spell the first name of the person, then press pound (or hash). For the letter Q, press seven. For the letter Z, press nine.
231	No message left.	N/A
232	...last...	N/A
233	...an outside caller at...	N/A
234	Message was sent from...	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
235	This matches the mailbox of...	N/A
236	...an unknown caller.	N/A
237	To turn off the envelope, press five.	To turn off the envelope, say “None” or press five.
238	To accept the changes, press pound (or hash).	To accept the changes, say “Accept” or press pound (or hash).
239	The envelope is turned off.	N/A
240	Please enter the application extension number.	Please say or enter the application extension number.
241	To select a day recording, press one.	To select a day recording, say “Day” or press one.
242	To select a night recording, press two.	To select a night recording, say “Night” or press two.
243	If this is the correct recording, press pound (or hash).	If this is the correct recording, say “Yes” or press pound (or hash).
244	Warning, no recording has been made.	N/A
245	Invalid application entered.	N/A
246	...is not a valid recording number.	N/A
247 - 252	Not used	Not used
253	This is a return receipt verifying the message sent has been heard.	N/A
254	I’m sorry. Private messages cannot be for warded.	N/A
255	This is the message center calling for...	N/A
256	If you are the correct person, press pound (or hash).	If you are the correct person, say “Yes” or press pound (or hash).
257*	...One	N/A
258*	...Two	N/A
259*	...Three	N/A
260*	...Four	N/A
261*	...Five	N/A
262*	...Six	N/A
263*	...Seven	N/A
264*	...Eight	N/A
265*	...Nine	N/A
266*	...Ten	N/A
267*	...Eleven	N/A
268*	...Twelve	N/A
269*	...Thirteen	N/A
270*	...Fourteen	N/A
271*	...Fifteen	N/A
272*	...Sixteen	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
273*	...Seventeen	N/A
274*	...Eighteen	N/A
275*	...Nineteen	N/A
276*	...Twenty	N/A
277*	...Thirty	N/A
278*	...Forty	N/A
279*	...Fifty	N/A
280*	...Sixty	N/A
281*	...Seventy	N/A
282*	...Eighty	N/A
283*	...Ninety	N/A
284*	...Hundred	N/A
285	This mailbox is currently full and cannot receive new messages.	N/A
286*	Oh	N/A
287	...of which are priority.	N/A
288	Priority message received...	N/A
289	Otherwise, press star.	Otherwise, say “No” or press star.
290	When you are finished entering mailbox numbers, press pound (or hash).	N/A
291	Otherwise, press pound (or hash).	Otherwise, say “Send” or press pound (or hash).
292	For PM, press two.	For PM, say “PM” or press two.
293	For Sunday, press one.	For Sunday, say “Sunday” or press one.
294	For Monday, press two.	For Monday, say “Monday” or press two.
295	For Tuesday, press three.	For Tuesday, say “Tuesday” or press three.
296	For Wednesday, press four.	For Wednesday, say “Wednesday” or press four.
297	For Thursday, press five.	For Thursday, say “Thursday” or press five.
298	For Friday, press six.	For Friday, say “Friday” or press six.
299	For Saturday, press seven.	For Saturday, say “Saturday” or press seven.
300	For Monday through Friday service, press one.	For Monday through Friday service, say “Week-days” or press one.
301	For all the days of the week, press two.	For all the days of the week, say “Days” or press two.
302*	Thousand	N/A
303	The hard disk is...	N/A
304	...percent full.	N/A
305	I’m sorry. The system is full and cannot record any messages.	N/A
306*	Zero	N/A
307	pound (or hash)	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
308	star	N/A
309	Group list number...	N/A
310	Please speak louder.	N/A
311	You have recorded for the maximum time permitted.	N/A
312	Not used	Not used
313	Resuming playback.	N/A
314	You have one priority message.	N/A
315	Use two digits for the hour and two digits for the minutes.	N/A
316 - 317	Not used	Not used
318	You have chosen not to have a password.	N/A
319	I'm sorry. This mailbox is currently in use and cannot be accessed. Please try again later.	N/A
320	Not used	Not used
321	If you are calling from a rotary-type telephone, please stay on the line and your call will be answered.	N/A
322	You can not reply to a system sent message.	N/A
323 - 324	Not used	Not used
325	Please select the days of the week.	N/A
326	To select individual week days, press three.	To select individual week days, say "Day" or press three.
327	Service is set for Monday through Friday.	N/A
328	Service is set for each day of the week.	N/A
329	Recording canceled.	N/A
330	The system was busy and could not deliver message notification at the specified time.	N/A
331	The system was unable to deliver message notification due to a programming error.	N/A
332	Extension ID...	N/A
333	...the operator.	N/A
334	...the operator...	N/A
335	Remote Messaging will call...	N/A
336	To change a cascade level, press one.	To change a cascade level, say "Level" or press one.
337	To select a message notification category, press four.	To select a message notification category, say "Category" or press four.
338	Please enter the cascade level you wish to program.	Please enter the cascade level you wish to program. Say or enter one through nine using the keys on your telephone.
339	...with a pager number of...	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
340	To change the pager dial string, press three.	To change the pager dial string, say “Pager” or press three.
341	To call an internal extension, press one.	To call an internal extension, say “Internal” or press one.
342	To call an outside number, press two.	To call an outside number, say “Outside” or press two.
343	Not used	Not used
344	Please enter the digits for your digital pager and wait for further instructions.	Please say or enter the digits for your digital pager and wait for further instructions.
345	The new pager number is...	N/A
346	Mailbox Number	N/A
347	New Message Count	N/A
348	Pause	N/A
349	You must program a telephone number first.	N/A
350	You have chosen not to have a pager number.	N/A
351	Cascade Level...	N/A
352	You did not enter a valid document number.	N/A
353	This document is currently unavailable.	N/A
354	... is not a valid document number.	N/A
355	You have already selected that document.	N/A
356	To cancel this message, press star.	To cancel this message, say “Cancel” or press star.
357	You have selected the maximum number of documents allowed.	N/A
358	To send your fax, press pound (or hash).	To send your fax, say “Send” or press pound (or hash).
359	You have selected one document.	N/A
360	You haven’t selected any documents yet.	N/A
361	You may select one more document during this call.	N/A
362	You may select up to...	N/A
363	... more documents during this call.	N/A
364	Selections cancelled.	N/A
365	The number that will appear on your fax cover sheet is...	N/A
366	You have chosen not to have your extension appear on your cover sheet.	N/A
367	If you would like to have your fax delivered in the future, press pound (or hash).	If you would like to have your fax delivered in the future, say “Delay” or press pound (or hash).
368	To re-enter the number, press three.	To re-enter the number, say “Re-enter” or press three.

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
369	If you wish to have your extension appear on your cover sheet, enter the extension followed by pound (or hash).	If you wish to have your extension appear on your cover sheet, say or enter the extension then say or press pound (or hash).
370	If you are calling from your fax machine, press one.	If you are calling from your fax machine, say “Yes” or press one.
371	Press the start button on your fax machine now.	N/A
372	All of our fax lines are currently busy.	N/A
373	Please try again later.	N/A
374	Your fax will be delivered to...	N/A
375	Your fax will be delivered shortly.	N/A
376	I am unable to communicate with your fax machine. Please make sure it is operating properly and try again.	N/A
377	*(For U.S. systems)* Please enter the 10 digit telephone number of your fax machine, including area code. *(For European systems)* Please enter the telephone number of your fax machine, including your national dialing code if required.	*(For U.S. systems)* Please say or enter the 10 digit fax number, including area code. *(For European systems)* Please say or enter the telephone number of your fax machine, including your national dialing code if required.
378	The number you enter may be up to twenty digits in length.	N/A
379	Your fax is scheduled to be delivered at...	N/A
380	Please enter a document number.	Please say or enter the document number.
381	When you have finished importing documents, press star.	When you are done importing documents, say “Finished” or press star.
382	Document number...	N/A
383	...is currently in use.	N/A
384	...already exists.	N/A
385	To replace this document, press pound (or hash).	To replace this document, say “Replace” or press pound (or hash).
386	Document received.	N/A
387	Press pound (or hash) to continue.	Say “Continue” or press pound (or hash) to continue.
388	To import a fax document, press four.	To import a fax document, say “Import Fax” or press four.
389	*(For U.S. systems)* If this is an international call, enter 011 followed by the country code and fax number. Then press pound. *(For European systems)* If this is an international call, enter 00 code followed by your country code and your fax number. Then press hash.	*(For U.S. systems)* If this is an international call, say or enter 011 followed by the country code and fax number. Then say or press pound/. *(For European systems)* If this is an international call, say or enter 00 code followed by your country code and your fax number. Then press hash.
390	I’m sorry. I’m not allowed to dial that number.	N/A
391	To receive your documents, please hang up and call from your fax machine.	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
392	The system does not have enough disk space to import your document.	N/A
393	The system is unable to import your document because it will exceed the Maximum Fax Library Size.	N/A
394	For recording options, press three.	For recording options, say “Options” or press three.
395	There are...	N/A
396	... calls ahead of you.	N/A
397	Your call should be answered within...	N/A
398	You have selected...	N/A
399	...documents.	N/A
400	Hello, you have reached the Inter-Tel voice processing system.	N/A
401	Hello, you’ve reached the Inter-Tel automated call handling system.	N/A
402	To change your message search order, press two.	To change your message search order, say “Order” or press two.
403	Your current message search order is first in first out.	N/A
404	Your current message search order is last in, first out.	N/A
405	To search for messages first in first out, press one.	To search for messages first in first out, say “First In” or press one.
406	To search for messages last in first out, press two.	To search for messages last in first out, say “Last In” or press two.
407	To change the search order for new messages, press one.	To change the search order for new messages, say “New” or press one.
408	To change the search order for saved messages, press two.	To change the search order for saved messages, say “Saved” or press two.
409	You have received a new message	N/A
410	You have received a new voice mail	N/A
411	You have received a new e-mail	N/A
412	You have received a new fax	N/A
413	You have received…	N/A
414 - 499	Not used	Not used
500	If you know your party’s extension number, enter it now.	If you know your party’s extension, please say or enter it now.
501	To leave a message, press one.	To leave a message, say “Leave Message” or press one.
502	To try another extension, press two.	To try another extension, say “Extension” or press two.
503	Please hold while your call is being transferred to...	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
504	Not used	Not used
505	To return to the menu, press two.	To return to the menu, say “Menu” or press two.
506	Your fax is scheduled to be delivered on...	N/A
507	The system already contains the maximum number of fax documents allowed.	N/A
508	The system does not have enough disk space to import any more documents.	N/A
509	The system is currently at or above the Maximum Fax Library Size.	N/A
510	Levels one through nine are currently disabled.	N/A
511	Your call will be answered in the order it was received.	N/A
512	There is one call ahead of you.	N/A
513	To change your transfer method, press five.	To change your transfer method, say “Transfer” or press five.
514	Your current transfer method is unannounced.	N/A
515	Your current transfer method is screened.	N/A
516	Your current transfer method is announce only.	N/A
517	To select unannounced transfers, press one.	To select unannounced transfers, say “Unannounced” or press one.
518	For screened transfers, press two.	For screened transfers, say “Screened” or press two.
519	For announce only, press three.	For announce only, say “Announce” or press three.
520	Who should I say is calling?	N/A
521	To replay the announcement, press one.	To replay the announcement, say “Replay” or press one.
522	To send this call to Voice Mail, press two.	To send this call to voice mail, say “Voice Mail” or press two.
523	To forward this call to another extension, press three.	To forward this call to another extension, say “Forward” or press three.
524	To accept this call, press pound (or hash).	To accept this call, say “Accept” or press pound (or hash).
525	To refuse this call, press star.	To refuse this call, say “Refuse” or press star.
526	You have a call from...	N/A
527	To change the time and date, press five.	N/A
528	The current time is...	N/A
529	To accept the current time, press pound (or hash).	N/A
530	To enter a new time, use two digits for the hour and two digits for the minutes.	N/A
531	The current date is...	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
532	To accept the current date, press pound (or hash).	N/A
533	To enter a new date, use two digits for the month, two digits for the day, and two digits for the year.	N/A
534	Invalid date.	N/A
535	Please enter the extension number.	Please say or enter the extension number.
536 - 537	Not used	Not used
538	For system prompt replacement or reinstatement, press six.	For system prompt replacement or reinstatement, say “Replacement” or press six.
539	Please enter the number of the system prompt.	Please say or enter the number of the system prompt.
540	The system prompt has been replaced by the following custom prompt...	N/A
541	The original system prompt is...	N/A
542	To replace this system prompt with a custom prompt, press three.	To replace this system prompt with a custom prompt, say “Custom” or press three.
543	...is not a valid system prompt number.	N/A
544	To delete the custom prompt and reinstate the system prompt, press nine.	To delete the custom prompt and reinstate the system prompt, say “System” or press nine.
545	To replace this custom prompt with a new custom prompt, press three.	To replace this custom prompt with a new custom prompt, say “New Prompt” or press three.
546	You have no deleted messages.	N/A
547	You have one deleted message.	N/A
548	...deleted messages.	N/A
549	To recover deleted messages, press two.	To recover deleted messages, say “Recover” or press two.
550	To recover this message, press seven.	To recover this message, say “Recover” or press seven.
551	Message recovered.	N/A
552	To purge this message, press nine.	To purge this message, say “Purge” or press nine.
553	Message purged.	N/A
554	To listen to deleted messages, press one.	To listen to deleted messages, say “Listen” or press one.
555	To recover all messages, press two.	To recover all messages, say “Recover” or press two.
556	To purge all messages, press three.	To purge all messages, say “Purge” or press three.
557	...messages recovered.	N/A
558	...messages purged.	N/A
559	...message recovered.	N/A
560	...message purged.	N/A
561	For message options, press five.	For message options, say “Options” or press five.

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
562	For more options, press nine.	For more options, say “More” or press nine.
563	To change the fax destination number, press one.	To change the fax destination number, say “Number” or press one.
564	Please enter the extension number of your fax destination, followed by the pound (or hash) key.	Please say or enter the extension number of your fax. Then say or press pound (or hash).
565	Your fax destination is...	N/A
566	You have chosen not to have a fax destination.	N/A
567	Invalid extension.	N/A
568	Fax destination saved.	N/A
569	Fax destination erased.	N/A
570	To cancel unheard sent voice mail messages, press one.	To cancel unheard sent voice mail messages, say “Unheard” or press one.
571	All sent messages have been heard.	N/A
572	Enter the destination mailbox.	Say or enter the destination mailbox.
573	This mailbox has one unheard message.	N/A
574	...unheard messages.	N/A
575	To append to the message, press two.	To append the message, say “Append” or press two.
576	To leave a Voice Mail message, press one.	To leave a Voice Mail message, say “Voice Mail” or press one.
577	To make a return call, press two.	To make a return call, say “Call Back” or press two.
578	Please hold while your call is being transferred to an outside number.	N/A
579	Enter the number of the remote mailbox.	Say or enter the number of the remote mailbox.
580	... followed by pound (or hash).	N/A
581	…has just listened to the original message.	N/A
582	This mailbox has...	N/A
583 - 586	Not used	Not used
587	To hear the recipients, press three.	To hear the recipients, say “Hear” or press three.
588 - 591	Not used	Not used
592	I’m sorry. I’m not allowed to dial that number.	N/A
593	Message received from an outside number with no caller ID information.	N/A
594	I’m sorry. I do not have valid Caller ID information.	N/A
595	…the network…	N/A
596	Your call is being handled by the Inter-Tel voice processing system.	N/A
597	You have no new e-mails.	N/A
598	... new e-mails	N/A
599	you have one new e-mail.	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
600	...saved e-mails.	N/A
601	You have one saved e-mail.	N/A
602	You have no new faxes.	N/A
603	... new faxes.	N/A
604	You have one new fax.	N/A
605	...saved faxes.	N/A
606	You have one saved fax.	N/A
607	The system has not been properly configured.	N/A
608	To listen to voice mails, press one.	To listen to voice mails, say “Voice Mail” or press one.
609	To listen to e-mails, press three.	To listen to e-mails, say “E-mail” or press three.
610	To listen to faxes, press six.	To listen to Fax messages, say “Facsimile” or press six.
611	You have no new messages.	N/A
612	with the subject...	N/A
613	I’m sorry, you cannot delete this e-mail.	N/A
614	To reply to all recipients, press one.	To reply to all recipients, say “Everyone” or press one.
615	To reply to just the sender, press two.	To reply to just the sender, say “Sender” or press two.
616	I’m sorry, this message was received from an unknown source.	N/A
617	Please enter the digits for the fax destination and wait for further instructions.	N/A
618	The system could not access your e-mail account with the known password.	N/A
619	...totalling	N/A
620	...pages	N/A
621	I’m sorry, you cannot delete this fax.	N/A
622	For voice mails, press one.	For voice mails, say “Voice Mail” or press one.
623	For e-mails, press two.	For e-mails, say “E-mail” or press two.
624	For faxes, press three.	For faxes, say “Facsimile” or press three.
625 - 629	Not used	Not used
630	voice mail	N/A
631	e-mail	N/A
632	Fax	N/A
633 - 644	Not used	Not used
645	...flagged...	N/A
646	...saved....	N/A
647	...answered...	N/A
648	...recent...	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
649	...deleted...	N/A
650	…one page…	N/A
651	...voice mails...	N/A
652	...e-mails...	N/A
653	...faxes...	N/A
654	This message is marked...	N/A
655	...flagged...	N/A
656	...saved...	N/A
657	...answered...	N/A
658	...recent...	N/A
659	...deleted...	N/A
660	...with attachment...	N/A
661	For your voice mail envelope, press one.	For your voice mail envelope, say “Voice Mail” or press one.
662	For your e-mail envelope, press two.	For your e-mail envelope, say “E-mail” or press two.
663	For your fax envelope, press three.	For your fax envelope, say “Facsimile” or press three.
664	For message filters, press six.	For message filters, say “Filters” or press six.
665	For the message subject, press three.	For the message subject, say “Subject” or press three.
666	...message subject...	N/A
667	...message filters...	N/A
668	For the number of pages, press three.	For the number of pages, say “Pages” or press three.
669	You cannot turn all fax envelope options off.	N/A
670	...number of pages...	N/A
671	You have no new voice mails.	N/A
672	Fax scheduled for delivery.	N/A
673	...new voice mails.	N/A
674	You have one priority voice mail.	N/A
675	You have one new voice mail	N/A
676	...saved voice mails…	N/A
677	You have one saved voice mail.	N/A
678	To listen to messages, press one.	To listen to messages, say “Messages” or press one.
679	The system is momentarily unable to process your request.	The system is momentarily unable to process your request.
680	To wait for resources to become available, press one.	To wait for resources to become available, say “Wait” or press one.

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
681	To cancel the request and return to a previous menu, press star.	To cancel the request and return to a previous menu, say “Cancel” or press star.
682	Not used	Not used
683	You have one unheard voice mail.	N/A
684	You have one heard voice mail.	N/A
685	… unheard voice mail…	N/A
686	… heard voice mail…	N/A
687	You have no unheard voice mails.	N/A
688 - 692	Not used	Not used
693	You have one priority e-mail	N/A
694 - 699	Not used	Not used
700	N/A	Please spell the first few letters of where you would like to search in the directory.
701	...hours...	N/A
702	...hour...	N/A
703	<10 second pause>	N/A
704	<five second pause>	N/A
705*	<one second pause>	N/A
706	Fax cancelled	N/A
707	phone number…	N/A
708	Not used	Not used
709	Your results match more than…	Your results match more than…
710	…entries.	…entries.
711	If you would like to browse these results, press one. To filter your search, press two.	If you would like to browse these results, please say “Yes” or press one. To filter your search, say “No” or press two.
712	If the result is the first name, press one. Otherwise, please tell me the first name by pressing the keys on your telephone.	If that name is the first name, say “Yes” or press one. Otherwise, please tell me the first name by saying, spelling, or pressing the keys on your telephone.
713	If the result is the last name, press one. Otherwise, please tell me the last name by pressing the keys on your telephone.	If that name is the last name, say “Yes” or press one. Otherwise, please tell me the last name by saying, spelling, or pressing the keys on your telephone.
715 - 716	Not used	Not used
717	…one saved voice mail	N/A
718	…one saved e-mail	N/A
719	…one saved fax	N/A
720	…one saved message	N/A
721	… unheard voice mails	N/A
722	… heard voice mails	N/A
723 - 726	Not used	Not used

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
727	…one heard voice mail	N/A
728 - 730	Not used	Not used
731	I'm sorry, I didn't understand your response.	N/A
732	The system prompt you have selected has two formats.	N/A
733	To select the standard format, press one.	To select the standard format, say "Standard" or press one.
734	To select the speech-recognition format, press two.	To select the speech-recognition format, say "Speech Recognition" or press two.
735	To replace the standard format, press one.	To replace the standard format, say "Standard" or press one.
736	To replace the speech-recognition format, press one.	To replace the speech-recognition format, say "Speech Recognition" or press one.
737 - 738	Not used	Not used
739	Speech recognition has been disabled for this call.	N/A
740	Not used	Not used
741	... messages…	N/A
742	...seen...	N/A
743 - 744	Not used	Not used
745	You have no saved voice mails.	N/A
746	Not used	Not used
747	You have no saved e-mails.	N/A
748	Not used	Not used
749	You have no saved faxes.	N/A
750	Not used	Not used
751	voice mail received...	N/A
752	E-mail received...	N/A
753	Fax received...	N/A
754	Priority voice mail received...	N/A
755	Priority e-mail received...	N/A
756	Not used	Not used
757	and seen	N/A
758	and recent	N/A
759	and answered	N/A
760	and deleted	N/A
761	Not used	Not used
762*	...Million...	N/A
763*	...Billion...	N/A
764*	...Trillion...	N/A
765*	...Minus...	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
766*	*(For U.S. systems)* ...Dollar... *(For European systems)*…Pound…	N/A
767*	*(For U.S. systems)* ...Dollars... *(For European systems)*…Pounds…	N/A
768*	*(For U.S. systems)* ...Cent... *(For European systems)*…Pence…	N/A
769*	*(For U.S. systems)* ...Cents... *(For European systems)*…Pence…	N/A
770	<Pause>	N/A
771*	O'clock	N/A
772*	...Point...	N/A
773	I was unable to connect to your voice mail server.	N/A
774	I was unable to connect to your e-mail server.	N/A
775	I was unable to connect to your fax server.	N/A
776	If you have changed your password, press one.	If you have changed your password, say "Password" or press one.
777	To continue without these messages, press pound.	To continue without these messages, say "Continue" or press pound.
778*	dash	N/A
779*	ampersand	N/A
780*	at	N/A
781*	left parenthesis	N/A
782*	right parenthesis	N/A
783*	underscore	N/A
784*	space	N/A
785*	Tilde	N/A
786*	a	N/A
787*	b	N/A
788*	c	N/A
789*	apostrophe	N/A
790*	lower A	N/A
791*	lower B	N/A
792*	lower C	N/A
793*	d	N/A
794*	e	N/A
795*	F	N/A
796*	exclamation point	N/A
797*	lower D	N/A
798*	lower E	N/A
799*	lower F	N/A
800*	g	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
801*	h	N/A
802*	I	N/A
803*	star	N/A
804*	lower G	N/A
805*	lower H	N/A
806*	lower I	N/A
807*	j	N/A
808*	k	N/A
809*	l	N/A
810*	pound (or hash)	N/A
811*	lower J	N/A
812*	lower K	N/A
813*	lower L	N/A
814*	m	N/A
815*	n	N/A
816*	o	N/A
818*	lower M	N/A
819*	lower N	N/A
820*	lower O	N/A
821*	p	N/A
822*	q	N/A
823*	r	N/A
824*	s	N/A
825*	lower P	N/A
826*	lower Q	N/A
827*	lower R	N/A
828*	lower S	N/A
829*	T	N/A
830*	u	N/A
831*	v	N/A
832*	question mark	N/A
833*	lower T	N/A
834*	lower U	N/A
835*	lower V	N/A
836*	w	N/A
837*	x	N/A
838*	Y	N/A
839*	z	N/A
840*	lower W	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
841*	lower X	N/A
842*	lower Y	N/A
843*	lower Z	N/A
844*	colon	N/A
845*	period	N/A
846*	comma	N/A
847*	forward slash	N/A
848*	backward slash	N/A
849*	plus sign	N/A
850	Please hold while I check your e-mail.	N/A
851	...one unreadable message	N/A
852	...unreadable messages	N/A
853	You have no unreadable messages	N/A
854	To manage unreadable messages, press six	N/A
855	unreadable message...	N/A
856	If you are finish, please press pound (or hash). Otherwise, press the next key to continue spelling.	N/A
857	Your quota grace limit has been reached.	N/A
858	please hold while I check your messages.	N/A
859	The message could not be played.	N/A
860	Not used	Not used
861	I'm sorry there are no network mailboxes available for Node...	N/A
862	I'm sorry. This message cannot be forwarded.	N/A
863	I could not connect to any of your remote servers. You will be redirected to the local voice mail view.	N/A
864 - 875	Not used	Not used
876	To record a standard greeting press one.	To record a standard greeting, say "Standard" or press one.
877	To copy the standard greeting to the speech-recognition greeting, press two.	To copy the standard greeting to the speech-recognition greeting, say "Copy" or press two.
878	To record a speech-recognition greeting press three.	To record a speech-recognition greeting say "Speech Recognition" or press three.
879	If you would like to copy this recording to the speech-recognition greeting, press one.	If you would like to use this recording as the speech-recognition greeting, say "Copy" or press one.
880	To record a new speech-recognition greeting, press three.	To record a new speech-recognition greeting, say "New" or press three.
881	ñ (pronunciation: en-yay)	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
882	I'm sorry. This system prompt is not available for replacement.	N/A
883	The following message could not be delivered to...	N/A
884	To disable Speech Recognition mailbox commands for this call, press eight.	To disable Speech Recognition mailbox commands for this call, say 'Disable' or press eight.
885	Speech Recognition disabled.	N/A
886	I could not create a secure connection to your remote server. You will be redirected to the local voice-mail view.	N/A
887	Please contact your system administrator.	N/A
888	E-Mail Reader was unable to retrieve your faxes from your e-mail server due to a login failure.	N/A
889	E-Mail Reader was unable to retrieve your e-mails from your e-mail server due to a login failure.	N/A
890	Please enter your e-mail account password. This is case sensitive. Using the keypad, you can scroll through symbols, numbers, and letters, both lower and upper case.	Please spell or enter your e-mail account password. This is case sensitive. You can speak symbols, numbers, and letters. For upper case letters, say 'upper' followed by the letter.
891	Using the keys on your telephone, please spell the last name of the person. Then press pound (or hash). For the letter Q, press seven. For the letter Z, press nine.	Please say the last name of the person you are trying to reach. Otherwise, using the keys on your telephone, please spell the last name of the person, then press pound (or hash). For the letter Q, press seven. For the letter Z, press nine.
892	You have no voice-mails	N/A
893	You have no e-mails	N/A
894	You have no faxes	N/A
895	...Network extension...	N/A
896	To enable Speech Recognition mailbox commands for this call, press eight.	To enable Speech Recognition mailbox commands for this call, say “Enable” or press eight.
897	There was an error accessing your messages; you will be returned to the main-menu	N/A
898	...Network Mailbox...	N/A
899	N/A	I think you said “delete.” If this is correct, please press one; if this is not correct, please press two.
900	Speech Recognition enabled.	N/A
901	No days selected.	No days selected.
902	On	On
903	Off	Off
904 - 906	Not used	Not used
907	Message dated…	Message dated…
908	E-Mail Reader was unable to retrieve your faxes and e-mails from your e-mail server due to a login failure.	E-Mail Reader was unable to retrieve your faxes and e-mails from your e-mail server due to a login failure.

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
909	To copy this message to additional mailboxes, press four.	To copy this message to additional mailboxes, say “Copy” or press four.
910	To spell a new name, press four.	To spell a new name, say “new name,” or press four.
911	Using the keys on your telephone, please spell the first name of the person. Then press pound (or hash). For the letter Q, press seven. For the letter Z, press nine.	Please say the full name of the person you are trying to reach. Otherwise, using the keys on your telephone, please spell the first name of the person, then press pound (or hash). For the letter Q, press seven. For the letter Z, press nine.
912	Using the keys on your telephone, please spell the last name of the person. Then press pound (or hash). For the letter Q, press seven. For the letter Z, press nine.	Please say the full name of the person you are trying to reach? Otherwise, using the keys on your telephone, please spell the last name of the person, then press pound (or hash). For the letter Q, press seven. For the letter Z, press nine.
913	When you are finished entering mailbox numbers, press star to exit.	When you are finished entering mailbox numbers, say finished or press star to exit.
914	Enter the destination mailbox or Group list number. For the Directory press pound (or hash). When you are finished entering the destination, press star to exit	Say or enter the destination mailbox or group list number. For the directory, say “Directory” or press pound (or hash). When you are finished entering the destination press star to exit.
915	... or if you are finished listening to the message, Hangup.	N/A
916	Subscriber access is not programmed yet. please contact the system administrator.	N/A
917	I am sorry, I could not complete your call at this time. Please try again later.	N/A
918	Message access is not configured. Please contact your phone system administrator.	N/A
919	To change your main menu e-mail reader count, press four.	To change your main menu e-mail reader count, say e-mail reader count or press four.
920	Main Menu E-Mail Reader Count is currently enabled.	N/A
921	Main Menu E-Mail Reader Count is currently disabled.	N/A
922	To enable, press one.	To enable, say “enable” or press one.
923	To disable, press one.	To disable, say “disable” or press one.
924	Main Menu E-Mail Reader Count has been automatically disabled, please go to the personal options menu to re-enable this feature.	N/A
926	For help with password special keys, press two.	For help with password special keys, say “help” or press two.
927	To repeat the following instructions press zero at any time.	To repeat the following instructions say “repeat” or press zero at any time.
928	Enter the destination mailbox or Group list number. When you are finished entering the destination, press star to exit.	Say or enter the destination mailbox or group list number. When you are finished entering the destination, press star to exit.
929	“equal to sign”	N/A

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
930	“greater than sign”	N/A
931	“less than sign”	N/A
932	“semi colon sign”	N/A
933	“percent sign”	N/A
934	“dollar sign”	N/A
935	“pipe sign”	N/A
936	“pound (or hash) sign”	N/A
937	I am sorry, you must enter a password for security reasons. Please, enter your new password, then press pound (or hash).	N/A
938	To listen to all additional messages, please press eight.	Say “all messages” or press eight to listen to all additional messages.
939	The locations of special characters are...For space, at, hyphen, underscore, equals sign, comma, greater than, and right parenthesis, use key “zero.” For exclamation point, pipe, apostrophe, colon, semi-colon, forward slash and backslash, use key “one.” For period, tilde, dollar, asterisk, ampersand, percentage, plus, less than and left parenthesis, use key “star.” For pound (or hash), use key “pound (or hash).”	N/A
940	This call routing application does not have an associated digit translation node. Please assign a digit translation node using the System Manager interface before attempting to assign an Audiotex greeting using the System Administrator interface.	N/A
941	E-Mail Reader encountered an error while trying to connect to your e-mail server. Please contact your system administrator for further assistance.	N/A
942	E-mail Reader was unable to connect to your e-mail server. The server may be down or there may be a configuration issue with your E-Mail Reader Profile. Please contact your system administrator for further assistance.	N/A
943	E-Mail Reader cannot connect to your e-mail server due to an encryption mismatch. Please contact your system administrator for further assistance.	N/A
944	I am sorry, your password must be at least four digits long. Please, enter your new password, then press pound.	N/A
945 - 946	Not used	Not used
947	To save this message, press nine.	To save this message, say “save” or press nine.
948	To delete this message, press seven.	To delete this message, say “delete” or press seven.

Table 4-4: *Voice Prompts: Non-ASR and ASR (Continued)*

Prompt ID	Content (Non-ASR)	Content (ASR)
949	To swap your key settings for save and delete, press three.	To swap your key settings for save and delete, say “swap” or press three.
950	Your current key settings are seven for save and nine for delete.	N/A
951	Your current key settings are nine for save and seven for delete.	N/A
952	To swap the key settings, press one.	To swap the key settings, say “swap” or press one.
953	Your settings have been changed.	N/A

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1. INTRODUCTION

1.1 The Inter-Tel® phone system, in addition to its sleek, modular appearance and easy installation and programming, provides many user-friendly features. To describe the system features, this chapter has been divided into the following sections:

- **Access to the Features**: Contains a list of the feature codes and their definitions.
- **System Organization**: Describes possible administrator assignments, attendant arrangements, and hunt groups (including UCD and ACD hunt groups).
- **Trunk Features**: Explains the trunk functions and programmable features, including: dual-tone multi-frequency (DTMF) or dial-pulse signaling; types of trunks (loop start, ground start, T1, T1/PRI, T1/E&M, DID, E1/PRI, DDI, Caller ID [CLIP - Calling Line Identity Presentation], DNIS, and ANI; trunk groups; outgoing access, allowed-answer, and ring-in assignments; call routing tables; day and night modes of operation; automatic route selection (ARS); trunk and toll restriction; area/office code number plans; emergency calls; and direct inward system access (DISA).

  NOTE: Ground start, T1/PRI, DID, DNIS, ANI, and area/office code number plans are not supported in Europe.

- **Station Instruments**: Describes the station instruments of the phone system. Available station instruments include digital, analog, and Executone phones, single-line sets, digital and analog Direct Station Selection/Busy Lamp Field (DSS/BLF) Units, and Mini-DSS Units.

  NOTE: Analog phone, Executone phones, and analog DSS/BLF Units are not supported in Europe.

- **Intercom, Outside, and General Call Processing Features**: Explain the use of the station features that are available to all station users, including intercom call features, outside call features, and general call processing features.
- **Attendant Features**: Explains the special attendant-only features and the use of the DSS/BLF Unit.
- **Administrator Features**: Explains the special administrator-only features.
- **Record Keeping and Maintenance Features**: Describes station message detail recording (SMDR), which is a system error reporting feature providing self-diagnostic information to make the system easier to service.

2. SOFTWARE LICENSE

2.1 As of v8.0, you can control the device capacity and feature availability of the system through a software license. All CPCs and v8.0 installations require a software license, which differs based on the CPU type, as described below.

- A CPC-Single (64) or CPC-Dual (128 or 256) requires a single software license.
- A 512-CPU, 1024-CPU (CPU/PCM-F), or CPU-ATM (PCM-A) system requires a Call Processing (CP) Server that has a special security key (part no. 827.9775) that supports the software license functionality. The software license is stored on the CP Server PC and works only with the individual special security key.

NOTE: Version 8.0 does not require feature PALs. In addition, because version 8.0 no longer detects PALs, you do not need to upgrade the ATM PALs.

2.2 See page 6-97 for a complete list of software licenses currently uploaded to the system.

A. SYSTEMS AND SOFTWARE LICENSES

2.3 CPCs and CP Servers are distributed without a software license and are defaulted to a CPC64 for the CPC and to a 512 for the CP Server. Because the product supports AKSCs and EKSCs, these cards are not auto-equipped.

2.4 If the system does not have a software license, the Minor Alarm LED flashes. In addition, Alarms 125 and 225 are generated and sent to Message Print and the System Administrator's phone. These alarms are generated every five minutes, and the system resets every four hours until the license issue is resolved. See page 7-20 for more information about these alarms.

B. CPU TYPES

2.5 If Database (DB) Programming connects to a system with a valid software license, you cannot restore a database that would invalidate the license. For example, if the system currently has a software license that supports a 512 system, you cannot restore a 128 database. If you need to change the CPU type or product type, you must first upload a new software license (see page 7-105 for details).

C. PREMIUM FEATURES

2.6 Premium features are no longer controlled by the number of available feature units; they are controlled by the software license. Because of this, you cannot restore a database with premium features that are not licensed. If you attempt to restore a database that has more premium features than the software license reflects, a warning message appears before the restore occurs. When this message appears, you must cancel the restore.

NOTE: If upgrading an existing system to version 8.0 or later, you must make sure the software license you request has the correct premium features enabled/disabled. See page E-1 for more details about upgrading a current system.

2.7 Because the software license dictates the premium features that are enabled/disabled, you can no longer enable/disable premium features in DB Programming in remote mode. If you need to add premium features, you must acquire a new license.

D. NUMBER OF LICENSES

2.8 The system stores up to five different licenses, each of which is unique based on the system version. For example, only one v8.0 software license can be stored on the CPC (or CP Server). If a new license that matches the version number of an existing license is loaded, the new license automatically overwrites the existing license.

2.9 In addition, if five licenses are currently stored and a sixth license is saved, the oldest license, based on the date and time the license was created, is overwritten. If, however, the oldest license is currently in use, the next oldest license is overwritten. In addition, because each license is unique, only one license for each version number can be stored at any time.

3. SYSTEM MANAGER SUPPORT

3.1 Versions 8.2 and later support System Manager, which is a server-based application that centralizes the management functions of the Axxess system and its peripheral products. To interface with System Manager:

- Call Processing requires a System Manager agent account for each node that connects to the System Manager server.
- DB Programming version 8.2 or later must be installed on the PC running System Manager Server.
- You must upload the Certification Authority (CA) certificate for System Manager. If you do not upload the CA certificate, you cannot access DB Programming through the System Manager Web interface.
- You must configure DB Programming to connect to System Manager.

3.2 System Manager's performance can suffer when connected to a network that includes 40 or more nodes. The performance degradation occurs when several systems send large amounts of Message Print data to System Manager. In an effort to eliminate this, if the system administrator plans on connecting 63 nodes or more to System Manager, a newer version of System Manager is needed. This new version supports the ability to enable or disable incoming Message Print messages.

3.3 For details about agent accounts and DB Programming requirements, see the *System Manager Installation Manual* (document part no. 835.2743).

4. ACCESS TO THE FEATURES

A. FEATURE CODES

4.1 Each of the station features is assigned a feature code. Using the station's keypad, these codes are entered to select trunks, process calls, and use special features. Complete explanations of the features and instructions for using them begin on the next page.

4.2 The codes shown on the following pages are the values that are assigned when the system is in the default state. If desired, they can be changed in DB Programming to any one- to five-digit value.

CAUTION

Changing a feature code may affect the accessibility of other features.

B. FEATURE BUTTONS

4.3 Phones have feature buttons that allow one-button dialing of feature codes. Information is programmed in the database to determine the arrangement of the feature buttons and their default values. If desired, some of the phone feature buttons can be designated as user-programmable buttons. See page 5-132 for more information.

C. SPECIAL BUTTON AND HOOKFLASH [RECALL IN EUROPE]

4.4 Depending on the feature and system programming, users can either enter feature codes immediately after lifting the handset or while on hook, or they must signal the system before entering the feature code. Phone users signal the system by pressing the Special button (which is represented by ∞ or SPCL on the phone). Single-line DTMF set users perform a hookflash (press and release the hookswitch quickly, or press **RECALL**) or a recall (press RECALL). If the user does not enter a code or begin dialing before the Dial Initiation timer expires, the system sends reorder tones.

4.5 There is a system-wide option called "SPCL Key Required For Feature Code Entry." If enabled, it requires phone users to always press the Special button before entering a feature code.

D. TRUNK ACCESS CODES

4.6 These feature codes are used for selecting trunks when placing outgoing calls.

Table 5-1. *Trunk Access Codes*

Feature Name	U.S. [Europe] Code	Definition
Automatic Route Selection (ARS)	92000	Allows the system to select the desired route for placing a call, as programmed in the database. This feature requires the ARS software license.
Trunk Group Access 1-208	92001-92208	Selects an available trunk from a programmed group of trunks for placing an outside call.
Emergency Calls	911 [999]	Entering this feature code selects an outgoing trunk and automatically dials the programmed emergency number, defaults to Trunk Group 1 and 911 [or 999 in Europe].
Outgoing Call	8	Selects an outgoing trunk according to the programmed outgoing access mode for that station. See PROGRAMMING, page 6-232. (Inter-Tel phones use the OUTGOING button or a CALL button.)

E. STATION FEATURE CODES

4.7 The following pages list the general feature codes. When a default feature button can be used in place of the feature code, it is indicated at the end of the explanation. See page 5-138 for a listing of the default phone feature buttons.

Table 5-2. *Station Feature Codes*

Feature Name	U.S. [Europe] Code	Definition
ACD Agent Login ACD Agent Logout ACD Agent Login/Logout Toggle	326 327 328	These feature codes allow an ACD hunt group member (agent) to log into and out of the ACD hunt group(s). The agent only receives calls through the ACD hunt group(s) while logged in.
ACD Agent Wrap-Up Terminate	329	When an ACD agent completes a call, no other ACD hunt group call rings at the station until the ACD Wrap-Up Timer expires or the agent enters this feature code to terminate the wrap-up session. As of version 8.21x, you can program a DSS/BLF button to indicate when an ACD hunt group member is in wrap-up mode. When programmed, the DSS/BLF button is lit solidly while the member is in wrap-up. When the wrap-up timer expires, the DSS/BLF button is unlit, and the agent is made available to take calls (see page 5-47).
Account Code – All Calls Following	391	Allows the station user to enter a forced or optional account code that applies to all calls following the entry of this feature code and appears in the SMDR. To disable the All Calls Following feature, the feature code is entered again without an account code.
Account Code – Optional	390	Allows the station user to enter an optional account code for SMDR reports during an outside call.

Table 5-2. *Station Feature Codes (Continued)*

Feature Name	U.S. [Europe] Code	Definition
Agent Help	375	The Agent Help feature allows a station user to request help from a designated "Agent Help Extension" during a two- or three-party call.
Agent Help Reject	376	When a request-for-help call rings, the Agent Help Extension can choose to join the call or enter this feature code to reject the request.
Answer (Ringing Call)	351	Answers the call that has been ringing or holding the longest at that station. See page 5-88 for the priority list. (Inter-Tel phones use the ANSWER button.)
Audio Diagnostics	320	Allows a user to generate diagnostics information for audio problems. After the Audio Diagnostics feature is initiated, the user is prompted to answer questions about the audio problems by pressing the associated buttons on his or her endpoint. Based on the user's selections, the system generates alarm 128 which appears on the administrator's endpoint. The collected data is then sent to Inter-Tel Technical Support via Automatic Diagnostics Delivery (ADD) if enabled.
Automatic CO Access On/Off	360	*(Inter-Tel Phones Only)* Allows the phone user to determine how ringing outside calls are answered: simply by lifting the handset or pressing the Speaker button (automatic answer); or by lifting the handset or pressing the Speaker button and pressing a CALL button, individual trunk button, or the ANSWER button. **NOTE:** This enables/disables the Automatic Answer CO flag (under System\Devices and Feature Codes\Stations\Local\<*station*>\Flags).
Automatic IC Access On/Off	361	*(Inter-Tel Phones Only)* Allows the phone user to determine how ringing intercom calls are answered: simply by lifting the handset (automatic answer); or by lifting the handset and pressing the IC button (or a CALL button, if there is no IC button). **NOTE:** This enables/disables the Automatic Answer IC flag (under System\Devices and Feature Codes\Stations\Local\<*station*>\Flags).
Automatic Trunk Answer	350	Using this feature code, station users with allowed answer can pick up trunks that are ringing into the system but that are not actually ringing at their stations. This feature does not pick up transferred calls or recalls that are ringing at the station.
Background Music On/Off	313	*(Inter-Tel Phones Only)* Turns ON and OFF background music heard through the phone speaker. (Analog phones have a default MUSIC button.)

Table 5-2. *Station Feature Codes (Continued)*

Feature Name	U.S. [Europe] Code	Definition
Barge-In	386	Accessed as an option from the Station Monitor feature (321) the Barge-In feature allows the supervisor to barge-in on a call to help the hunt group member/agent.
CO Hookflash (Recall)	330	Sends a timed hookflash (recall) over the trunk while on an outside call (includes conference calls).
Call Forward All Calls	355	Immediately forwards all calls to another station or to an outside telephone number. (Inter-Tel phones use the FWD button.)
Call Forward If Busy	357	Immediately forwards all calls to another station or to an outside telephone number when the station is in use.
Call Forward If No Answer	356	Forwards all calls to another station or to an outside telephone number if not answered within a predetermined time.
Call Forward If No Answer/Busy	358	Forwards all calls to another station or to an outside telephone number if not answered within a predetermined amount of time or immediately if the station is in use.
Call Logging	333	Allows digital/IP endpoints and IP softphone users to view the missed, received, and dialed call logs on their endpoint. If the endpoint does not support Call Logging or if the Enable Call Logging flag is not enabled, when the user enters this feature code the endpoint displays an error message and receives a corresponding error tone. Non-display endpoints only receive the error tone. See page 5-180 for details.
Change Language	301	A station user can change the assigned language for the station by entering the Change Language feature code while the station is idle.
Conference	5	Connects from three to four parties in a conference. A conference consists of any combination of inside and outside parties. (Inter-Tel phones use the CNF button.)
Data	340	*(Inter-Tel Phones Only)* Allows operation of a data device attached to a phone that has a PC Data Port Module and a Modem Data Port Module (also requires a modem-equipped data device). (Analog phones have a default DATA button.)
Default Station	394	This single feature code cancels account codes for all calls following, do-not-disturb, manual call forwarding, background music, ring intercom always, and queue requests; restores handsfree mode, pages, hunt group calls, and system forwarding; and returns phone volumes to default values.
Directory	307	*(Display Phones Only)* Allows display phone users to search for extension numbers or system speed-dial numbers. The number can then be dialed, if desired. *This feature requires the Directories software license.*

Table 5-2. *Station Feature Codes (Continued)*

Feature Name	U.S. [Europe] Code	Definition
Display Outside Party Name On/Off	379	*(Display Phone Only)* Toggles the display between the caller's name and number while connected to a CO call that has outside party name information. If there is no outside party name or the Expanded CO Call Information On Displays flag is disabled, the user hears a burst of reorder tone and sees the CANNOT ACCESS FEATURE display. If the Display Outside Party Name On/Off feature code is programmed in a user-programmable button with a lamp, the lamp is lit when the outside party name is enabled and off when the outside party number is enabled.
Display Time/Date	300	*(Display Phones Only)* Temporarily displays the system date and time, username, and extension number during a call or when other displays are shown. **NOTE:** For SIP (Session Initiation Protocol) endpoints, this feature code displays the IP address of the endpoint, not the time/date.
Do-Not-Disturb Do-Not-Disturb Cancel Do-Not-Disturb On/Off	370 371 372	Halts all intercom calls, transferred calls, and pages to the station. The cancel code returns the station to normal operation. The ON/OFF code can be used to turn do-not-disturb on or off. (Inter-Tel phones use the DND button.)
Do-Not-Disturb Override	373	*(Inter-Tel Phones Only)* If enabled in the database, allows the phone user to break through another station's do-not-disturb mode when placing an intercom call.
Enhanced Speakerphone Enable	310	*(Digital Phones Only)* When entered at a digital phone, this feature code enables the enhanced speakerphone as described on page 5-124. (Digital phones can also use the ∞ + Speaker buttons.)
Feature Key Default	395	*(Inter-Tel Phones Only)* Phones have user-programmable feature buttons that can be set to enter feature codes. This code returns the user-programmable buttons to the database default values.
Group Listen	312	*(Inter-Tel Phones Only)* Allows a user to transmit a conversation over the phone speaker while in handset or headset mode.
Handsfree On/Off	319	*(Inter-Tel Phones Only)* Disables/enables the phone's handsfree intercom answering. Incoming intercom calls ring as private calls if handsfree answering is disabled.
Headset Enable Headset Disable Headset On/Off	315 316 317	*(Inter-Tel Phones Only)* Signals the system that a headset has been connected to the phone when entering the enable code. The disable code returns the phone to normal operation. The on/off feature code can be used to toggle the feature ON or OFF.

Table 5-2. *Station Feature Codes (Continued)*

Feature Name	U.S. [Europe] Code	Definition
Hold – Individual	336	Places a call on hold so that it can be picked up directly at that station or through a reverse transfer from any other station. (Inter-Tel phones use the HOLD button.)
Hold – System	335	Places an outside call on system hold. It can be picked up directly at any phone that has an individual trunk button and has allowed-answer and/or outgoing access for that trunk or by the station that placed it on hold. (If used on conference or intercom calls, the system places the call on individual hold.)
Hunt Group Remove Hunt Group Replace Hunt Group Remove/Replace	322 323 324	Removes the station from its assigned hunt group(s) or places it in again. Does not affect non-hunt group calls. The remove/replace feature code can be used to toggle the feature.
LCD Contrast Adjustment	303	*(Models 8662, 8622, 8620, 8660, and 8560 Phones Only)* Adjusts the LCD contrast on the display. The phone must be idle to use this feature.
Message	365	This feature code is used for leaving and retrieving a message waiting indication at a called station or the called station's message center. Depending on how the message was left, the called station user either retrieves the message from his/her message center or from the station that left the message. (Inter-Tel phones use the MSG button.)
Message – Cancel	366	Allows the station user to cancel a message waiting indication that the user left at another station.
Message – Cancel Current	368	Cancels a message waiting indication that is waiting at the station without requiring the user to respond to it. (Phones can use the asterisk [*] button while viewing the message.)
Message – Silent	367	Leaves a message waiting indication at a station without first placing an intercom call. (This is the code that should be used by analog voice mail units connected to the phone system.)
Mute On/Off	314	*(Inter-Tel Phones Only)* Turns the microphone on or off during a call. If muted, the phone user can hear the other party, but the party cannot hear the phone user. (Phones can also use the MUTE button.)
Page	7	When followed by a paging zone code (0-9 or 0-49), it allows announcements to be made through phone speakers and any external paging speakers in the page zone. (Analog phones have a default PAGE button.)
Page On/Off	325	*(Inter-Tel Phones Only)* Halts pages through the phone speaker or allows them to be received again.

Table 5-2. *Station Feature Codes (Continued)*

Feature Name	U.S. [Europe] Code	Definition
Program Bit Rate	393	*(Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones Only)* Allows an Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone user to change the bit per second (bps) rate of an attached PCDPM's serial port.
Program Keys	397	*(Inter-Tel Phones Only)* User-programmable feature buttons and station speed-dial buttons can be programmed using this feature code.
Program Station Password	392	Programs the Station password for the Remote Programming feature. The password can be changed by entering the Program Station Password feature code at the station or when using the Remote Programming feature.
Queue Request	6	Requests (or cancels) an automatic callback when a busy trunk or station becomes available. (Analog phones have a default QUEUE button.)
Record-A-Call	385	If the system is programmed with a Record-A-Call application, the stations can be programmed to use the Record-A-Call feature. It allows users to enter a feature code whenever they want to record an ongoing call in their designated Record-A-Call mailbox. Users can retrieve the recorded messages later, just as they would any other mailbox messages.
Redial	380	Redials the last outside telephone number dialed or saved at the station (up to 48 digits). Also used to save numbers at phones programmed for last number saved. (Inter-Tel phones use the REDIAL button).
Redirect Call	331	Allows the phone user to route ringing outside, intercom, and camped on calls to another station, hunt group, outside number, voice mail, or Do-Not-Disturb.
Reminder Message Reminder Message Cancel	305 306	*(Inter-Tel Phones Only)* Sets reminder messages that signal the station at specific times. Or, cancels all reminder messages for the station.
Remote Programming	359	Allows a user to place a phone in do-not-disturb mode or forward the station's calls, either from another station or through DISA.
Reverse Transfer (Call Pick-Up)	4	Picks up a call ringing or holding at a station or hunt group
Review Keys	396	*(Inter-Tel Phones Only)* User-programmable feature buttons and station speed-dial buttons can be viewed using this feature code.
Ring Intercom Always On/Off	377	Enables/disables the feature that allows the station to always place private (non-handsfree) intercom calls.
Ring Tone Selection	398	*(Inter-Tel Phones Only)* Selects the type of ringing alert tone that is heard from the phone.

Table 5-2. *Station Feature Codes (Continued)*

Feature Name	U.S. [Europe] Code	Definition
Routing Off	304	Disables System OAI Offering Control for third-party applications. (You are prompted for a password.) After you disable routing, you cannot enable it again (i.e., only the third-party application can enable routing).
Steal	387	Accessed as an option from the Station Monitor feature (321) the Steal feature allows the supervisor to steal (take away) the call from the hunt group member/agent.
Station Monitor	321	*(Hunt Group Supervisors Only)* Allows a designated hunt group supervisor to monitor a call of anyone in the associated hunt group.
Station Speed Dial Station Speed Dial Programming	382 383	Dials/programs one of the 10 station speed-dial numbers when followed by a location code (0-9). (Inter-Tel phones use the Station Speed Dial button plus a location code for programming and dialing, or they can program speed-dial buttons for one-button dialing.)
Switch Keymap	399	*(Inter-Tel Phones Only)* Allows a phone user to switch between standard and alternate keymaps.
System Forward Enable System Forward Disable System Forward On/Off	352 353 354	Enables or disables the database-programmed System Forwarding feature for this station. The on/off feature code can be used to toggle the feature ON or OFF. This feature requires the System Forwarding software license. See page 6-97 in PROGRAMMING.
System Speed Dial	381	Dials one of the 1000 (5000 if using a CP Server) system speed-dial telephone numbers when followed by a location code (000-999 or 0000-4999). Also used for reviewing system speed-dial numbers. (Inter-Tel phones use the SYS SPD button.)
Transfer to Hold	346	Transfers a call to another station and places it on individual hold so that it does not ring or send call waiting signals until it recalls.
Transfer to Ring	345	Transfers a call to another station or to an outside telephone number. (Inter-Tel phones use the TRANSFER button.)

F. ADMINISTRATOR FEATURE CODES

4.8 The following feature codes may be used only at designated administrator stations. Non-administrator station users hear reorder tones if they attempt to use these feature codes.

Table 5-3. *Administrator Feature Codes*

Feature Name	U.S. [Europe] Code	Definition
Automatic Diagnostics Enable/Disable	9823	Enables/disables the Automatic Diagnostics Delivery (ADD) feature. Use this code only when directed to do so by Technical Support.
Clear Network Alarm	9851	Clears network-wide alarms on every node in the network, but does not affect system alarms. The Clear Network Alarm feature code may be entered on any node in the network, but the Send Network Alarms flag must be set for the administrator to clear alarms on other nodes in the network.
Clear System Alarm	9850	Clears all local and network-wide system alarm displays on your node.
Compression On/Off	9982 [9182]	Compresses call processing messages sent to DB Programming, speeding up transfers. You should not disable this feature unless instructed to do so by Inter-Tel personnel.
Compression Statistics	9981 [9181]	Dumps various statistics related to the DB Programming compression algorithm. This feature should be used only when directed to do so by Inter-Tel personnel.
Diagnostics On/Off	9900 [9100]	Enables or disables system diagnostics mode so that the database can be frozen and/or unfrozen, message/reset logs can be printed, and devices seized using the diagnostic feature codes.
Enable Network Day	9862	Places any or all remote nodes into day mode (you are prompted for the node number).
Enable Network Night	9861	Places any or all remote nodes into night mode (you are prompted for the node number).
Modem Disable	9867	Disables the modem on the Call Processing Card (CPC). If disabled, the modem is placed into DND. In this state, the modem rejects all calls and does not allow camp ons.
Modem Enable	9866	Enables the CPC modem. When enabled, the modem can be used for DB Programming or as a serial port.
Modem Reset	9869	Resets the CPC modem, dropping any call that is currently connected to the modem. **NOTE:** Do not reset the modem if it is connected to a DB Programming session. If you do, the programming session is dropped, potentially corrupting the database. For this reason, Inter-Tel strongly recommends that the System Administrator assign the modem extension to a DSS/BLF button.

Table 5-3. *Administrator Feature Codes (Continued)*

Feature Name	U.S. [Europe] Code	Definition
Night Ring On/Off	9860	Places system in night mode or cancels night mode. While in night mode, the night toll and trunk restriction lists are used and the night transfer relays on the OPC are activated.
Periodic Diagnostics On/Off	9825	Periodic diagnostics extends and improves the functionality and checks and reconciliations performed on various resources in the system.
Program Database	9932 [9132]	Can be used for programming station, system, and trunk parameters.
Program System Speed Dial	9801	Programs system-wide speed-dial numbers when followed by location codes (000-999 or 0000-4999).
Set Network Date/Time	9810	Programs the time and date for all nodes in the system. (Local time and date can also be reset by individual nodes.)
Set Time/Date	9800	Programs system time and date.
Synchronize Network Time	9811	Synchronizes the minutes past the hour across the network. **NOTE:** If a node's time is off by more than 30 minutes, synchronizing the minutes may cause the hour to change. Also, network time is automatically synchronized every day at 12:30 AM (00:30), using the time setting on the node with the lowest number.

G. DIAGNOSTICS MODE FEATURE CODES

4.9 The Diagnostics Mode feature code (9900 or 9100) must be entered at the administrator's station to enable system diagnostic mode before these feature codes can be used:

Table 5-4. *Diagnostics Mode Feature Codes*

Feature Name	U.S. [Europe] Code	Definition
Diagnostic - ASAI Snoop Off	9926 [9126]	This feature code turns off the ASAI output to Message Print. ASAI is the protocol the system uses to talk to DB Programming and the AVDAP. Turning this feature on helps the Inter-Tel Engineers debug the messaging between Call Processing and DB Programming or Call Processing and the AVDAP.
Diagnostic - ASAI Snoop On	9927 [9127]	This feature code turns on the ASAI output to Message Print. ASAI is the protocol the system uses to talk to DB Programming and the Voice Processor. Turning this feature on helps the Inter-Tel Engineers debug the messaging between Call Processing and DB Programming or Call Processing and the AVDAP.
Diagnostic - Dump Extension	9922 [9122]	The system dumps the internal data structures for that extension to message print. If you press the pound [hash in Europe] button (#) you are prompted for a command. The command allows you to enter an alphanumeric string of the structure to dump. There are many times when an Inter-Tel Engineer may have the field technician dump out a device or structure when debugging a problem.
Diagnostic - Dump Node Information	9936 [9136]	The user can now easily dump node information to Message Print using a single feature code. Pressing the Dump Node Information feature code dumps specified node information to Message Print for diagnostic purposes. When the system prompts for a node number, enter the desired node number (1 to 99) or enter zero (0) for all nodes within a network. If the node does not exist, an error message saying INVALID NODE NUMBER appears and prompts the user for a node number again. After entering a node number, the system displays a confirmation message on the phone display. This helps the user to analyze which nodes are up and which nodes are down (the word 'down' does not mean that the node is completely down, it simply means the node is unreachable). For Example: • If all nodes are up, the display shows ALL NODES ARE UP X. The 'X' represents the total number of the nodes. • If node 3 and 5 of a 5-node network are down, the display shows # NODES DOWN 2 3 5.
Diagnostic - Dump System Diagnostics	9933 [9133]	Allow a system administrator to dump system diagnostic information to Message Print.

Table 5-4. *Diagnostics Mode Feature Codes (Continued)*

Feature Name	U.S. [Europe] Code	Definition
Diagnostic - Heap Statistics	9947 [9147]	This feature code outputs miscellaneous heap information to Message Print as well as putting up a message on the phone that indicates the percentage of available dynamic heap memory. This feature code is useful in determining if the system is losing heap memory and how quickly the system may be losing it.
Diagnostic - ISDN View	9948 [9148]	This feature code toggles through the three different ISDN view output modes. • Entering the feature code the first time puts the ISDN view feature into headers only mode. In this mode the system outputs all ISDN messages to message print in header format (e.g., it does not contain any ISDN information elements). • Entering the feature code the second time puts the ISDN view feature into full mode. In this mode the system outputs all ISDN message to message print in full format (e.g., each ISDN information element). • Entering the feature code one more time turns this feature off.
Diagnostic - Major Reset	9962 [9162]	This is used by Inter-Tel software developers and cannot be used in Beta or Production software.
Diagnostic - Mark As Leaks	9945 [9145]	This is used by Inter-Tel software developers and cannot be used in Beta or Production software.
Diagnostic - Mark As Quiescent	9946 [9146]	This is used by Inter-Tel software developers and cannot be used in Beta or Production software.
Diagnostic - Minor Reset	9964 [9164]	This is used by Inter-Tel software developers and cannot be used in Beta or Production software.
Diagnostic - Network Freeze Zone System Histories Diagnostic - Network Unfreeze Zone System Histories	9939 [9139] 9989 [9189]	The system fault history for any freeze zone in the network can be halted (frozen) or re-enabled using these feature codes when diagnostics mode is enabled. The fault history can then be extracted from each zone and used by service personnel when troubleshooting the system.
Diagnostic - Network Groups	9963 [9163]	Allows an administrator to verify that the Network Groups on the local node are programmed properly. When 9963 is entered at an administrator's endpoint, the phone system initiates pings from each Inter-Tel IP device on the local node and determines if the other devices respond to the ping. If a device does not respond to the ping or if a firewall is detected, the system issues a Message Print message.
Diagnostic - Print Auxdata	9972 [9172]	(Not Programmable) Sends a report to a designated printer or file that shows system reset history information to be used for troubleshooting purposes.
Diagnostic - Print Message Log	9975 [9175]	(Not Programmable) Sends a report to a designated printer or file that lists system error messages to be used for troubleshooting purposes.
Diagnostic - Print Network Log	9976 [9176]	This feature code prints the network log to message print. This feature is useful in determining the system of a networked system.

Table 5-4. *Diagnostics Mode Feature Codes (Continued)*

Feature Name	U.S. [Europe] Code	Definition
Diagnostic - Query Node Traffic	9978 [9178]	Using this feature code you can query the status of various devices on the system based on the status of the traffic flags in DB Programming. The output is sent to Message Print.
Diagnostic - SIP View	9987 [9187]	Allows the user to change the system wide SIP output value. Options include No Output, Headers and Full Output.
Diagnostic - Show Version	9928 [9128]	To check the call processing software version at an Administrator's phone, you can enable diagnostics mode, then enter this feature code to view the version and date of the call processing software. **NOTE:** For SIP endpoints, this feature code displays the firmware version of the endpoint, not the Call Processing version.
Diagnostic - Spare 1-3	9910-9912 [9110-9112]	This is used by Inter-Tel software developers and cannot be used in Beta or Production software.
Diagnostic - System History	9974 [9174]	This is used by Inter-Tel software developers and cannot be used in Beta or Production software.
Diagnostic - View Displays	9983 [9183]	This is used by Inter-Tel software developers and cannot be used in Beta or Production software.
Seize Device	9973 [9173]	Used during troubleshooting to seize a specific trunk or station by indicating the board number, port number, and device number.
System History—Freeze System History—Unfreeze	9993 [9193] 9998 [9198]	The system fault history can be halted (frozen) or re-enabled using these feature codes when diagnostics mode is enabled. The fault history is used by service personnel when troubleshooting the system

H. EXTENSION NUMBERS

4.10 Extension numbers are recognized as feature codes by the system. When the system is in the default state, the extension numbers are assigned as follows:

Table 5-5. *Extension Numbers*

Attendant	0
Stations	1000-1999
Modem (CPC Only)	1500
Hunt Groups	2000-2074 or 2000-2299
Voice Processing Applications	2500-2999
Page Zones	9600-9609 or 9600-9649 (Used only for programming a feature button to select a page zone.)
External Page Ports	91000-91003 [99000-99003 in Europe]
Individual Trunks	94000-95999 (Determined by card slot and device number. Used for programming and troubleshooting.)

5. ADMINISTRATOR STATIONS

5.1 Any phone (attendant or non-attendant) can be assigned as an administrator. Administrator stations provide the following services by using the feature codes shown on page 5-21:

- Place the local node or other nodes in the network in night mode or day mode
- Set the date and time of the local node
- Set the network date and time and re-synchronize clocks in the network.
- Make database changes (see page 5-276 for a list of programming areas)
- Program system speed-dial numbers on the local node
- Receive and clear displayed system and network alarms
- Use diagnostic mode features to:
  - — Freeze and unfreeze database history on local nodes or other nodes in the system
  - — Print error logs
  - — Seize specific devices for troubleshooting purposes

5.2 Administrator features are described in detail beginning on page 5-273.

5.3 Any phone can be designated as an administrator station through DB Programming (see page 6-209 in PROGRAMMING). All designated administrator stations should be equipped with display phones to show system alarms and make programming easier.

6. ATTENDANT STATIONS

NOTE: For information on ***automated*** attendants, see page 4-16 in VOICE PROCESSING FEATURES.

6.1 Attendant stations can be called by dialing 0 at the stations they serve. Usually, they are also programmed to provide these services:

- Central operators for incoming calls
- Message centers
- Recall stations for unanswered calls

6.2 Any station can be designated as an attendant station during DB Programming (see page 6-209 in PROGRAMMING). Attendant stations should be equipped with display phones to show recall sources.

6.3 If using an Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone, an attendant's station can also be used with a PC Data Port Module (PCDPM) and single or tandem DSS/BLF Units for one-button intercom dialing of extension numbers and for constant station/hunt group status indications. (Non-attendant Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phones can also be equipped with DSS/BLF Units.) Up to 20 DSS/BLF units can be equipped per system, up to four per Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone (see page 5-152).

NOTE: Model 8520 and 8560 Phones also support the Mini-DSS, which provides 16 user-programmable buttons. The Mini-DSS does not affect the number of DSS/BLF Units that can attach to the phone.

6.4 A hunt group can be assigned to serve as an attendant. However, the individual stations in the hunt group are not required to be programmed as attendant stations, and the database does not reflect that the individual stations serve as an attendant for any other stations. (This feature is especially useful when a voice mail hunt group is assigned as an attendant.)

A. NETWORK AND LOCAL PRIMARY ATTENDANTS

6.5 One attendant can be designated as the primary attendant who can receive unsupervised outside call recalls, hunt group recalls, and calls that cannot be matched to patterns in call routing tables. (See page 6-319 in PROGRAMMING for more information.)

6.6 When phone systems are installed in a network, there are two types of primary attendants:

- **Node Attendants**: There can be a primary attendant for each node.
- **Network Primary Attendant**: There can be a network primary attendant programmed on each node. (There is usually one Network Primary Attendant shared by all of the nodes.) When the network needs to direct a call to an attendant, it attempts to direct the call to the network primary attendant first. If the network primary attendant is unavailable, it directs the call to the attendant on the node where the call originated.

6.7 If there is no network or local primary attendant, calls that would normally go to the primary attendant (as described in paragraph 6.6 above) are handled as follows:

- If the system has seized the call, but it has not been sent to a station, the call is disconnected.
- If the call has been sent to a station, it remains at the station and rings until answered.
- If the call is not seized and not sent to a station, the caller hears ringing until the caller hangs up. The call does not ring at any station.

B. ONE-ATTENDANT OR MULTIPLE-ATTENDANT OPERATION

6.8 Each node can be set up with one attendant or several attendants, as follows:

- **One Attendant**: One attendant provides all of the attendant services listed in paragraphs 6.1 and 6.5. All trunks (except private trunks) are usually programmed to ring in at this attendant's station.
- **Multiple Attendants**: Any or all stations can be programmed as attendants. (For example, there may be one or more attendants to serve each department in a business.) Trunks are programmed to ring at any or all attendant stations. Multiple attendants can be arranged in a hierarchy. That is, one attendant may be the attendant for another. In this case, the "serving" attendant is reached by dialing 0 at the "served" attendant.

C. ATTENDANT RECALL

6.9 When a call is placed on hold or is transferred from one station to another, the Hold and Transfer timers limit the amount of time the call may remain unattended. After that time, the call recalls the station that transferred it or placed it on hold, and the Recall timer is started. If the call remains unanswered at the station until the Recall timer expires, it recalls the station's attendant and the Abandoned Call timer starts. (If there is no attendant, the call continues to recall at the station that transferred it or placed it on hold.) If the attendant station is busy, the call camps on, and the display shows the source of the recall. If the call is not answered before the Abandoned Call timer expires, the system disconnects the call.

6.10 If an attendant station transfers a call and it is not answered before the Attendant Transfer timer expires, it recalls the attendant station.

6.11 If a station user transfers or forwards an outside call to an outside telephone number, it becomes an "unsupervised" CO-to-CO call because no inside parties are involved. The CO-to-CO call is limited by the Unsupervised CO timer. When the timer expires, the call recalls the primary attendant station and causes the CNF button to flash. (Display phones show UNSUPERVISED CNF RECALL.) This serves two purposes:

- It allows the attendant to monitor the length of CO-to-CO calls. When a CO-to-CO call recalls, the attendant can disconnect the call or allow it to continue.
- If the callers hang up before the attendant receives the recall, the system may not have disconnected the trunks because a disconnect was not received from the central office [local exchange in Europe]. The attendant must disconnect the call.

6.12 When a recall rings at the attendant's station, a CALL or individual trunk button flashes at a medium rate.

6.13 If the attendant has calls forwarded, recalls from stations follow internal call forward requests. Recalls do not forward to outside telephone numbers but recall the attendant's station until they are answered or the Abandoned Call timer expires. Placing the attendant's station in do-not-disturb mode does not block recalls or direct ring-in calls.

6.14 If there is not a primary attendant, recalls remain at the stations and ring until answered or disconnected by the system.

6.15 **To answer a hold or transfer recall at an attendant station**: When you see a hold or transfer recall display (XXXX RCL FROM YYYY) and hear a recall ringing, lift the handset. Or, do one of the following:

- **Outside call recall**: Press the medium-flashing CALL button or individual trunk button. Or, press the ANSWER button. (If more than one trunk is recalling, pressing the ANSWER button selects the outside call indicated on the display.)
- **Intercom call recall**: Press the IC button or the **ANSWER** menu button. (If you are busy when the intercom call recalls, it camps on. The IC button flashes at the medium rate, but you do not hear recall ring signals.)
- **Conference call recall**: Press the flashing CNF button to connect with the conference call. The CNF button flashes slowly, and the display shows CONFERENCE IN PROGRESS. *If the parties are still talking,* press the CNF button again and hang up to return the parties to their conversation. The CNF button flutters. You can enter the conference at any time by pressing the fluttering CNF button. If the Hold timer expires, the conference recalls your station again. *If the parties have hung up,* hang up to disconnect the call.

7. EXTENSION LISTS

7.1 An extension list is a group of intercom numbers or trunk numbers located on a single node (off-node devices cannot be included in extension lists). These lists are used when programming features that use common devices. For example, a group of phones could be assigned to the same paging zone, have ring-in for the same trunk groups, and share common toll restrictions. With an extension list, the programmer would have to enter only one extension list number instead of repeatedly entering all of the station numbers for each list.

7.2 There are several types of extension lists:

- **Agent ID**: Includes only ACD hunt group Agent IDs. If no Agent IDs have been created, this option cannot be used.
- **Keyset**: Includes only phone intercom numbers.
- **Single-Line**: Includes only single-line set intercom numbers.
- **Stations**: Include phone and/or single-line intercom numbers.
- **Trunk Group**: Includes only trunk group access numbers.

7.3 Station and Agent ID extension lists can be included in hunt groups. For example, a hunt group can be programmed to send calls first to an individual station, then to an extension list (where it rings all stations on the list), and then to another individual station or an extension list.

7.4 The number of extension lists allowed on the system is determined by the number of entries in the lists. There can be a combined total of up to 2500 entries (station or trunk extensions) in all of the extension lists. (There can be up to 5000 entries in a triple- or quad-chassis system.)

NOTE: When using an extension list for ring-in or hunt groups, do not exceed 30 stations per list. The system can send ring signal to up to 30 stations.

7.5 In addition to the programmable extension lists, versions 7.0 and later system software support two auto extension lists:

- PP051: Auto: All Stations
- PP052: Auto: All Keysets

7.6 When a station or voice mail application is equipped or unequipped, the system automatically adds or removes that extension in the appropriate list(s). You cannot delete or renumber the auto extension lists. You can only add or remove extensions from the lists in on-line monitor mode (not recommended).

7.7 The two auto extension lists default with all existing stations and voice mail applications. In a default database, *PP051: Auto: All Stations* is assigned to Day/Night Outgoing Access and Emergency Outgoing Access for the following:

- CO Trunk Groups
- Node Trunk Groups
- Node IP Connection Groups

NOTE: Extension list *PP051: Auto: All Stations* is assigned to Day/Night Outgoing Access for version 7.0 and later default databases and newly created CO/Node Trunk Groups only. If you convert an existing database to 7.0 or later, the extension list currently assigned to Day/Night Outgoing Access (e.g., P0001) is used, not PP051.

8. ASSOCIATED DEVICES AND REFERENCES

8.1 The Associated Devices and References feature allows you to see the associated devices for a station, mailbox, and hunt group. This feature also allows you to query various groups in the database to locate the associated references to the extension. To search for associated devices or references, right-click the station, mailbox, or hunt group and select **Associated Devices and References**.

8.2 The following table shows the groups that you can query in the database for references, the type of information included in the groups, and the associated fields.

Table 5-6. *Associated Devices and References*

Group	Type of Information	Fields
Voice Processor Info	Applications AVDAP Information Call Routing Announcements Local Mailbox (Primary/Alternate Cascade Levels) Group List Members	Alternate Message Source Attendant Transfer Recall Destination Dial-0 Day Destination Type (Mailbox/Extension) Dial-0 Night Destination Type (Mailbox/Extension) System Administrator Mailbox Fax Delivery Destination Device (Extension)
Station	Call Routing Entries Stations Emergency Day Outgoing Access List Emergency Night Outgoing Access List Forwarding Points	Ring-In (Extension) Attendant Message Center Transfer Recall Destination Device (Extension)
Hunt Group	Hunt Group Supervisors Hunt Group ACD Agent IDs Hunt Group Members Hunt Groups	Device (Extension) Announcement (Extension) Overflow (Extension) Recall (Extension)
Trunk Group	CO Trunk Group Day Answer Access List CO Trunk Group Day Multiple Ring In List CO Trunk Group Night Answer Access List CO Trunk Group Night Multiple Ring In List Trunk Group Day Outgoing Access List Trunk Group Night Outgoing Access List CO Trunk Groups	Device (Extension) Day Single Ring In (Extension) Night Single Ring In (Extension)
Programmable Keys	Station Programmable Keys Single Line Default Programmable Keys Keyset Default Programmable Keys Keyset KeyMap Keys	Device (Extension)

9. HUNT GROUPS

9.1 The Hunt Group feature permits calls to be placed to a group of stations and to be automatically transferred to an available station within the group. Up to 75 hunt groups (300 with a CP Server) can be programmed in the database. Hunt group lists can contain individual stations and/or extension lists. Non-ACD hunt group stations must reside on the same node; off-node devices must be ACD hunt group members.

9.2 The order in which hunt group stations receive incoming calls is determined by a list stored in the database. A station or extension list can appear in a single hunt group more than once, and it can appear in multiple hunt group lists, if desired.

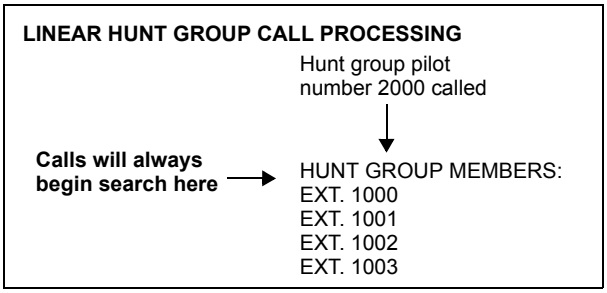
9.3 Hunt groups have their own extension numbers (defaults to 2000-2074 or 2000-2299). Individual stations within the hunt group can be called using their assigned extension numbers.

9.4 Two other premium feature hunt group types are also available: UCD Hunt Groups and ACD Hunt Groups. For UCD information, see page 5-37. For ACD, see page 5-44.

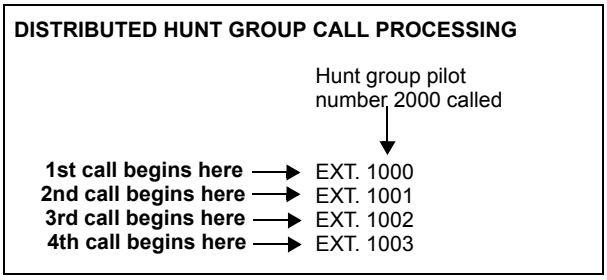
A. HUNT GROUP CALL DISTRIBUTION

9.5 When an intercom or outside call is transferred or rings in to the pilot number, it circulates through the hunt group in linear or distributed order until answered, as described below.

- **Linear order**: Incoming calls always start circulating by ringing at the first station (or extension list) on the hunt group list that is stored in the database.



- **Distributed order**: To even out the call load, distributed order shifts the starting point of each call. When a station user receives a call, the next station or extension list in the hunt group receives the next incoming call. If the receiving station is busy (or all stations in the extension list are busy), or if there is no answer before the No Answer Advance timer expires, the call goes to the next station or extension list on the hunt group list.



NOTE: See page 5-44 for information on the additional distribution types available in ACD hunt groups.

9.6 If an extension list is included in a hunt group, a call rings all stations on the extension list when it reaches that point in the hunt group list. Therefore, to create an "all ring" type of hunt group, you can program the hunt group as either linear or distributed and then assign an extension list as the only hunt group member.

NOTE: A CP Server is required to support off-node members of an all-ring hunt group.

B. HUNT GROUP CALL PROCESSING

9.7 Hunt group stations receive the following indications when a call is ringing in:

- If an outside call is ringing, the station designated to receive the call first shows ring flash on the associated individual trunk button (if it has one) ***or*** a CALL button until the call is answered or the No Answer Advance timer expires and the call moves to the next station.
- If hunt group camp on is allowed and all stations on the hunt group list are unavailable (busy, in do-not-disturb, or with hunt group remove enabled), an intercom or outside call camps on, and the stations receive the following indications:
  - *Busy stations:* Receive hunt group camp-on tones (if enabled) and display. If there is an individual trunk button associated with the trunk, it shows ring flash. (There is a programmable station flag that can disable the camp-on tones for hunt group calls. See PROGRAMMING, page 6-211.)
  - *Stations with Hunt Group Remove enabled:* Receive camp-on tones and display. If there is an individual trunk button associated with the trunk, it shows ring flash.
  - *Stations with Do-Not-Disturb enabled:* Receive no camp-on indications, but if there is an individual trunk button associated with the trunk, it shows ring flash.
  - *The first station that becomes available:* Receives ringing and a flashing trunk or Call button. The camp-on and ring flash indications end at the other stations. (The associated trunk button is steadily lit at those stations to show that the trunk is busy.)
- If hunt group camp on is ***not*** allowed and all stations on the hunt group list are unavailable, the hunt group members do not have any camp-on indications. (See page 6-168 for camp-on flag programming information.) Callers hear one of the following:
  - *Intercom callers* hear busy signals.
  - *Callers on a trunk that does not have to be seized for the phone system to return busy signal* (such as a T1 E&M or trunk with E&M equipment) hear busy signals.
  - *Callers on a trunk that has to be seized for the phone system to return busy signal* (such as a non-T1/E1 loop-start) hear ringing, and the call is sent to the primary attendant.
  - *ISDN callers* hear busy signals.

9.8 Stations within the hunt group can receive direct trunk ring-in, intercom, forwarded, or transferred calls to their individual extension numbers without affecting other stations in the hunt group.

9.9 Trunk groups can be programmed to ring in directly to either a pilot number or extension number(s). If assigned to a pilot number, ring in for the trunk group cannot be assigned to any other extension number(s).

9.10 If an outside call rings in to a pilot number that does not have hunt group station assignments, the call is sent to the primary attendant. If a station user attempts to transfer a call to an invalid pilot number, the call is placed on individual hold, and the station user hears reorder tones. A station user attempting to place an intercom call to an invalid pilot number hears reorder tones, and the display shows NO MEMBERS IN HUNT GROUP.

9.11 Hunt group programming affects the Call Forwarding feature in the following ways:

- **Hunt group calls follow unconditional forward**: If a station in a hunt group is in the unconditional call forward mode to another station, calls to the hunt group follow the station forwarding request. A forwarded hunt group call rings at the forwarding destination until the No Answer Advance timer expires. If it is not answered before the timer expires, the call returns to the hunt group and continues circulating through the hunt group list.
- **Conditional forward dependent on timer interaction**: Stations that are busy and have their calls forwarded conditionally (no answer, busy, or unavailable) to another station receive the hunt group call (if the station is not busy) until the Forward No Answer timer expires. At this point, if the hunt group's No Answer Advance timer has not expired, then the call is forwarded. After the No Answer Advance timer expires, the call circulates to the next station in the hunt group list.
- **Hunt group calls are not forward to some destinations**: Hunt group calls are not forward to voice mail, outside numbers, or system forwarding paths.
- **Announcement and overflow stations can forward hunt group calls**: If an announcement or overflow station has call forward enabled, hunt group calls follow the forward, and the forwarding destination station acts as the announcement or overflow station.
- **Hunt groups can receive forwarded calls**: Stations can forward calls to a hunt group's pilot number.

9.12 Hunt groups can be assigned as message centers and/or alternate message sources for individual stations.

C. HUNT GROUP TIMERS

9.13 Several timers are programmed on a hunt group-by-hunt group basis, as shown in the following table.

Table 5-7. *Hunt Group Timers*

Timer	Default	Range	Purpose
No Answer Advance	18 seconds	1-255 seconds	Determines the amount of time a call rings at a hunt group station or extension list (unanswered) before advancing to the next station or extension list on the hunt group list.
Announcement	18 seconds	1-255 seconds	*(UCD Hunt Groups Only)* Determines the amount of time a call remains unanswered before it is sent to the hunt group's announcement station. UCD Hunt Groups software licenses are required.
Overflow	72 seconds	1-255 seconds	*(UCD Hunt Groups Only)* Determines the amount of time a call circulates through the hunt group (unanswered) before being sent to the hunt group's overflow station. This timer is started when the Announcement timer expires or, if there is no announcement station, when the call is received by the hunt group. It is restarted each time the call leaves the overflow station. UCD Hunt Groups software licenses are required.
Recall	180 seconds	1-65535 seconds	Determines the amount of time a call circulates through the hunt group (unanswered) before being sent to the hunt group's recall destination station. This timer is started when the call is received by the hunt group.
Wrap-Up Duration	15 seconds	1-65535 seconds	*(ACD Hunt Groups Only)* Each time an agent ends an ACD hunt group call, the ACD Wrap-Up Duration timer starts. From the moment this timer starts until that timer expires, the agent does not receive calls through any ACD hunt group; however, calls in an agent's queue that were initiated before the Wrap-Up Duration timer starts are delivered to the agent for servicing. In the Wrap-Up state, the agent can receive non-ACD hunt group calls, direct ring-in calls, and transferred calls. The default value is 15 seconds, and the range is 1-65535 seconds, or up to 18 hours.
Average Connect Time Per Call	60 seconds	1-10,000 seconds	*(UCD Hunt Groups Only)* A Voice Processing Unit application announcement station's message can be programmed to include the caller's queue position and/or estimated wait time. The estimated wait time is based on the Average Connect Time Per Call multiplied by the number of calls ahead of the caller in the queue, divided by the number of available hunt group members (*avg. connect time per call* x *no. of waiting calls* ÷ *available members*). (See page 5-39 for more information.) UCD Hunt Groups software licenses are required.

D. RECALL DESTINATION STATIONS

9.14 An incoming call to a hunt group immediately registers the Recall timer if there is a recall destination identified for that hunt group. If a call is not answered by a hunt group station before this timer expires, the call exits the hunt group and begins ringing at the recall destination station. The call remains at this destination until it is answered or the caller hangs up.

9.15 If there is no recall destination station, the call remains in the hunt group until it is answered or the caller hangs up.

9.16 If the recall destination is a Voice Processing application and the system cannot communicate with the Voice Processing PC, the call remains in the hunt group until it is answered or the caller hangs up.

E. HUNT GROUP REMOVE/REPLACE AND DO-NOT-DISTURB

9.17 Hunt group members can temporarily stop hunt group calls from ringing at their stations by entering the Hunt Group Remove feature code as described below. (If a station is assigned to more than one hunt group, this halts calls from all hunt groups. Hunt group assignments cannot be removed individually.) Do-not-disturb can also be used to halt hunt group and other calls to the station (see page 5-257 for details).

NOTE: This feature has no effect on ACD hunt groups. See page 5-45 for ACD login/logout information.

9.18 When the Hunt Group Remove feature is enabled, the user still receives the camp-on display and tone, and the individual trunk button flashes, if one exists, for calls to the hunt group. (The station continues to receive calls placed to its extension number.) Hunt group overflow and announcement stations cannot block hunt group calls using this feature.

9.19 If a phone user has programmed a button for entering the Hunt Group Remove/Replace feature code, and if that button has a lamp, the lamp is lit whenever the station is removed from the hunt group.

9.20 To remove the station from its hunt group(s) or replace the station in its hunt group(s):

Phone: While on or off hook, enter the Hunt Group Remove feature code (322) to halt hunt group calls or enter the Hunt Group Replace feature code (323) to return the station to its hunt group(s). Or, use the Hunt Group On/Off feature code (324) to toggle the hunt group mode on or off. You hear a confirmation tone, and the display shows ACCEPT (or DIVERT) HUNT GROUP CALLS. *If off hook,* hang up.

Single-Line Set: Lift the handset and enter the Hunt Group Remove feature code (322) to halt hunt group calls or enter the Hunt Group Replace feature code (323) to return the station to its hunt group(s). Or, use the Hunt Group On/Off feature code (324) to toggle the hunt group mode on or off. You hear dial tone.

F. UCD HUNT GROUPS

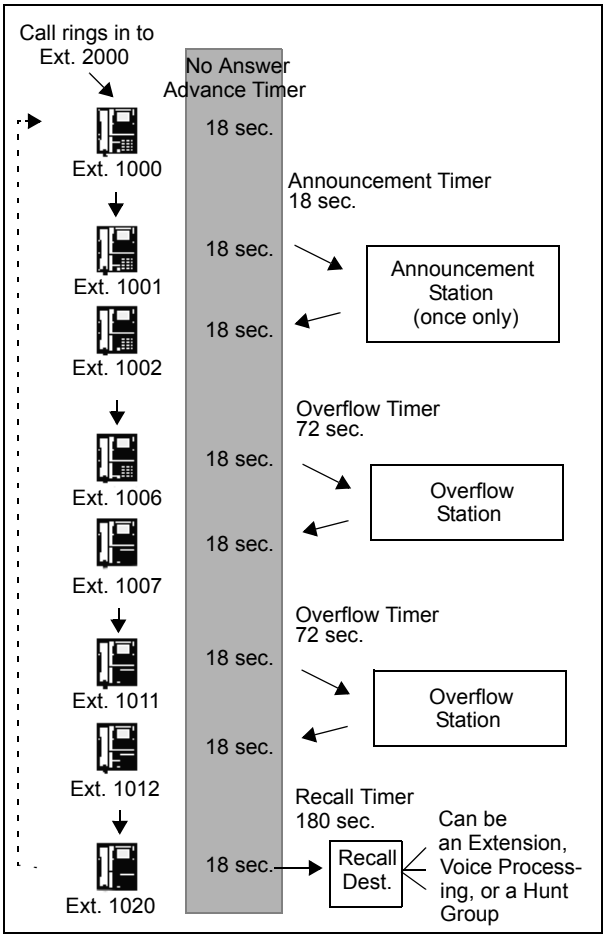
NOTE: This feature requires the UCD Hunt Groups software license.

9.21 Uniform Call Distribution (UCD) hunt groups provide additional features to improve hunt group efficiency. The features added when UCD is enabled are as follows:

- Announcement and overflow stations pick up unanswered calls when the hunt group stations are busy.
- Hunt group priority ranking places calls to one hunt group before another at stations that are members of more than one hunt group.
- Hunt group supervisors can monitor outside calls of any member of the hunt group.

Announcement and Overflow Stations

9.22 Announcement and overflow stations are individual stations that receive unanswered calls when all of the hunt group stations are unavailable. (They are not usually included in the hunt group list.) Announcement and overflow stations can be separate stations, or they can be the same station. Also, each of the hunt groups can have the same announcement and overflow stations, or they can be assigned different stations. In an Inter-Tel system network, the announcement and overflow stations can be off-node devices (see page 6-156 and Appendix A — Networking and for more information about networks and off-node devices).



9.23 **Announcement Stations**: When a call rings in or is transferred to a hunt group, the Announcement and Recall timers are started. The incoming call circulates through the hunt group until it is answered or the Announcement timer expires. When this timer expires, the unanswered call is picked up by the designated announcement station which is a playback device that answers the call and plays a message. Meanwhile, the call continues circulating through the hunt group (unless it was sent to a Voice Processing application and then transferred to a station). If it is answered by an available hunt group station while the announcement station is connected to the call, the call leaves the announcement station. Each direct ring-in call transfers to the announcement station only once. If it remains unanswered when the Overflow timer expires, the call is sent to the overflow station. Or, if there is no overflow station, the call is sent to the recall destination station when the Recall timer expires.

9.24 **Overflow Stations**: The Overflow timer is started when the Announcement timer expires or, if there is no announcement station, when the call is received by the hunt group. If an incoming hunt group call is unanswered when the Overflow timer expires, the call is picked up by an overflow station. The overflow station is a playback device that answers the call and plays a message. Meanwhile, the call continues circulating through the hunt group (unless it was sent to a Voice Processing application and then transferred to a station). If the call is answered by an available hunt group station while the overflow station is connected to the call, the call leaves the overflow station. The Overflow timer restarts each time the unanswered call leaves the overflow station. An unanswered call returns to the overflow station each time the Overflow timer expires until the call is answered by a hunt group station or it is sent to the recall destination station.

9.25 The following table shows the path that an incoming hunt group call follows for all possible combinations of announcement and overflow stations.

Table 5-8. *Hunt Group Call Path*

Ann. Station	Over Flow Station	Call Path
No	No	Call remains in hunt group.
Yes	No	Call goes to announcement station only once, after Announcement timer expires.
No	Yes	Call goes to overflow station after each expiration of Overflow timer.
Yes	Yes	Call goes to announcement station once (after Announcement timer expires) then goes to overflow station after each expiration of Overflow timer.

9.26 If an announcement or overflow station has call forward enabled, hunt group calls follow the forward, and the forwarding destination station acts as the announcement or overflow station.

Voice Processing Applications as Announcement or Overflow Stations

9.27 When a Voice Processing application is used as an announcement or overflow station, calls to the application automatically stop circulating through the hunt group if the caller selects a valid digit translation option (such as pressing a digit to select an extension from a Call Routing Announcement menu), and the call is transferred. This allows the application to send the call to other stations without the call being "pulled back" into the hunt group when a hunt group station answers the call. However, if the caller does not dial a valid digit translation option, the call is pulled back if a hunt group member answers.

NOTE: Standard Display, Associate Display, Axxess Basic Digital, and Eclipse Basic Digital Phones do not send DTMF tones on intercom calls and, therefore, cannot stop circulating through the hunt group by dialing a digit.

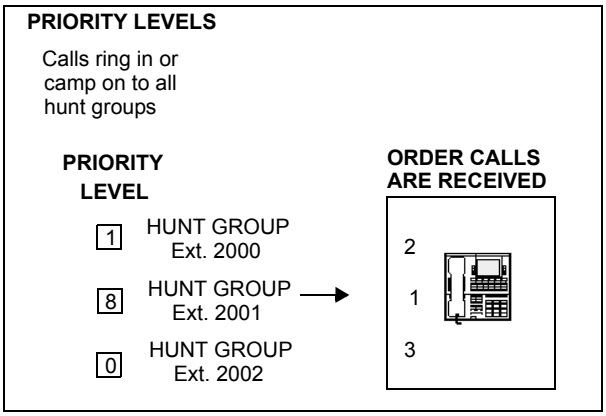
9.28 A Call Routing Announcement application's message can be programmed to include the caller's queue position and/or estimated wait time. The queue position announcement tells the caller how many calls are ahead of his or her call. This includes calls being served and waiting calls (however, all calls being served count as one call). The estimated wait time is based on a programmed Average Connect Time Per Call multiplied by the number of calls ahead of the caller in the queue, divided by the number of available hunt group members (*avg. connect time per call* × *no. of waiting calls* ÷ *available members*). See pages 4-21 and page 6-356 for details.

9.29 If a Voice Processing application is used as an announcement or overflow station or as the recall destination, and the system is unable to communicate with the Voice Processing PC, outside calls are not be sent to the announcement or overflow application. They continue to camp on to the hunt group.

UCD Hunt Group Priority List

9.30 Some stations may be members of more than one UCD hunt group. For this reason, hunt groups are assigned a "priority level." The priority level determines which hunt group's calls should be received first when calls ring in or camp on to several hunt groups at once.

9.31 UCD priority levels range from 0 (low) to 75 (high). If a station is a member of multiple hunt groups that have the same priority level, calls received by those hunt groups are queued in the order they were received by the phone system. (Non-UCD hunt groups are treated as if they have a priority level of 0.)



UCD Hunt Group Supervisors and Station Monitoring

9.32 The Station Monitor feature allows hunt group supervisors to monitor the calls of anyone in a specified hunt group. It can be useful in training or in evaluating the performance of hunt group members.

NOTE: Station Monitor and Barge-In features are available for UCD and ACD hunt groups, not just UCD hunt groups.

9.33 As a courtesy, hunt group members should be notified in advance that their calls may be monitored. In addition, a programmable option can be enabled that sends a tone to the station being monitored whenever the hunt group supervisor joins an ongoing call. See page 6-262 in PROGRAMMING. (Note that call monitoring may be illegal in some locations. It is up to the end user to ensure that use of this feature is in compliance with local laws.)

9.34 In DB Programming, each hunt group can have one or more stations assigned as the hunt group supervisor(s). An extension list can be assigned as the supervisor to provide multiple supervisors. The supervisor is usually not a member of the hunt group. If the supervisor is a member of the hunt group, the Hunt Group Remove/Replace feature can be used at any time without affecting the Station Monitor ability. If desired, one station can be assigned as the supervisor for more than one hunt group.

NOTE: The Barge-In/Monitor feature can be used across nodes, as long as all nodes are running v9.0 or later. Only ACD hunt groups using Agent IDs can set up a supervisor to monitor across nodes.

ALSO: You can add off-node stations as hunt-group supervisors.

9.35 To monitor a hunt group member's call, the supervisor enters the Station Monitor feature code (321) and dials an extension number. The supervisor is then connected to the call and can hear both parties, but cannot be heard by either one. If the monitored call is terminated, transferred, or placed on hold by the hunt group member, the monitor function is terminated.

9.36 In the associated hunt group, the supervisor may monitor any active intercom or CO-to-intercom call (both hunting and non-hunting), including incoming, outgoing, and DISA-to-intercom calls. Conference calls and calls that do not involve hunt group members cannot be monitored.

9.37 If the supervisor attempts to monitor a station that is not on an active call that allows monitoring, the system sends reorder tones, and the supervisor must enter the feature code again to try another number. If the supervisor attempts to monitor a station that is not in the hunt group or an idle station in the hunt group, the system sends reorder tones and cancels the Station Call Monitor feature.

9.38 Station monitoring requires conferencing circuits. If resources are not available when a supervisor attempts to monitor a station, the supervisor's display shows NO CNF CIRCUITS AVAILABLE, and the monitor is not allowed. Multiple supervisors can monitor the same station, providing that a conference circuit is available for each supervisor.

9.39 The supervisor station cannot use the Agent Help feature while monitoring a call.

9.40 To monitor a hunt group call (designated supervisor only):

1. **To use the speakerphone:** While on hook, enter the Station Call Monitoring feature code (321). You hear a confirmation tone, and the Speaker button lights. The display shows ENTER EXT TO MONITOR.

   To use the handset: Lift the handset and enter the Station Call Monitoring feature code (321). You hear a confirmation tone. The display shows ENTER EXT TO MONITOR.

2. Dial the extension number (or press the lit station speed-dial or DSS/BLF button) of the station to be monitored. You are automatically connected to the call, and the display shows MONITORING *(username)*.

3. **To monitor another hunt group member's call:** Enter the Station Call Monitoring feature code (321), and dial the extension number (or press the speed-dial button).

   To terminate the monitor feature: *If off hook,* hang up. *If on hook,* press the Speaker button.

9.41 In addition, if running system versions 7.004 and later or 8.002 and later, you can use the Record-A-Call and Station Monitor features simultaneously.

9.42 To use the Record-A-Call and Station Monitor features simultaneously:

1. Initiate a monitor on the desired device (feature code 321 by default).
2. Have the monitored station start a Record-A-Call session (feature code 385 by default).

9.43 Because you cannot monitor a station that is currently recording a call, you must start the monitor before the Record-A-Call session is initiated. In addition, supervisors cannot initiate a Record-A-Call session if they are monitoring a station; only the monitored device can start a Record-A-Call session. If you attempt to monitor a station that is actively recording a call or if you attempt to record a call while monitoring a station, the phone displays CANNOT ACCESS FEATURE.

NOTE: ATM systems do not support the simultaneous use of Record-A-Call and station monitoring.

Barge-In

9.44 For systems running software v9.0 and later, the Barge-In feature adds new functionality to the existing Station Monitoring feature. When silent monitoring a call, the supervisor now has the following options:

- **Barge-In**: Allows the supervisor to barge-in on a call to help the hunt group member/agent (default feature code 386).
- **Record**: Allows the supervisor to record the call to review it later (default feature code 385). The supervisor may also hang up and continue to record the call until one of the parties on the call hangs up or puts the call on hold. The record option allows the supervisor to record several calls at once and have them delivered to his or her mailbox.
- **Steal**: Allows the supervisor to steal (take away) the call from the hunt group member/agent (default feature code 387).
- **Join and Record**: *(For 6-line display endpoints only.)* Allows the supervisor to join and record the call simultaneously. This feature is useful if the supervisor wants to review the call later.

9.45 Supervisors must be monitoring a call before they can access any of the Barge-In features (barge-in, steal, record, or join and record). Like the Station Monitor feature, Barge-In requires conference circuits. If resources are not available when a supervisor attempts one of these features, the system sends reorder tones and the supervisor sees NO CNF CIRCUITS AVAILABLE on his or her display. A monitor/Barge-In call is rejected if:

- A hunt group member/agent is on a conference call.
- The hunt group member/agent is on a peer-to-peer (P2P) call.
- A hunt group member/agent is getting agent help.
- Another supervisor has already joined the call.

9.46 When the above situations occur, the supervisor see SILENT MONITOR REJECTED on his or her display.

9.47 The Station Monitor and Barge-In features are supported on remote nodes. Supervisors may monitor a hunt-group member/agent as long as they are designated as a supervisor of the group that the member/agent is logged into.

9.48 There are two system flags (see page 6-259) and a system timer (see page 6-329) associated with the Barge-In feature in DB Studio. These programmable fields allow you to control the Barge-In notification tone and display. For troubleshooting information, see page 7-37.

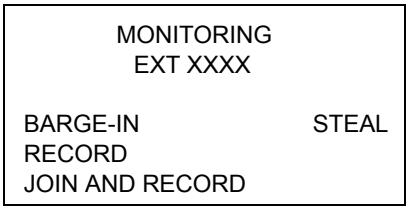
9.49 To monitor and/or barge in on a call:

NOTE: Only 6-line display endpoints have access to the menu buttons. The 2-line or non-display endpoints must use the feature codes to access the Barge-In features.

1. **To use the speakerphone:** While on hook, enter the Station Monitor feature code (321). You hear a confirmation tone, and the Speaker button lights. The display shows ENTER EXT TO MONITOR.

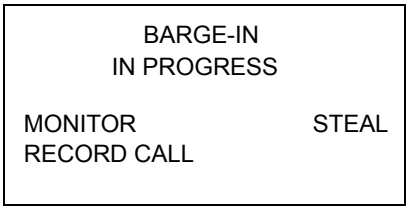
   To use the handset: Lift the handset and enter the Station Monitoring feature code (321). You hear a confirmation tone. The display shows ENTER EXT TO MONITOR.

2. Dial the extension number (or press the lit station speed-dial or DSS/BLF button) of the station to be monitored. You are automatically connected to the call, and the display shows MONITORING (*extension number*).



3. *While silent monitoring a call*, you have the following options:

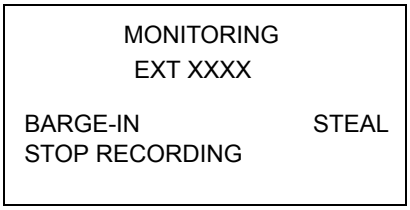
   - *To barge in on a call,* press the **BARGE-IN** menu button or enter the Barge-In feature code (386). On display endpoints, the display shows BARGE-IN IN PROGRESS. While on a call, you can monitor, record, or steal the call by entering the appropriate feature code or by pressing the appropriate menu button.



     To return to silent monitoring of the call, enter feature code 386 again, or press the **MONITOR** menu button.

To record the call, enter the Record-A-Call feature code (385), or press the **RECORD CALL** button.

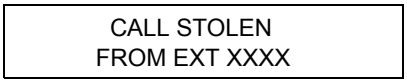
- *To record a call,* press the **RECORD** menu button or enter the Record-A-Call feature code (385). On display endpoints, the display shows MONITORING (*extension number*). While on a call, you can monitor or steal the call by entering the appropriate feature code or by pressing the appropriate menu button.



To stop recording the call, enter feature code 385 again or press the **STOP RECORDING** menu button.

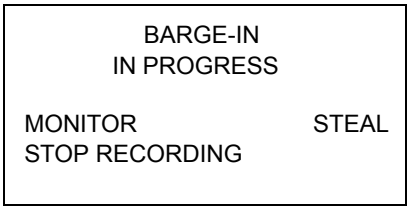
- *To steal a call,* press the **STEAL** menu button, or enter the Steal feature code (387). On display endpoints, the display shows CALL STOLEN FROM (*extension number*). The hunt group member's display shows CALL STOLEN BY (*extension number*). After the supervisor steals the call, the call becomes a regular call. While on a call, you can record the call by entering feature code 385 or by pressing the **RECORD** menu button.

Supervisor's Display



Hunt Group Member's Display

- *To join and record a call,* press the **JOIN AND RECORD** menu button or enter the Join feature code (386), followed by the Record-A-Call feature code (385). On display endpoints, the display shows BARGE-IN IN PROGRESS. While on a call, the supervisor can monitor, stop recording, or steal the call by entering the appropriate feature code, or by pressing the appropriate menu button.



To stop recording the call, enter feature code 385 again or press the **MONITOR** or **STOP RECORDING** menu button.

4. **To terminate any of the features:** *If off hook*, hang up or enter the specific feature code of the feature you want to terminate. *If on hook*, press the Speaker button or enter the specific feature code of the feature you want to terminate. On display endpoints, the display shows BARGE-IN TERMINATED or SILENT MONITOR TERMINATED (if you are silent monitoring at the time of termination).

OR

G. ACD HUNT GROUPS

NOTE: This feature requires the ACD Hunt Group software license. To take full advantage of all hunt group features, the UCD Hunt Groups software license should also be licensed.

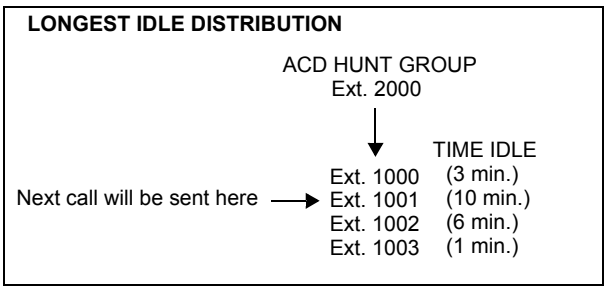
9.50 Automatic Call Distribution (ACD) can be programmed to distribute hunt group calls to equalize call time or call count among the available members. If the System OAI Events premium feature is licensed, ACD hunt groups can also be programmed to send call information records that can be processed by an external device connected to a system serial port (such as Contact Center Suite).

9.51 ACD hunt groups can use the standard hunt group features described on page 5-32, and/or the UCD features described on page 5-37 (if the UCD Hunt Groups premium feature is licensed).

ACD Call Distribution

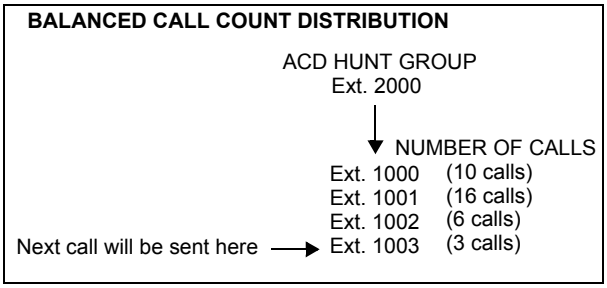
9.52 When an intercom or outside call is transferred or rings in to the ACD hunt group, it can circulate in linear or distributed order (as described on page 5-32) or using one of the following ACD distribution methods:

- **Longest Idle**: Sends an incoming call to the station that has not been involved in a call to this hunt group for the longest period of time. (It does not count calls that were received through other hunt groups, direct ring-ins, or transfers.



NOTE: The Restart ACD Idle Time Upon Login flag affects how the longest idle value is calculated (see page 6-169).

- **Balanced Call Count**: Sends each incoming call to the station that has received the fewest calls through this hunt group, balancing the call load. (It does not count calls that were received through other hunt groups, direct ring-ins, or transfers.)



NOTE: When an agent logs in to a balanced call count ACD hunt group (or comes out of Do-Not-Disturb mode), the system checks the average call count for that ACD hunt group and assigns that average count to the station logging in. This prevents the agent who is logging in from receiving all of the incoming calls until the call count of the lowest agent catches up with his call count. For example, if there are five agents logged in whose call counts are 5, 5, 5, 10, and 10, the average call count of 7 is assigned to the next agent who logs in.

9.53 If an extension list is included in an ACD hunt group set for Longest Idle or Balanced Call Count distribution, it treats each station in the extension list as a separate agent; it does not ring all of the stations on the list at once. If the hunt group is set for linear or distributed order, a call rings at all stations on an extension list at once when the call reaches that point in the hunt group list.

ACD Agent IDs

9.54 ACD hunt group members are referred to as "agents." Agents log in to the ACD hunt group to receive calls and log out to halt ACD hunt group calls. An ACD hunt group can be programmed to circulate calls to agents in two ways:

- **Agent IDs**: Assigns each agent an Agent ID number which the agent enters during the login procedure (described below). The hunt group calls are routed to logged in agents, according to their Agent ID number instead of their station extension. Because the Agent ID is not associated with any station extension, the agent can use any station in the system to log in and does not have to use the same station every time.
- **Members**: Sends calls to the stations where agents are logged in, using a list of stations instead of IDs.

> **NOTE:** For optimum system performance, there should not be more than 1000 Agent IDs in any hunt group and no more than 2000 Agent ID entries in all hunt groups combined.

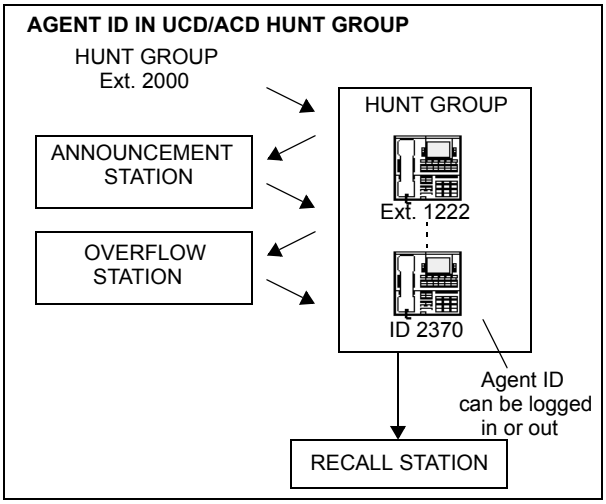
9.55 ACD Agent IDs can be included in Extension Lists which allow several ACD Agents to receive a call at once.

9.56 When a call camps on to an ACD hunt group that uses Agent IDs, only the agents currently logged in to the hunt group receives camp-on indications.

9.57 ACD hunt group supervisors receive visual camp-on displays if they are programmed as members of the hunt group and have the ACD Agent Logout feature enabled.

ACD Agent Login/Logout

9.58 ACD hunt group members are referred to as "agents." Agents can log in to and out of the ACD hunt group at any time. While logged in, the agent receives calls through the ACD hunt group. When the agent is logged out, calls to that ACD hunt group bypass the station. (The Hunt Group Remove/Replace feature, described on page 5-36, does not have any effect on ACD hunt group calls when entered by an ACD agent. However, placing the agent's station in do-not-disturb mode halts all ACD and non-ACD hunt group calls.)



9.59 There are three feature codes that can be used for logging in to and out of the ACD hunt groups: ACD Agent Login, ACD Agent Logout, and ACD Agent Login/Logout. The first two perform only one operation. The third (Login/Logout) is a toggle feature code that logs the station in or out of *all* associated ACD hunt groups at once. If the toggle feature code is programmed in a user-programmable button with a lamp, the lamp is lit when the agent is logged in to an ACD hunt group and available to receive calls, and it is unlit when the agent is logged out.

9.60 To log out of all of your ACD hunt groups at once using the login/logout feature code:

While on or off hook (single-line users must lift the handset), enter the ACD Agent Login/Logout feature code (328). The display shows AGENT LOGGED OUT OF ALL ACDS, and you hear a confirmation tone. Hang up if off hook.

9.61 To log in to all of your ACD hunt groups at once using the login/logout feature code:

1. While on or off hook (single-line users must lift the handset), enter the ACD Agent Login/Logout feature code (328). *If you were already logged in,* the display shows AGENT LOGGED OUT OF ALL ACDS, and you hear a confirmation tone. *If you were not logged in,* continue to the next step.
2. The display shows AGENT LOGIN AGENT ID. Do one of the following:

   — **To log in to all ACD hunt groups for which you have an Agent ID:** Enter your Agent ID. The display shows AGENT LOGGED INTO ALL ACDS. You are logged into *only the ACD hunt group(s) that use the Agent ID that you entered.* Hang up if off hook. *If another agent is already logged in at this station,* you hear reorder tones, and the display shows DIFFERENT AGENT ID ALREADY USED. You must have the other agent log out before you can use that station.

   — **To log in to all of your ACD hunt groups that do not use Agent IDs:** Press **#**. The display shows AGENT LOGGED INTO ALL ACDS. You are logged into ***only the ACD hunt group(s) that do not use Agent IDs.*** Hang up if off hook.

9.62 To log in to one or more hunt groups using the ACD agent login feature code:

1. While on or off hook (single-line users must lift the handset), enter the ACD Agent Login feature code (326). The display shows AGENT LOGIN ACD NUMBER.
2. Enter the pilot number of the desired ACD hunt group. (Or, you can press **#** to log in to several ACD hunt groups at once.)

   If you entered an invalid hunt group number, the display shows NOT AN ACD HUNT GROUP, and you hear reorder tones.
3. The display shows AGENT LOGIN AGENT ID. Then do one of the following:

   — **To log in to ACD hunt group(s) using Agent IDs:** Enter your Agent ID. The display shows AGENT LOGGED INTO ALL ACDS. You are logged into ***only*** the ACD hunt group(s) that you requested in step 2 ***that use the Agent ID that you entered in this step.*** *If another Agent ID is already logged in at this station,* you hear reorder tones, and the display shows DIFFERENT AGENT ID ALREADY USED. You must have the other agent log out before you can use that station.

   — **To log in to ACD hunt group(s) that do not use Agent IDs:** Press **#**. The display shows AGENT LOGGED INTO ALL ACDS. You are logged into *only* the ACD hunt group(s) that you requested in step 2 *that do not use Agent IDs.*

   If you were already logged in, the display shows ALREADY LOGGED INTO *(hunt group).*

If you are not a member of the entered hunt group, the display shows NOT AN AGENT IN THIS HUNT GROUP, and you hear reorder tones. Start over.

4. Hang up if off hook. Repeat this procedure to log in to additional ACD groups, if necessary.

9.63 To log out of an ACD hunt group:

1. While on or off hook, enter the ACD Agent Logout feature code (327). One of the following occurs:

   If you were logged in to only one hunt group, you hear a confirmation tone and the display shows AGENT LOGGED OUT OF *(hunt group).* The procedure is complete. Hang up if you were off hook.

   If you were logged in to more than one hunt group, you hear a confirmation tone, and the display shows AGENT LOGOUT ACD NUMBER. Go to the next step.

   If you were not logged in to any hunt group, the display shows NOT LOGGED INTO ANY ACD GROUP, and you hear reorder tone.

2. Dial the pilot number of the desired ACD hunt group or press pound [hash in Europe] (#) to log out of all of your ACD hunt groups. You hear a confirmation tone, and the display shows AGENT LOGGED OUT OF *(hunt group).*

   If you were not logged in to the hunt group number that you entered, the display shows NOT LOGGED INTO *(hunt group).* Start over.

3. Hang up if off hook.

ACD Agent Wrap-Up Timer

9.64 Each time an agent ends an ACD hunt group call, the station goes into "wrap-up" mode, and the Wrap-Up timer is started. Until that timer expires, the agent does not receive another call through any ACD hunt group. (However, the agent can receive non-ACD hunt group calls, direct ring-in calls, and transfers.)

9.65 The range of the Wrap-Up timer is 1 to 65535 seconds. The default value is 15 seconds. It is programmed individually for each ACD hunt group.

9.66 If an agent wants to terminate the wrap-up mode before the timer expires, the agent can use the following procedure.

9.67 To terminate ACD agent wrap-up:

While on or off hook (single-line users must lift the handset), enter the ACD Wrap-Up Terminate feature code (329). The display shows ACD WRAP-UP TERMINATED. (*If you were not logged in to an ACD hunt group,* the display shows CANNOT ACCESS RESERVED FEATURE.)

9.68 In addition, you can now program a DSS/BLF button to indicate when an ACD hunt group member is in wrap-up mode. When programmed, the DSS/BLF button is lit solidly while the member is in wrap-up. When the wrap-up timer expires, the DSS/BLF button is unlit, and the agent is made available to take calls. To use this feature, the agent's phone must have an available, user-programmable button.

9.69 To program a DSS/BLF for agent wrap-up at an agent phone:

1. Enter the Program Keys feature code (397 by default).
2. Press the button to program.
3. Enter the ACD Agent Wrap-Up Terminate feature code (329 by default). The button is now programmed for wrap-up indications.

9.70 Because the button is programmed with the ACD Agent Wrap-Up Terminate feature, agents can also press the button to be removed from wrap-up mode. When the agent terminates the wrap-up time, the lamp turns off.

Wrap-Up Mode for Holding ACD Calls

9.71 There is a system flag called "Wrap-Up Mode For Holding ACD Calls." If enabled, it places an ACD agent's station in wrap-up mode when an ACD call is placed on hold. (However, the ACD Wrap-Up Duration timer is not activated.) This prevents the agents from receiving additional ACD hunt group calls after they place an ACD call on hold while the station is idle. (The station can still receive non-ACD calls, as usual.) If the flag is disabled, the agent is available to receive additional ACD calls as soon as an ACD call is placed on hold. In the default state, this flag is *disabled.* This flag is programmed as described on page 6-262.

9.72 This flag applies to any type of hold including individual, system, transfer, and conference-wait hold. If the call is terminated or if it is reverse transferred by another station, the ACD agent's station is made available to receive incoming ACD calls.

9.73 If the agents want to terminate the wrap-up mode while an ACD call is on hold, they can use the procedure given in paragraph 9.67.

Automatic Connect Flags

9.74 When the **Auto Connect** flag is enabled (in Hunt Group Related Programming\ACD Agent IDs, see page 6-264) for an agent ID, and the agent is using a headset, ACD hunt group calls are automatically connected following a short ring burst. (This feature does not work if the agent is not using a headset.)

9.75 When the ACD agent logs in, however, the first call rings until the ACD agent answers it. For subsequent calls, the agent hears the ring burst in the headset, and the call is automatically connected.

9.76 The Auto Connect flag overrides the station's Transfer-To-Connect station flag (see page 6-213). It is disabled by default.

9.77 When the ACD agent removes the station from DND, the call may or may not ring until the agent answers it. This is dependent on the **Allow Immediate ACD Auto Connect after DND** flag (under System\Flags). If this flag is enabled, the agent is automatically connected to all calls, including the first one received after exiting DND. If this flag is disabled, the first call rings until the agent answers it, but subsequent calls are automatically connected. By default, this flag is *disabled*.

ACD Serial Port

9.78 The database contains a list of system serial ports. One or more of the serial ports can be programmed to provide call and device information to an external device. The information can then be processed by the device to generate reports on call traffic, answer times, and other management issues. For more information, refer to the manual provided with your Open Architecture Interface product (such as the Contact Center Suite).

H. ACD HUNT GROUPS SPAN NODES

9.79 There is a software license called Remote Automatic Call Distribution Hunt Groups. It allows ACD hunt groups to span nodes. Node-spanning ACD hunt groups can have either members or ACD Agent IDs.

- **Members**: ACD hunt group members may now include off-node phones, off-node single lines, and Hunt Group Member extension lists.
- **ACD Agent IDs**: Unlike members, there are no off-node Agent IDs. Agent IDs are considered global throughout all nodes in which the Agent exists. This means that if you create Agent ID 100 on one node, you must also create Agent ID 100 on all other nodes that have a station that a user may want to login to the hunt group using that ID.

9.80 The Remote Automatic Call Distribution Hunt Groups premium feature must be licensed on the node where the hunt group resides as well as any node that wants to have users login to the node-spanning hunt group. This feature also requires the ACD Hunt Group software license. Because of this fact, there are several cases that must be considered. The following table shows the different displays and results for logging into and out of ACD hunt groups with the different combinations of the premium feature and the ACD Hunt Group enabled/disabled:

Table 5-9. *ACD Hunt Groups Span Nodes*

STN Node	Hunt Grp Node	Login Display	Login Result	Login "All" Display	Login "All" Result
Yes	Yes	AGENT LOGGED INTO HUNT GROUP XXXX	Agent logged into hunt group.	AGENT LOGGED INTO ALL ACDS	Agent logged into all local and remote groups.
Yes	No	CANNOT ACCESS RESERVED FEATURE	Agent not logged into group.	AGENT LOGGED INTO ALL ACDS	Agent logged into all local hunt groups, but not remote where feature disabled.
No	Yes	CANNOT ACCESS RESERVED FEATURE	Agent not logged into group.	AGENT LOGGED INTO ALL ACDS	Agent logged into all local, but no remote groups.
No	No	CANNOT ACCESS RESERVED FEATURE	Agent not logged into group.	AGENT LOGGED INTO ALL ACDS	Agent logged into all local, but no remote groups.

DB Programming

- ACD hunt groups have options allowing the addition of phone off-node devices, single line off-node devices, and hunt group member extension lists containing local or off-node stations or both.
- If the Remote Call Distribution Hunt Groups software license is not licensed, the off-node options do not appear in the list when programming members for ACD hunt groups.

Operational Changes

- The Station Monitor feature remains limited to a single node. Members who are logged in to a remote node or who are logged in to the supervisor's node from a remote node cannot be monitored.
- Hunt groups with remote members rely on network links between nodes. Node availability and software version compatibility affect hunt group log-in procedures as follows:
  - Hunt group members attempting to log in to all hunt groups are logged in to only those hunt groups on nodes that are reachable. The display on the user device only indicates log in results for reachable nodes.
  - Members logging out of all hunt groups are logged out of groups on reachable nodes immediately and are automatically logged out of groups on unreachable nodes as soon as the link to the node is restored. The phone displays MESSAGE PENDING in this case.
  - When users attempt to log in to a remote hunt group on a node running an earlier version of system software, the phone display shows NOT AN ACD HUNT GROUP.
  - If users on a node running an earlier version of system software try to log into a hunt group on a node, the display shows the message NOT AN AGENT IN THIS ACD GROUP.
- The ACD Wrap-Up functions perform as in previous versions with the exception of priority. Due to timing issues involved in sending messages across the network, the following prioritization method is used for hunt groups with remote members:
  - The first criterion is priority level. The hunt group with the highest priority gets the available agent first.
  - In the event of a tie in priority, hunt groups located locally take priority over those located on remote nodes.
  - In cases where ties in priority occur within the local node, the hunt group with the longest camped on call receives the available agent.
  - Given off-node ties in priority, the first hunt group to camp on receives the available agent.

Maintenance of Node-Spanning Hunt Groups

- With node-spanning hunt groups, deleting an agent from a hunt group can be done from the node on which the hunt group resides or from a remote node.
  - — If an agent is deleted on the node containing hunt group(s) of which he is a member, then the agent is removed from all hunt group(s) on that node and is logged out.
  - — If the agent is deleted from a remote node, the agent is logged off from that node, but remains a member of any hunt group(s) located on other nodes.
- Deletion of ACD Hunt Groups in previous versions resulted in all member stations being logged out. This applies to off-node, logged-in stations also. When the hunt group is deleted, all remotely logged-in stations are logged out.
- If the extension number of an ACD Hunt Group is changed, any local or remote stations that are logged in reflect the change. If links are down at the time of the change, remote nodes do not receive the change and an import/export process is necessary to update the information on the affected nodes.
- Removing an Agent ID from a hunt group agent list logs out any station, local or remote, that is logged in to that hunt group using that Agent ID.
- With the new node-spanning features of ACD hunt groups, changing Agent IDs can be done from two possible locations: on the node where the group resides or on a node where the logged-in station resides.
  - — When changing an Agent ID on the node where the hunt group resides, the remotely logged-in station using that Agent ID is logged out even though the Agent ID still exists on the remote node. The hunt group then contains the new Agent ID in its removed list, making the new Agent ID available for log-in use.
  - — When changing an Agent ID on the node on which the logged-in station resides (a node remote from the hunt group), the remote member is logged out and the old Agent ID no longer exists on that node. The hunt group on the original node contains the old Agent ID in its removed list, because this ID still exists on that node.
- During a database restore on a node containing ACD hunt groups that use Agents IDs, all Agents IDs are placed into each hunt group's removed list. All stations are logged out during the restore.
- When the extension number of a logged-in station is changed, the new extension is logged in with the same Agent ID and is not disconnected. This applies to stations logged in both remotely and locally.

I. ANALOG VOICE MAIL HUNT GROUPS

NOTE: This feature requires the Analog Voice Processor Hunt Groups software license.

9.81 A hunt group can be designated as a "voice mail hunt group" to enable it to contain the multiple ports of an analog voice mail unit (not Voice Processing). Each voice mail unit port is programmed in the distribution list like a regular hunt group station. With this feature, incoming calls to the voice mail unit can be sent to a single pilot number where they can be processed even if one port is busy or out of service.

9.82 If the voice mail hunt group is assigned as the message center for a station, it is called after the Message Wait timer expires. When the voice mail unit answers the call, the called party's mailbox number is automatically dialed.

NOTE: Analog voice mail units should use the Silent Message feature code (367) instead of the Message feature code (365).

9.83 If the voice mail hunt group is a system forwarding point (see the system forwarding description on page 5-220) or if a station is forwarded to the voice mail hunt group, the mailbox number that is dialed when the voice mail unit answers is the original destination station's extension number. (For example: The principal station is called, the call is sent to the first system forwarding point, and then is sent to a voice mail forwarding point where the principal station's extension number is dialed.

9.84 The voice mail hunt group can also be assigned as the alternate message source for each of the voice mail ports. Then, when a port leaves a message at a station, and the station user responds to the message, the hunt group is called instead of the individual port. (For more information on alternate message sources, see page 5-189.)

J. GROUP CALL PICK-UP

9.85 When the Group Call Pick-Up feature is enabled, a call ringing in to a hunt group or one of its stations can be picked up at any other station. Users can enter the Reverse Transfer feature code (4) and dial a hunt group's pilot number to pick up a call that is ringing in to the hunt group's pilot number or to any station within that hunt group.

9.86 The system follows the priority list shown below to determine which call is picked up. The system follows the hunt group list (always beginning with the first station on the list) to check each station in the hunt group and then the overflow station for one type of call at a time. If there is more than one call of the same type at the selected station, the call that was received by the station first is picked up. Holding calls and queue callbacks cannot be picked up.

1. Calls to the hunt group pilot number
2. CO calls
   a. Direct ring-in calls
   b. Transferred calls
   c. Recalls
   d. Camped on calls
3. IC calls
   a. Direct ring-in calls
   b. Transferred calls
   c. Recalls
   d. Camped on calls

NOTE: Group Call Pick-up can only retrieve calls from stations that are currently logged in to the hunt group. You cannot use this feature to pick up calls from members who have logged out using the Hunt Group Remove feature code. Also, Group Call Pick-up cannot be used on ACD Hunt Groups that use Agent IDs. It can only be used on hunt groups that use lists of extensions. If the ACD Hunt Group flag is enabled, the Group Call Pick-up flag is dimmed.

ALSO: Group Call Pick-up does not affect DSS lamp status any differently than standard hunt group calls or individual calls.

9.87 To pick up a call that is ringing at a hunt group:

1. Lift the handset, and/or enter the Reverse Transfer feature code (4). You hear a single progress tone.
2. Dial the pilot number of the hunt group where the call is ringing.

   Phone: If the system is programmed for direct connection of reverse transferred calls, you are immediately connected. If not, press the flashing CALL, individual trunk, or IC button.

   Single-Line Set: You are automatically connected.

10. TRUNK FEATURES

10.1 This section explains the trunk functions and programmable features. Trunk capacities are explained in SPECIFICATIONS on page 2-4.

CAUTION

While this system is designed to be reasonably secure against CO trunk misuse by outside callers, there is no implied warranty that it is not vulnerable to unauthorized intrusions, toll fraud, or unintended toll charges. If the central office [local exchange] does not provide supervision, it does not disconnect the call when one party hangs up, making it possible for a caller to remain connected to a CO trunk circuit. If this happens, and the outside caller remains connected, toll charges could continue to accrue until the outside caller hangs up. Or, if this happens, and the outside caller begins dialing, the call could be placed through the phone system and would then be billed to the system's owner. The system cannot check this type of call for toll restriction and may not register the call in SMDR. This problem could arise when a call is connected to a station or when a call is in an unsupervised conference.

A. DUAL-TONE MULTI-FREQUENCY (DTMF) OR DIAL-PULSE SIGNALING

10.2 Trunks can be designated for DTMF or dial-pulse signaling through DB Programming (see PROGRAMMING, page 6-254).

10.3 Phones and DTMF single-line sets can be used on either type of trunk. While on a dial-pulse trunk, a station user can switch from dial-pulse to DTMF signals by pressing the pound [hash] (#) or asterisk (*) buttons. After the pound [hash] or asterisk button is pressed, all digits that follow during that call are sent out as DTMF tones. (This is useful when calling computerized services that require a DTMF telephone, such as voice mail.)

B. TYPES OF TRUNKS AVAILABLE

10.4 The Inter-Tel system supports the following types of trunks:

- **Loop start trunks on the Loop Start Card (LSC) or Loop Start/Ground Start Card (LGC)**: Loop start trunks support the Caller-ID feature. (See page 5-72 for details.)
- **Ground start trunks on Loop Start/Ground Start Card (LGC)**: Ground start trunks on an LGC support incoming and outgoing calls. (See page 5-72 for details.)
- **T1 trunks on the T1 Card (T1C) or T1/E1 card** *(U.S. only)*: (See page 5-55 for more T1 information.) The phone system supports the following types of T1 trunks:
  - — *Loop-start:* Supports incoming and outgoing calls and Caller-ID.
  - — *Ground-start:* Supports incoming and outgoing calls. T1 ground start trunks do not support Caller ID [CLIP].
  - — *DID:* Supports DID, ANI, and DNIS call routing. (See page 5-72 for details.)
  - — *Basic E&M:* Supports DID, ANI, and DNIS call routing as well as basic networking (such as ring in to DISA and connection of two systems). For E&M information, see page 5-66. See page 5-72 for DID, ANI, and DNIS information.
  - — *OPX:* A single-line set can be connected to the system via a T1C or T1/E1 card connection.
- **Primary Rate T1 trunks on the T1/PRI or T1/E1/PRI card** *(U.S. only)*: (See page 5-55 for more T1 information.) The phone system supports ISDN Primary Rate trunks on up to eight T1 cards in the system.

- **Primary Rate ISDN trunks on the E1/PRI or T1/E1/PRI card** *(Europe only)*: (See page 5-65 for more T1 information.) The phone system supports ISDN Primary Rate trunks.
- **Basic Rate ISDN Trunks on the BRS card**: (See page 5-67 for more information.) The Inter-Tel system supports ISDN Basic Rate trunks.
- **DID trunk interface on a Single-Line Adapter (SLA)**: Each SLA can be configured to support up to two DID trunk interfaces [DID trunk is not applicable to Europe].

10.5 Because loop start trunks frequently do not provide supervision, unsupervised trunk-to-trunk calls (such as unsupervised conferences and forward to an outside number), are limited by the Unsupervised CO timer. When this timer expires, the call recalls the primary attendant who can determine whether the trunks are still in use or if the call should be disconnected. (See page 5-28 for attendant recall information.) If the trunk connections are ground start or DID trunks, this timer is not needed as they supply reliable disconnection signals.

C. T1 TRUNKS AND PRIMARY RATE T1 TRUNKS (U.S. ONLY)

NOTE: This requires the Advanced CO Interfaces software license and a T1 Card (T1C), T1 Primary card (T1/PRI), or T1/E1/PRI card.

Standard T1 Trunks

10.6 The term "T1" refers to a specific digital method of transmitting voice and data; it is the basic 24-channel time-division multiplex, pulse code modulation system used in the United States. Because each T1 trunk actually consists of 24 individual circuits (or channels) multiplexed together, it is often less expensive to purchase a single T1 trunk than it is to purchase multiple individual trunks.

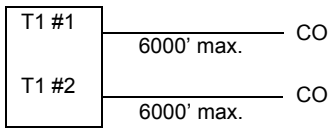
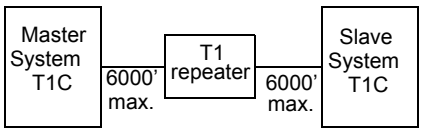
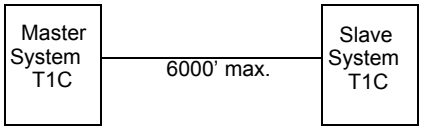
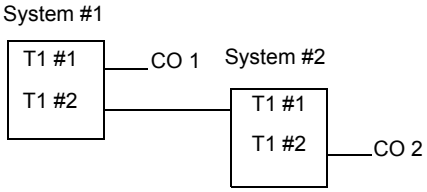
10.7 T1 trunks can be programmed to provide the following services:

- **Loop-Start Interface**: Loop-start T1 trunks can provide the same services as regular loop-start trunks, including Caller-ID. See page 5-72 for Caller-ID information.
- **Ground-Start Interface**: Calls can be placed and received using ground-start T1 trunks.
- **Direct Inward Dialing (DID) Interface**: Direct inward dialing (DID) allows an outside party to dial into the system without attendant intervention. See page 5-70 for more information on DID.
- **E&M Interface**: E&M trunks, also called "tie" lines, are special trunks that tie two distant telephone systems together. They allow the users of either telephone system access to the users and resources of the other telephone system. See page 5-66 for more information on E&M trunks.
- **Off-Premises Extensions (OPX)**: Off-premises extensions are remote single-line DTMF sets that are connected to the T1 or T1/E1 card through telephone company OPX circuits (or customer-provided circuits).

10.8 Fractional T1 (FT1), which uses fewer than the standard 24 circuits, can be used on the Inter-Tel system. If FT1 is used, the unused circuits of the T1 or T1/E1 card must be unequipped.

10.9 See Figure 5-10 on page 5-56 for examples of T1 installations.

Table 5-10. *T1 Trunk Applications*

System-to-CO 	In this application, the T1 cards are connected directly to the central office. The line buildout is programmed to support the distance to the CO (or nearest repeater). Both cards draw reference clock from the public network and are slave clocks. One card is designated as the system reference clock.
System-to-System with repeater(s) 	In this application, two Inter-Tel systems are connected through a T1 repeater. Both T1 cards have the line buildout programmed to support the distance to the T1 repeater. One card is designated as the master clock and the other is the slave clock (that receives reference from the master clock).
System-to-System without repeaters 	In this application, two Inter-Tel systems are connected directly to each other. The T1 cards have the line buildout programmed to support the distance between the Inter-Tel systems. One card is designated as the master clock and the other is the slave clock (that receives reference from the master clock).
Sample T1 Network 	In a T1 network that involves multiple CO and T1 card connections, the line buildout is determined by each connection. In the example shown here, the two T1 cards connected to the CO receive reference clock from the public network and are slave clocks. The two that are connecting the Inter-Tel systems do not draw reference from the public network. T1 #2 on System #1 is the master clock and T1 #1 on System #2 is a slave clock.

T1 OPX Disconnect Flag

10.10 In previous software versions, some sites had difficulties when using a loop start T1 line configured for OPX to connect two Inter-Tel systems and performing "semi-networking" applications such as intercom access, lighting message lamps, and shared voice mail. The reason for the difficulty is that the T1 OPX does not send a disconnect signal to the loop start trunk, which caused trunks to be "locked up."

10.11 For T1 OPX stations, there is now a flag called "Send T1 OPX Disconnect Flash" that sends a proprietary disconnect signal from the T1 OPX to the loop start trunk. (The "A" bit is toggled high for the duration of the SL Disconnect Flash Duration timer.) At default, this flag is disabled. It is programmed on a station-by-station basis and is available only to single-line stations.

10.12 To use this station flag, you must have a T1 or T1/E1 card equipped with the proper firmware. To remind you, a warning appears in DB Programming whenever the flag is changed. The warning states that the flag has no effect unless the correct T1 firmware is installed on the card. If your T1 or T1/E1 card does not have the proper firmware, there is an upgrade kit available (see REPLACEMENT PARTS for part numbers).

NOTE: This flag does not affect T1 channels configured for Loop Start. Therefore, an Inter-Tel system can be on the receiving end with T1 Loop Start channels and recognizes the disconnect. However, it cannot ***send*** the disconnect unless it has system version 4.3 or later software and has T1 OPX enabled.

Primary Rate B-Channel T1 Trunks

10.13 The Inter-Tel system also supports primary-rate B-channels. Each T1/PRI card can have up to 23 B-channels. Each B-channel is programmed as an individual trunk in the database.

10.14 The capacities for PRI operation (T1/PRI or T1/E1/PRI) is as follow:

Table 5-11. *PRI B-Channel T1 Trunk Capacities*

CPU	Number of Cards
CPC/CPU 64 and 128	2
CPC/CPU 256 and CPU 512	8
CP Server and ATM	28

NOTE: As of version 8.0, the capacities in a CPC64 or CPC128 are limited by the software license.

10.15 The phone system is designed to support only AT&T 4ESS Custom, AT&T 5ESS Custom, National ISDN 2, and DMS-100 switch types or Private Networking. *No other switch types are currently supported by the Inter-Tel system.*

NOTE: Primary Rate trunks that are not subject to toll restriction do not return dial tone. This is because these unrestricted trunks are treated as if they are connected to another Inter-Tel system. Therefore, no dial tone is provided, no digits are collected, and the setup is immediate.

10.16 B-channel trunks can be combined with other types of trunks in any trunk group.

10.17 If a user attempts to place an outgoing call using an individual B-channel trunk, the system sends a Setup Request to the network that requests that B-channel. If the network specifies a different B-channel (possibly because of a glare condition), the system releases the call and the user hears busy tone. The call camps on to the selected B-channel trunk. If the same situation occurs when the user selects a B-channel through a trunk group or ARS, the system simply moves to the next available trunk instead of camping on.

10.18 The CO Hookflash (Recall) feature cannot be used on B-channel trunks.

10.19 The available ISDN services are shown in the chart below. They are divided into two categories: services supported by the system and unsupported services. If the B-channel trunks provide an unsupported service, the system still functions properly. However, it does not make use of the service.

Table 5-12. *Available ISDN Services*

Supported by the Inter-Tel System	Unsupported
Automatic Number Identification (ANI)	D-Channel Backup
On-Demand B-Channel Selection	Flexible Billing
Dialed Number Identification Service (DNIS)	Network Ring Again
Calling Party Number	Non-Facility Associated Signaling (NFAS)
Call-By-Call Service Selection	Primary Rate Inter face Nodal/Centrex Express
	PRI & Private Line Integration
	Switched Digital Services
	User-To-User Information
	Wide Area ACD

10.20 Primary Rate Call Type: All Primary rate ISDN calls have a specific number type (International, National, Network, or Local/Subscriber) and numbering plan field (ISDN, Telephony, or Private) indicated in the call setup message to the public network peer. In certain cases, front end equipment interfacing with the ISDN user (CPE) peer requires a specific number type and/or numbering plan. The number type and numbering plan are programmable per call type for each equipped primary rate card. The operation of this feature is transparent to the user. After the number type and numbering plan are programmed, the Primary rate call setup message includes this information for each call. To set the Primary Rate Call Type for a T1/E1 card, select System—Cabinets, then double-click on the card (or right-click and select Explore Card). Then select **Call Type**.

Trunk Group PRI Call By Call Feature

10.21 System trunk groups provide limited access to PRI Call By Call services. Each trunk group that contains B-channels can be assigned a PRI Call By Call feature to use for outgoing calls. The supported Call By Call features include switched digital circuit services, foreign exchange, TIE services, local exchange, OUTWATS, inter-exchange carrier services, and custom AT&T and Nortel private networks, features, and services. If the customer wants to use more than one PRI Call By Call feature, there can be multiple trunk groups, each programmed to use a different feature. The customer then can select the Call By Call feature by selecting the trunk group associated with the feature.

10.22 When the trunk group has a programmed Call By Call feature, all channels within the specified trunk group indicate the PRI Call By Call feature code in the outgoing ISDN message. Each outgoing ISDN call sends a SETUP message on the Primary Rate D-channel that contains the B-channel (voice channel) assignment, called digits information, and the network-specific facility. The PRI Call By Call feature is specified within the network-specific facility, ISDN information element which contains the Call By Call feature code.

10.23 To properly program the PRI Call By Call feature, the installer must analyze the customer's ISDN outgoing call traffic and configure the amount of B-channels needed to support a particular Call By Call feature or service. In the service order, the installer must request the Call By Call service for the PRI span or a subset of channels.

10.24 The installer selects the Call By Call feature during trunk group programming by selecting the desired PRI Call By Call feature. After the trunk group has been created and assigned to a Call By Call feature, the appropriate ISDN B-channels are programmed into the trunk group.

10.25 The station user dials ARS or a PRI trunk group extension. The system selects the trunk group's programmed Call By Call feature. The system automatically inserts the Call By Call feature into the ISDN call control message and transmits the message to the public network. The Public Network accepts or rejects the requested Call By Call feature in the ISDN message.

10.26 The PRI Call By Call feature supports the following services:

- Call By Call Inactive
- AT&T WATS Band
- AT&T Banded OUTWATS
- AT&T Foreign Exchange
- AT&T TIE Trunk
- AT&T Local Operator
- AT&T Carrier Operator
- AT&T MEGACOM 800
- AT&T MEGACOM
- AT&T INWATS
- AT&T WATS Maximal Band
- AT&T ACCUNET Switched Digital
- AT&T International LDS
- AT&T International 800
- AT&T Electronic Tandem Network
- AT&T Private Virtual Network
- AT&T DIAL-IT 900 / MultiQ
- AT&T National ISDN INWATS

- AT&T Unbanded OUTWATS
- Nortel Private Network
- Nortel INWATS
- Nortel OUTWATS
- Nortel Foreign Exchange
- Nortel TIE trunk

Purchasing T1 Services

10.27 When purchasing a T1 trunk, there are several variables involved, depending on the T1 application required. (See SPECIFICATIONS, page 2-49, for details.) The variables include the following:

- **Card framing type**: The type of framing scheme used by the T1 trunks connected to the card can be D4 Superframe (normally used for voice transmissions) or Extended Superframe (ESF is usually used for data transmissions).
- **Zero code suppression scheme**: The T1 trunk zero-suppression scheme (that limits the number of consecutive zeroes in transmissions) for the trunks on the T1 or T1/E1/PRI card can be AMI (Bit 7), Bipolar Eight Zero Substitution (B8ZS), or None. The T1/PRI card supports B8ZS or None.
- **ISDN switch type**: *(Used for PRI-equipped cards only)* The phone system supports primary-rate B-channels. However, the system is designed to support only AT&T 4ESS Custom, AT&T National ISDN 2, and Private Networking switch types. *No other switch types are supported by the system.*
- **Line build-out (LBO)**: The LBO attenuation of the T1 trunk connected to the card must be designated in programming. This value is determined by the distance to the nearest public network T1 repeater.
- **Reference clock programming**: If the T1 Card is connected to the public network, the card is automatically designated as a slave clock (the public network always acts as the master clock and one card provides the system reference clock). If the card is not connected to the public network, but is instead connected to another T1 Card or a card in another telephone system, it can be a master clock or a slave clock (in relation to the card on the other end).
- **Timers**: *(Timers are programmed only for PRI-equipped cards.)* The ISDN timer default values have been carefully selected to ensure proper system operation under most circumstances. Occasionally, one or more of the timers may need to be adjusted.
- **CO Provides Progress Tones**: *(Used for PRI-equipped cards only)* When the system provides local ringback, some central offices (particularly in Europe) take too long to send the system the connect message when a call is received. When the CO connects a call too slowly, the user can answer before the caller is on the line, causing the caller to miss the greeting (such as “Welcome to Inter-Tel. How may I help you?”). If the CO Provides Progress Tones flag is enabled, the system does not provide local ringback. Instead, the system connects the call to the line and the caller hears CO ringback. That way, when the called party answers, the calling party is already on the line. The flag is *disabled* by default.

NOTE: This flag should be enabled for PRI spans that are used for placing outgoing calls from another node (e.g., Node 2 uses ARS to place outgoing calls using the PRI span on Node 1). If it is not enabled, some of the calls from Node 2 may not go out to the public network properly. Enable this flag for **all** nodes in a system.

- **Connect On Call Proceeding**: *(Used for PRI-equipped cards only)* This flag affects outgoing Primary Rate ISDN calls. If it is enabled, the system connects the B-channel as soon as the Call Proceeding message is received from the CO interface. Some CO interfaces, especially SS7 interfaces, do not send progress indicators when they are playing tones or announcements on the B-channel. They play the tone or announcement prior to sending the progress message, after the call proceeding. In this case, if the flag is *disabled,* the user experiences audio clipping. If the flag is enabled, the channel is opened when the call proceeding message is received and the system does not have to wait for a progress indicator (indicating alerting, busy, reorder tones or announcements).
- **Operator System Access**: *(Used for PRI-equipped cards set for National ISDN2 only)* When National ISDN 2 is used, there is an option to enable Operator System Access. Operator System ACCESS (OSA) is required by some central offices. If this flag is enabled, the user is allowed to request access to an operator services system.

NOTE: If using ARS with National ISDN 2, and OSA is enabled, you must remove Dial Rule #2 Echo Toll Field from the Operator and International Operator Route groups. (See page 6-291.)

- **Send International Toll Digits**: *(Used for T1/PRI or E1/PRI trunks only)* If enabled (default), the system sends international toll digits (for example, 011) in the dial string in addition to the number type. If disabled, the toll digits are stripped from the dial string for international calls using ISDN trunks. Enable this flag if the CO ignores the number type.

10.28 When ordering Primary Rate Interface services, see the following chart for the information your carrier needs to know:

Protocol	Custom ISDN or National ISDN 2
Signaling	Clear Channel 64Kbs
Framing	BSF
Coding	B8ZS
Jack Type	RJ48C
Data	Normal (not inverted)
Glare	CPE Yields on Incoming Calls

Terminology

10.29 Some common terms associated with Primary Rate and ISDN include the following:

- **2B+D**: A 2B+D circuit is divided into two 64 kbps B-channels that can carry voice, video, or data. It also has a 16 kbps D-channel for low-speed data and signaling. There are two types of 2B+D interfaces that use different wiring: the "U" interface uses a single twisted pair and the "S/T" interface uses two twisted pairs. This is also referred to as "Basic Rate Interface (BRI)."
- **23B+D**: A 23B+D circuit is divided into 23 B-channels (64 kbps each) that can carry voice, data, and video simultaneously. It also has a 64 kbps D-channel for out-of-band signaling, call control, and packet data. (**NOTE**: The Inter-Tel system does not support the use of the D-channel for packet data.). This interface uses two twisted-pairs of wire. This is also referred to as "Primary Rate Interface (PRI)."
- **30B+D**: This is the standard European ISDN interface. A 30B+D circuit is divided into 30 B-channels (64 kbps) that can carry voice, data, and video simultaneously. It also has a 64 kbps D-channel for out-of-band signaling and call control, and one framing channel. This interface uses two twisted-pairs of wire.
- **Automatic Call Number Identification (ANI)**: Automatic Numbering Identification (ANI) information identifies the caller's telephone number. The system receives a specified number of digits. *ANI* is another type of ANI that does not require a specified number of digits. The system receives a star (*) before the ANI digits to signal the beginning of the caller's telephone number. Then there is another star after the digits to signal the end of the ANI information.
- **B-Channel**: The B-channel, or "bearer" channel, is the basic component of ISDN interfaces. A B-channel can transmit or receive voice or data at up to 64,000 bits per second (64 kbps).
- **B-Channel Negotiation**: *(Currently not supported)* This allows the phone system to request the channel that is used for each call. When the phone system requests a specific B-channel for an outgoing call, the network decides which channel is used. However, if the channel selected by the network is not the requested-channel, the phone system releases that channel (if seized) and attempts to re-initiate the call using the channel specified by the network. (For an incoming call, the phone system does need to request a channel.)
- **Bandwidth**: The bandwidth of a device determines the frequency range it can handle. Different types of communications use different bandwidths. For example, telephone communication requires a relatively narrow bandwidth, while video requires a wide bandwidth.
- **Basic Rate Interface (BRI)**: Also called 2B+D, BRI provides two 64 kbps B-channels that can carry voice or data and a 16 kbps D-channel for low-speed data and signaling. This can be used for homes and Centrex sites that do not require PRI.
- **Calling Party Number Service**: This service provides the calling party's line number (not billing number) to the called party. *This is not supported in the current system software.*
- **D-Channel**: The data channel, or D-channel, carries the signaling information at 64 kbps for the B-channels in a Primary Rate Interface or at 16 kbps in a Basic Rate Interface.

- **D-Channel Backup**: Because the D-channel supports the other channels in the ISDN network, loss of the D-channel would cause loss of access to the ISDN circuits. D-channels can be backed up to provide protection in the event of D-channel loss. D-channel backup is especially important in Non-Facility Associated Signaling (NFAS), because the D-channel supports all of the B-channels in the ***entire*** network. *This is not supported in the current system software.*
- **Dedicated Service**: Dedicated service reserves a group of B-channels for a specific function, such as outgoing calls. The system uses trunk group programming to determine how the channels are used. Therefore, the system would require a separate trunk group for each function. The installer can create "PBX-provided" dedicated service on call-by-call circuits through DB Programming (such as incoming only or accessible only to a group of users) or the service provider can determine the dedicated service.
- **Dialed Number Identification Service (DNIS)**: DNIS identifies the number that the caller dialed to reach your location. The system receives a base number and a specified number of digits that identify the dialed number. *DNIS* is a form of DNIS that does not have a specified number of digits. The system receives a star (*) before the DNIS digits to signal the beginning of the dialed number. Then there is another star after the digits to signal the end of the DNIS information.
- **Flexible Billing**: Flexible billing allows a 900 subscriber to modify a call's billing rate while speaking with the caller. *This is not supported in the current system software.*
- **H0 and H11**: These switched digital service types combine contiguous B-channels into a single bit stream. H0 combines six channels into a 384 kbps bit stream. H11 combines 24 channels into a 1536 kbps bit stream. Applications for these services include video conferencing and LAN connections. *This is not supported in the current system software.*
- **Integrated Services Digital Network (ISDN)**: ISDN is a network of channels that can provide simultaneous voice, data, and video communication.
- **Multi-Rate ISDN**: This type of switched digital service allows the user to dial up, on demand, any number of combined B-channels. *This is not supported in the current system software.*
- **N X 64 kbps**: This switched digital service type combines the B-channels into a single bit stream. Applications for this service include video conferencing and LAN connections. *This is not supported in the current system software.*
- **Network Ring Again**: Network ring again is like the system queue feature that allows a caller to queue on to a busy station and then be called when the station is available. However, network ring again allows queuing between two PBXs that are on the same ISDN network. *This is not supported in the current system software.*
- **Non-Facility Associated Signaling (NFAS)**: When a site uses several Primary Rate Interface circuits, the D-channel on one circuit can support the B-channels on other circuits, providing one additional B-channel per circuit (that is, 24 B-channels instead of 23B+D). This is called Non-Facility Associated Signaling or NFAS. This arrangement also requires D-channel backup. *This is not supported in the current system software.*
- **On-Demand B-Channel Selection**: This feature allows on-demand allocation of B-channels. With on-demand B-channel selection, all B-channel trunks can be assigned to one trunk group and then programmed to be used as needed (ARS, outgoing access, etc.). This feature takes full advantage of the Inter-Tel system features (trunk groups, ARS, and call routing) and allows maximum on-demand use of the bandwidth. This can be combined with Dedicated Service.

- **Primary Rate Interface (PRI)**: This is the ISDN circuit that provides B-channels that can carry voice, data, and video simultaneously. It also has one D-channel for out-of-band signaling and packet data. Note that the Inter-Tel system does not support the use of the D-channel for packet data. See also 23B+D and 30B+D.
- **Primary Rate Interface Nodal/Centrex Express**: This service type allows communication between a PBX system and a Centrex system. *This is not supported in the current system software.*
- **Private Line Integration**: This allows the system to have private lines using individual channels. The D-channel is not used on the private lines. *This is not supported in the current system software.*
- **Switched Digital Services**: Switched digital services provide ISDN over a single telephone line by combining several B-channels into a single bit stream. Switched digital service types include H0, H11, Multi-rate, and NX 64 kbps. *This is not supported in the current system software.*
- **Two-Stage Caller Identification (*ANI*DNIS*)**: This is a two-stage address service that provides both the caller's telephone number and the dialed number. It combines the *ANI* and the *DNIS* features described above.
- **User-To-User Information (UUI)**: This type of ISDN service allows two users to share and manipulate data information over the D-channel while maintaining a voice connection on a B-channel. For example, both parties can view the same document, discuss it, edit it and see the changes as they are made. *This is not supported in the current system software.*
- **Wide Area Automatic Call Distribution (ACD)**: Like the system ACD feature, this service can distribute calls to provide faster service. However, Wide Area ACD allows the calls to be sent to other PBXs in the network. *This is not supported in the current system software.*

Overlap Sending and Receiving

NOTE: Currently, the overlap sending/receiving protocols are supported with PRI Net 5 and BRI Net 3 switch types, which are prevalent in the European and Mexico markets.

10.30 With overlap sending, the phone system can send some or no called number digits in the setup message and send additional digits (or overflow digits) in subsequent information messages. You can also program the system to use the overlap sending protocol immediately. If this option is selected, the setup message contains no called number digits, and all digits are sent in subsequent information messages as the digits are dialed.

10.31 With overlap receiving, the phone system can receive some or no called number digits in the setup message, followed by overflow digits in information messages. The phone system does not route the call based on these digits until it receives an indication from the network provider that all digits have been sent. If the network provider does not send such an indication, the phone system waits until the Overlap Receiving Timeout Timer has expired. After this timer has expired, the call is routed based on the digits received.

10.32 Prior to version 8.224, the phone system sent/received all Called Number digits in the ISDN setup message for outgoing calls. This process of sending/receiving digits is referred to as "en bloc," however, this is not compatible with some network providers in the European ISDN market. To accommodate these markets, version 8.224 now supports the "overlap" sending method, which allows the phone system to send digits in setup and information messages.

10.33 For details about programming overlap sending and receiving, see page 6-141.

D. PRIMARY RATE ISDN TRUNKS (EUROPE ONLY)

NOTE: This requires the Advanced CO Interfaces software license and an E1 Primary Card (E1/PRI), or T1/E1/PRI Card.

Primary Rate B-Channel E1 Trunks

10.34 The system supports primary-rate B-channels. Each E1C can have up to 30 B-channels. Each B-channel is programmed as an individual trunk in the database.

10.35 The capacities for PRI operation (E1/PRI or T1/E1/PRI) is as follow:

Table 5-13. *PRI B-Channel E1 Trunk Capacities*

CPU	Number of Cards
CPC/CPU 64 and 128	2
CPC/CPU 256 and CPU 512	8
CP Server and ATM	28

NOTE: As of version 8.0, the capacities in a CPC64 or CPC128 are limited by the software license.

10.36 B-Channel trunks can be combined with other types of trunks in any trunk group.

10.37 If a user attempts to place an outgoing call using an individual B-Channel trunk, the system sends a Setup Request to the network that requests that B-Channel. If the network specifies a different B-channel (possibly because of a glare condition), the system releases the call, and the user hears busy tone. The call camps on to the selected B-Channel trunk. If the same situation occurs when the user selects a B-Channel through a trunk group or ARS, the system simply moves to the next available trunk instead of camping on.

10.38 The CO Hookflash (Recall) feature cannot be used on B-Channel trunks.

10.39 The ISDN card supports DDI and CLIP.

Service Flexibility

10.40 ISDN B-channels are placed into trunk groups to control ring-in and outgoing access.

10.41 All B-channels could be placed in the same group or, where available, they can be split for various services. For example, if your network provider supports this, one ISDN link could contain any combination of the following:

- Various lines from the “Main Number” of the company
- Various lines for DDI
- Various single dedicated channels for faxes, modems, etc.

10.42 Each group can contain trunks from different connections.

E. E&M TRUNKS (U.S. ONLY)

NOTE: This feature is available only on T1 or T1/E1 card circuits.

10.43 E&M trunks are special trunks that tie two distant telephone systems together. They allow the users of either telephone system access to the users and resources of the other telephone system.

10.44 E&M trunks, like standard trunks, may be programmed to ring in to call routing tables, individual stations, multiple stations, or hunt groups, or as a DISA call. (If programmed to ring in at a single station, the E&M call follows any programmed forward.)

10.45 When a call is placed over an E&M interface, one of the following things happens:

- If programmed for "Immediate" start type, the calling system immediately begins sending the dialed digits to the receiving system.
- If programmed for "Wink" start type, the systems perform a "handshake" to allow the receiving system to signal that it is ready to receive the digits dialed by the other system. (This is the default start type.)
- If programmed for "Delay Dial" start type, the calling system waits until its E&M Dial Delay timer expires before sending any digits to the receiving system.
- If programmed for "Dial Tone" start type, the calling system waits until it receives dial tone from the other system before sending any digits.

10.46 E&M trunks on the system can be programmed to support DID (see page 5-70. Other options include DNIS and ANI (see page 5-72).

10.47 E&M trunks are programmed into trunk groups. The trunk group programming determines day and night mode outgoing access for the trunks.

10.48 When the system is in the default configuration, all E&M trunks are configured for DTMF signaling. If necessary, some or all of the trunks can be reprogrammed for dial-pulse signaling through DB Programming.

10.49 The DID/E&M [DDI in Europe] Receive Busy Instead Of Camp-On station flag determines whether E&M and DID callers receive busy signal or receive ringback and camp on when calling a busy station. In the default state, busy tones are disabled and the callers hear ringback while camped on to the called station. This flag is programmed on a station-by-station basis (see page 6-210 in PROGRAMMING).

10.50 The following programming is required to use the E&M feature:

1. Make sure the Advanced CO Interfaces premium feature is licensed (see page 6-97 in PROGRAMMING).
2. Install and configure a T1 Card (T1C) or T1/E1 card, as described on page 6-126 in PROGRAMMING. Configure the appropriate circuits as "E&M" and assign extension numbers to the trunks.
3. Program the T1 or T1/E1 card as required. See page 6-126 (T1 Programming) in PROGRAMMING.
4. Program a Call Routing Table to be used by the E&M trunk group. See page 6-319 (Call Routing Tables) in PROGRAMMING. Also see page 5-89 for more information regarding Call Routing Tables. If it is used for DID service, set up the table using the DID numbers.

5. Create a trunk group that contains the E&M trunk(s) and program it to ring in to the Call Routing Table created above. See page 6-228 (Trunk Groups) in PROGRAMMING.
6. Program the individual trunks. See page 6-244 (Individual Trunk Programming) in PROGRAMMING. For E&M Trunk programming instructions. If marked for DID service, program the numbers to match the Call Routing Table created above.

F. BASIC RATE INTERFACE (BRI) TRUNKS

NOTE: This requires the Advanced CO Interfaces software license and a BRS or BRU card.

10.51 ISDN BRI trunks enable the system to connect to video, voice, and data devices such as video conference units, ISDN modems, LAN bridges, and ISDN telephones. The BRI cards provide a cost-effective alternative to PRI for large and small systems requiring ISDN services.

10.52 BRI trunks provide the following basic rate features:

- Support BRI stations that connect to data/video devices behind the PBX via BRS card with an S interface and a BRU card with a U interface.
- Support BRI trunks that connect to the network via BRS cards with an NT1 device.
- Support data/video switching from behind the PBX out any U interface trunk (e.g., PRI trunk and BRI trunk) to the public network.

10.53 BRI stations operate like other stations with the following exceptions:

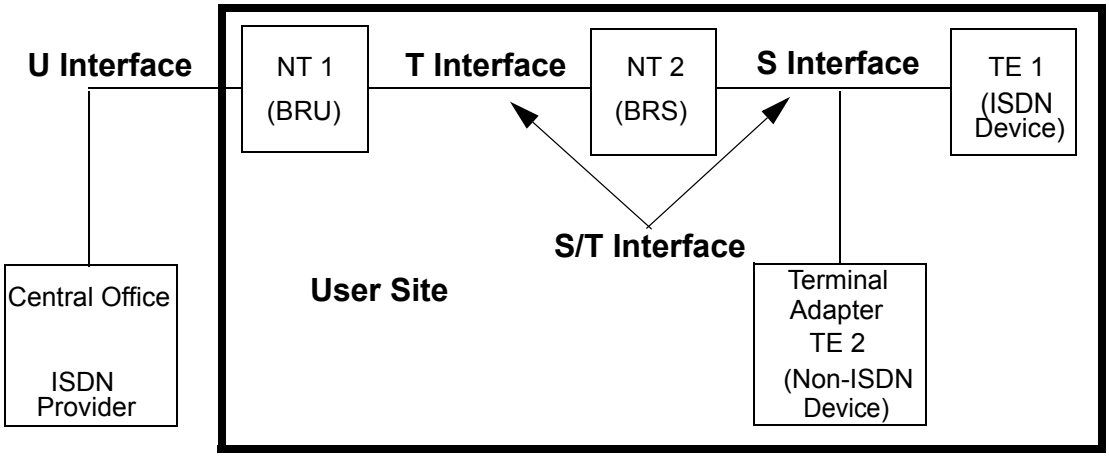
- BRI station ports do not support passive bus configuration (one ISDN device per BRS port)
- A BRI station does not have the ability to perform features while connected on a call that require a hookswitch or SPCL button, including Individual Hold, System Hold, Transfer, Record-A-Call, and Conference.
- The BRI station cannot have a mailbox, but can have an extension ID
- A message cannot be left for a BRI station.
- Centrex options are not supported behind the BRS or BRU card.
- Caller ID [CLIP] information does not include the caller's name (i.e., only the caller's number is available).

10.54 A BRI station cannot place a call on both B-channel stations with a single setup message. The bonding must be performed outside the D-channel signaling. In other words, a video conference unit or data device must place two separate calls and perform the bonding at the external device level. The routing user information provided on the D-channel at this time is call setup and call termination messages only.

Basic Rate Interface Terminology

10.55 The following terms are defined to clarify the functional aspects of the two different types of BRI cards available for the system (see the diagram following the definitions):

BRI ISDN Reference Points



- **Terminal Equipment (TE)**: A general term for a device at the user end of a telephone connection. BRS cards are connected to either a Network Terminator (NT) or a user device. The BRS port is the TE when it is connected to a Network Terminator (NT). If a user device is connected to the BRS port, then that device is the TE.
- **Terminal Equipment Type 1 (TE1)**: An ISDN device that terminates the user end of a telephone connection, such as an ISDN telephone, an ISDN modem, an ISDN LAN bridge, or a video conference unit.
- **Terminal Equipment Type 2 (TE2)**: A non-ISDN device that terminates the user end of a telephone connection, such as an analog telephone, fax machine, or analog modem.
- **Terminal Adaptor (TA)**: A device that terminates the ISDN connection and interfaces to an analog user device, such as an analog telephone, fax machine, or modem.
- **Network Terminator (NT)**: A general term for a device on the customer end of a connection to the Central Office [local exchange] that resides between the user's terminating ISDN device (TE or TA) and the Central Office [local exchange]. For systems with BRS cards, the NT is the BRS port. In the U.S., the NT device is provided by the customer. In other countries this device is often supplied by the ISDN provider.
- **Network Terminator 1 (NT1)**: The primary terminating device that resides between the Central Office [local exchange] and the customer site. The Central Office [local exchange] side connection to this device is called the U interface. The customer side connection is called the T interface. A BRU port has a built-in NT1 on each port. With BRI (U.S. only), the NT1 is normally provided by the local service company or consumer. With PRI, NT1 is not normally provided by the local service company or consumer, instead the ISDN provider or Central Office [local exchange] provides the NT1for the customer location.
- **Network Terminator 2 (NT2)**: This terminating device resides between the primary terminating device (NT1) and the user's ISDN device (TE or TA). The side of the connection that links to the NT1 is called the T interface (see above), and the side that connects to a user device is called the S interface. A BRS port is an NT2 device and so is the T1/E1, T1C, and E1C.

- **U Interface (User Interface)**: A connection coming into the system from an external transmission system (ISDN provider), which is typically a Central Office [local exchange]. (BRI U interface is a two-wire connection and PRI U interface is a four-wire connection.) All BRI ISDN devices with a U interface normally have an NT1 built-in to interface directly to the Network (ISDN provider or Central Office/Local Exchange).
- **T Interface (Trunk Interface)**: A four-wire connection between the NT1 that terminates the transmission coming from the Central Office [local exchange] and the NT2 that connects to the user's TE. This interface is located at the customer site.
- **S Interface (Station Interface)**: A four-wire connection between the NT2 and the user's TE. This interface is located at the customer site.
- **S/T Interface (Station/Trunk Interface)**: Because the BRI Station card (BRS) can be programmed for either a station port (S) or a trunk port (T) the interface, it is referred to as the S/T Interface.

10.56 The following programming is required to use the BRI feature:

1. Make sure the Advanced CO Interface premium feature is licensed. If this feature is not enabled, the warning message "Online Ignored for Board - Check Advanced CO Interfaces" appears in Message Print when the card is installed. (see page 6-97 in PROGRAMMING). If this occurs, upload a software license that supports this premium feature.
2. Choose a slot and equip it for either a BRS or BRU card using System\Cabinets in DB Programming. (see page 6-136 in PROGRAMMING)
3. In DB Programming, equip the ports to be either station or trunk ports. Leave unused ports set to **None**.
4. Program the PRI or BRU ports for either BRI station or BRI trunks depending on the device being installed.
5. Program the established BRI station and trunk ports.
6. Program the individual BRI devices.

NOTE: Refer to the manufacturer's manual for the external device being installed.

7. Program the reference clock information for each of the BRI trunk ports in System\Cabinets\Reference Clock List.
8. Install the BRS cards in the selected slot in the equipment chassis, and connect the external power supply, if required. Also connect the 50-pin Amp connection to the front of the card and terminate the cable to the MDF (Main Distribution Frame). (see page 3-131 in INSTALLATION)
9. Install the BRU card in the selected slot in the equipment chassis, and connect the 50-pin Amp connection to the front of the card, and then terminate the cable to the MDF.
10. Connect the individual BRI devices to the chosen ports.

NOTE: If a BRS or BRU card is installed on a system using CPU 128 or CPU 64, a Network Clock Card (NCC) must also be installed. If the system uses a CPC, however, a NCC is not required.

G. DIRECT INWARD DIALING (DID) TRUNKS (U.S. ONLY)

10.57 Direct Inward Dialing is available on T1 or T1/E1 card circuits and Single-Line Adapter (SLA) interfaces. (See page 2-39 in SPECIFICATIONS for more information.) E&M trunks on T1 or T1/E1 cards can also be designated as DID trunks.

10.58 Direct inward dialing (DID) allows an outside party to dial into the system without attendant intervention. To gain direct access to the system, an outside party dials a number that was assigned by the telephone company to the DID trunks installed on the phone system. The system then provides ring signal to the proper station(s), hunt group, or DISA according to the programmed ring-in for that DID number.

10.59 DID trunks are purchased with blocks of numbers. When a caller dials one of the numbers, the central office rings the trunk that contains that number. When the system answers the call, it handshakes with the central office and receives digits that tell it which DID number was dialed. The system then routes the call according to the call routing programmed for that number.

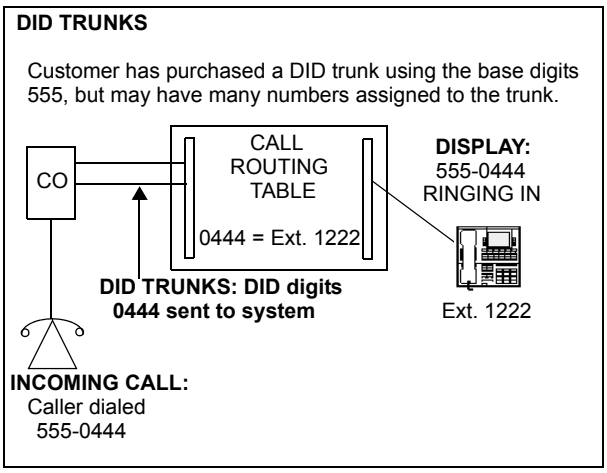
10.60 When a call is placed to a DID trunk, one of three things happens:

- If the receiving DID trunk is programmed for "Immediate" start type, it immediately begins receiving the dialed digits to the receiving system.
- If programmed for "Wink" start type, the DID trunk performs a "handshake" to signal that it is ready to receive the incoming digits. (This is the default start type.)
- If programmed for "Delay Dial" start type, the DID trunk performs a handshake and pause before receiving any digits.

10.61 Each DID trunk is programmed with a base number and is programmed to collect a set number of digits. The base number is made up of the digits that are dialed by the caller that are *not* redialed by the central office after the handshake. For example, if the dialed number is 961-9000 and the base number is 9619, the central office would send only 000 after the handshake. The system would then send the call to the ring-in destination associated with 000 in the trunk's call routing table. The number sent to the phone display could combine the base number and the collected digits to show 961-9000 or can be programmed to show a name. (See page 5-13 for a description of the Display Outside Party Name On/Off feature code.)

10.62 DID trunks are programmed into trunk groups. However, because DID trunks do not allow outgoing calls to be placed, the outgoing access programming for the trunk group is ignored. If a user attempts to seize a DID trunk for an outgoing call, the user hears reorder tones and the phone display shows OUTGOING ACCESS DENIED.

10.63 The DID/E&M [DDI in Europe] Receive Busy Instead Of Camp-On station flag determines whether E&M and DID callers receive busy signal or receive ringback and camp on when calling a busy station. In the default state, busy tones are disabled and the callers hear ringback while camped on to the called station. This flag is programmed on a station-by-station basis (see page 6-210 in PROGRAMMING).



10.64 The following programming is required to use the DID feature:

1. Make sure the Advanced CO Interfaces premium feature is licensed (see page 6-97 in PROGRAMMING).
2. If installing DID on a T1 or T1/E1 card, do the following:
   a. Install and configure a T1 Card, as described on page 6-126 (System Configuration) in PROGRAMMING. Configure the appropriate circuits as "DID" and assign extension numbers to the trunks. (If using an E&M trunk, see the instructions on page 5-66.)
   b. Program the T1 Card as required. See page 6-126 (T1 Programming) in PROGRAMMING.
3. If installing DID using a Single-Line Adapter (SLA), do the following:
   a. Install the SLA on a keyset card circuit and configure the card, as described on page 6-118 (System Configuration) in PROGRAMMING.
   b. Program the station circuit as required. See page 6-200 (Individual Station Programming) in PROGRAMMING.
4. Program the individual trunks. See page 6-244 (Individual Trunk Programming).
5. Create a trunk group that contains the DID trunks. Program the trunk group to ring in to a Call Routing Table. (Note the Call Routing Table number. It is programmed in step 6.) If there are only DID trunks in the trunk group, or the trunks are used only for incoming calls, you can flag the trunk groups as "Incoming Only." See page 6-236 (Trunk Groups) in PROGRAMMING.
6. Program a Call Routing Table to be used by the DID trunk group. See page 6-319 (Call Routing Tables) in PROGRAMMING. Also see page 5-89 for more information regarding Call Routing Tables. Set up the table using the DID numbers to match the programming performed in step 4.
7. Enable call information for phone displays. See Caller ID [CLIP] on the next page for more information.
8. Because DTMF receivers are required to receive DID and ANI information, you should enable additional receivers in OPC programming (see page 6-117 in PROGRAMMING).

H. CALLER ID [CLIP], DNIS, AND ANI

NOTE: These features require the Advanced CO Interfaces software license.

10.65 The Inter-Tel system supports Caller ID [CLIP], ANI, and DNIS. These features provide information about the source of the call.

NOTE: ANI is not supported in Europe.

- **Caller ID [CLIP]**: Caller ID [CLIP] information gives the caller's telephone number and/or name.
- **ANI**: Automatic Numbering Identification (ANI) information identifies the caller's telephone number. The system receives a specified number of digits.
- ***ANI***: This is a form of ANI that does not have a specified number of digits. The system receives a star (*) before the ANI digits to signal the beginning of the caller's telephone number. Then there is another star after the digits to signal the end of the ANI information.
- **DNIS**: Dialed Number Identification Service (DNIS) identifies the number that was dialed to reach your location. The system receives a base number and a specified number of digits that identify the dialed number.
- ***DNIS***: This is a form of DNIS that does not have a specified number of digits. The system receives a star (*) before the DNIS digits to signal the beginning of the dialed number. Then there is another star after the digits to signal the end of the DNIS information.
- ***ANI*DNIS***: This is a two-stage address service that provides both the caller's telephone number and the dialed number. It combines the *ANI* and the *DNIS* features described above.

10.66 The network retains trunk and outside party information and passes that information along when a call is routed from node to node.

10.67 Using these features and the call routing tables, each incoming call can be identified and sent to selected destinations. For example, calls identified with Caller ID [CLIP] or ANI from specific regions of the country can be sent to the appropriate sales representative or calls from specific locations can be sent to selected individuals.

10.68 With a U.S. system, using DNIS information responses to advertisements using one 800 number can be sent to one hunt group and calls from other 800 numbers can be sent to other hunt groups. Or, they can all be sent to the same destination with different identifying names for the phone displays (such as Magazine Ad, TV Ad, etc.).

10.69 Call information services, Caller ID [CLIP], ANI, and DNIS, must be enabled for the individual trunks (see "service type" programming on page 6-253). Loop start trunks on LSCs and LGCs can use Caller ID [CLIP]. E&M and DID trunks can use ANI and DNIS. Caller information is not available on ground start trunks.

10.70 As of version 8.1 [8.2 in Europe], the phone system supports the transmission of Caller ID [CLIP] for central office (CO) calls to single-line phones that terminate on Single-Line Cards (SLCs or SLC16s). This feature uses Automatic Number Identification (ANI) or Caller ID [CLIP]. After programmed, on-hook single-line phones display the calling party's phone number and name (if available) when receiving an incoming CO call. The Caller ID [CLIP] information appears after the first ring. The Caller ID [CLIP] information is also displayed if the single-line phone receives a transferred call from another station that has calling party information.

10.71 In v9.0 and later, the system supports Caller ID [CLIP] for intercom (IC) calls on a single-line device. When a single-line device receives an incoming call, if the system has adequate caller ID resources, the single-line device displays the calling party's username and IC extension. This feature allows the single-line device to use the caller ID information to perform certain functions. This feature is supported by single-line endpoints including telephones and fax machines connected to a Single Line Card (SLC) or SLC-16 in the U.S., or an SLC-16+ in the U.K. The single-line device connected to the SLC must support caller ID or Calling Line Identity Presentation [CLIP] as defined by current industry-standard specifications. This feature is not supported on Single-Line Adapters (SLAs).

NOTE: There may be some third-party devices that are supposed to support Caller ID, but actually do not. Because an extension is shorter than a 10-digit number that the device would normally receive, the device may not be capable of receiving the extension-based Caller ID information. Also, some third-party devices may display the extension in a telephone number format rather than as an extension number. For example, an extension of 11023 may appear as "110-23," or "- 1-1023," etc.

10.72 Because Caller ID [CLIP] transmitters are required to send the information to the single-line phone, you must upgrade the firmware on an Options Card (OPC) or 16-port Digital Keyset Card (DKSC-16+) that has an available digital signal processor (DSP). Each DSP on a DKSC-16+ provides eight (four if using version 8.1) Caller ID [CLIP] transmitters, and each DSP on an OPC provides six transmitters.

NOTE: For U.K. installations only, as of version 8.224 (U.K. v8.225), each DSP on the OPC provides an additional six caller ID transmitters. If necessary, you can upgrade the firmware on an existing OPC with the A-Law OPC Upgrade Kit (part no. 828.1671).

10.73 Single-line phone Caller ID [CLIP] requires the following:

- Calling party information services from the CO
- Advanced CO Interfaces software license
- Latest firmware on the DSPs that provide the Caller ID [CLIP] transmitters
- CO trunks programmed for Caller ID [CLIP]
- Installed and programmed single-line phones that terminate on an SLC, SLC-16, or SLC-16+
- Caller ID - CO/IC flags enabled in DB Studio
- Extended Ring Cadence flag enabled

Phone Displays

10.74 Whether call information appears on the phone display, and what information appears, is determined by station flags in the database. The flags are:

- **Expanded CO Call Information On Displays**: This station flag determines whether call information (trunk name or call information) appears at the phone. If it is enabled, the Outside Call Party Information Has Priority flag (described on the next page) determines what appears on the display. If it is disabled, the programmed trunk group username appears on the display. In the default state, this flag is *enabled.*
- **Outside Party Call information Has Priority**: If the Expanded CO Call Information flag is enabled (as described on the previous page), this flag determines what information appears on the display. If enabled, any call that is received on a trunk that provides outside call information, for example, Caller ID [CLIP] or ANI, is identified on the phones display with the call information. If disabled, the display shows the DID or DNIS [DDI in Europe] information for the call (if available). In the default state, is it *enabled.*
- **Display Outside Party Name**: If the Display Outside Party Name station flag is enabled, the phone user can switch between the outside party name and number when connected to a CO call with outside party information. The phone user enters the Display Outside Party Name On/Off feature code (379). In addition, the enhanced ring-in displays provide the user with more information such as both Caller ID [CLIP] name and number if available, or tell the user if a Caller ID [CLIP] number is blocked or out-of-area. This is a phone-only station flag. In the default state, this flag is enabled.

10.75 The system provides the user with advanced displays for direct ring-in calls, if the station has “Expanded CO Call Information On Displays” and “Outside Party Call Information Has Priority” station flags enabled (by default they are enabled).

10.76 A typical direct ring in display would look like the following: “TRNK GRP 1 RINGING IN.” However, with the enhanced displays, the system attempts to display the name of the outside caller on line 1 and the number of the outside caller on line 2.

10.77 If the user has a direct ring in from a trunk or DID to an extension, the display shows both the name and number. If a trunk rings in to a CRA, then is transferred, the display shows the name or number on the first line, and shows where the call is transferred from on the second line (which is where the number would show if it were a direct ring). The user can then use the feature code 379 to toggle between name and number.

10.78 When a CO call rings into the system *(in the U.S. only)*, the system uses the following criteria to determine the name that appears on the display phones. The criteria are listed from highest priority to lowest priority.

Display Line 1

10.79 The following criteria are used to display the name on the top line of the ring-in display. This assumes that all of the necessary station flags are properly set and usernames are correctly filled in or left blank (at the programmer’s discretion) for Call Routing Tables and Trunk Group Names.

1. Outside Party Name provided by the Desktop Application, if available.
2. Outside Party Name provided by Caller ID, if enabled at the station level.
3. Outside Party Name provided by system speed-dial. If the collected number matches a number in a speed-dial bin, the system uses the name of the corresponding speed-dial bin.
4. Outside Party Number provided by the Desktop Application.

5. Outside Party Number provided by Caller ID, if enabled at the station level.
6. Outside Party Number provided by ANI service, if enabled at the station level.
7. Call Routing Table Name, if the name is not blank.
8. Trunk Group Name, if the name is not blank.
9. Default Trunk Group Name (TG XXXXX).

Display Line 2

10.80 The following criteria are used to display the number on the bottom line of the ring-in display. This assumes that all of the necessary flags are properly set.

1. Outside Party Number provided by the Desktop Application, if available.
2. Outside Party Number provided by Caller ID, if enabled at the station level.
3. Outside Party Number provided by ANI service, if enabled at the station level.
4. Number Absence Reason, such as Caller ID "OUT-OF-AREA" or "PRIVATE" (blocked) message.
5. RINGING IN display.

10.81 ANI and Caller ID information is also sent to the Voice Mail application to provide the caller's telephone number as part of the message envelope *(in the U.S. only)*.

10.82 Caller Information features require the following special programming *(in the U.S. only)*:

- The Advanced CO Interfaces feature license is required.
- Caller information features require OPC programming (see page 6-117 in PROGRAMMING). Caller ID uses Caller ID receivers. Because DTMF receivers are required to receive DID and ANI information, you should enable additional receivers.
- Caller information display station flags must be enabled in DB Programming (see page 6-211 in PROGRAMMING).
- The call information services must be enabled for the individual trunks (see page 6-253 in PROGRAMMING).

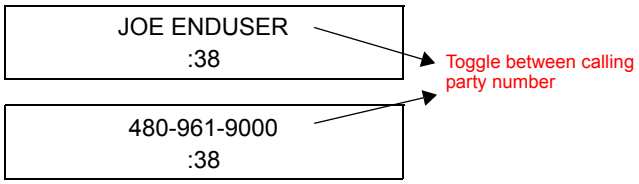
Display Caller ID Name and Number Enhancement

10.83 Prior to v10.0, the system allowed the endpoint to display only the incoming CO caller's name or number on the first line and the call time elapsed on the second line. There is a station flag named "Display Outside Party Name" that the endpoint user could access to toggle between showing the caller's name or number on the first line of the display.

10.84 System software 10.0 or later includes a system flag named "Display Caller ID Name and Number" in DB Programming, which allows you to adjust the display on an endpoint for an incoming CO call to always show caller ID name and number (see page 6-260). When this flag is enabled, the endpoint user does not see the elapsed call timer that shows the duration of the call. When this flag is enabled, the outside caller's name appears on the first line and the caller's number appears on the second line of the endpoint display. The following example shows how the display appears when this flag is disabled and when it is enabled.

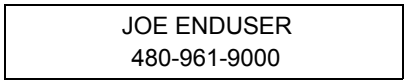
ENDPOINT DISPLAY FOR CALLER ID v9.1 AND EARLIER

"Display Outside Party Name" station flag is enabled



ENDPOINT DISPLAY FOR CALLER ID V10.0 AND LATER

"Display Caller ID Name and Number" system flag is enabled



Caller ID Forwarding

10.85 Caller ID forwarding allows Call Processing (CP) to send Caller ID information from any incoming services to the public or private network using Primary Rate Interface (PRI) or Basic Rate Interface (BRI) Integrated Services Digital Network (ISDN) circuits. In the European market Caller ID is known as Calling Line Identification [CLID]. When Caller ID is enabled, the system includes information that identifies the caller to the public or private network.

10.86 The Caller ID information can be an endpoint user name or extension, caller number (calling party number or name), or incoming Caller ID information. The system software can process any incoming caller ID information and then resend this information through an ISDN circuit with some limitations. In other words, caller ID received through an ISDN, analog line, or T1 circuit can be processed and sent out to an ISDN circuit. Caller ID information can also be forwarded between nodes.

10.87 Caller ID information presented by the system results from any one of the following events:

- A call that is originated from an endpoint on the system.
- A call that is redirected from an external call.
- The system programmable fields in DB Programming under System\Devices and Feature Codes\CO Trunk groups\<*trunk group*>:
  - Send Station Caller ID to Attached PBX (see page 6-239)
  - Propagate Original Caller ID (see page 6-240)
  - Wait for ISDN Caller ID Information (see page 6-241)

10.88 The following fields have been added to DB Programming under System\Devices and Feature Codes\Stations\<*station*> to support Caller ID Forwarding.

- Calling Party Name (see page 6-207)
- Propagate Original Caller ID on Transfer Flag (see page 6-212)

10.89 The Propagate Original Caller ID on Transfer flag (for voice mail) has been added in Voice Processor\Devices\Applications\<*Voice Mail application*>. See page 6-369.

10.90 See page 7-36 for troubleshooting information.

Caller ID for Line Type

10.91 Based on the caller ID settings, the following calls are supported with the processing of the Caller ID number or name.

Table 5-14. *Caller ID for Line Type*

Incoming Call Type	Outgoing Call Type	The System Sends:
PRI	PRI	Caller ID name and number
	BRI	Caller ID number only[1]
BRI	PRI	Caller ID number only[1]
	BRI	Caller ID number only[1]
T1 Automatic Number Identification (ANI)	PRI	Caller ID number only[2]
	BRI	Caller ID number only[1]

Table 5-14. *Caller ID for Line Type (Continued)*

Incoming Call Type	Outgoing Call Type	The System Sends:
T1 Loop Start with Caller ID	PRI	Caller ID name and number
	BRI	Caller ID number only[1]
Analog Loop Start with Caller ID	PRI	Caller ID name and number
	BRI	Caller ID number only[1]

1. BRI only supports caller ID number and not name. Caller ID name cannot be received or sent on a BRI circuit.

2. ANI is the same as Caller ID number or Calling Number. If the ANI number is received, this can be processed on an outgoing ISDN circuit.

10.92 When the caller ID name is sent to the public telephone network, the name may be removed or replaced by the CO.

10.93 Incoming calls that have their caller ID information blocked are not sent by the system. Instead, the system passes the generic block name (Private, Restricted, Out-Of-Area, or Blocked). The system can only send caller ID information it receives. If the incoming call does not provide both name and number, the information sent only includes what was received.

NOTE: When the caller ID name is sent to the public telephone network, the name may be removed or replaced by the CO.

Network Considerations

10.94 Keep the following considerations in mind for networked systems:

- An intercom call that routes between nodes then out through an ISDN line that is using a Display message does not send the endpoint's user name for the caller ID name, but sends the endpoint's calling party number instead.
- An intercom call that routes between nodes then out through an ISDN that is using a Facility message sends the endpoint's user name for the caller ID name.

10.95 The reason for the user name condition above is for network compatibility for older systems. The user name information between nodes is not sent immediately.

Caller ID Forwarding Scenarios

10.96 The following scenarios describe how incoming and outgoing calls are connected through a PRI to another PBX using the Caller ID Forwarding feature.

10.97 Scenario #1: An extension or application places an outbound call.

Table 5-15. *Caller ID Forwarding for Outbound Calls*

Type of Call	When:	Is Set to:	The System Sends:
Outbound External Call	Calling Party Number and/or Name	No entry	No Caller ID information
	Propagate Original Caller ID	--[1]	
	Send STN Caller ID to Attached PBX	No	
	Propagate Original Caller ID on Transfer	--[1]	
	Calling Party Number and/or Name	an outside telephone number and/or name	Calling Party Number and/or Name
	Propagate Original Caller ID	--[1]	
	Send STN Caller ID to Attached PBX	No	
	Propagate Original Caller ID on Transfer	--[1]	
	Calling Party Number and/or Name	No entry or an outside telephone number/ name[1]	User name and extension from the calling extension
	Propagate Original Caller ID	--[1]	
	Send STN Caller ID to Attached PBX	Yes	
	Propagate Original Caller ID on Transfer	--[1]	

1. This flag or entry has no affect on the situation because the other flags that are enabled have priority, the call condition does not apply, or caller information is not provided.

10.98 Scenario #2: An extension transfers an inbound or outbound call to an outside number or to an extension that is forwarded to an outside number.

Table 5-16. *Caller ID Forwarding for Non-Voice Mail Transfer Calls*

Type of Call	When:	Is Set to:	The System Sends:
External Call Transfer	Calling Party Number and/or Name	No entry	No Caller ID information
	Propagate Original Caller ID	No	
	Send STN Caller ID to Attached PBX	No	
	Propagate Original Caller ID on Transfer	No	
	Calling Party Number and/or Name	an outside telephone number and/or name	Calling Party Number and/or Name
	Propagate Original Caller ID	--[1]	
	Send STN Caller ID to Attached PBX	No	
	Propagate Original Caller ID on Transfer	No	
	Calling Party Number and/or Name	No entry or an outside telephone number/name[1]	User name and Extension from the calling extension. Calling Party Number is not sent
	Propagate Original Caller ID	--[1]	
	Send STN Caller ID to Attached PBX	Yes	
	Propagate Original Caller ID on Transfer	--[1]	
	Calling Party Number and/or Name	No entry or an outside telephone number/name[1]	Original incoming Caller ID information to the external call
	Propagate Original Caller ID	Yes	
	Send STN Caller ID to Attached PBX	No	
	Propagate Original Caller ID on Transfer	--[1]	

1. This flag or entry has no affect on the situation because the other flags that are enabled have priority, the call condition does not apply, or caller information is not provided.

10.99 Scenario #3: A Voice Mail application transfers an inbound call to an extension that is forwarded to an outside number.

Table 5-17. *Caller ID Forwarding for Voice Mail Transfer Calls*

Type of Call	When:	Is Set to:	The System Sends:
Inbound External Call	Calling Party Number and/or Name	No entry or an outside telephone number/ name[1]	User name and Extension from the calling extension. Calling Party Number is not sent.
	Propagate Original Caller ID	--[2]	
	Send STN Caller ID to Attached PBX	Yes	
	Propagate Original Caller ID on Transfer	--[2]	
	Calling Party Number and/or Name	No entry or an outside telephone number/ name[1]	Calling Party Number and/or Name
	Propagate Original Caller ID	Yes	
	Send STN Caller ID to Attached PBX	No	
	Propagate Original Caller ID on Transfer	No	
	Calling Party Number and/or Name	No entry or an outside telephone number/ name[1]	User name and extension from the calling extension
	Propagate Original Caller ID	Yes	
	Send STN Caller ID to Attached PBX	No	
	Propagate Original Caller ID on Transfer	Yes	

1. This field does not apply to this type of device. Voice Mail applications do not have Calling Party Number field.
2. This flag or entry has no affect on the situation because the other flags that are enabled have priority, the call condition does not apply, or caller information is not provided.

10.100Scenario #4: Inbound call is immediately routed to an outside number using Single Ring-in Destination, Station Forwarding, and OAI Deflect.

Table 5-18. *Caller ID Forwarding for Incoming Unanswered Calls*

Type of Call	When:	Is Set to:	The System Sends:
Inbound External Call	Calling Party Number and/or Name	No entry or an outside telephone number/ name[1]	No Caller ID information
	Propagate Original Caller ID	No	
	Send STN Caller ID to Attached PBX	--[2]	
	Propagate Original Caller ID on Transfer	--[2]	
	Calling Party Number and/or Name	No entry or an outside telephone number/ name[1]	Original incoming Caller ID information to the external call
	Propagate Original Caller ID	Yes	
	Send STN Caller ID to Attached PBX	--[2]	
	Propagate Original Caller ID on Transfer	--[2]	

1. The Calling Party Field is only sent if the endpoint places an outbound call or if an endpoint intercoms another extension that is forwarded to an outside number. Incoming calls routed to an endpoint that are forwarded or processed by OAI from an endpoint to an outside number do not provide Calling Party Number. The Calling Party Number is not sent if an incoming unanswered call immediately routes to an outside number through Call Routing Table or directly to a trunk group or through station forward of any type.
2. This flag or entry has no affect on the situation because the other flags that are enabled have priority, the call condition does not apply, or caller information is not provided.

Scenario #5: An extension calls another extension that is routed to an outside number through station forward, OAI, or Send to Destination. Same Node or remote Node intercom calls are processed the same.

Table 5-19. *Caller ID Forwarding for Intercom Calls*

Type of Call	When:	Is Set to:	The System Sends:
Intercom Call	Calling Party Number and/or Name	No entry	No Caller ID information
	Propagate Original Caller ID	--[1]	
	Send STN Caller ID to Attached PBX	No	
	Propagate Original Caller ID on Transfer	--[1]	
	Calling Party Number and/or Name	An outside telephone number/ name	See footnote [2]
	Propagate Original Caller ID	--[1]	
	Send STN Caller ID to Attached PBX	No	
	Propagate Original Caller ID on Transfer	--[1]	
	Calling Party Number and/or Name	No entry or an outside telephone number/ name[1]	User name and extension from the calling extension. Calling Party Number is not sent.
	Propagate Original Caller ID	--[1]	
	Send STN Caller ID to Attached PBX	Yes	
	Propagate Original Caller ID on Transfer	--[1]	

1. This flag or entry has no affect on the situation because the other flags that are enabled have priority, the call condition does not apply, or caller information is not provided.
2. The Calling Party Field is only sent if the endpoint places an outbound call or if an endpoint intercoms another extension that is forwarded to an outside number. Incoming calls routed to an endpoint that are forwarded or processed by OAI from an endpoint to an outside number do not provide Calling Party Number. The Calling Party Number is not sent if an incoming unanswered call immediately routes to an outside number through Call Routing Table or directly to a trunk group or through station forward of any type.

Caller ID Enhancements

10.101 Because some COs are now requiring that calls have Caller ID set, this release adds the "Calling Party Number" and "Calling Party Name" fields to the following settings in DB Programming.

- **Voice Mail**: Allows subscribers to specify the Caller ID [CLID] to use when the voice mail makes an outgoing call. This field is located in Voice Processor\Devices\Applications\<*application*> (see page 6-369).
- **Trunk Group**: Allows Caller ID information to be used on outbound calls through trunk groups. This field is located in System\Devices and Feature Codes\CO Trunk Groups\<*trunk group*> (see page 6-207).

10.102 If the Caller ID is not set on the call, the CO rejects the call when it comes from voice mail, or the called party may not receive incoming Caller ID on redirected calls.

10.103 This release also adds the following fields:

- **Emergency Calling Party Number** under System\Devices and Feature Codes\Stations\<*station*> (see page 6-209).
- **Force Trunk Group Calling Party Name and Number** under System\Devices and Feature Codes\CO Trunk groups\<*trunk group*> (see page 6-241).

10.104 For node networking, if the connected nodes are running Axxess system software earlier than v11.0, the older version nodes ignore all the new fields and function as it does currently.

Call Processing Forwarding Priorities

10.105 The Call Processing forwarding priorities are designed to provide maximum flexibility without burdening the calls that do not require Caller ID.

10.106 If the Emergency Calling Party Number is configured, Call Processing uses the Emergency Calling Party Number as the outgoing ISDN Caller ID [CLID]. If the Emergency Calling Party Number is not configured, Call Processing uses the station Calling Party Number, trunk group Calling Party Number, or station extension depending on the "Propagate ID on Transfer" and "Send Station Caller ID to Attend PBX" flags. See "Caller ID Forwarding" on page 77 for details about these fields. See Figure 5-1 on page 5-85 for station level Caller ID Forwarding call flow.

10.107 If the voice mail Caller ID [CLID] is configured, Call Processing uses the voice mail Caller ID [CLID] when the Message Notification Retrieval application makes an outgoing call. When a Call Routing Application (CRA) makes an outgoing call, Call Processing forwards the Caller ID [CLID] based on the "Propagate Original Caller ID on Transfer" flag. See "Caller ID Forwarding" on page 77 for details about this field. See Figure 5-2 on page 5-86 for voice mail level Caller ID Forwarding call flow.

10.108 The trunk group Calling Party Number and Calling Party Name are used only if no other Calling Party Number can be determined. Call Processing uses the Calling Party Number and Calling Party Name that is defined in the trunk group only if the original Caller ID [CLID], station, or voice mail level Caller ID [CLID] information is not set. The trunk group Caller ID [CLID] has the least precedence than other Caller IDs. The following flowcharts show how Call Processing forwards the Caller ID [CLID] based on the configuration.

Figure 5-1. *Station Level Caller ID Forwarding Flowchart*

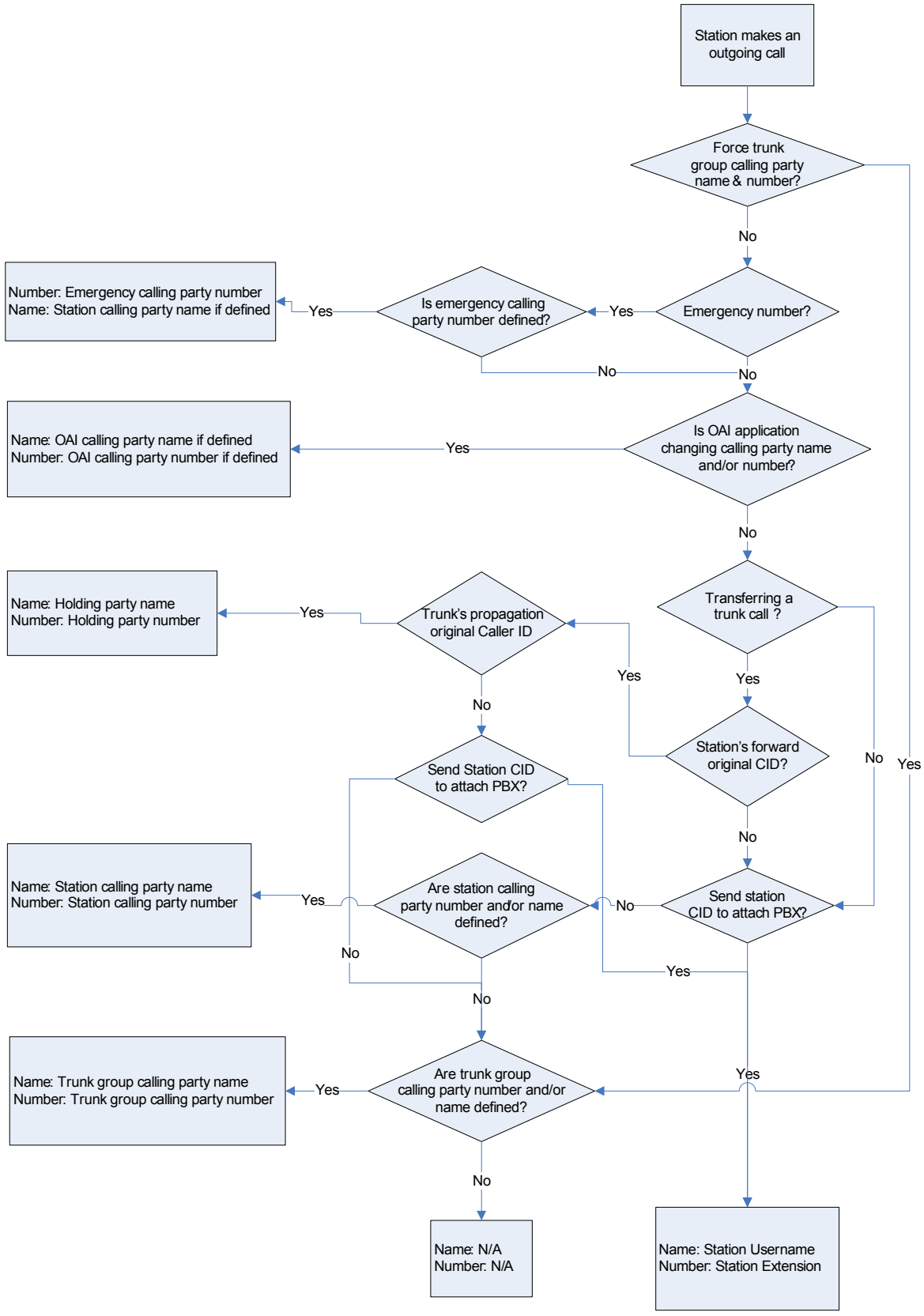
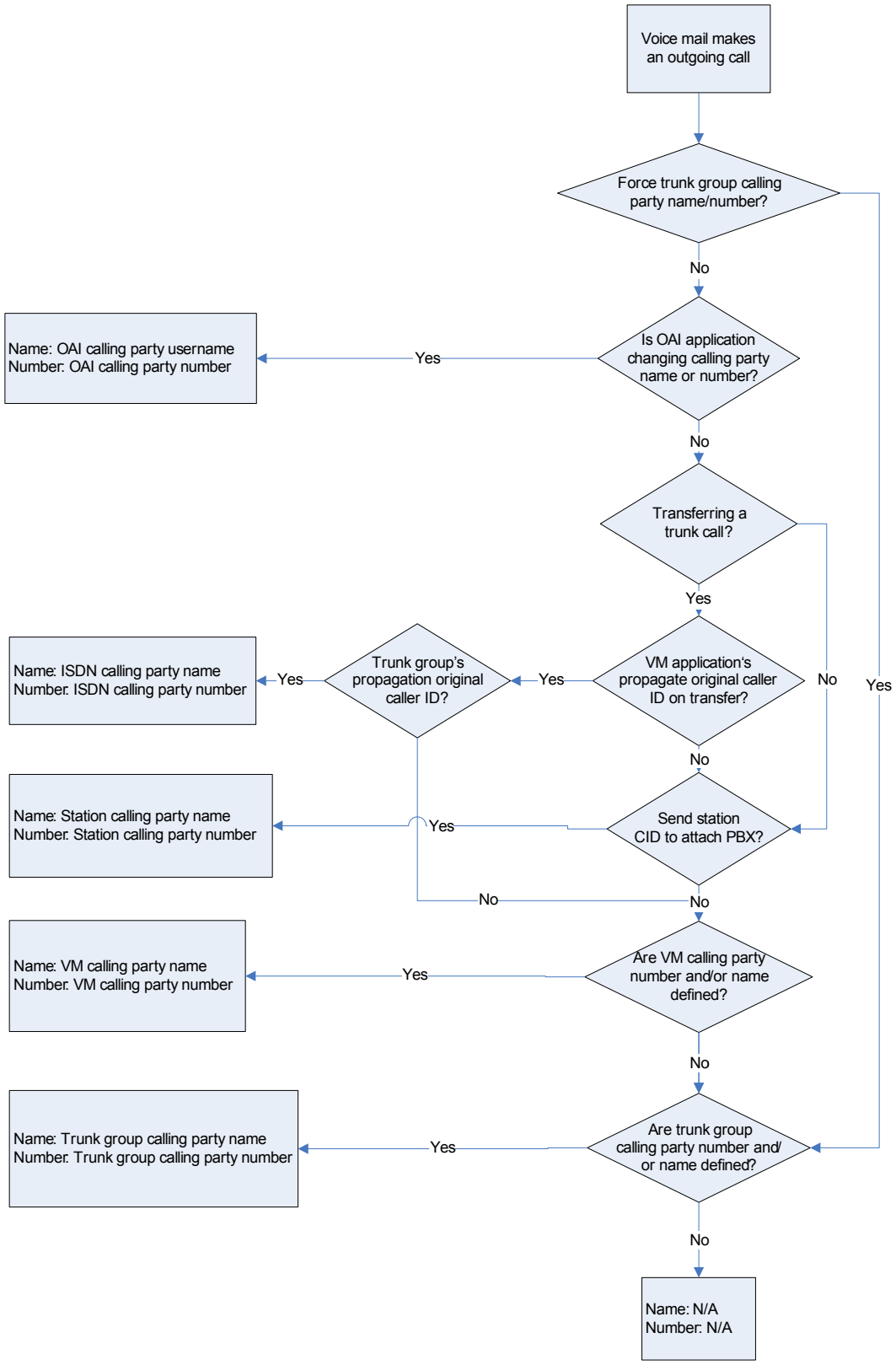


Figure 5-2. *Voice Mail Level Caller ID Forwarding Flowchart*



I. TRUNK GROUPS

10.109 Each trunk is assigned to one of 104 possible trunk groups (208 in a triple- or quad-chassis system). Trunk group feature codes and trunk group buttons are used to select a trunk in one of the programmed trunk groups. Each trunk must be assigned to a trunk group. For example, all local trunks could be in one group, while another group could contain WATS trunks that are used for long distance calling. Unused trunks can be placed in a single trunk group that is labeled "unused."

10.110 All trunks in a trunk group must reside on the same node as the trunk group.

10.111 The trunk group is programmed to process outgoing calls in linear or distributed order:

- **Linear Trunk Group**: Requests for an outgoing trunk are always processed beginning with the highest numbered trunk circuit on the list and move through the list until an available trunk is found.
- **Distributed Trunk Group**: The first request is processed beginning with the highest numbered trunk circuit on the list. The next request begins with the second highest numbered trunk, and each subsequent request begins one trunk lower on the list. When the end of the list is reached, requests begin again with the highest numbered trunk on the list.

10.112 The camp-on feature can be enabled or disabled for each trunk group. If the camp-on feature is disabled, callers hear continuous busy signals when all trunks in the trunk group are in use or unavailable. If it is enabled, callers can camp on and wait for an available trunk. See page 6-237 for programming information.

J. NODES AND NODE TRUNK GROUPS

10.113 **Node trunk groups**: Are made up of PRI circuits that are programmed for private network use. When a PRI-equipped card is programmed for networking (Private Networking switch type), the system automatically creates a node trunk group. (All B-Channels residing on the same T1/PRI or E1/PRI card are in the same node trunk group.)

10.114 **Node programming**: Contains a list of node trunk groups used to access another node. For each node in the network, you must define the routes to every other node. For example, in a network with four nodes, you would define three routes for each node (one to each of the other three nodes). For details, see page 6-243 in PROGRAMMING and the information in Appendix A — Networking.

10.115 The nodes are programmed to select their node trunk groups in linear or distributed order:

- **Linear**: The node first attempts to route through the first node trunk group listed in the node. If the node is unable to route through that node trunk group, it attempts to route through the second node trunk group listed in the route. The node continues to attempt to route through subsequent node trunk groups in the list until it successfully routes or exhausts all node trunk groups in the list.
- **Distributed**: The node shifts the first node trunk group it attempts to use. For example, if the node routed the previous call or communication through the first node trunk group listed, the node routes the second call or communication through the second node trunk group listed.

10.116 To prevent circular or excessive routes, the system limits the number of "hops." A call or communication can take up to 10 hops to other nodes. (However, 2-3 hops maximum is recommended.)

K. OUTGOING-ACCESS, ALLOWED-ANSWER, AND RING-IN ASSIGNMENTS

10.117 Each trunk group has programmed lists of stations for outgoing-access, allowed-answer, and ring-in assignments for day and night modes.

- **Outgoing-Access**: Permits the station user to place calls using trunks in that trunk group. Each station has a default outgoing access code programmed in the database. When the user presses the OUTGOING button, presses an idle CALL button, or enters the Outgoing Call feature code (8), the system automatically selects an outgoing trunk. (See page 6-232, in PROGRAMMING, for more information.)

  NOTE: Because the network does not allow users to directly access trunks on other nodes, each trunk group's Outgoing Access can contain only stations on the local node. (Users must use ARS to access trunks on other nodes.)

- **Allowed-Answer**: Permits the station user to answer incoming calls on the trunks in that trunk group (even if the station does not have ring-in assignment for that trunk group). Stations cannot have allowed-answer assignment for trunk groups on other nodes. On phones, the individual trunk button flashes (if one exists) to indicate the ringing call. If a station is programmed with allowed-answer access only (no ring-in) for a trunk group, direct ring-in calls on that trunk group can be answered by entering the Automatic Trunk Answer feature code (350) or pressing the flashing individual trunk button.

- **Ring-In**: Assigns ring-in destinations which can be one or more stations, modems, extension lists, or Voice Processing applications; to a hunt group; to DISA; or to a call routing table. A trunk can ring in to a station or Voice Processing application on another node. The ring in destination(s) for the trunk group receive direct ring-in calls on trunks in that trunk group. On phones with ring in, the individual trunk or CALL button flashes and the display indicates a ringing call. Allowed answer for the trunk group is automatically assigned to a station with ring in.

10.118 When phones have outgoing-access or ring-in assignments, the associated individual trunk buttons show the status of their trunks.

10.119 Stations that do not appear on any of the lists cannot place or directly receive outside calls; they are limited to intercom calls, conferences, transferred calls, and retrieving calls on system hold. (A call on system hold can only be picked up at the station that placed it on hold or at a station that has an individual trunk button and has allowed-answer and/or outgoing access for that trunk.)

10.120 A private trunk group (with one or more trunks) can be established by programming outgoing-access, ring-in, and allowed-answer permission for the trunk group to only one station.

L. ANSWER FEATURE CODE

10.121 For calls that are ringing or holding at the station, the user may enter the Answer feature code (351) or press the ANSWER button. When more than one call is ringing or holding, the following priority list determines which call is answered first:

- Ringing calls (ring ins, recalls, callbacks, or transfers) are answered in the order they were received.
- Calls on individual hold are answered in the order they were placed on hold.

M. CALL ROUTING TABLES

NOTE: The Advanced CO Interfaces software license is required to use Call Routing Tables.

10.122A trunk group can be programmed to ring in to a call routing table to utilize the information received from DID DNIS, ANI, and Caller ID [DDI and CLIP in Europe]. This added information allows the station user to receive information about the callers (such as location, name, or which advertisement they saw) on the phone display. (See page 6-237 for programming information.)

10.123 Call routing tables include the following:

- **DID** *(U.S. Only)*: Sends a string of digits to indicate the desired destination and includes E&M trunks. Each DID digit string can be programmed to ring in to a different destination and display a different name.
- **DDI** *(Europe Only)*: Sends a string of digits to indicate the desired destination. Each DDI digit string can be programmed to ring in to a different destination and display a different name.
- **ANI/Caller ID [CLIP in Europe]**: Identifies the source of the call. Each incoming call can be identified and sent to selected destinations. For example, calls from specific regions of the country can be sent to the appropriate sales representative or calls from specific telephone numbers can be sent to selected individuals.
- **DNIS** *(U.S. Only)*: Identifies the number that was dialed to reach your location. Using this number, the call routing table can send the call to the desired destination. For example, responses to advertisements using one 800 number can be sent to one hunt group and calls from other 800 numbers can be sent to other hunt groups. Or, they can all be sent to the same destination with different identifying names for the phone displays (such as Magazine Ad, TV Ad, etc.).

10.124 A call routing table consists of fields for dialing patterns strings, ring-in types, ring-in destinations, and names. The fields are used for identifying and analyzing the source of the call and directing it to the proper destination.

- **Call Routing Key**: Determines which information the call routing table uses to direct the call: Trunk Number or Outside Party Number. If you select Trunk Number, the system looks for the dialed number provided by DID or DNIS [DDI in Europe]. If you choose **Outside Party Number**, the system checks for the ANI or Caller ID [CLIP] information that identifies the source of the call.
- **Pattern Strings**: Determines call routing based on a comparison of the pattern strings in the table against the digits received from the trunk interface for the incoming call. If the digits match one of the pattern strings in the table, the call is routed according to the ring-in type and destination fields associated with that pattern. If a match is not found, the call is sent to the primary attendant. However, there are two wildcard patterns that allow any number (+) or an empty pattern to match (E).
- **Ring-In Type and Destination**: Identifies the call routing destination and ring-in type. The call can ring in to a single extension number (station, hunt group, Voice Processing application, individual trunk, trunk group, modem, or ARS), an extension list, another call routing table, a destination based on collected digits, or as a DISA call (which allows the caller to select a destination).

NOTE: When programming a call routing table that rings in to another call routing table, make sure the two call routing tables do not send calls to each other, creating a "loop."

- **Name**: Determines how the call is identified at display phones. It can contain up to 12 characters. The name can be a word or number that identifies the call source. If the name is left blank, the system uses the priority list shown on page 5-72 to determine what appears on the phone.

10.125 The system can have up to 15 call routing tables. There can be a total of 900 patterns (more if using a CP Server) in all of the call routing tables combined. To program a call routing table, see the information on page 6-319 in PROGRAMMING.

NOTE: When programming Call Routing Table (CRT), if you are running system software v9.0 or later you can copy one or more patterns to a target Call Routing Table (see page 6-320). If the target CRT has a list of patterns, the copied patterns are added to the end of the existing list.

10.126(*For the U.S. only*) Each individual trunk, not the trunk group, is programmed to collect digits using Caller ID [CLIP], DID, ANI, or DNIS. If the trunk is programmed to collect digits, but the trunk group does not use call routing tables, the system routes the call based on the trunk group programming and ignores the collected digits. (However, the collected digits appear in SMDR and can be used by the Desktop Interface.) If the trunk is not set up to collect digits and the trunk group uses call routing tables, the system uses the call routing destination for "no-digit" calls.

Call Routing on a Network

10.127 Call routing operates as follows on an Inter-Tel phone system network:

- You cannot route a call from one Call Routing Table to a Call Routing Table on another node. When a Call Routing Table sends a call to another node, it cannot be routed to any additional nodes.
- Trunks that ring in to call routing tables can only ring in to tables that are located on their local node. They cannot ring in to a call routing table on another node.
- The network attempts to optimize routes so that calls do not route through a node more than one time (looping). However, the network does not have the ability to dynamically reroute. That is, if a node trunk group that was previously unavailable becomes available after the network has already chosen another route, the network does not reroute using that node trunk group.
- The network tries to optimize routes when calls have been transferred or reverse transferred across several nodes (possibly forming loops).

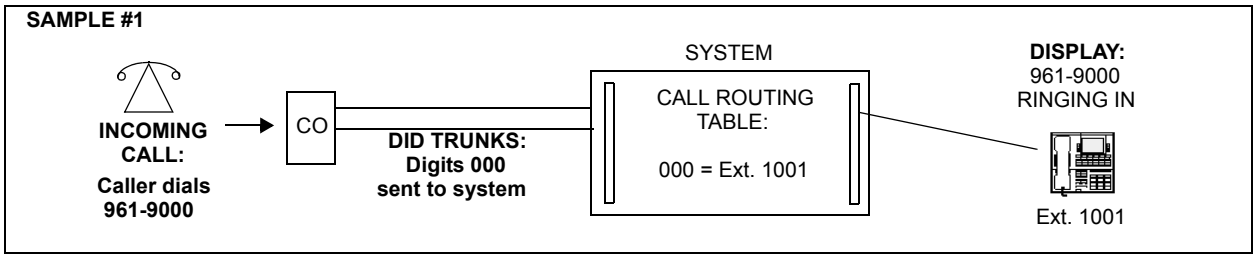
Sample Call Routing Tables

10.128 Several sample call routing tables for U.S. and European systems are shown on the following pages. (Actual tables would have more than the sample entries; these are abbreviated lists to show how the tables work.)

Table 5-20. *Sample Call Routing Tables (U.S.)*

Sample #1: This is an example of a DID routing table. (An actual DID table would have more than four entries; this is an abbreviated list.) Each DID number is identified by the three digits that are sent by the central office (000-003). The base number of the DID numbers is 9619. Therefore, the names of each of the patterns combine the base number and the additional three digits to form the complete number for display (961-9000 to 961-9003).

Pattern	Ring-In Type	Name	Ring-In Destination
000	Single	961-9000	Extension Number 1001
001	DISA	961-9001	Not Applicable
002	Extension List	961-9002	Extension List P004
003	Single	961-9003	Hunt Group Number 2550



Sample #2: This is an ANI table for various area codes. If the incoming ANI information begins with one of the indicated area codes, it is routed to the designated destination and the display shows the state where the call was placed.

Pattern	Ring-In Type	Name	Ring-In Destination
602+	Single	Arizona	Extension Number 2001
505+	DISA	New Mexico	Not Applicable
801+	Extension List	Utah	Extension List P005
904+	Single	Florida	Hunt Group Number 2560

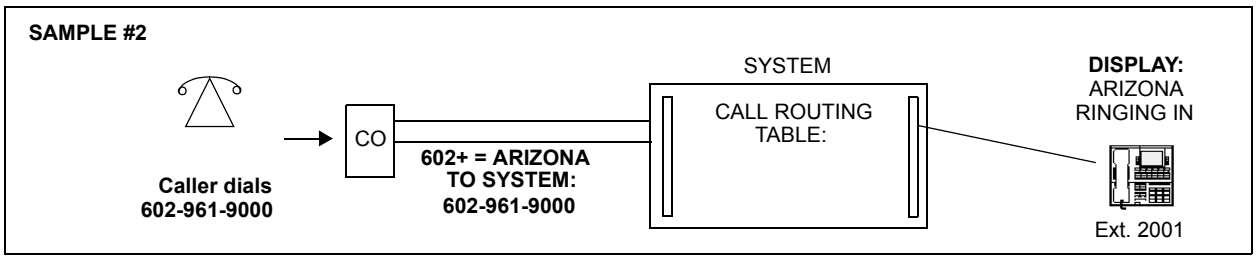
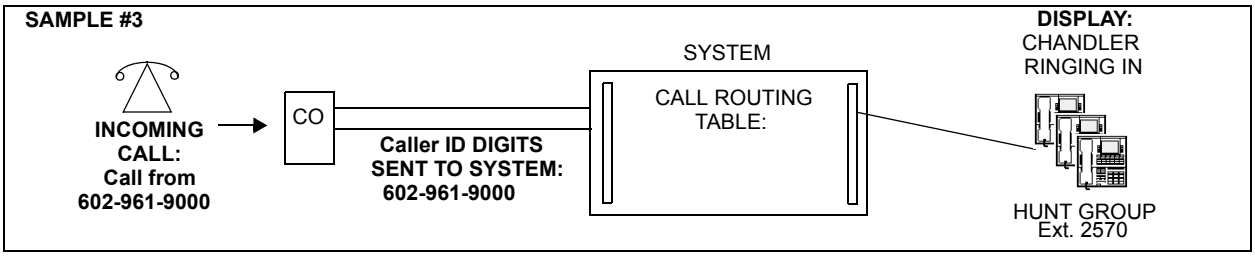


Table 5-20. *Sample Call Routing Tables for the U.S. (Continued)*

Sample #3: This table is set up for Caller ID. It is like the ANI table except that it uses office codes instead of just area codes and includes one entry for a complete telephone number. However, note that if the station caller information flags are enabled, the Caller ID information received from the central office overrides the call routing table's name on the phone display (see the priority list shown on page 5-72).

Pattern	Ring-In Type	Name	Ring-In Destination
602961+	Single	Chandler	Hunt Group Number 2570
602893+	DISA	Phoenix	Not Applicable
602255+	Extension List	Phoenix	Extension List P006
6022549400	Single	Phoenix Branch	Extension Number 2030



Sample #4: This is an example of a DNIS application pattern. If the DNIS digits match 8006695858, the call came in on the Technical Support 800 number and is sent to the Technical Help hunt group. The display shows that the call is from "Help Line."

Pattern	Ring-In Type	Name	Ring-In Destination
8006695858	Single	Help Line	Hunt Group Number 2001

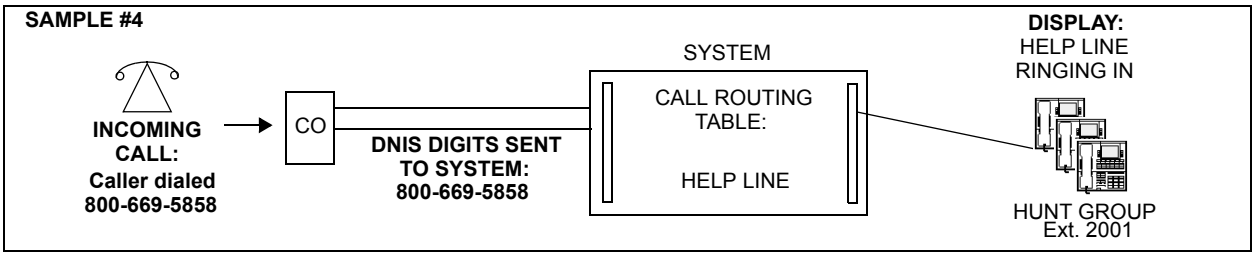


Table 5-20. *Sample Call Routing Tables for the U.S. (Continued)*

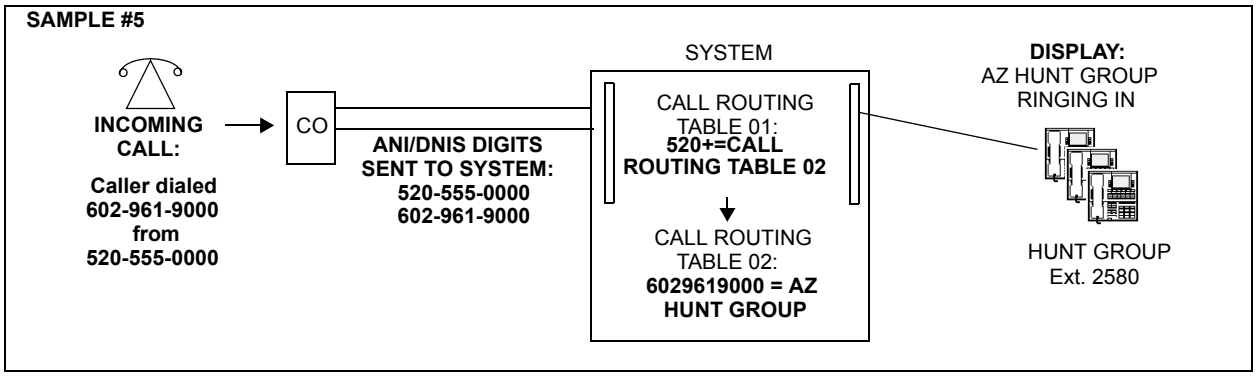
Sample #5: This table has a ring-in destination that sends calls to another call routing table. It also uses two-stage caller identification (*ANI*DNIS*). In this example, a caller from area code 520 who dials the number (602) 961-9000 would be routed to the AZ Hunt Group. If the caller in the 520 area code dialed any other number in area code 602, the call would be routed to extension 1002.

TABLE 01 – CALL ROUTING KEY: Outside Party Number from 520 area code

Pattern	Ring-In Type	Name	Ring-In Destination
520+	Call Routing Table	Table 2	Call Routing Table 02

TABLE 02 – CALL ROUTING KEY: Trunk Information calling 602+ number

Pattern	Ring-In Type	Name	Ring-In Destination
6029619000	Single	AZ Hunt Group	Hunt Group Number 2580
602+	Single	Arizona	Extension Number 1002



Sample #6: For call routing to the public network, this table is programmed to send callers who dial a special number (940-1431) to an outside line.

Pattern	Ring-In Type	Name	Ring-In Destination
6029401431	Single	CR To Public	Trunk Group 92001

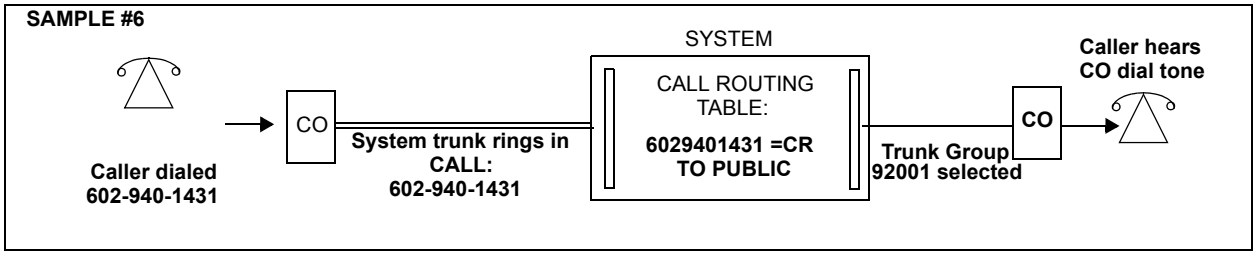


Table 5-20. *Sample Call Routing Tables for the U.S. (Continued)*

Sample #7: For call routing to another node in the system network, this table is programmed to send callers who call Node 1 using 602-961-9000 to a hunt group on Node 2. It also identifies the call as being from the "Sales Line."

NODE 1/TABLE 01

Pattern	Ring-In Type	Name	Ring-In Destination
6029619000	Single	NODE 2	Node 02

NODE 2/TABLE 01

Pattern	Ring-In Type	Name	Ring-In Destination
6029619000	Single	SALES LINE	Hunt Group Number 2501

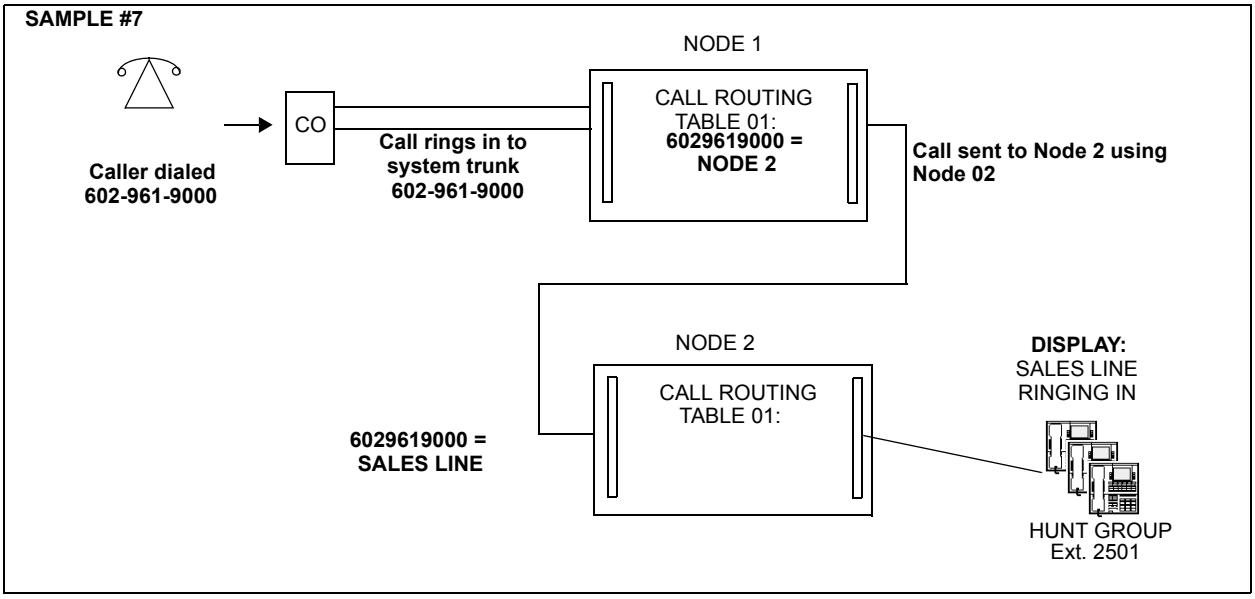
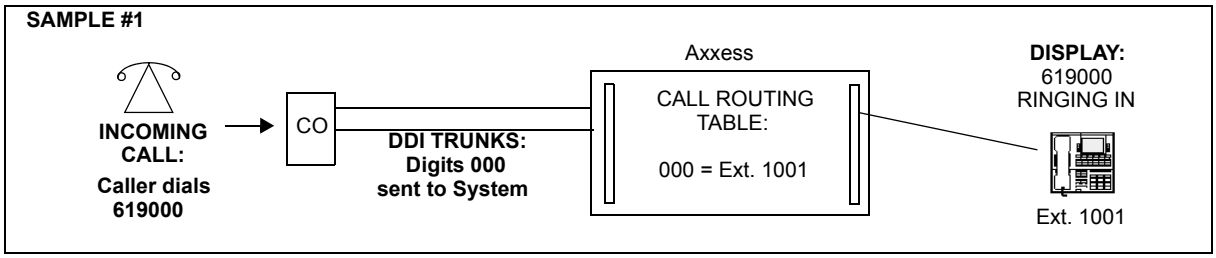


Table 5-21. *Sample Call Routing Tables (Europe)*

Sample #1: This is an example of a DDI routing table. (An actual DDI table would have more than four entries; this is an abbreviated list.) Each DDI number is identified by the three digits that are sent by the local exchange (000-003). The base number of the DDI numbers is 619. Therefore, the names of each of the patterns combine the base number and the additional three digits to form the complete number for display (619000 to 619003).

Pattern	Ring-In Type	Name	Ring-In Destination
000	Single	619000	Extension Number 1001
001	DISA	619001	Not Applicable
002	Extension List	619002	Extension List P004
003	Single	619003	Hunt Group Number 2550



Sample #2: This is an CLIP table for various STD codes. If the incoming CLIP information begins with one of the indicated STD codes, it is routed to the designated destination, and the display shows the area where the call was placed.

Pattern	Ring-In Type	Name	Ring-In Destination
0181+	Single	Outer London	Extension Number 2001
0171+	DISA	Inner London	Not Applicable
01933+	Extension List	Wellingborough	Extension List P005
01536+	Single	Kettering	Hunt Group Number 2560

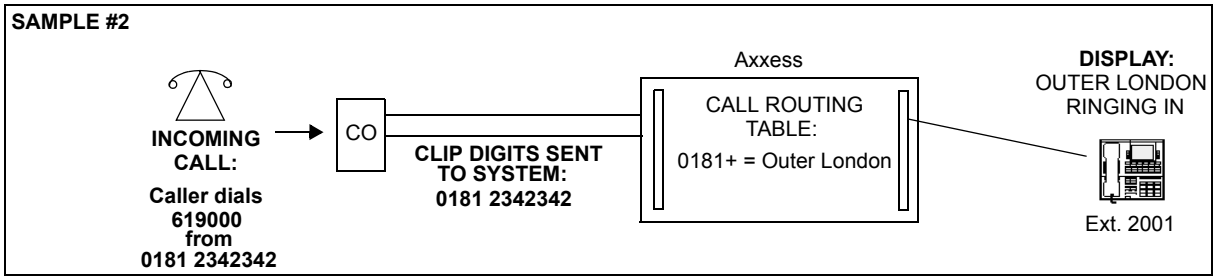


Table 5-21. *Sample Call Routing Tables for Europe (Continued)*

Sample #3: This is a two-stage routing plan. The first stage collects selected CLIPs and sends them to designated extensions (e.g., 01933402332 is delivered to Hunt Group 1, and 01933402206 is delivered to the fax machine, extension 1007.) Other identified calls pass to table 02 to operate as DDI. Unidentified calls pass to 1000. In this example, a caller from STD code 01933 who dials the number 01536619000 would be routed to Sales. If the caller in the 01933 STD code dialed any other number in STD code 01536, the call would be routed to extension 1002.

TABLE 01 – CALL ROUTING KEY: Outside Party Number

Pattern	Ring-In Type	Name	Ring-In Destination
01933402332	Single	Main Ring-in	Hunt Group Number 2000
01933402206	Single	Fax Machine	Extension Number 1007
01933+	Call Routing Table	Table 2	Call Routing Table 02
E	Single	Unidentified	Extension Number 1000

TABLE 02 – CALL ROUTING KEY: Trunk Information

Pattern	Ring-In Type	Name	Ring-In Destination
01536619000	Single	Sales	Hunt Group Number 2580
01536+	Single	Reception	Extension Number 1002

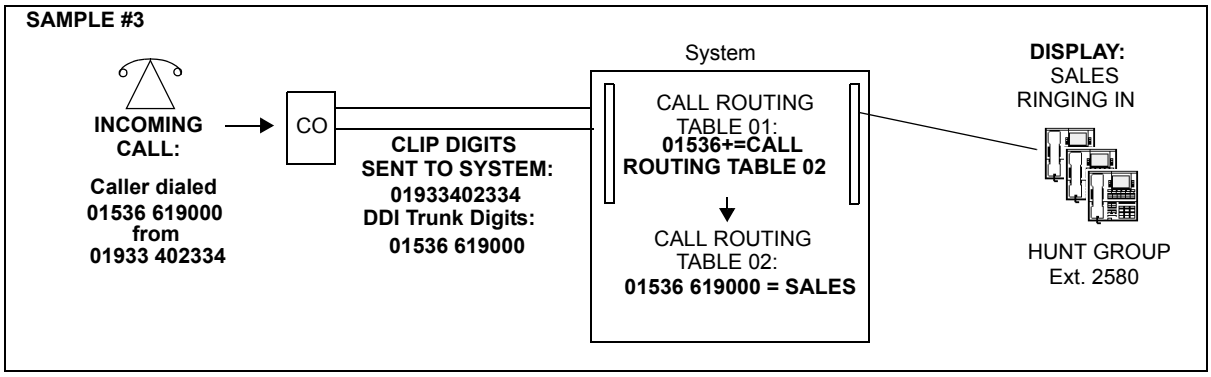


Table 5-21. *Sample Call Routing Tables for Europe (Continued)*

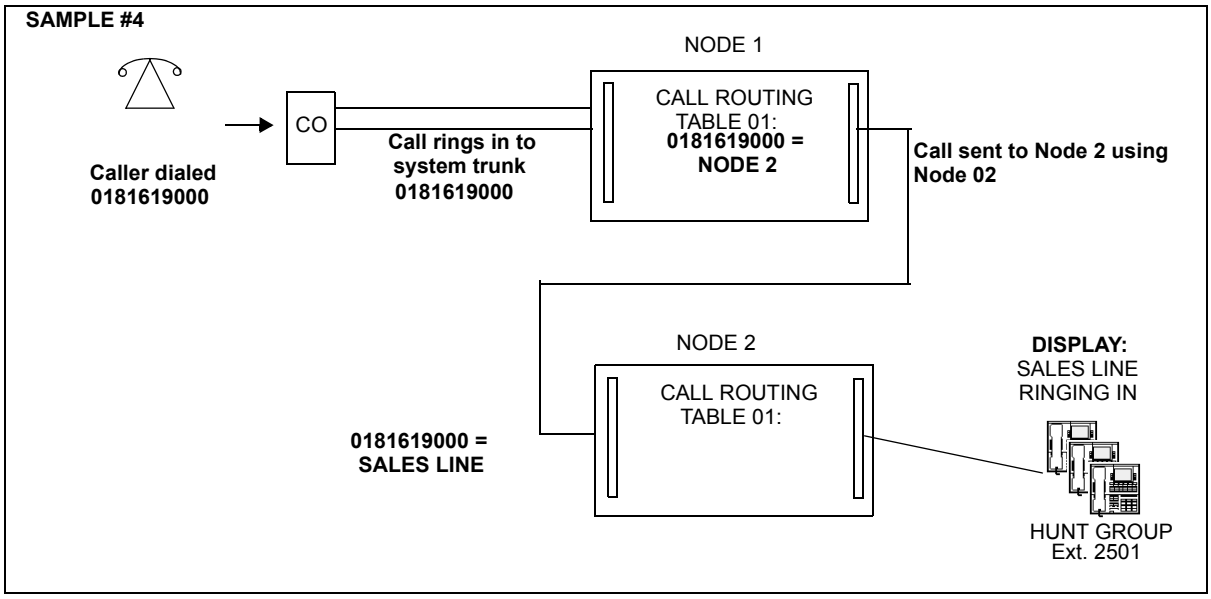
Sample 4: For call routing to another node in the network, this table is programmed to send callers who call Node 1 using 0181619000 to a hunt group on Node 2. It also identifies the call as being from the "Sales Line."

NODE 1/TABLE 01

Pattern	Ring-In Type	Name	Ring-In Destination
0181619000	Single	NODE 2	Node 02

NODE 2/TABLE 01

Pattern	Ring-In Type	Name	Ring-In Destination
0181619000	Single	Sales Line	Hunt Group Number 2501



Music-On-Hold for Call-Routing Tables

10.129 Inter-Tel Axxess Call Processing manages the audio connections for incoming Public Switched Telephone Network (PSTN) calls. Currently, when a device places a call on hold, the trunk settings determine how it provides hold audio for the outside party. This determination is based on the settings at the Central Office (CO) trunk group to which the trunk resides. The settings include "Music-On-Hold," "Audio on Transfer to Ring," "Audio on Transfer to Hold," and "Audio on Hold for Transfer Announcement." Each of these fields allows for various options, including Silence, Tick Tone, Ringback, Board, Music-On-Hold, and Use Next Device's Audio Source. See page 6-324 for details.

10.130 Call Processing (CP) v11.0 modifies this functionality such that CO trunk calls routed through Call Routing Tables (CRTs) use the matching CRT entry's "Music-On-Hold Profile" to overwrite the settings at the CO Trunk Group.

10.131 Table 5-22 shows an example of how the Music-On-Hold Profiles are handled when three CRT entries are chained.

Table 5-22. *An Example of the Music-On-Hold (MoH) Profiles for Chained CRTs*

	CRT #1	CRT #2	CRT #3	Result
MoH Profile IDs	None	None	None	Use the CO Trunk Group's MoH settings
	1	None	None	Use MoH Profile ID 1
	1	2	3	Use MoH Profile ID 3

10.132 If the matching CRT entry does not have a "Music-On-Hold Profile" set or the call is not routed through a CRT, the trunk determines the audio setting based on the CO Trunk Group settings as in previous versions. Therefore, the "Music-On-Hold Profile" in the matching CRT entry has priority over the audio settings in the CO Trunk Group.

10.133 If CRT entries are "chained" (a CRT entry may point to another CRT), the last matching CRT entry in the chain with a Music-On-Hold Profile set to anything other than "None" will take precedence. If the last matching CRT entry has a Music-On-Hold Profile set to "None," the associated CO Trunk Group settings apply.

10.134 When the Music-On-Hold selection is set for a specific call, that Music-On-Hold setting will persist through the duration of the call even if that call is forwarded, moved or transferred. The following case scenarios describe how this feature works:

- Several businesses share a single phone system. Due to the layout and size of the companies involved, each business does not require its own CO Trunk Group. However, each business in the group owns a specific subset of the Direct Inward Dialing (DID) numbers available. These businesses require their own Music-On-Hold played for callers. Using the new feature, each business would have a Music-On-Hold capable board assigned and would specify that board within the CRT entry for the matching called number.
- A business sells several different products. Customers call different numbers based on the product of interest. Management would like customers to hear Music-On-Hold based on the product in question. This feature would allow the specific CRT entry to dictate which board provides the Music-On-Hold for the incoming calls.

10.135 To support this functionality, there is a new "Music-On-Hold Profiles" folder in v11.0 DB Programming. This field provides the same Music-On-Hold fields that currently exist in a CO Trunk Group. Each profile also contains a profile ID and name. You may then create or modify an entry in a CRT to have a specific called number or caller number use the "Music-On-Hold Profile," instead of the CO Trunk Group settings.

N. AUTOMATIC ROUTE SELECTION (ARS)

NOTE: This feature requires the ARS software license.

10.136ARS is a money-saving feature that allows the system to be programmed to select the least expensive route for placing outgoing calls. It can be used for placing outgoing calls and transferring or forwarding calls to outside telephone numbers. Stations can be restricted to using only ARS for placing outgoing calls. Also, because users do not have direct access to trunks on other nodes, ARS is the only way users can place calls using the other nodes' trunks.

10.137ARS is programmed using route groups and facility groups. (See page 6-291 in PROGRAMMING for programming information.)

- **Route Groups**: Contains dialing patterns and facility groups.
  - — The dialing patterns are used to determine the calls that will be routed through the route group. For example, with a U.S. system, the default dial pattern for Route Group 1 is N+ (any number of digits beginning with digit 2-9). If a number is dialed that begins with 1, it will not be routed through this route group. With a European system, the default dial pattern for Route Group 1 is N+ (any number of digits beginning with digit 2-9, 345+, 0500+, 0645+, and 0800+).
  - — Each route group has an ordered list of facility groups that contains lists of local trunk groups and/or nodes. There can be 100 facility groups in the system. The facility groups should be programmed so that the least-expensive route is checked and, if available, is selected first. If the least-expensive facility group is not available the system checks the other groups in the list until it finds an available trunk.
- **Facility Groups**: Contains trunk group/node lists and dial rules.
  - — The trunks can include local trunk groups or nodes.
  - — The dial rules tell the system what to dial. The system can have up to 32 dial rules, 26 of which are programmable. Each facility group can use up to 32 dial rules. For example, with a U.S. system, if the selected route group requires that the number contain "1" but no area code, the dial rules include the 1 and drop the area code. The modified telephone number can contain up to 32 digits. (If SMDR is enabled, the modified telephone number, not the digits dialed, will appear in the SMDR call record.) When programming ARS, you can use preset dial rules or create new dial rules that add up to 16 digits each. The pre-programmed dial rules are:

    Dial Rule #1 – Echo Equal Access: This non-programmable dial rule includes the equal access digits (10XXX or 101XXXX) in the number, if dialed.

    Dial Rule #2 – Echo Toll Field: The toll field (1, 0, 01, or 011) is included in the number, if dialed. This dial rule is non-programmable.

    Dial Rule #3 – Echo 3 Digits After Toll Field: With this non-programmable dial rule, the three digits after the toll field are included in the number. (These digits are usually the area code or country code.)

    Dial Rule #4 – Echo Local Address: This non-programmable dial rule allows ARS to dial the telephone number that was dialed by the station user.

Dial Rule #5 – Echo Account Code: This non-programmable dial rule causes the system to dial the account code that is associated with the call, when it is available. The account code can be entered using any of the account code types, including All Calls Following, as long as the system receives the account code before the end of dialing. (To use an optional account code, the code must be entered before the number is dialed.)

Dial Rule #6 – Echo Extension Number: This non-programmable dial rule tells the system to include the extension number of the station or Voice Processing application that is being used to place the call.

Dial Rule #7 – Add #: This dial rule adds a pound [hash] (#) to the number dialed. This dial rule can be programmed.

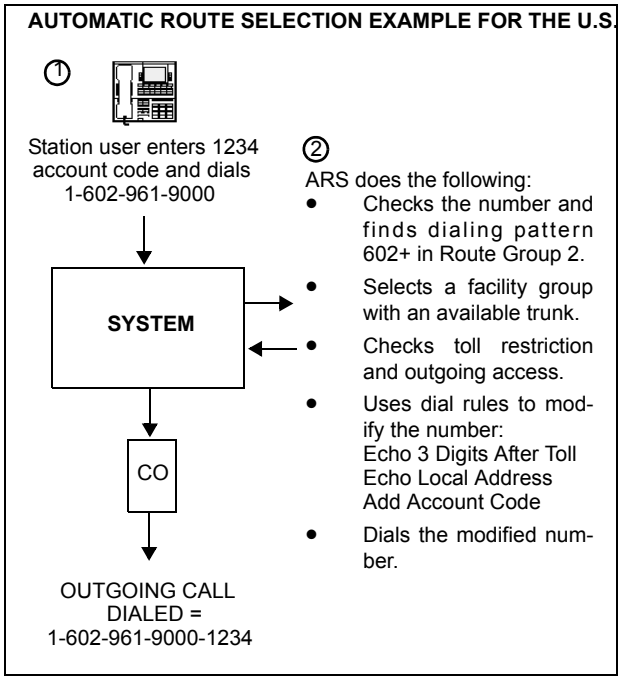
10.138 When ARS is selected, the user dials the number (including the area code, if needed, for the U.S. only), and the system does the following:

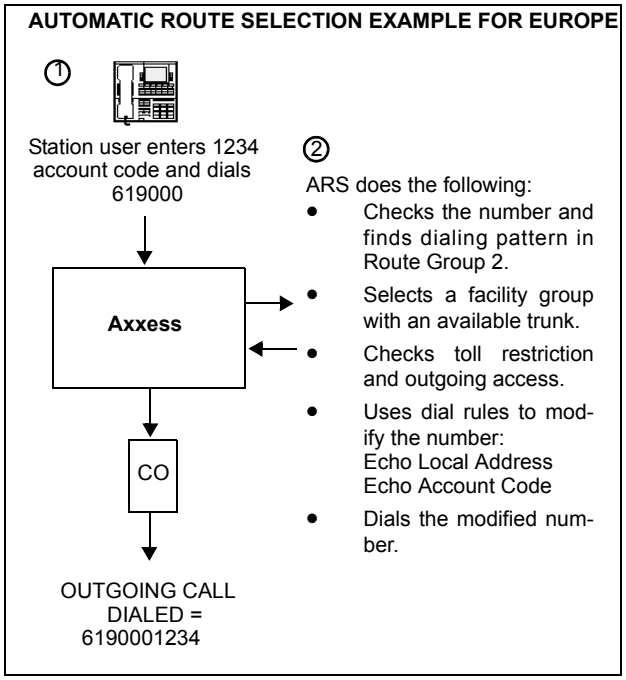
1. **Checks the dialed number and matches the dialing pattern to a route group**: The system checks the route groups in numerical order and selects the first group that applies to the dialing pattern of the number that was dialed.

   NOTE: If the Emergency Call feature is programmed to use ARS, the emergency calls will always go through Route Group 1, regardless of route group programming.

2. **Selects a facility group**: If all of the trunks are busy, and the station is enabled for ARS camp on, the call will camp on to the facility group until a trunk is available.
3. **Checks for toll restrictions and outgoing access**: Before the number is modified by the facility group's dial rules, the system checks the station toll restrictions and outgoing access to determine whether the call is allowed. (All calls placed using ARS are toll restricted, regardless of whether the selected trunk is subject to toll restriction. However, Emergency Call feature calls are never toll restricted.) If allowed, the system continues to the next step. If not allowed, the system sends reorder tones and the call is not placed.
4. **Adds or deletes digits according to the facility group chosen**: Each facility group has a programmed set of dial rules that tells the system what to dial. For example, with a U.S. system, if the selected route group requires that the number contain "1" but no area code, the dial rules include the 1 and drop the area code; with a European system, if the selected route group requires that the number contain a toll field, the dial rules include the toll field.
5. **Dials the modified telephone number**: If the number is allowed, the system seizes an idle trunk in one of the selected trunk groups, waits for the Dialing Wait After Connect timer to expire, and then dials the number.

10.139 When ARS is used, the station user hears dial tone when the feature code is entered (manually or automatically using a CALL button, the ARS button, or the **OUTGOING** button). The user then dials the number and hears silence until the trunk is seized and dialing is completed. (The user will not hear the digits being dialed.) The call appears under an available CALL button or trunk button.





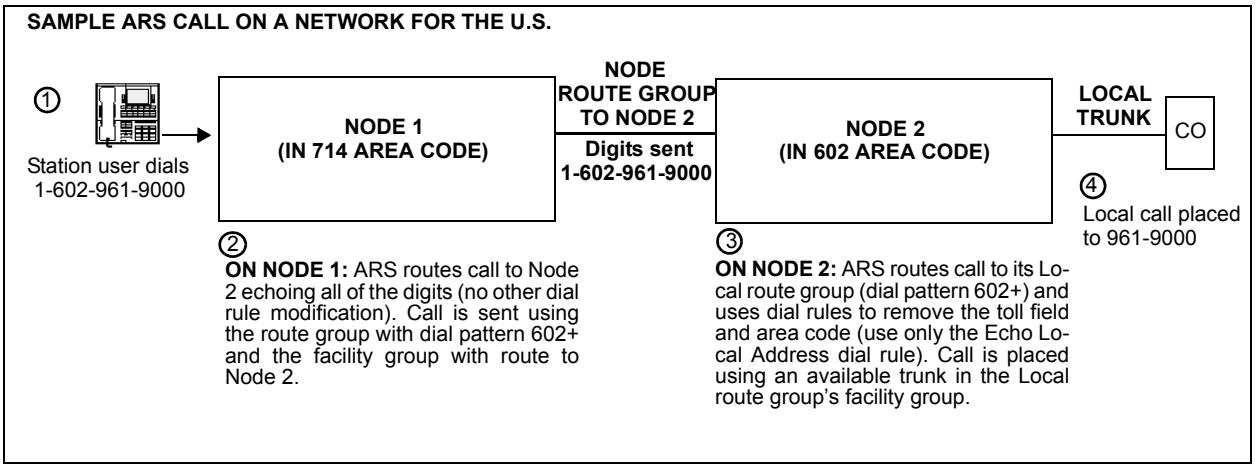
ARS on a Network

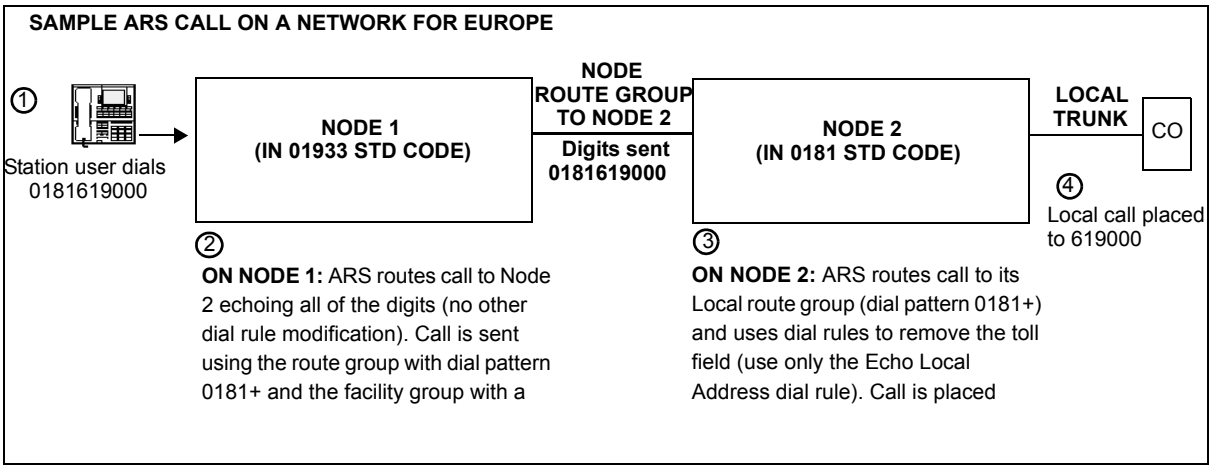
10.140 Each node has its own ARS programming. All trunk groups and nodes in a facility group must reside on the same node as the facility group.

10.141 Using ARS is the only way a user can access trunks on other nodes. If the system routes an outgoing call to another node using an intermediary node, the intermediary node's ARS cannot use the nodes in its facility group to move the outgoing call to another node. This eliminates the system from getting into an infinite loop searching for a node to route the outgoing call.

10.142 When determining toll restriction for an ARS outgoing call, the network checks the station toll restriction based on the database information on the node on which the station resides, not the node that contains the trunk which the system uses to place the call. The system does not check the trunk COS for ARS calls.

10.143 For the purposes of the phone call cost display and SMDR output, the network computes call cost using the factors and rates on the node on which the outgoing trunk resides. In other words, if a station on Node A dials what is considered a long-distance number on Node A, and the call is routed to Node B where the number is considered local, the phone will see the local call cost rate which equals the value on Node B.





O. TOLL RESTRICTION

10.144 The following features provide toll restriction on the Inter-Tel system. (They are described in further detail in this section.)

- **Trunk Group Toll Restriction**: Designates a trunk group as "subject to toll restriction" or "not subject to toll restriction" in DB Programming. If subject to toll restriction, station class of service is checked when the trunk group is selected for placing an outgoing call.
- **Exemption From ARS-Only Restriction**: Designates each trunk group as "exempt from automatic route selection (ARS) only." If exempt, station users with the ARS-Only class of service can select the trunk group directly.
- **Absorbed Digits**: Allows trunk groups that are subject to toll restriction to "absorb" (ignore) the first digit(s) dialed. This allows the system to handle the dialed digits just as they would be by the local telephone company or PBX to which the system is connected.
- **Station And Trunk Group Classes Of Service**: Restricts or allows certain digit patterns when an outside call is dialed. (Trunk group COS is not checked when ARS is used.)
- **Toll Strings**: Allow the programmer to define dialing patterns which are abbreviated to single character "wildcards." These programmable wildcards are used frequently in the toll restriction class of service programming and in other system areas such as ARS route group programming, trunk group absorbed digits programming, and others. ***Note that changing these dialing patterns will affect all of the areas where they are used.***

Trunk Groups Subject to Toll Restriction

10.145 Each trunk group can be designated as "subject to toll restriction" or "not subject to toll restriction" in DB Programming. All trunk groups are subject to toll restriction when the system is in the default state.

- When a station user selects a trunk group that is designated as "subject to toll restriction," the system checks the database for that station's class of service, and call cost is determined by the digits that are dialed (see page 5-305).
- When a trunk group is chosen that is "not subject to toll restriction," the station's COS is not checked (except ARS-Only restriction).

10.146 The exceptions to trunk group toll restriction are as follows:

- **Emergency Call Feature**: Uses the designated trunk access code to seize a trunk and dials the programmed emergency number regardless of any station or trunk group toll restrictions. (See page 5-112 for details.)
- **Automatic Route Selection (ARS)**: Designates all calls using ARS (except Emergency Call feature calls) as always subject to toll restriction, regardless of the selected trunk group's toll restriction programming. Note that only the station's COS is checked; the trunk group's COS is not.
- **Network ARS Calls**: Determines toll restriction for an ARS outgoing call by checking the station toll restriction based on the database information on the node on which the station resides, not the node that contains the trunk which the system uses to place the call. The system does not check the trunk COS for ARS calls.

10.147 Trunk groups are often programmed as "not subject to toll restriction" to allow station users to have access to reduced-cost long distance carriers [private wires in Europe], or to use ringdown lines, dictation machines, voice mail systems, and/or other auxiliary equipment. When programming unrestricted trunk groups, one of five call cost factors can be selected to designate calls as free, local, toll local, toll long distance, or operator, international - station, or international operator [local, national, operator, or international in Europe]. The selected call cost is then used for all calls that are placed using that trunk group.

10.148 Note that whether or not a trunk is marked as "subject to toll restriction," dialing is ***not*** required to hold the trunk. For example, when a user seizes a line to make an outgoing call (via any method), if there is an incoming call on the line for which ring-in had not yet been detected, the user can talk with the incoming caller without dialing any digits. (That is, the call will not be dropped when the Dial Initiation timer expires.) However, if the trunk is marked as "subject to toll restriction" and the user attempting to make the outgoing call dials one or more digits before realizing that there is an incoming call, and the digits dialed represent a restricted number for the station and trunk group being used, the call ***will*** be dropped (unless the "Drop Incomplete Outgoing Calls" flag is disabled as described on page 6-260 in PROGRAMMING). In other words, the system has protection for "glare" situations between incoming and outgoing calls, but the protection is overridden if the outgoing caller dials a restricted number.

Trunk Group Exemption from ARS Only

10.149A trunk group may be designated as "exempt from Automatic Route Selection (ARS) Only." This allows users with the ARS-Only class of service (see COS 01 on page 5-107) to directly select specified trunk groups by pressing the appropriate trunk group buttons or by entering the appropriate trunk group access codes.

10.150 This capability is necessary if trunks are connected to auxiliary equipment, such as analog voice mail systems, dictation equipment, or ring-down lines [private wires in Europe]. When such trunks are designated as exempt from ARS Only, stations with the ARS-Only COS (and allowed access) can use the special facilities.

10.151 The default assignment for all trunk groups is "*not exempt* from ARS Only." This means that, by default, stations with the ARS-Only COS are denied direct access to the trunk groups.

Area/Office Code Numbering Plans and Local Area Codes (U.S. Only)

10.152 If any office codes in the local area code have a 0 or 1 as the second digit, the local office codes are probably used as area codes elsewhere, and the system requires special programming to allow toll restriction to work properly. During System-Wide Information DB Programming, the programmer can specify the following non-standard numbering plan information:

- **Office Codes Used as Area Codes**: Area codes in other locations are the same as office codes within the system site's area code.
- **Area Codes Used as Office Codes**: Office codes within the system site's area code are the same as area codes assigned to other areas.
- **Local 7/10 Digit Dialing**: When this flag is enabled, outgoing calls are identified as having reached the end of dialing if the first digits are not a toll field, equal access field, or a local area code. This function speeds up placement of local seven-digit calls in an area where some local calls require ten digits.

NOTE: The chassis within an ATM-based system cannot span area codes unless the area codes are all considered local. You can, however, program an ATM node to require ten-digit dialing on all local calls. This makes ten-digit dialing optional for all dual chassis residing in the home area code. Metropolitan areas with overlapping area codes would, however, have to dial ten digits for home area code calls.

- **Toll Digit Allowed On Toll Local Calls**: This option applies only if the area and office codes overlap. Callers in the site's area code usually dial a 1 when placing a call within the local area code(s).
- **Toll Digit Required On Toll Long Distance Calls**: This option applies only if the area and office codes overlap. Callers in the site's area code must dial a 1 when placing a call outside of the local area code(s).

10.153 The database requires that the programmer enter a "home" area code for the system site. It also allows up to 16 home area codes for non-toll calling. The system refers to these area codes for toll restriction and call cost.

Absorbed Digits

10.154 Trunk groups that are subject to toll restriction can be programmed to "absorb" (ignore) the first digit(s) so that the digits dialed are handled by the system just as they would be by the local telephone company [network provider in Europe] or PBX to which the system is connected. There are two applications for this capability: PBX installations and installations in areas where part of the local office code is absorbed by the central office (CO) or local exchange. Trunk groups that connect to outside trunks in PBX installations are referred to as "PBX trunk groups." Trunk groups that use trunks for which the CO absorbs part of the local office code are referred to as "local trunk groups."

10.155 When an Inter-Tel system is installed behind a PBX, users must dial special PBX codes to select the CO trunks. The system trunk group that is used for PBX trunk access must be programmed to absorb PBX digits. The PBX access codes are then programmed as the absorbed digit patterns for that PBX trunk group. Up to 50 absorbed digit patterns are allowed, with a maximum of 48 digits in each.

10.156 When a PBX trunk group is used and an absorbed digit string is ***not*** dialed, the system behaves as though the trunk group is "not subject to toll restriction" and does not perform toll restriction and call cost functions. The call is considered to be an internal PBX call; it is allowed no matter what digits are dialed. The call cost rate is "PBX," and the call is designated as free (000) in the SMDR report.

10.157 When a PBX trunk group is used and an absorbed digit string ***is*** dialed, the system absorbs the PBX access code and behaves as though the trunk group is "subject to toll restriction," using the remaining digits for toll restriction and call cost processing. When the Redial, Speed Dial, or Call Forward features are used, the system automatically inserts a short pause after the absorbed digit string to provide for the delay that occurs while a trunk is seized.

10.158 In some rural areas, specific digits (dialed as all or part of the local exchange) are absorbed by the central office [local exchange], thus reducing the number of digits required to dial local calls. These digits may also be "repeatable." That is, they are absorbed if dialed more than once. To determine if a central office [local exchange] absorbs digits and whether the digits are repeatable, contact the telephone company [network provider].

10.159 When using this type of trunk, the system must be programmed to recognize the absorbed digits. That is, the absorbed digits must be programmed as an absorbed digit string for any trunk groups that include those trunks. For each local trunk group, only one string is allowed, with a maximum of 48 digits. The only characters allowed in the string are digits (0-9), pound (hash) sign (#), parentheses ( ), brackets ([ ]), greater than (>) and less than (<) symbols, and the wildcards. (See page 5-111 for wildcard information.)

10.160 If the system is not programmed to recognize the absorbed digits for a PBX or local trunk, the system may be left open for users to bypass toll restriction. A user could bypass toll restriction by dialing the digits that are absorbed by the CO before dialing a toll number. If the system does not know that the first digits are absorbed, it will not recognize a toll digit that follows them. However, if the digits absorbed by the CO are programmed as an absorbed digit pattern for the trunk group, the system will recognize them, absorb them, and then recognize any digits that follow as possible toll digits.

10.161 When dialed, absorbed digits (PBX or local) appear in the SMDR record if they are not suppressed during SMDR programming. Repeatable (local) absorbed digits will appear in SMDR only once, even if dialed repeatedly.

10.162 Absorbed-digits are programmed on a trunk group-by-trunk group basis. In the default state, no trunk group absorbs digits. See page 6-233 in PROGRAMMING for more information.

Equal Access (U.S. Only)

10.163 Under the terms of the final divestiture agreement, Bell Operating Companies provide equal access to all long distance companies. The equal access provision requires each Bell Operating Company to modify existing switching equipment to make it possible for customers to have direct access to all available carriers serving their area code. Prior to equal access, routing long distance calls to long distance companies other than AT&T required a lengthy dialing process usually involving a local access number, authorization number, and public network number.

10.164 As central offices are converted to provide equal access, customers served by those offices may choose one of two options for the routing of long distance calls. The options are:

- **Presubscription**: Normal “1+” or “0+” dialing procedures may be used and a switching arrangement automatically routes calls to the predesignated long distance company.
- **Non-Presubscription**: A long distance company is not predesignated and a five or six digit equal access code identifying a long distance company (10XXX or 101XXXX) is required to be dialed preceding the desired “1+” or “0+” long distance telephone number to designate the long distance company to handle the call.

10.165 Presubscription does not limit long distance calling to the predesignated long distance company. An access code (10XXX or 101XXXX) may be dialed to direct a call to the long distance company of choice. Presubscription also does not affect the way an operator may be contacted. Any existing non-presubscription long distance company operator is called by dialing the carrier’s access code followed by “0.”

Class Of Service (COS)

10.166 Each station and each trunk group that is "subject to toll restriction" is assigned toll restriction classes of service (COS) that restrict or allow certain dialing patterns on outside calls.

10.167 The classes of service (COS) are programmed individually for stations, Voice Processing applications, and trunk groups. There are separate COS designations for day and night modes. A station or trunk group can be completely unrestricted or can have any combination of the classes of service.

10.168 The first nine [seven in Europe] COS designations (01-09 or 01-07) have default values. COS designations (10-16 or 08-16) do not have default values and are blank. All of the classes of service are programmable. Each may be specified as an "allowed" or "denied" class of service, dialing patterns can be programmed for each, and each can be assigned to stations and trunk groups as needed. (See page 6-281 in PROGRAMMING for COS programming information.) The default values of COS for U.S. and European systems are as follows. The toll string wild cards and non-programmable wildcards are described and listed on page 5-111.

For U.S. Systems:

- **COS 01 – ARS Only**: *(This is a station COS only. It is not used for trunk groups. Also, it cannot be used unless the ARS premium feature is licensed.)* Calls can only be placed using the Automatic Route Selection (ARS) feature when this restriction is assigned. The user will hear reorder tones when attempting to place a call using any other method. A restricted user can still select individual trunks if the trunks are designated as "exempt from ARS Only" (as described on page 5-104), were transferred, were placed on hold, or are recalling or ringing. Trunk restriction determines which trunks in the ARS route group can be selected by the station or Voice Processing application.
- **COS 02 – Deny Area/Office**: This restriction is divided into eight user groups to allow the use of varying area/office code restriction tables. This is useful for reducing restrictions for some of the stations, Voice Processing applications, or trunk groups while increasing restrictions for others. Each station, application, and trunk group is assigned a day mode and a night mode user group in DB Programming. Within each user group, area codes can be designated as restricted, allowed, or extended. Restricting an area code prevents users from placing calls to that area code. Allowing an area code allows all office codes within that area code. Designating an area code as extended allows the programmer to determine which office codes (up to 800) are allowed or restricted within that area code. For each user group, 800 area codes can be marked as allowed or restricted in the database list, and up to six area codes can be marked as extended.
- **COS 03 – Deny Operator**: Calls to numbers that match the dial patterns for this class of service (defaults to [Q]RN+, [Q]R0, and [Q]RE) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 04 – Deny Toll Access**: Calls to numbers that match the dial patterns for this class of service (defaults to [Q]TN+ and [Q]TE) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 05 – Deny International**: Calls to numbers that match the dial patterns for this class of service (defaults to [Q]I+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.

- **COS 06 – Deny Equal Access**: Calls to numbers that match the dial patterns for this class of service (defaults to Q+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 07 – Deny Local Calls**: Calls to numbers that match the dial patterns for this class of service (defaults to N+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 08 – Denied Numbers**: Calls to numbers that match the dial patterns for this class of service (defaults to 1900NXXXXXX+ and 976XXXX+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 09 – Allowed Numbers**: Calls to numbers that match with the dial patterns for this class of service, defaults to 1(800, 888, 877, 866, 855, 844, 833, and 822)NXXXXXX+, are allowed, even if number also matches a dial pattern in a restricted class of service that is assigned to the station, Voice Processing application, or trunk group being used.

For European Systems:

- **COS 01 – ARS Only**: *(This is a station COS only. It is not used for trunk groups. Also, it cannot be used unless the ARS premium feature is available.)* Calls can only be placed using the Automatic Route Selection (ARS) feature when this restriction is assigned. The user will hear reorder tones when attempting to place a call using any other method. A restricted user can still select individual trunks if the trunks are designated as "exempt from ARS Only" (as described on page 5-104), were transferred, were placed on hold, or are recalling or ringing. Trunk restriction determines which trunks in the ARS route group can be selected by the station or Voice Processing application.
- **COS 02 – Deny Operator**: Calls to numbers that match the dial patterns for this class of service (defaults to R+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing Unit application, or trunk group being used.
- **COS 03 – Deny Toll Access**: Calls to numbers that match the dial patterns for this class of service (defaults to TN+, TE, 010+, and T1+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing Unit application, or trunk group being used.
- **COS 04 – Deny International**: Calls to numbers that match the dial patterns for this class of service (defaults to I+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 05 – Deny Local Calls**: Calls to numbers that match the dial patterns for this class of service (defaults to N+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 06 – Denied Numbers**: Calls to numbers that match the dial patterns for this class of service (defaults to 0891+ and 0898+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 07 – Allowed Numbers**: Calls to numbers that match with the dial patterns for this class of service, defaults to 0345+, 0500+, 0645+, and 0800+, are allowed, even if number also matches a dial pattern in a restricted class of service that is assigned to the station, Voice Processing application, or trunk group being used.

10.169 Calls are only restricted if the dialed patterns match the denied pattern exactly and that is the only class of service you have. Allowed patterns, however, always override denied patterns, even if the numbers are similar.

10.170 For example, with a U.S. system, the following table demonstrates what is allowed or denied when the restricted pattern is 976+ and the allowed pattern is 976-1111.

Table 5-23. *COS Example (U.S.)*

Number Dialed	Allowed or Denied?
97 (partial match)	Allowed
9761234 (exact match with “denied” pattern)	Denied
9761111 (exact match with “allowed” pattern)	Allowed

10.171 With a European system, the following table demonstrates what is allowed or denied when the restricted pattern is 619+ and the allowed pattern is 619000.

Table 5-24. *COS Example (Europe)*

Number Dialed	Allowed or Denied?
61 (partial match)	Allowed
619123 (exact match with “denied” pattern)	Denied
619000 (exact match with “allowed” pattern)	Allowed

10.172 System speed-dial numbers can be programmed to bypass COS restrictions on a system-wide basis. If the option is not enabled, all system speed-dial numbers are subject to toll restriction.

10.173 Because a condition may exist where a critical or life threatening situation needs to be reported, using the Emergency Call feature code will override all toll restrictions and trunk access programming.

10.174 In the event that the classes of service for the station or Voice Processing application and the trunk group conflict, the decision to restrict or allow a number is based on the following ordered checks. Note that these ordered checks apply to ***all*** calls to outside telephone numbers, including calls forwarded to outside telephone numbers.

1. *Station or Voice Processing application COS is checked.* If the digits dialed ***exactly*** match a pattern included in a “denied” class of service assigned to the station or Voice Processing application and not with a pattern included in an “allowed” class of service assigned to the station or Voice Processing application, the number is restricted.
2. *If ARS is not being used and forward to an outside number is not being programmed, trunk group COS is checked.* If the digits dialed ***exactly*** match a pattern included in a “denied” class of service assigned to the trunk group and not with a pattern included in an “allowed” class of service assigned to the trunk group, the number is restricted.
3. *If ARS is being used and a forward to an outside number is being programmed, station or Voice Processing application COS is checked.* If the digits dialed ***partially*** match a pattern included in a “denied” class of service assigned to the station or Voice Processing application and not with a pattern included in an “allowed” class of service assigned to the station or Voice Processing application, the number is restricted.
4. *If a forward is being programmed to an outside number, trunk group COS is checked.* If the digits dialed ***partially*** match with a pattern included in a “denied” class of service assigned to the trunk group and not with a pattern included in an “allowed” class of service assigned to the trunk group, the number is restricted.
5. *If none of the above cases apply,* the number is allowed.

10.175 Note that if the trunk group is "not subject to toll restriction," neither the trunk group nor station COS is checked, unless the call was placed using ARS. All ARS calls are subject to ***station*** toll restriction only.

Toll String Dialing Pattern Wildcards

10.176 Toll strings are dialing patterns that are abbreviated to single character "wildcards." These programmable wildcards are used frequently in the toll restriction class of service programming and in other parts of the system, including ARS route group programming, trunk group absorbed digits programming, and others.

NOTE: Changing these dialing patterns will affect all of the other parts of the system where they are used.

10.177 Toll strings can contain any digit 0-9, #, or *. In addition, a variety of special characters may be entered into the database to reflect particular digit strings, hookflashes (recalls), or special digit strings (as described below).

10.178 Each of the following toll string wildcards can be reprogrammed and/or renamed to meet the customer's requirements. The programmable toll string wildcards for U.S. and European systems are as follows:

For U.S. Systems:

- **Operator Access (R)**: Represents the digit string that is required to reach a telephone company operator. In the default state, this dialing pattern is 0.
- **Toll Access (T)**: Represents the digit(s) required when using long distance service. In the default state, this toll string is 1.
- **International (I)**: Represents the digits required for international dialing. In the default state, the dialing patterns for this toll string are 011 and 01.
- **Equal Access (Q)**: Represents the digits required for access to secondary carriers using equal access dialing. In the default state, this toll string is 101XXXX, 10NXX, and 100XX.
- **Any Toll String (S)**: Indicates that any of the designated patterns can be dialed at that point in the digit string. In the default state, the dialing patterns for this toll string are International (I), Operator Access (R), and Toll Access (T).

For European Systems:

- **Operator Access (R)**: Represents the digit string that is required to reach a network provider operator. In the default state, this dialing pattern is 1XX.
- **Toll/National Access (T)**: Represents the digit(s) required when using long distance service. In the default state, this toll string is 01 and 0.
- **International (I)**: Represents the digits required for international dialing. In the default state, the dialing pattern for this toll string is 00.
- **Any Toll String (S)**: Indicates that any of the designated patterns can be dialed at that point in the digit string. In the default state, the dialing patterns for this toll string are International (I), Operator Access (R), and Toll Access (T).

Non-Programmable Wildcards

10.179 The following chart shows the special characters that may be used when specifying dialing patterns. These characters are not programmable.

Table 5-25. *Non-Programmable Wildcards*

Character	Meaning
X	Any digit 0-9
A	Any keypad entry (0-9, #, *)
N	Any digit 2-9
Z	Any digit 0 or 1
B	# or *
H	A hookflash (recall)
E	End of dialing; the pattern will not match if any digits are dialed beyond this point
+	Any additional dialing will be accepted, from this point in the string, with no further checking for a match. This also means that no further dialing is required beyond this point.

10.180 Unless otherwise noted, each wildcard character represents exactly one character position in the dialed number. For example, 258999X will accept 258999 plus any digit 0-9.

10.181A character inside brackets indicates an optional pattern within another pattern. For example, with a U.S. system, the International Access character "I" could be defined as 01[1]. The 01 is followed by an *optional* 1. To add an optional equal access code to the beginning, the pattern would be [Q]01[1]; with a European system, when CLI suppression (141) is being used, the [141] would need to be entered in all restrictions to ensure checking is performed after this string. The operator string R will block all 141 calls if selected.

10.182A range of numbers within parentheses, such as (00-99), indicates a range of digits within a pattern. The digits on either side of the hyphen and all digits that fall within the numerical range are included in the match. The digit strings on either side of the hyphen must be the same length, and the only digits that may appear in the range are 0-9 (#, *, pauses, and hookflashes/recalls are not allowed).

10.183 Patterns within angle brackets, such as <9>, indicate repeatable patterns within patterns. In other words, no matter how many times the digit string within the brackets is dialed, the system will consider the dialed digits to match the pattern. Note that a repeatable pattern is an entire pattern; no other characters are allowed before or after a repeatable pattern. In other words, a repeatable pattern cannot be included within any other pattern.

P. EMERGENCY CALL FEATURE

NOTICE

It shall be the responsibility of the entity or person(s) completing installation and maintenance of hardware or software described herein to research, comply with and be responsible for the specific governmental rules and regulations regarding Emergency Outgoing Access 911 [999 in Europe] of the geographic location in which such functions are performed.

10.184 The phone system immediately allows access to local emergency facilities whenever a station user enters the Emergency Call feature code (defaults to 911 or 999). When activated, the Emergency Call feature will select a trunk or route the call based on the device programming. The digit(s) required to call the local emergency services facility (defaults to 911 or 999) are then automatically dialed. Because a condition may exist where a critical situation needs to be reported, this feature will override all toll restrictions and trunk access programming.

10.185 In versions prior to 7.0, when a user dialed 911 [999] at intercom dial tone, the system selected an idle trunk using the trunk access code programmed for this feature (defaults to Trunk Group 1) or an individual line code (direct trunk access). As of version 7.0, however, the emergency extension is programmable on a per-station basis (see page 6-203). In addition, Day/Night Emergency Outgoing Access lists are provided to validate the extension making the emergency call. When a user dials 911 [999] at intercom or CO dial tone, emergency outgoing access is granted based on how system programming and what the user dialed, as described below.

- *If the user dials the emergency feature code from a station with a trunk or trunk group programmed as the Emergency Extension,* the Emergency Call feature routes the call based on the station's Emergency Extension. The trunk or trunk group does *not* validate the station originating the emergency call against the trunk group's Emergency Day/Night Outgoing Access List. An idle trunk is seized, and the system automatically dials Emergency Number 1.
- If the user dials the emergency feature code from a station with ARS programmed as the Emergency Extension:

  a. The Emergency Call feature routes the call based on the station's Emergency Extension.

  b. The call is routed to Route Group 1.

  c. The call tries the first facility group in Route Group 1.

  d. The trunk group or node trunk group validates the station originating the emergency call against the group's Emergency Day/Night Outgoing Access List.

  e. *If the call is denied,* it tries the next member in the facility group. If each member denies the call, the call tries the next facility group in Route Group 1's list.

     If everything is denied due to Emergency Outgoing Access, the call is routed once again to Route Group 1. The call then tries the first facility group in Route Group 1. The trunk group or node trunk group does *not* validate the station originating the emergency call against the group's Emergency Day/Night Outgoing Access List. If no trunks are available, the call tries the next member in the facility group. If all of the members are unavailable, the call camps-on to Route Group 1 until a trunk is available.

- *If the user accesses a trunk or trunk group and dials any Emergency Number (1-10),* the trunk or trunk group does *not* validate the station originating the emergency call against the trunk group's Emergency Day/Night Outgoing Access List. An idle trunk is seized, and the system automatically dials the emergency number.
- If the user accesses ARS and dials any Emergency Number (1-10):

  a. The call is routed to Route Group 1.

  b. The call tries the first facility group in Route Group 1.

  c. The trunk group or node trunk group validates the station originating the emergency call against the group's Emergency Day/Night Outgoing Access List.

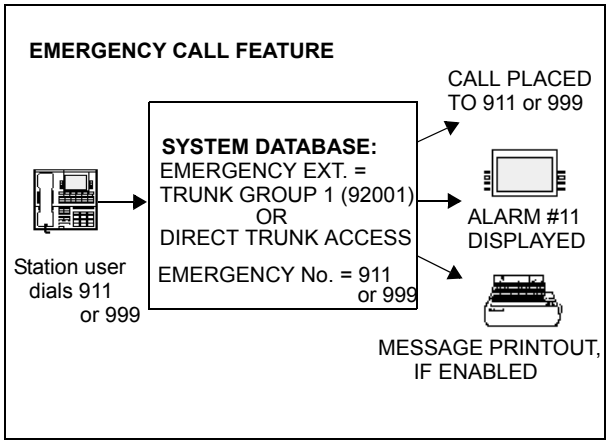
  d. *If the call is denied,* the call tries the next member in the facility group. If each member denies the call, the call tries the next facility group in Route Group 1's list.

  If everything is denied due to Emergency Outgoing Access, the call is routed once again to Route Group 1. The call then tries the first facility group in Route Group 1. The trunk group or node trunk group does *not* validate the station originating the emergency call against the group's Emergency Day/Night Outgoing Access List. If no trunks are available, the call tries the next member in the facility group. If all of the members are unavailable, the call camps-on to Route Group 1 until a trunk is available.

NOTE: Inter-Tel highly recommends that only local trunks be installed and used for emergency trunk access and that only local trunks be programmed in Route Group 1. Programming a node in Route Group 1 may cause the system to access a trunk on a separate node when ARS is used for emergency access. If this occurs, Emergency Outgoing Access is no longer validated.

10.186 The system allows the Dialing Wait After Connect timer to expire and then dials the digit string programmed in the database as the emergency number.

10.187 At the time the call is processed, a minor alarm will be generated by the system and sent to all administrator stations. Also if the Message Print option is enabled, as described on page 6-274 in PROGRAMMING, the alarm message is sent to the designated output port.



Emergency Outgoing Access Lists

10.188 As of version 7.0, the system supports two outgoing access lists called *Emergency Day Outgoing Access* and *Emergency Night Outgoing Access*. These lists, which are similar to the Day/Night Outgoing Access lists, are programmable for CO Trunk Groups (see page 6-228) and Node IP Connection Groups (see page 6-185). These lists default to the new Auto Extension List *PP051: Auto: All Stations*.

Ten Emergency Numbers

10.189 The system can store up to ten emergency numbers that are dialed when the Emergency Call feature is used.

10.190 If the system is installed in an area where 911 [999] service is not available, it is suggested that the number for the local police or fire department or the telephone company [network provider] operator be substituted.

10.191 If the database programmer renders this feature inoperative by removing all trunk access and/or by removing the dialed digit string, the database program will present a warning message indicating that the feature has been disabled and requiring the programmer to acknowledge or change this condition before being allowed to continue.

10.192 Like all other feature codes, the 911 [999] feature code can be changed to another 1- to 5-digit code, if necessary, but it is not recommended.

10.193 An emergency number will be blocked if there are no trunks or emergency numbers programed in the database, or all trunks in a trunk group are busy.

10.194 If the Emergency Call feature is programmed to use ARS, route group 1 (local calls) will be used even if its dial patterns are reprogrammed.

10.195 Emergency calls, by default, use the first local trunk group and will not be sent using node trunk groups on other nodes. However, when ARS is used to place an emergency call, Route Group 1 is used, even if it contains nodes. This means that the network ***can*** access a trunk on a node other than the user's node if the user accesses ARS and dials the emergency number. ***Local trunks must be installed and used for emergency number trunk access and nodes should not be used in Route Group 1.*** When a user places an emergency call, every administrator in the network receives an emergency alarm.

Emergency/911

10.196 Inter-Tel recommends that the system be provisioned to have an appropriate number of Emergency/911 IP resources reserved for both IP endpoints and IP trunks. For instance, if an IP endpoint makes an Emergency/911 call that must go out an IP trunk, sufficient IP resources for the both the IP endpoint and IP trunk need to be reserved to ensure the call goes through. If an Emergency/911 call uses ARS to route the call through another node, then the required IP networking IP resources also need to be sufficiently reserved on each node the call passes through to ensure non-blocking. This provisioning includes IP resources for IP trunks if that is the destination.

NOTE: The system is designed so that an Emergency/911 call has priority over all other IP resource reservations. As long as an IP resource is idle, it will be seized even though not reserved for Emergency/911.

Calling Party Number Field

10.197 Each station can be programmed to send an identifying number when a call is placed. The "Calling Party Number field is located in Devices and Feature Codes - Stations - Individual Station programming (see page 6-207). In the U.S., this information is required for emergency 911 calls in some states. You can program any number up to 48 digits in the Calling Party Number field. However, check with your service provider to determine their specific requirements for this field. This number is sent in the ISDN setup message in the Calling Party Number Information Element. In addition, the phone system also sends the extension number of the station in the Calling Party Number Subaddress Information Element. The CO should ignore this information element if it does not support it.

NOTE: There is no default number for this field. It is up to the programmer to supply the correct Emergency Calling Party Number for each station.

10.198 If an off-premises extension is used for dialing 911 [999], the 911 [999] operators will see Calling Party Number or the address of the location of the phone system, not the off-premises address. Off-premises personnel should be prepared to give the correct address and other pertinent information if it is not programmed as the Calling Party Number.

Q. IP DEVICES AND LOCAL LOOP SUPPORT

10.199 System versions 7.0 and later support IP-based solutions for providing local Public Switched Telephone Network (PSTN) connectivity using the IPRC card and a SIP Gateway.

CAUTION

The SIP gateway must have Disconnect Supervision enabled so that it can detect when the PSTN disconnects the call. The SIP gateway then sends a BYE command and terminates the call. This applies to both the Audiocodes and Quintum SIP Gateways.

Session Initiation Protocol (SIP) Gateway

10.200 The system supports SIP (Session Initiated Protocol) trunks to reach the CO. SIP trunks allow the system to communicate with the CO via SIP-enabled gateways. As the SIP protocol becomes more and more popular, it is important to be able to communicate to SIP gateways in the IP-centric world.

10.201 The following new SIP gateways are available through the Inter-Tel CommSource® division:

- Quintum® AFT 400 SIP gateway
- AudioCodes™ MP-114 SIP gateway

10.202 These gateways replace the AudioCodes MP-104 SIP Gateway that has gone end-of-life in October 2006. Both gateways support analog Foreign eXchange Office (FXO) ports. The setup procedure is different for each device; the MP-114 gateway uses a Web interface called the AudioCodes Web administrator, while the AFT 400 gateway uses a Configuration Manager program and a Command Line Interface (CLI). Existing Axxess systems must upgrade to v10.0 or later to support the gateways. For more information, refer to the *Inter-Tel Axxess/5000 Quintum SIP Gateway Installation Manual* (document part no. 835.3123) or the *Inter-Tel Axxess/5000 AudioCodes MP-114 SIP Gateway Installation Manual* (document part no. 835.3202).

10.203SIP trunks support the following functionality:

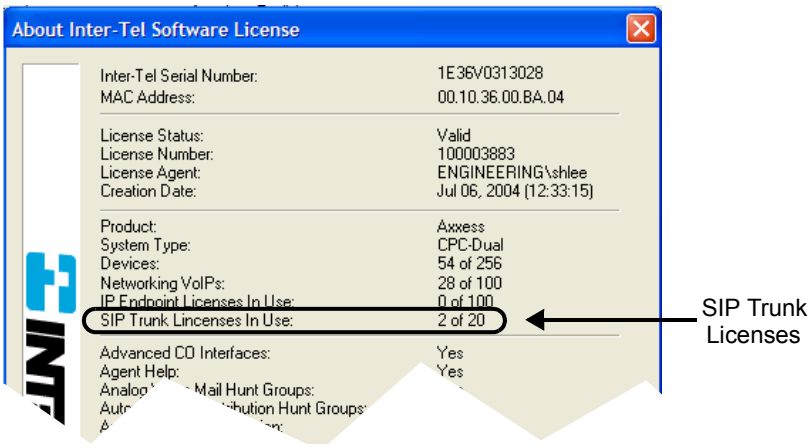
- They are transparent to the system user because SIP trunks work like any other CO trunk in the system.
- They support transferring trunks, putting trunks on hold, and connecting trunks to conferences, similar to other CO trunks in the system.
- They support making and receiving calls by any endpoint.
- They support peer-to-peer audio by IP endpoints.
- They reside in CO trunk groups just like other trunks so that SIP trunk calls can be routed using Automatic Route Selection (ARS).
- They support 911 calls (like MGCP trunks).

10.204SIP trunks require:

- An IPRC with v8.1.x or later firmware.
- A SIP trunk software license (part number 840.0267).

NOTE: SIP trunks do not support the Direct Inward System Access (DISA) or speed dial features.

10.205Licensing: The **About Inter-Tel Software License** information in DB Studio displays the current SIP trunks licenses as well as the SIP trunk licenses in-use as shown below.



10.206 If a new license is uploaded to the system that has more SIP trunks than the old license, the system will not reset, however, any SIP trunks that were previously programmed without an available license on an IPRC will not come online until that particular IPRC is reset. If the license is downgraded and the number of SIP trunks licenses is less than the number of currently in-use SIP trunks in the system, the system will reset and all active calls will be dropped. If a SIP trunk attempts to come online without a license, alarm 127 appears on administrator endpoints in the system and a message print is generated (see page 7-17).

10.207Installing, Programming, and Troubleshooting: For instructions about DB Studio programming for SIP trunks, see page 6-327. For SIP trunks troubleshooting information, see page 7-61.

R. DAY AND NIGHT MODES

10.208 There are separate lists in the database for class-of-service, outgoing-access, allowed-answer, emergency outgoing access, and ring-in assignments for day and night modes. When an administrator enters the Night Ring feature code, the system uses the night mode lists. When the system is in day mode, the day lists are used. (Active calls are not affected when the mode is changed.)

10.209 Night mode also affects the night transfer relays on the Options Card (OPC). The relays are activated when the system is placed in night mode. See page 2-67 in SPECIFICATIONS for details.

10.210 Administrators can place any or all remote nodes into night mode or day mode. The default feature code for Enable Network Night is 9861. The default feature code for Enable Network Day is 9862.

NOTE: The Night Ring On/Off feature code (9860) affects only the node on which the administrator resides. See page 5-273 for a complete explanation of administrator features.

S. DIRECT INWARD SYSTEM ACCESS (DISA)

NOTE: SIP trunks do not support the Direct Inward System Access (DISA) feature.

10.211DISA is a programmable feature that allows an outside party to dial into the system from an external DTMF telephone and then dial extension numbers, hunt group pilot numbers, and off-node device extensions. (DISA callers do not have access to outgoing trunks or page zones.)

10.212 Any of the trunk groups can be programmed to receive incoming DISA calls in day and/or night mode. When not in use for DISA, the trunk group can be used for placing outgoing calls by stations with outgoing access permission.

NOTE: Due to the natural characteristics of the trunk, the volume level of DTMF tones transmitted over the trunk may be substantially reduced before reaching the phone system. This natural degradation in tone volume may adversely affect the reliability of the DISA feature. Other factors which can affect DISA performance are trunk noise and the quality and strength of the DTMF tones generated by the off-premises phone itself. If the system cannot recognize a DTMF digit, the call is automatically sent to the primary attendant.

10.213 When a DISA user calls a station extension number, the call rings as a direct ring-in call, even if the called station is busy or in do-not-disturb. The DISA caller hears music or ringing (determined by a system-wide option in DB Programming) until the Transfer Available or Transfer Busy timer expires. Then, if the call is not answered, it recalls the called station's attendant. If the called station is forwarded, the call follows the programmed forward.

10.214 When a hunt group pilot number receives a call through DISA, the call rings or circulates according to how the hunt group is programmed (e.g., linear or distributed).

10.215 If a caller dials a valid hunt group pilot number that has no members assigned to it, the call automatically rings at the primary attendant station until the Abandoned Call timer expires. If the call is not answered before the Abandoned Call timer expires, the call is disconnected.

Security Codes

10.216DISA trunks can be assigned security codes (up to eight digits in length) that are required for access to the system. The installer can program separate codes for each DISA trunk group to be used during day and/or night modes.

NOTE: To prevent unauthorized access to the outgoing trunks, ***all*** trunk groups using DISA should have a security code.

10.217 During DB Programming, the programmer can determine the number of times that a caller may unsuccessfully attempt to enter a security code and/or dial an extension number. If the user does not succeed within the determined number of attempts, the call will be disconnected (if the security code is invalid) or transferred to the primary attendant (if the extension number is invalid).

Using DISA

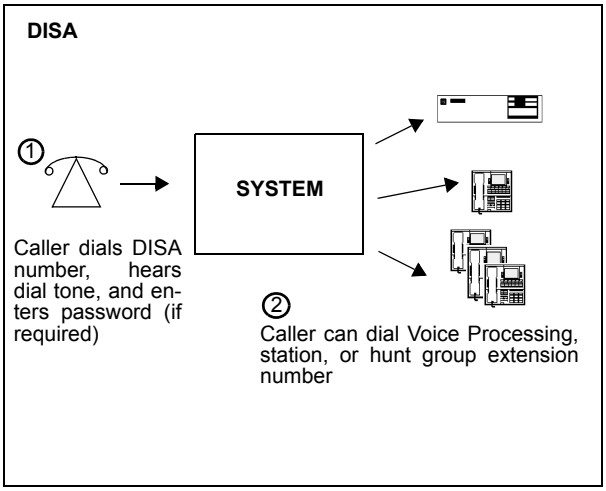
NOTE: If DTMF decoders are unavailable when a DISA call is received, the incoming DISA call is automatically sent to the primary attendant.

10.218 To use DISA:

1. From a DTMF telephone, dial the telephone number of the DISA trunk. When the call is answered by the system, you hear system intercom dial tone.

   If you hear a single progress tone, enter the appropriate (day or night) DISA security code, followed by pound [hash] (#).

2. Dial an extension number or a hunt group pilot number. (To call the primary attendant, dial 0.) You hear music or ringing until the call is answered.



11. PHANTOM DEVICES

11.1 Phantom device functionality has been enhanced in v11.0. The new phantoms are fully functional virtual devices on the system. The following scenarios describe phantom configuration uses. You can configure a phantom device as a:

- **Call-Park Extension**: You may set up one or two (or more) phantoms to be extensions at which a call can be parked.
- **Stand-Alone Device**: Users could have a cell phone to use when they are not in the office but still need to be on the office switch. You could program a phantom for them and the user (or you) could forward the phantom to the cell phone, using Unified Communicator® (UC). That way, callers could still call an extension on the office switch and be able to get the user on his or her cell phone. In the extreme case, all devices on the switch could be phantoms for users with cell or other phones that are not physically connected to the switch.
- **Monitoring Device**: You may put a phantom device in a hunt group and program an endpoint button to map to that phantom extension. Therefore, a user can monitor when the phantom is ringing, etc., and even answer the call either through a reverse transfer or a secondary extension key.
- **General Mailbox**: Phantoms can have associated mailboxes. When a caller calls the phantom, if the phantom is set up to go to voice mail, the caller can leave a message in this "general" mailbox. However, this configuration is not much different from an unassociated mailbox, with the exception that a phantom mailbox can be accessed off-node (whereas an unassociated mailbox cannot). Note that unassociated mailboxes can be accessed off-node, if a caller calls the appropriate voice mail application and enters the unassociated mailbox extension.

11.2 Phantoms can function with UC to perform advanced call routing tasks without the need for a real desk phone. They can also have a true status, such as idle, Do-Not-Disturb (DND), ringing, etc. so that they could be placed in hunt groups and actually ring. The statuses that do not apply to phantom devices are as follows:

- Unplugged (phantoms have no hardware and will always be online)
- Busy/Permanently Busy (phantoms cannot answer a call)

11.3 The user must have UC or another OAI application (for example, Contact Center Suite or Attendant Console) to use the phantom feature. Because the phantoms do not have corresponding hardware devices, all programming that cannot be done in DB Programming must be done from OAI/UC.

11.4 An OAI application can have phantoms perform almost all operations as an endpoint. The exceptions are listed below. These exceptions must be enforced because phantoms do not have voice paths. Phantoms cannot:

- Answer calls from an OAI application.
- Make calls from an OAI application.
- Be in a conference.
- Reverse-transfer calls.

11.5 Some OAI applications, such as Attendant Console, can send feature code requests to devices. The following feature codes (U.S./Europe) are not available for phantoms:

- Answer (default fc 351)
- Auto Line Answer (350)
- Barge-In (386)
- Change Language (301)
- Conference (5)
- Emergency Calls (911/999)
- Group Listen On/Off (312)
- Handsfree On/Off (319)
- Headset Off (316)
- Headset On (315)
- Headset On/Off (317)
- Hookflash (330)
- Background Music On/Off (313)
- Page (7)
- Page On/Off (325)
- Ring Tone Selection (398)
- Queue Request (6)
- Record-A-Call (385)
- Redial (380)
- Reverse Transfer (4)
- Seize Device (9973/9173)
- Steal (387)
- System Hold (335)
- Transfer to Ring (346)
- Transfer to Hold (346)

11.6 Phantoms use up the same software resources as a regular endpoint or IP endpoint, with the exception of the voice paths (phantoms do not have voice paths). Therefore, they can have an impact on system performance. The default capacities for phantoms are shown in the following table.

Table 5-26. *Phantom Devices Default Capacities*

System Type	Maximum Phantoms
CPC	64
CPS	256

11.7 Phantoms in v11.0 or later DB Programming are defaulted to the not-in-DND status. When upgrading an old database to v11.0 or later, if there were phantom devices in the old database, their DND statuses will not be changed after the conversion.

11.8 See the related topics:

- Programming (see page 6-196)
- Troubleshooting (see page 7-50)

12. STATION INSTRUMENTS

NOTE: For details about installing and programming analog phones and Executone® IDS and Wave phones, see a version 7.0 or earlier *Installation and Maintenance Manual*.

12.1 To allow system flexibility and cost efficiency, a variety of station instruments can be used on the Inter-Tel System. They are:

- 8-, 12-, and 24-Line AIM Analog Phones
- 8-, 12-, and 24-Line Inter-Tel/DVK Analog Phones
- 12-, and 24-Line GMX Analog Phones
- 8-, 12-, and 24-Line IMX Analog Phones
- IDS Model 160

NOTE: The system does not support the Model 160's serial port, which IDS customers typically use for TAPI applications. Customers may use System Open Architecture Interface (OAI) or the Professional or Associate Display Phones instead.

- IDS Model 64 (not applicable to Europe)
- IDS Model 32 (not applicable to Europe)
- IDS Model 18 (not applicable to Europe)
- IDS Model 12 (not applicable to Europe)
- IDS Agent Terminal (not applicable to Europe)
- 29-Key Wave Phone (not applicable to Europe)

NOTE: The 28-Key Wave Phone is not supported.

- 17-Key Wave Phone (not applicable to Europe)
- 6-Key Wave Phone (not applicable to Europe)
- Agent Terminal (not applicable to Europe)

NOTE: Similar to the Model 160, the system does not support the Agent Terminal's serial port. Also, only the primary headset jack is supported.

- Axxess Basic Digital, Standard Display, and Executive Display Phones
- Eclipse Basic Digital, Associate Display, and Professional Display Phones (not applicable to Europe)
- Model 8500, 8520, and 8560 Phones
- Digital DSS/BLF Unit – single or tandem
- Mini-DSS

- Industry-standard single-line, dual-tone multi-frequency (DTMF) or pulse-tone sets (Version 2.x or later software is required for pulse-tone sets)

NOTE: The volume adjustment range of non-AIM and non-DVK analog phones may not be suitable in all user environments. In such cases, Inter-Tel recommends using AIM and/or DVK phones, if possible.

- Inter-Tel hard and soft IP and multi-protocol endpoints (See the *IP Devices Installation Manual* for information about IP endpoints.)

12.2 Digital Keyset Cards (DKSC or DKSC-16) are used to support Axxess Basic, Eclipse Basic, Standard Display, Associate Display, Executive Display, Professional Display, Model 8500, Model 8520, and Model 8560 Phone stations. Digital Keyset Card circuits can be equipped with a Single-Line Adapter (SLA) and programmed to support two single-line stations (instead of a phone). DSS/BLF Units are connected to Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phones that are equipped with PC Data Port Modules (see page 5-146 for details). The DKSC has eight circuits, and the DKSC-16 has 16 circuits.

12.3 Analog Keyset Cards (AKSC) support the analog AIM, DVK, GMX, ESP, and IMX phones. An Analog Keyset Card can be programmed to support dual-circuit 8-line phones on a single phone circuit.

NOTE: GX Phones (and analog DSS/BLF Units in Europe) are not supported on the phone system.

12.4 Each Executone Keyset Card (EKSC) provides up to 12 circuits for connecting Executone phones and 48-key DSS Consoles to the phone system.

12.5 The Single-Line Card (SLC) provides up to eight circuits for connecting on-premises AC-ringing single-line DTMF or pulse-tone sets and playback devices. (Version 2.x or later software is required for pulse-tone sets.) The SLC-16 can support up to 16 on-premises AC-ringing single-line devices.

12.6 System software will support any configuration of station instruments with the limitation that there must be a minimum of one keyset card. The actual configuration is defined by the type of cards and the number of circuits in use on the system.

A. SPEAKERPHONES

NOTE: The network allows handsfree intercom calls when calling from one node to another, unless the call camps on to the node before being sent to the other node.

Executive Display and Professional Display Phone Speakerphone

12.7 Each Executive Display and Professional Display Phone has a built-in, integrated speakerphone that allows on-hook dialing and handsfree operation on outside calls and inside (intercom) calls. The Executive Display and Professional Display Phone speakerphone can be used in standard mode or in enhanced mode which allows for full-duplex operation over a limited volume range (the range depends on trunk characteristics, typically the range is 1 to 3). Enhanced speakerphones are explained on the next page.

12.8 The Executive Display and Professional Display Phone's speaker can also provide background music (if an external music source is installed) and is used to receive pages.

12.9 A programming option can be used to disable the phone's speakerphone capabilities (see page 6-211 in PROGRAMMING). Or, the user can disable the speakerphone for intercom calls using the Handsfree On/Off and/or Ring Intercom Always On/Off feature codes as described on page 5-174.

NOTE: On certain handsfree-to-handsfree intercom calls, voice volume levels may cause feedback to occur. If this happens; one station user should pick up the handset. Two Display Phones cannot be in enhanced mode while on a call together; one phone must be off hook to allow the other phone to enter enhanced speakerphone mode. If both phones are on hook, there is no benefit to having one phone in enhanced mode.

Other Digital and Executone Phone Speakerphones

12.10 Standard Display, Associate Display, Axxess Basic Digital, Eclipse Basic Digital, Model 8520, Model 8560, and Model 8500 Phones have a built-in speaker and microphone that allow the user to answer intercom calls handsfree, dial while on hook, monitor calls, and listen to background music.

12.11 Standard Display, Associate Display, Axxess Basic Digital, Eclipse Basic Digital, Model 8520, Model 8560, and Model 8500 Phones can be programmed in individual station programming to allow the use of system resources for standard and enhanced speakerphones. Enhanced mode allows for full-duplex operation over a limited volume range (the range depends on trunk characteristics, typically the range is 1 to 3). Enhanced speakerphones are explained on the next page.

12.12 To provide standard and enhanced speakerphone resources for Standard Display, Associate Display, Axxess Basic Digital, Eclipse Basic Digital, Model 8520, Model 8560, and Model 8500 Phones, the system must be equipped with DKSC-16/DKSC-16+ cards and/or at least one Options Card. Each DKSC-16/DKSC-16+ provides DSP circuitry for up to 10 switched speakerphones for Standard Display, Associate Display, Axxess Basic Digital, Eclipse Basic Digital, Model 8500, Model 8520, and Model 8560 Phones. The Options Card DSP circuitry must be designated in DB Programming for system-wide speakerphone resources. (There are up to four DSPs on each Options Card. Each DSP can be programmed to support up to eight speakerphones. See page 6-118 in PROGRAMMING.) DKSC-16/DKSC-16+ and Options Card speakerphone resources are available system-wide to Standard Display, Associate Display, Axxess Basic Digital, Eclipse Basic Digital, Model 8560, Model 8520, and Model 8500 Phones on a first-come, first-served basis.

12.13 Executone IDS Keyset Models 160, 64, 32, 18, and the Wave phones have a speaker and a microphone for handsfree intercom reply. They do not need any additional speakerphone system resources. Executone IDS Keyset Model 12 keysets and the Agent Terminal have only a speaker. They do not have a microphone.

12.14 Each time a speaker/microphone keyset user presses the Speaker button while on hook, the standard switched speakerphone is activated, and the DSP resource (DKSC-16/DKSC-16+ or Options Card) is assigned to that phone for the length of the call. If a phone user requests the speakerphone function when DSP speakerphone resources are not available, the user will hear reorder tones, and the display will show NO SPKRPHN AVAIL STAY OFF HOOK. If the user attempts to change to the speakerphone and hangs up during a call when speakerphones are unavailable, the call will be forced into a monitor-only mode. The user will hear the other party but will not be allowed to transmit to them. In this mode, the user's MUTE button will be lit, and the display shows NO SPKRPHN AVAIL COME OFF HOOK. Pressing the lit MUTE button will result in a burst of reorder tone; you must lift the handset to speak.

Enhanced Speakerphones for Digital Phones

NOTE: In general, IP phones and phones that connect to IP calls (i.e., calls that are routed through an IPRC) cannot enable enhanced speakerphone mode. However, sometimes the system may not detect that a call is connected over an IP network. If this occurs, the system may allow users to enable enhanced speakerphone mode, which will affect the call quality.

12.15 To use an enhanced speakerphone on a digital phone, the user establishes a call and presses ∞ or SPCL and then the Speaker button (or presses ∞ or SPCL and enters the Enhanced Speakerphone feature code – 310). There will be about one second of white noise at this point. (Both parties should avoid speaking during this white noise.) This white noise is used to calibrate the echo-canceling circuitry. After the white noise, there will be a confirmation tone (Executive Display and Professional Display Phone users will hear the tone, but the other party will not. On a Standard Display, Associate Display, Axxess Basic Digital, Eclipse Basic Digital, Model 8520, Model 8500, or Model 8560 Phone, both the user and the other party will hear the tone.) Then the user will be connected in an enhanced handsfree call. If the speakerphone cannot be calibrated correctly, due to trunk conditions, the enhanced speakerphone will not be enabled and the display will show CANNOT ENABLE ENHANCED MODE.

NOTE: It is normal for an outside party to hear a minimal amount of his or her own echo when speaking to a digital phone in enhanced speakerphone mode. This is usually acceptable to most callers on local calls and on most long-distance calls. However, on calls established using an IP Networking Module or IPRC and some T1 and long-distance calls involving significant delays (for example, cross-country T1 calls, overseas calls or calls via satellite), the amount of echo may be objectionable and the enhanced speakerphone mode should not be used on these types of long-distance calls, including those connected via an IP Networking Module.

12.16 If an enhanced speakerphone call is placed on hold and then taken off hold on an Executive Display and Professional Display Phone, it remains an enhanced speakerphone call. If it is placed on hold at a Standard Display, Associate Display, Axxess Basic Digital, Eclipse Basic Digital, Model 8520, Model 8560, or Model 8500 Phone, the call is returned to standard speakerphone mode.

12.17 A station can only have one enhanced speakerphone call. For example, if an Executive Display or Professional Display Phone user makes an enhanced speakerphone call, puts it on hold, and establishes another enhanced speakerphone call, the first call will be turned back into a standard speakerphone call.

12.18 A digital phone user can set up a conference and then, once it is established, press ∞ and the Speaker button (or press ∞ and enter the Enhanced Speakerphone feature code – 310) to have an enhanced speakerphone call in the conference.

NOTE: On certain handsfree-to-handsfree intercom calls, voice volume levels may cause feedback to occur. If this happens, one station user should pick up the handset. Two Display Phones cannot be in enhanced mode while on a call together; one phone must be off hook to allow the other phone to enter enhanced speakerphone mode. If both phones are on hook, there is no benefit to having one phone in enhanced mode.

Analog Phone Speakerphones (Europe Only)

NOTE: The volume adjustment range of non-AIM and non-DVK analog phones may not be suitable in all user environments. In such cases, Inter-Tel recommends using AIM and/or DVK phones, if possible.

12.19 Every Analog Phone station has a built-in, integrated speakerphone that allows the user to dial while on hook and allows handsfree operation on outside calls and inside (intercom) calls. In addition, the Analog Phone speaker may provide background music (if an external music source is installed) and is used to receive pages.

12.20 A programming option can be enabled that disables the speakerphone (as described on page 5-174), or the user can disable the speakerphone using a feature code. This prevents the station from using the speakerphone on outgoing intercom calls and all outside calls. If the option is enabled, the speakerphone can still be used for receiving handsfree intercom calls, but the user must lift the handset to speak when placing intercom calls and when placing or receiving outside calls. The MUTE button is lit when the speaker is activated for outside calls or outgoing intercom calls. Call monitoring and on-hook dialing are not affected.

12.21 Analog Phones cannot use the shared speakerphone resources or the Enhanced Speakerphone option that is provided for the Digital Phones.

NOTE: On certain handsfree-to-handsfree intercom calls, voice volume levels may cause feedback to occur. If this happens, one user should pick up the handset.

Speakerphone Best Practices

12.22 Keep in mind the following guidelines when using the standard mode:

- Talk within four feet of your phone.
- Keep your phone at least four feet away from noise generators (e.g., PC fans). Speakerphones perform better in a quiet office environment.
- Use any angle your model 8500-series endpoint supports as the angle makes very little difference to levels and performance even though the microphone is “hidden” when vertical.
- Avoid using the speakerphone as a “conference room” speakerphone (e.g., a Polycom conference phone). Conference room speakerphones, such as a Polycom, typically have as many as three to five microphones that stretch 10 feet or more end-to-end to pick up the voices of all people in the room. An Inter-Tel endpoint has just one microphone and works best when you and or other parties are within four feet of the phone. If you are in a conference room where there are multiple parties further than four feet from the phone, it is best to use a stand-alone speakerphone designed specifically for that purpose.

- Avoid using the standard mode when the far-end party is in a "noisy" environment—typical with cell phone callers.
- Avoid using the standard mode when the call is a conference call with multiple parties because the likelihood of one or more of the parties having background noise is high.
- Standard speakerphone operation is half-duplex or "switched" and switches to the "loudest" party. Half-duplex means that if both parties talk at the same time, only one party (the loudest party) will be heard.
- The far-end talk "level" is based on your speaker volume setting (1 to 8). The louder you set the volume, the more difficult it becomes to interrupt the far-end party.
- As a general rule, the far-end talk level is "louder" than the near-end talk level giving the far-end party priority when talking.
- The farther away you are from the phone, the lower the talk level.
- The speakerphone adjusts to ambient "noise" level in the room, which is why you need to keep the phone away from noise generators (such as PC fans) and talk within four feet of your speakerphone.

12.23 Keep in mind the following guidelines when using the enhanced mode:

- Follow the same guidelines as those listed for the "standard" speakerphone, and, in addition to the guidelines for an enhanced speakerphone.
- Inform the far-end party about the burst of white noise before you enter the enhanced mode.
- Use volume levels 1-3 for full-duplex operation.
- Avoid using the enhanced mode on calls with significant delay.
- Avoid using the enhanced mode on volume levels 4 through 8 since the speakerphone transitions back to half-duplex operation.
- The enhanced speakerphone provides full-duplex operation at lower volume settings (1 through 3) but transitions to half-duplex operation at higher volume settings (4 through 8). Full-duplex means both parties can talk at the same time and hear each other.
- The enhanced mode does not perform acoustic echo cancellation, so the far-end parties always hear their own acoustic echo.
- The far-end party will hear all near-end background noises.
- The far-end party must be on a handset for full-duplex operation.

B. LIQUID CRYSTAL DISPLAY (LCD)

Menu Keys

12.24 Some phones have feature-related menus that change with the function being performed by the phone (intercom call, outside call, programming, etc.). These lines have menu selection keys next to them or below them, that are used for menu features.

12.25 All Executive Display, Professional Display, and Model 8560 Phones have a six-line display. Each line has 16 characters. (Illustrations of the Executive Display Phone, Professional Display Phone, and Model 8560 displays are shown on the following page.) The top two lines are for standard call information and show the same displays as other display phones. The remaining four lines are used for displaying menu options. These lines have menu selection buttons on both sides (eight buttons) that are used for menu features. When there are two prompts on a line, like PREVIOUS and NEXT, the button next to the word is the one that will select that feature. When there is only one prompt on a line, like DO NOT DISTURB, then pressing the button on either side of the prompt will select it.

12.26 There is a system-wide flag, called "Alternate Keyset Menu Displays," that determines the display that Executive Display, Professional Display, and Model 8560 Phone users will see during an outside call. In the default state, the display shows the menu (shown right) with Transfer-To-Hold on the first menu and Flash (Recall) on the "More..." menu.

HOLD	SYS HOLD
TRANSFER	CNF
TRANSFER-TO-HOLD	
MUTE	MORE...

12.27 If the Alternate Keyset Menu Displays flag is enabled, the Executive Display, Professional Display, and Model 8560 Phone users will see the menu with Flash or Recall (shown right) on the first menu and Transfer-To-Hold in the "More..." menu.

HOLD	SYS HOLD
TRANSFER	CNF
FLASH/RECALL	
MUTE	MORE...

NOTE: Version 7.0 or later system software is required to support the menu display and buttons on the Model 8560.

12.28 Enabling the Alternate Keyset Menu flag allows Centrex (STAR) features to be accessed more easily, using the **FLASH** (**RECALL**) menu option.

Figure 5-3. *Executive and Professional Phone Display*

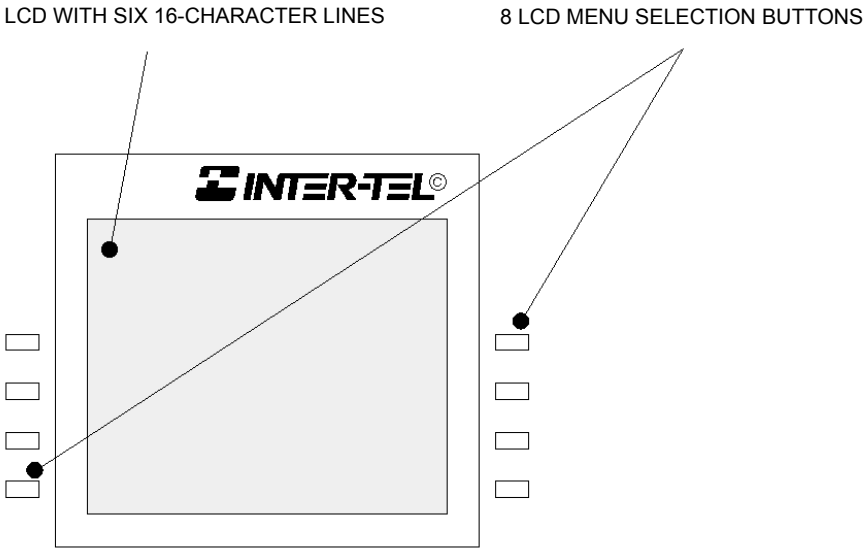
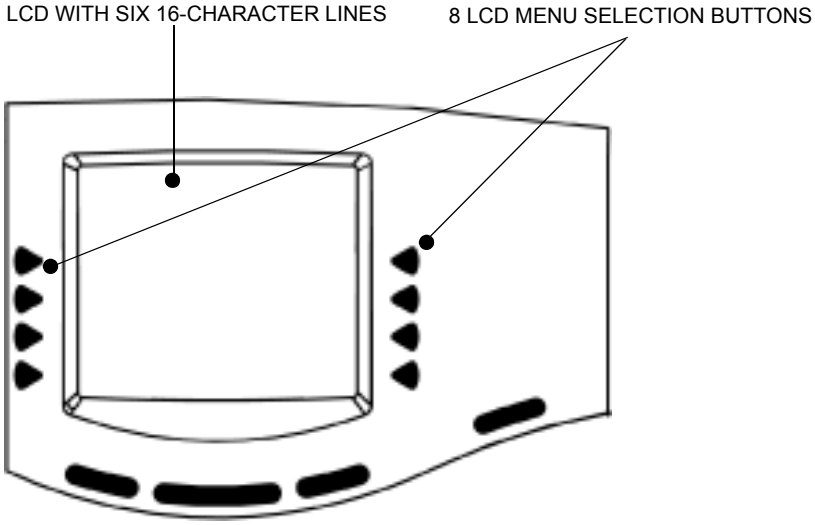


Figure 5-4. *Model 8560 Phone Display*



Two-Line Phone Liquid Crystal Display (LCD)

12.29 Analog phones, some Executone IDS and Wave Phones, and Standard Display, Associate Display, and Model 8520 Phones have two-line displays with 16-characters per line.

12.30 Older phone types are available in standard or display model. The non-display phones can be converted to display phones by installing LCD Units. Installation instructions are in the INSTALLATION section of the manual. (Model 8520 Phones are not available in a standard model.)

12.31 When the phone is not in use, the LCD shows the station extension number, user name, time of day, and date. Other displays include reminder messages, do-not-disturb messages, numbers dialed, call sources, elapsed time of calls, current call costs, error messages, station status, programming messages, etc. The displayed information is described throughout the instructions in this section of the manual.

LCD Contrast Feature

12.32 The LCD contrast is adjustable to eight different settings for the following endpoints:

- Model 8662
- Model 8622
- Model 8620
- Model 8660
- Model 8560
- Model 8520 (adjusted with a contrast control wheel)

NOTE: The LCD Contrast feature is supported on version 7.0 or later systems only.

12.33 To adjust the LCD contrast on a Model 8662, 8622, 8620, 8660, and 8560 phone:

1. While the phone is idle, press the Special button (∞) and enter the LCD Contrast feature code (303).
2. *EITHER,* Select **LIGHTER** or **DARKER** on the menu buttons.

   OR, Use the Volume button to make your selection.

   OR, Press the desired level (1-8).
3. Press the Speaker button or **ACCEPT** to save your change.

12.34 The Model 8520 Phone has a contrast control wheel that adjusts the contrast level. This wheel is located on the back of the phone and can be adjusted while viewing the display.

12.35 To adjust the LCD contrast on a Model 8520 phone:

1. While the phone is idle, locate the contrast control wheel on the upper-right section of the phone.
2. Adjust the contrast of the LCD screen by turning the wheel clockwise (lighter) or counterclockwise (darker).

Extension Number, Username, Time and Date Display

12.36 When a display phone is idle, the extension number, username, time, and date are displayed. While on a call, in do-not-disturb, or unconditionally forwarded, other displays are shown. In these circumstances, use the following procedure to redisplay the date, time, user name, and extension number for five seconds.

12.37 To display the date, time, username, and extension number:

Press the Special button (∞) and enter the Display Time/Date feature code (300). The date, time, username, and extension number are displayed for five seconds.

Node and Circuit Number on Power-Up

12.38 To aid in installation and troubleshooting procedures, display phones show their station identification for five seconds when power is turned on, when the system is reset, or when the phone line cord is first plugged in (or removed and replaced). The extension number and assigned user name appear on the top line of the display, and the node number and circuit number appears on the bottom line (for example, circuit 1.1.1 on node 2 would appear as 2:1.1.1). After the five-second identification display, the phone returns to the appropriate display, depending on its current status.

All Transient Displays Station Flag

12.39 The "All Transient Displays" station flag, when enabled, will allow all transient call handling displays to appear on the phone display. If it is disabled, the CALL TRANSFERRED TO XXXX and MESSAGE LEFT FOR XXXX transient displays will not appear. (See page 6-209 in PROGRAMMING.)

12.40 Each transient display is shown for five seconds before another display can take its place. Disabling the flag allows high-traffic stations (such as attendants) to change displays more quickly after a message is left by the station user or after a call is transferred.

12.41 This station option cannot be programmed by the user. It is a database-programmable flag.

DKTS Alternate Transient Display Timer Station Flag and Timer

12.42 As an alternative to (or in addition to) disabling the call transfer and message transient displays, as previously described, the programmer can use a combination of a station flag and a timer to control ***all*** of the station's transient displays (not just the call transfer and message displays). The "DKTS Alternate Transient Display Timer" station flag, if enabled, allows all transient displays to be controlled by the "DKTS Alternate Transient Display" timer. (See page 6-211 in PROGRAMMING.)

12.43 The "DKTS Alternate Transient Display" timer can be set to 1-255 tenths of a second (0.1-25.5 seconds). The default value is 10 tenths of a second (1 second). This timer affects only those phone stations with the "DKTS Alternate Transient Display Timer" flag enabled.

Transient Call Indication on Call Answer Station Flag

12.44 This programmable station flag determines if the station user will see a call display when answering a call by pressing a secondary extension button or by reverse transferring. If enabled, the display will tell if the call was ringing, recalling, transferred, or holding at the other station.

C. FEATURE KEYS

Hot Keypad Keys vs. SPCL Key Required for Feature Code Entry Flag

12.45 Unless programmed to use the Special button ∞ or SPCL when entering a feature code, all phones have "hot" keypad buttons that allow dialing of feature codes, extension numbers, and trunk access codes without pressing the Special button or lifting the handset. As soon as a keypad digit is pressed, the digit is recognized by the phone system.

12.46 There is a system-wide option called "SPCL Key Required For Feature Code Entry" that, if enabled, requires phone users to always press the Special button before entering a feature code.

12.47 Regardless of programming, if a phone user presses the Special button while on hook, the display shows ENTER FEATURE CODE. The user can enter the desired code or press a menu button.

Keymaps

12.48 A keymap group determines the layout of the buttons for all the stations assigned to that group. If any of the buttons in the keymap group are user-programmable or undefined, they can be changed by the station user. There can be up to 250 different keyset keymap groups (when using a 1024 or ATM system). Each node contains its own keymap programming. See PROGRAMMING, page 6-220, for more details. (There are also 20 maps for DSS/BLF Units.)

12.49 Each phone is assigned a standard keymap and can also have an alternate keymap. See page 5-137 for information about switching between standard and alternate keymaps.

12.50 When the system is in the default state, all phones are assigned to the default Keyset Keymap 1. There are no individual pre-programmed keymaps for each of the specific phone types. (Previous software had an Executive/Standard map and a Basic map. If you convert from a previous version to the current version, those keymaps will be preserved.) Keymap #1 also acts as the template for new keymaps, regardless of the phone type. For example, if keymap #1 is an Executive Display keymap with the default Hold button programmed as a Forward button, any newly created keymaps that have a Hold button, even Standard Display keymaps, will reflect this change. Because keymap #1 is used as a template, do not change this keymap. The DSS keymaps do not use a template.

12.51 As part of the installation procedure, the installer must make sure the phone buttons are labeled to match their functions.

12.52 The buttons in each keymap can be assigned the following functions:

- **Accept Key**: *(This button type is not required for digital phones. It is intended for analog phone use.)* This button is used to accept an entry when using numeric mode. (Digital phones use the Volume button, by pressing both ends at once.)
- **Cancel Key**: No further programming is necessary. This button will be used to cancel a feature or cancel an entry when in numeric mode.

- **Call Key**: Up to 10 CALL buttons can be assigned in each keymap. These buttons are used for call access. They are not associated with specific trunks. Instead, they are assigned to calls in the order that the calls are placed or received. In the following example, the user has three CALL buttons.

Table 5-27. *Call Keys*

Action	Call Key 1	Call Key 2	Call Key 3
User places call #1	Shows status of call #1		
Call #2 rings in while call #1 is active	Shows status of call #1	Shows ring flash for call #2	
Calls #1 and #2 on hold, call #3 placed	Shows hold flash for call #1	Shows hold flash for call #2	Shows status of call #3

If there are more calls in progress than there are CALL buttons (e.g., if another call rings in to the phone in the example), the new call will not have an available call button and will camp on to the station until a CALL button becomes available; it will then appear under the available button. Pressing an idle CALL button will enter the programmed Outgoing Call feature for that station. If there is no IC button programmed in the keymap, intercom calls will appear under CALL buttons.

NOTE: All keymaps should have at least one CALL button if there is not an individual trunk button for every trunk.

- **Down Key**: *(This button type is not required for digital phones. It is intended for analog phone use.)* No further programming is necessary. This button can be used scrolling backward through displays. When the phone is idle, it can be used for adjusting volume, in place of the Volume button. (Digital phones use the Volume button.)
- **DSS/BLF Key**: A DSS/BLF button will dial the associated extension number when pressed. If assigned to a button with a lamp, it will also show status of the associated station, as described for DSS/BLF Units on page 5-152. DSS/BLF buttons cannot be changed by the station user.
- **Feature Key**: These are programmed with feature codes. The feature code appears in the same button location on all stations assigned to the keymap and cannot be changed by the phone user.
- **Forward Key**: The forward button can be programmed with any of the Call Forwarding feature codes. The forwarding code used by the button can be changed by the phone user.
- **Hunt Group Key**: Like a DSS/BLF button, this button dials a hunt group pilot number when pressed and, if assigned to a button with a lamp, will show the status of that hunt group.
- **Individual Trunk Key**: This button is associated with one trunk. It can be used for placing outgoing calls, and for picking up calls on hold and answering incoming calls on that trunk.

- **Intercom (IC) Key**: The IC button allows access to intercom channels for placing and answering calls. If an IC button is not assigned, an intercom call will appear under a CALL button.
- **Next Key**: *(This button type is not required for digital phones. It is intended for analog phone use.)* This button is used for scrolling forward through displays. (Digital phones use the Volume button.)
- **Page Zone Key**: A page zone button is programmed with one of the page access codes (9600-9609 or 9600-9649 depending on the software version) for quick access to page zones (0-9 and 0-49 respectively).
- **Previous Key**: *(This button type is not required for digital phones. It is intended for analog phone use.)* This button is used for scrolling backward through displays. (Digital phones use the Volume button.)
- **Programmable Key**: Up to 35 user-programmable buttons can be designated and given default values in the keymap database. They can then be reprogrammed to enter any of the trunk group access codes, extension numbers, or feature codes by the individual phone users as desired. Or, they can be programmed as system or station speed-dial bins, or as secondary extension buttons (see below). The phone user or programmer can return the buttons to default values at any time.
- **Save Key**: *(This button type is not required for digital phones. It is intended for analog phone use.)* This button is used for saving volume levels or programmed phone features. (Digital phones use the Volume button, by pressing both ends at once.)
- **Secondary Extension Key**: This type of button allows a station user to have one-button access to calls ringing or holding at another station. It should be assigned to a button with a lamp so that the secondary extension button can show the call status. If desired, a station can have several secondary extension buttons, each associated with a different station. For details, see page 5-139.
- **SPKR Key**: This button is used for turning the phone speaker or headset on and off.
- **Speed-Dial Keys**: Each keymap can have up to 10 speed-dial buttons. Each speed-dial button can be programmed by the phone user to dial one digit string. If assigned to a button with a lamp, the lamp will show the status of an extension number if it is associated with that button.
- **Trunk Group Key**: These buttons are associated with specific trunk group access codes and are used for placing outgoing calls on trunks in the selected trunk group. If assigned to a button with a lamp, the lamp will be lit if all trunks in the trunk group are in use.
- **Undefined Key**: Any button can be left undefined if it will not be needed. It can be programmed, like a user-programmable button, as described above. If the phone user or programmer returns the phone to default values, the button will return to being "undefined."

- **Up Key**: *(This button type is not required for digital phones. It is intended for analog phone use.)* No further programming is necessary. This button can be used for scrolling forward through displays. When the phone is idle, it can be used for adjusting volume, in place of the Volume button. (Digital phones use the Volume button.)

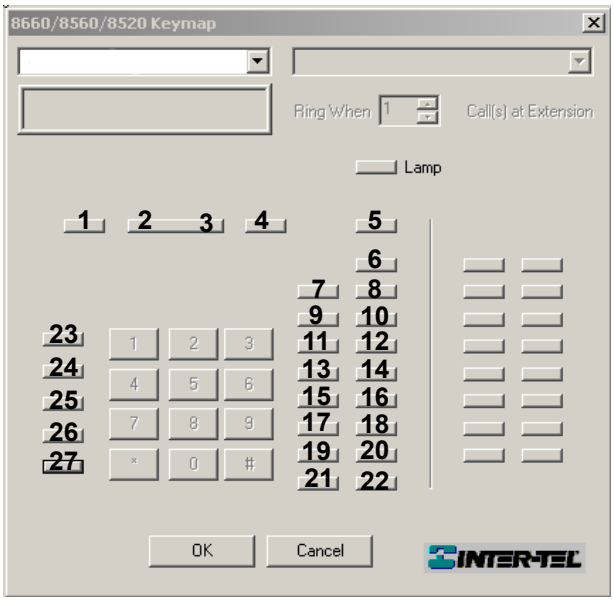
NOTE: The volume adjustment button on the digital phones also can be used for scrolling through displays. However, the analog phone volume buttons are used ***only*** for adjusting volume; analog phones need to have Previous and Next buttons programmed in their keymaps to scroll through displays. They also need an Accept button to accept entries when in numeric programming mode. This does not apply to Executone IDS and Wave Phones.

Model 800 Digital Phones and Button Locations

12.53 Because the Model 8500, 8520, and 8560 Phones have additional buttons and lamps, programming the buttons on a system using a software version prior to 7.0 may be challenging. The figures below provide a cross-reference guide for button locations based on the software version.

12.54 The buttons of the following v7.0 or later keymaps have been assigned a number, and their locations are cross-referenced on the v6.X keymaps. For example, button 5 (Message button by default) on the 8662/8660/8620/8560/8520 keymap corresponds to button 5 (Message button by default) on the Executive keymap.

Figure 5-5. *8662/8660/8620/8560/8520 Keymap (v7.0 and Later)*



NOTE: To program the 8660, 8560, and 8520 (27 buttons) on a system using software v3.0-6.X, use the Executive keymap (24 buttons), as shown in the following figure. Buttons designated with xxx do not correspond to any buttons on the v7.0 and later phones and will not function on earlier software versions.

Figure 5-6. *Executive Keymap (v6.x)*

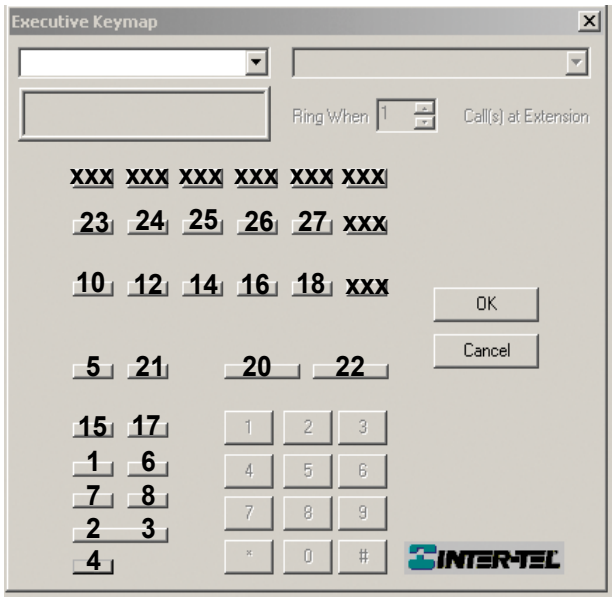
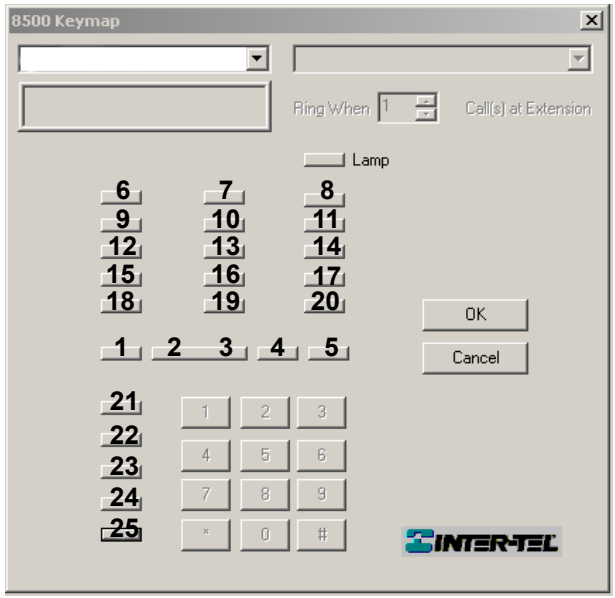
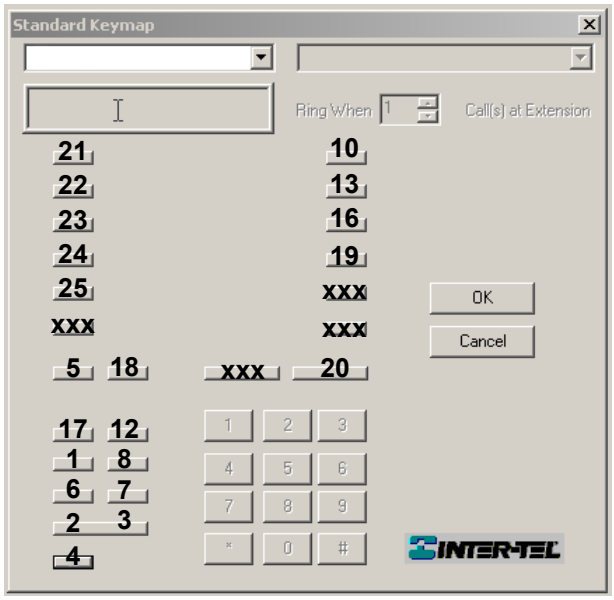


Figure 5-7. *8500 Keymap (v7.0 and Later)*



NOTE: To program the 8500 (24 buttons) on a system using software v3.0-6.X, use the Standard keymap (24 buttons), as shown in the following figure. Buttons designated with xxx do not correspond to any buttons on the v7.0 and later phones and will not function on earlier software versions.

Figure 5-8. *Standard Keymap (v6.x)*



Standard/Alternate Keymap Switching

12.55 In DB Programming, each phone is assigned a standard keymap and can also have an alternate keymap. The user can switch between the keymaps by entering a feature code. (The maps can also be switched in the database using the Alternate Keymap station flag; see page 6-209 in PROGRAMMING.)

12.56 Having two keymaps provides the user with access to more features or trunks. In most situations, the user would have one button programmed with the Switch Keymap feature code (399) in each keymap so that it would be easy to change the keymap. (If the user-programmed button has a lamp, the lamp is lit when the alternate keymap is active.)

12.57 When keymaps are switched, any user-programmable buttons in the maps retain their user-programmed values; they are not returned to default during the switch. However, if the keymaps do not have the same user-programmable buttons in the same positions, the location of individual buttons may change when the keymaps change.

12.58 If a user switches keymaps during an outside call, the call will appear on the appropriate trunk or CALL button for that keymap. The button assignment may not be the same for both keymaps. For example, it could appear under a CALL button on one map and an individual trunk button on the other.

12.59 If a button is not available in the currently active keymap when a call rings in, the user will still see the correct display, hear ringing, and can answer as usual.

12.60 To switch between keymaps when the station is idle:

While on hook, enter the Switch Keymap feature code (399). The display shows STANDARD KEYMAP IS ACTIVE or ALTERNATE KEYMAP IS ACTIVE.

12.61 To switch between keymaps during a call:

While on a call, press ∞ or SPCL and enter the Switch Keymap feature code (399). The display momentarily shows STANDARD KEYMAP IS ACTIVE or ALTERNATE KEYMAP IS ACTIVE.

Message Lamp

12.62 Model 8500, 8520, and 8560 Phones have a Message Lamp located on the upper, right-hand corner of the phone. This lamp is programmable and is defaulted to the Message feature code (365). If you change the feature code associated with this lamp, it will flash when the feature is active. For example, if you program the lamp for Individual Hold (336), the lamp will flash when a call is held at the station.

NOTE: Version 7.0 or later system software is required to support the Message Lamp.

12.63 Professional Display, Associate Display, and Eclipse Basic Digital Phones also have a Message Lamp, but it is not programmable. Instead, it flashes whenever the MSG button flashes.

Speaker Buttons

12.64 On Executive Display, Professional Display, Standard Display, Associate Display, Axxess Basic Digital, and Eclipse Digital Phones, on AIM and Inter-Tel/DVK Analog Phones, and on some Executone IDS and Wave Phones, handsfree mode is controlled by a Speaker (SPKR) button. On Model 8500, 8520, and 8560 Phones, an illustration of a speaker (🔊) represents the Speaker button. On other Executone IDS and Wave Phones, the HF button controls handsfree mode.

Hold Buttons

12.65 On Executive Display, Professional Display, Standard Display, Associate Display, Axxess Basic Digital, and Eclipse Basic Digital Phones, on AIM Inter-Tel/DVK Analog Phones, and on some Executone IDS and Wave Phones, the Hold feature is controlled by a Hold (HOLD) button. On Model 8500, 8520, and 8560 Phones, an illustration of a phone with the handset off-hook (📞) represents the Hold button.

Default Phone Feature Buttons

12.66 When the system is in the default state, all phones are in the same keymap group and have the following feature buttons. Feature code descriptions start on page 5-14. (The Special button is represented by keycaps with ∞ or SPCL on the phone.)

Table 5-28. *Default Phone Feature Buttons*

Button	Function
CALL BUTTONS 1-4	Selects an outgoing trunk, if idle, or answers a call on the button.
IC	Selects a voice channel or picks up a holding or camped-on intercom call.
DND	Programmable Button 1: Do-Not-Disturb On/Off (372)
PROGRAMMABLE BUTTONS 2-6, 14-16, 18, and 21-35	Undefined values. Can be programmed with any feature code, trunk access code, or extension number, or can be programmed as a station or system speed-dial bin.
CNF	Programmable Button 7: Conference (5)
HOLD	Programmable Button 8: Individual Hold (336)
TRANSFER	Programmable Button 9: Transfer to Ring (345)
SYS SPD	Programmable Button 10: System Speed Dial (381)
REDIAL	Programmable Button 11: Redial (380)

Table 5-28. *Default Phone Feature Buttons (Continued)*

Button	Function
PAGE	Programmable Button 12: Page (7)
QUEUE	Programmable Button 13: Queue Request (6)
MUSIC	Programmable Button 17: Background Music (313)
PAGE ZONE 1	Programmable Button 19: Selects Page Zone 1 for making a page.
STN SPEED DIAL	Programmable Button 20: Station Speed Dial (382)
FWD	Call Forward All Calls (355) Also used for canceling Call Forward.
SPECIAL	Signals the system to expect a feature code.
ANSWER	Answer (351)
OUTGOING	Selects a trunk for placing an outgoing call using the station's Outgoing Call feature (8).
SPEAKER	Speakerphone On/Off
MSG	Message (365) – Also retrieves messages when the flashing MSG button is pressed.
MUTE	Microphone Mute On/Off (314) (This is a fixed-function button on digital phones.)
DATA	Data (340)
STATION SPEED DIAL BUTTONS 0-9	Used for storing and dialing one station speed-dial number each.
UNDEFINED	Analog phones and some Executone IDS and Wave Phones have several undefined buttons that can be programmed as desired.

Secondary Extension Appearances

12.67 The Secondary Extension Appearances feature allows calls ringing or on individual hold at one “primary” station to appear on a “secondary extension key” at another station. For example, a secretary or assistant can have a secondary extension button for a supervisor’s station, or one station can have several secondary extension buttons, each associated with a different primary station. The primary station and the secondary station must reside on the same node. (See page 6-221 for key programming information.)

12.68 A secondary extension button functions as follows:

- When a call is ringing or holding on any CALL button at the primary station, pressing the secondary extension button will answer the call. (This includes recalls, transfers, callbacks, and hunt group calls.)
- If the secondary extension button is unlit, pressing the button places an intercom call to the primary station. If the button is flashing, an intercom call can still be placed without answering the incoming call by pressing the pound [hash] (#) button before the secondary extension button, by pressing a DSS button, or by manually dialing the primary station’s extension number.

12.69 There are two programmable options that determine the type of call indications the station user will receive for calls on a secondary extension button:

- **Ring When n Calls At Extension**: Allows the station to receive a burst of ringing when "n" number of calls are present at the associated primary station. The ring signal will repeat periodically as long as "n" number of calls are present at a primary station. (This period is determined by the DKTS Secondary Extension Key Alerting Tone timer.) If "n" is set to 0, the secondary station will never receive the burst of ringing. This option is programmed individually for each secondary extension button, during keymap programming.
- **Transient Call Indication On Call Answer**: Determines if the station user will see a call display when answering a call by pressing the secondary extension button or reverse-transferring the call. If enabled, the display will tell if the call was ringing, recalling, transferred, or holding at the primary station.

12.70 The secondary extension button can be located on the phone or on a DSS/BLF Unit. There are two ways to make a secondary extension button:

- Create a secondary extension button in the keymap and determine which station it will represent. This will be the same for all stations assigned to that keymap.
- Set the default value of a user-programmable button to "secondary extension key" in the keymap. This allows each station user assigned to that keymap to determine the associated station for his or her secondary extension button. (See page 5-156 for key programming procedures.)

12.71 User-programmable buttons cannot be made into secondary extension buttons by the station user; they can only be created in the database. Also, the station user cannot change the button into any other ***type*** of button (such as a speed-dial or feature button), even though it is assigned to a user-programmable button.

12.72 Any phone button can be designated as a secondary extension button, but buttons with lamps are recommended so that the secondary extension button can show call status. The table on page 5-141 shows the definition for each possible lamp flash rate.

12.73 The secondary extension button shows only calls that appear under the CALL buttons at the primary station. Any calls on individual trunk buttons at the primary station are considered private calls and will not appear on the secondary extension button.

12.74 Normally, the secondary extension button will not show intercom calls. However, if the primary station does not have an IC button, intercom calls will appear under a CALL button and, therefore, will appear on the secondary extension button. If the primary station is a single-line set, ***all*** calls will appear on the secondary extension button.

12.75 After answering a call on the secondary extension button, the station user can transfer it back to the primary station using one of the following methods:

- **Transfer to hold**: Press the secondary extension button, announce the call, if desired, and then hang up.
- **Transfer to ring**: Press TRANSFER and then the secondary extension button. You can announce the call, if desired, before hanging up to complete the transfer.

12.76 Because the calls that appear on the secondary extension button are active only at the primary station, they will not follow call forwarding or system forwarding paths at the station with the secondary extension button.

D. LIGHT-EMITTING DIODE (LED) INDICATIONS

12.77 The phone CALL buttons, trunk buttons, speed-dial buttons, hunt group buttons, and some of the feature buttons contain light-emitting diodes (LEDs). The LED flash rates, shown below, indicate the status of the stations, trunks, and features. The rates are shown in interruptions per minute (IPM).

Table 5-29. *Endpoint LED Indications*

	Steady 0 IPM	Slow 30 IPM	Medium 120 IPM	Fast 240 IPM	Flutter 60/960 IPM
DATA	Auto-dial modem is dialing and the data port is unavailable. (Red LED)			Ring signal is being sent to the data device. (Red LED)	Data call is in progress and accessible. A continuous flutter indicates that the data device is off hook. (Red LED)
IC		Intercom call is in progress. (Green LED*)	Recalling intercom call is camped on. (Green LED*)	Intercom call is camped on (waiting). (Green LED*)	Intercom call is on hold. (Green LED*)
SPEAKER	Speakerphone is on or ready for use. Or, station programming is in progress. (Red LED)				
CNF		Conference is in progress. (Green LED*)	Conference is recalling your station. (Green LED*)		A conference party is on conference wait hold. (Green LED*)
MUTE	Microphone is muted. (Red LED)				
FWD	Calls are being forwarded. (Red LED)				
MSG	Station is in alpha-numeric mode. (Red LED)		Message is waiting. (Red LED)		
DND	Station is in do-not-disturb. (Red LED)				
SPEED DIAL or DSS	Station is busy or unplugged. (Red LED)	Station is in do-not-disturb. (Red LED)		Station has a call ringing in. (Red LED)	Continuous flutter indicates that the station was left off hook. (Red LED)
HUNT GROUP	All stations in hunt group are unavailable. (Red LED)			Hunt group has a call camped on. (Red LED)	
CALL		Call is active at your station. (Green LED*)	Call is recalling from hold or transfer. (Green LED*)	Call or queue callback is ringing in or camped on (waiting). (Green LED*)	Call is on hold. (Green LED*)
TRUNK	Trunk is in use at another station, in a conference at your station, or is unplugged. (Red LED)	Trunk is in use at your station. (Green LED*)	Trunk is recalling from hold or transfer. (Red LED)	Trunk is ringing in or camped on (waiting). (Red LED if on another station or green if at your station*) Or, busy trunk queue callback to your station. (Green LED*)	Trunk is on hold. (Red LED if on another station or green if at your station*)

*This applies only to phones with red/green LEDs.

12.78 Red and Green LEDs: Inter-Tel digital phones have red and green LEDs in the button lamps. When both colors are used, a green lamp generally indicates that the call is active, ringing, or holding at your station (see below for a possible exception for direct ring-in calls). When the lamp is red, the call is active, ringing, or holding at another station. Trunk access, allowed-answer permission, and the feature being used (for example, System Hold or Call Transfer) determine if you can pick up a (red) call from another station.

NOTE: Green LEDs are not applicable on multi-protocol endpoints.

12.79 There are two system-wide programming options that control these LEDs. (See page 6-258 in PROGRAMMING.)

- **Allow Green LEDs**: Gives the installer the option of having all red/green phones and red-only phones operate the same, or allowing the red/green phones to use the green LEDs. If this flag is enabled, the dual-color phones will use both the red and green LEDs. If it is disabled, the phones will operate in red-only mode, like the phones without green LEDs. In the default state, the flag is disabled.
- **Use Green LEDs For Direct Ring**: Determines whether direct ring-in calls cause a red or green lamp. If the flag is enabled, direct ring-in calls to your station will be green. If disabled, all direct ring-in calls will cause the LED to be red. In the default state, this flag is enabled.

NOTE: The Allow Green LEDs flag must be enabled in order for this flag to operate.

E. HEADSETS

12.80 For convenience, an electret headset can be used at any phone. The Speaker button is used to connect and disconnect calls when the headset is attached. See SPECIFICATIONS, page 2-75, when selecting a headset for use on the phone.

12.81 Some headsets have a power-saver mode that disables the headphone after a period of silence. To prevent these headsets from missing portions of incoming calls, there is a station flag called “Send Alert Burst To Headset.” Enabling this flag allows the phone to generate a tone that activates the headset before connecting to an incoming call. In the default state, it is disabled. See page 6-211 in PROGRAMMING.

12.82 The new Model 8520 and 8560 Phones have headset jacks on the back of the phones. This allows you to have the headset and the handset connected to the phone at the same time. You must still, however, enable/disable the headset using the Headset Mode feature code.

Connecting a Headset

12.83 To connect a headset to a Model 8520 or 8560 Phone:

1. Insert the headset plug into the vacant headset jack on the back of the phone.
2. While on hook, enter the Headset On feature code (315) or the Headset On/Off feature code (317). The display shows HEADSET MODE ON.

12.84 To connect a headset to all other phones:

1. Remove the coiled handset cord from the base of the phone. Leave the handset in the cradle.
2. Insert the headset plug into the vacant handset jack.
3. While on hook, enter the Headset On feature code (315) or the Headset On/Off feature code (317). The display shows HEADSET MODE ON.

12.85 To disconnect the headset from a Model 8520 or 8560 Phone:

1. Unplug the headset cord from the back of the phone.
2. While on hook, enter the Headset Off feature code (316) or the Headset On/Off feature code (317). The display shows HEADSET MODE OFF.

12.86 To disconnect the headset from all other phones:

1. Unplug the headset cord from the base of the phone.
2. Plug the coiled handset cord into the base of the phone.
3. While on hook, enter the Headset Off feature code (316) or the Headset On/Off feature code (317). The display shows HEADSET MODE OFF.

"Headset Connect Tone" Station Flag

12.87 The "Headset Connect Tone" flag will send a single tone to a headset, instead of normal phone ring tones, when a call is ringing at the station (non-handsfree intercom call, outside call, queue callback, or reminder message). The tone is sent only once, when the call is received, and will not repeat even if the call continues to ring unanswered. It is intended as a "wake up" tone for headsets that have a power-saving function that allows the device to "sleep" if idle for a set period. The flag is programmed on a station-by-station basis. By default the flag is disabled.

F. VOLUME CONTROLS

12.88 On some analog phones, speaker volume and ring tone volume are controlled by two thumbwheels on the right side of the phone.

12.89 On all digital phones, AIM and Inter-Tel/DVK Analog Phones, and Executone IDS and Wave Phones, volume is controlled by pressing volume control buttons on the phone.

- On the Executive Display, Standard Display, and Axxess Basic Phones, an illustration on the button (that looks like this) represents higher and lower volume levels. Press the lower end in place of the down arrow and the higher end in place of the up arrow.
- On the Professional, Associate, and Eclipse Basic Phones, the button has up and down arrows, representing higher and lower volume levels.
- On Model 8500, 8520, and 8560 Phones, illustrations of speakers (and) represent higher and lower volume levels. Press the end of the button to lower the volume and the end of the button to raise the volume.
- The analog phones and Executone IDS and Wave phones have separate VOL UP and VOL DOWN buttons.

12.90 The Standard Display, Associate Display, Axxess Basic Digital, Eclipse Basic Digital, Model 8520, and Model 8500 Phones have a volume range of 1 to 7, with db levels of -12, -8, -4, 0, +4, +8, and +12 respectively. The Executive Display, Professional Display, and Model 8560 Phones have a volume range of 1-8, with db levels of -12, -8, -4, 0, +3, +6, +9, and +12 respectively.

NOTE: The volume adjustment button on the digital phones can also be used for scrolling through displays. The analog phone and Executone IDS and Wave Phone volume buttons are used only for adjusting volume; they need to have PREVIOUS and NEXT buttons programmed in their keymaps to scroll through displays. For information about programming the analog phones and Executone phones, see a version 7.0 or earlier *Installation and Maintenance Manual*.

12.91 To change volumes on a digital phone:

1. Press the appropriate button to raise or lower the volume, using one of the procedures listed below. A confirmation tone will alert you when you have reached the highest or lowest possible volume. On display phones, the display shows VOLUME LEVEL X. The "X" represents the level and changes as the volume is raised or lowered.

   — **To change handset intercom voice volume:** Adjust the volume while on an off-hook (handset) intercom call.

   — **To change handsfree intercom voice and paging volume:** Adjust the volume while on an on-hook (speakerphone) intercom call.

   — **To change handset outside call voice volume:** Adjust the volume while on an off-hook (handset) outside call.

   — **To change handsfree outside call voice volume:** Adjust the volume while on an on-hook (speakerphone) outside call.

   — **To change headset intercom voice volume:** *(Model 8520 and 8560 Phones Only)* Adjust the volume while using the headset and on an intercom call.

   — **To change headset outside call voice volume:** *(Model 8520 and 8560 Phones Only)* Adjust the volume while using the headset and on an outside call.

   — **To change background music volume:** Adjust the volume while listening to music through the phone speaker.

   NOTE: The selected background music volume level is automatically saved.

   — **To change alerting tone (ring tone) volume:** Adjust the volume while the phone is ringing or idle. Or, while on hook, press both edges of the Volume button on a digital phone or the pound (hash) button (**#**) on all other phones to hear the currently programmed volume, then adjust it up or down.

   NOTE: The selected alert tone volume level is automatically saved.

   — **To change handset progress tone volume:** Press the desired end of the Volume button while listening to intercom dial tone through the handset.

   — **To change speakerphone progress tone volume:** Press the desired end of the Volume button while listening to intercom dial tone through the speakerphone.

2. **To save the setting on Executive Display, Standard Display, Basic Digital, Model 8560, Model 8520, and Model 8500 Phones:** Press both edges of the Volume button at the same time. Display phones show VOLUME SAVED.

   To save the setting on Professional Display, Associate Display, and Eclipse Basic Digital Phones: Press the center of the button. Display phones show VOLUME SAVED.

   If this step is not performed, the phone will return to the previous volume level when the call or tone ends (except for the ringing alert tone or background music volume, which is automatically saved at the most recent volume level).

12.92 To return a phone to default volume levels:

While on or off hook, enter the Default Station feature code (394). *If off hook,* hang up. The display shows STATION DEFAULTED.

NOTE: Entering this feature code also cancels do-not-disturb, manual call forwarding, background music, ring intercom always, all calls following account code, headset mode, and queue requests; and restores handsfree mode, pages, hunt group calls, and system forwarding.

G. SELECTABLE RING TONE

12.93 The tone of the phone ring signals (all tones except call waiting) can be changed by the user to create distinctive ringing. If stations are placed close together, changing the tone makes each station's ring easier to recognize. On all phones, the tone is changed by entering a feature code and a code, as described below.

12.94 There are ten ring tone options (0-9). Selection 0 will turn off the ring signals.

NOTE: Executone phones have only eight distinct ring tones. Settings 1 and 9 are actually the same tone. However, you can still use setting 0 to disable ringing.

12.95 To change phone ring tone:

1. While on hook, enter the Program Ring Tone feature code (398). The display shows SELECT RING TYPE (0-9) OR SCROLL.
2. Select the desired tone.
   - — **To select a specific ring tone:** Enter a number 1-9 for the desired ring tone. The display shows RING TYPE X SELECTED.
   - — **To scroll through the tones:** Press the low end of the Volume button or the **PREVIOUS** button to scroll backward through the tones. Press the high end of the Volume button or the **NEXT** button to scroll forward. (You cannot scroll to selection 0, you must enter it as described below.) The display shows RING TYPE X SELECTED.
   - — **To turn off ringing:** Enter **0** or press the **RINGER OFF** menu button. The display shows RING TYPE OFF SELECTED.
3. Lift and replace the handset or press the pound (hash) button (**#**), both ends of the Volume button, or the **ACCEPT** button to exit. The selected tone will be heard when the phone rings (unless ringing was disabled). The display shows RING TYPE X SAVED.

H. DATA PORT MODULES

Digital Display Phone Data Port Modules

12.96 There are two types of Data Port Modules that can be connected to Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones:

- **PC Data Port Module (PCDPM)**: A digital device that is connected to the Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone and supports data transmission between the phone and the phone system for DSS/BLF, off-hook voice announce, Desktop Interface, and/or serial port functions. A PCDPM is required if a Modem Data Port Module or DSS/BLF Unit will be installed, or if the user wants to receive OHVA calls. A PCDPM is not required for the Mini-DSS.

  NOTE: A phone serial port should not be used as the Voice Processing communications port.

- **Modem Data Port Module (MDPM)**: An analog device that can be used with a PCDPM and connects to a modem for data applications or to a single-line set. The port itself has an individual extension number and complete station programming setup including its own trunk access and toll restriction information. (See page 5-269 for data device information.)

12.97 Special programming is required to enable the Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone circuit to support some of the PCDPM and MDPM functions. See page 6-118 in PROGRAMMING for details.

12.98 Station users can adjust the bit rate (bps) of the serial port on the PCDPM connected to the phone, if it is not configured for a DSS/BLF unit. The rate can be set to 300, 600, 1200, 2400, or 4800 (1200, 2400, or 4800 if it is a system serial port).

12.99 To change the bit rate of your PCDPM:

1. While on hook, enter the Program Bit Rate feature code (393). If the port is not configured for a DSS/BLF unit, the display shows ENTER BIT RATE. (*If it is configured for DSS/BLF,* the display shows CANNOT PROGRAM BIT RATE.)
2. Enter a valid rate (300, 600, 1200, 2400, or 4800). Executive Display, Professional Display, and Model 8560 Phone users can press the appropriate menu button in the display. The display shows BIT RATE UPDATED. (*If the entered bit rate was already in effect,* it shows NO UPDATE PERFORMED.)

NOTE: For information about analog phones and Executone phones, see a version 7.0 or earlier *Installation and Maintenance Manual*.

I. DIGITAL DISPLAY PHONE DESKTOP INTERFACE

NOTE: The Desktop Interface software license is required if attaching Desktop Interface devices to a digital phone.

12.100 Desktop Interface is a *station-level* communications protocol that allows a PC, attached to a PCDPM on an Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone, to emulate a phone for call handling. This protocol allows devices such as PC phones, operators' consoles, ACD supervisors, and integrated contact management tools to be developed for the Inter-Tel phone system.

12.101 There can be as many Desktop Interface devices on the system as there are Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and/or Model 8560 Display Phone circuits. The only limitation is on Desktop Interface devices that require DSS/BLF status updates (such as operators' consoles). The system can support up to 20 DSS/BLF status requests from external devices at one time. Therefore, if the device requires continuous DSS/BLF status updates, only 20 can be installed on the system at one time. Devices that require one-time or infrequent DSS/BLF updates are not limited by this capacity, except that only 20 requests for updates can be processed at one time. There is a limit of four virtual DSS devices per system because of the number of requests they require and the amount of information that is shared with the system.

12.102 Another Software license, System OAI, is a *system-level* communications protocol that allows command-oriented tools to be developed for the phone system. System OAI is available in two levels: Events and Third Party Call Control. System OAI Events provides call status information to an external device. Its primary use is for automatic call distribution products such as the Contact Center Suite. System OAI Third Party Call Control is a TSAPI protocol that allows system command-oriented tools to be used on the phone system. Contact Inter-Tel for more information regarding products and development materials for these features.

NOTE: Level 2 System OAI applications can be used on networked systems. Note that some System OAI Level 2 applications must be used in conjunction with the CT Gateway and/or Transceivers to process the network information correctly. For high-capacity systems, such as an ATM-based system, the CT Gateway is recommended. In addition, version 8.0 or later software requires CT Gateway version 3.0 or earlier. See the *CT Gateway Installation Manual* for more information.

J. AGENTSET

12.103 The AgentSet is used in place of a phone with an attached PCDPM. It is connected to a Digital Keyset Card (DKSC or DKSC-16/DKSC16+) port just as a phone would be. It has a jack for plugging in an Inter-Tel handset or headset. It also has a serial connector (DB9) for attaching it to a PC. The AgentSet does not require a transformer or external power source.

12.104 To operate properly, the AgentSet *must* be programmed for headset mode. This can be done in one of the following ways:

- Enable Headset Mode via Station Programming in the system database
- At the AgentSet, ***unplug the headset or handset*** (to take the AgentSet off hook) and enter the Headset On/Off feature code (317) to enable headset mode. Then plug in the headset or handset.

NOTE: There is nothing to prevent the user from disabling headset mode using the Headset On/Off feature code at the station.

12.105 Because the AgentSet does not have feature buttons, a dialing pad, or a speaker, it must be connected to a PC that is equipped with an OAI product that emulates a phone. The dialing software is used for dialing and feature access.

12.106 The AgentSet also requires an Inter-Tel headset or handset. Only Inter-Tel equipment should be used. Other manufacturer's equipment may not be compatible with the AgentSet.

12.107 If the user wants to receive audible signals at the headset, there is a new station flag that will send a single tone to the headset when a call is ringing. (See page 5-143.)

K. SINGLE-LINE SETS

12.108 Industry-standard, single-line DTMF or pulse-tone sets can be installed on the system and/or used as an off-premises extension (OPX). (Version 2.x or later software is required for pulse-tone sets.) On-premises single-line circuits can be used for installing AC-ringing single-line sets, playback devices, and other single-line devices.

12.109 The single-line sets are connected to Single-Line Cards (SLCs) or 16-Port Single Line Cards (SLC-16s). Or, they can be connected to Single-Line Adapter (SLA) circuits, which are in turn connected to available Digital Keyset Card (DKSC or DKSC16) circuits. Off-premises stations can be connected to SLAs, T1 Cards (T1Cs), or T1/E1 cards.

- Each SLC can support up to eight on-premises single-line stations and each SLC-16 can support up to sixteen stations. The SLC and SLC-16 will support message lamps on single-line sets if the optional Lamp Option Card is installed on the SLC or SLC-16.
- Each SLA can support up to two single-line sets or DID [DDI in Europe] devices.
- T1 Card (T1C) and T1/E1 card circuits can be used for installing off-premises single-line stations *(U.S. only)*.

Single-Line DTMF Sets

12.110 Single-line DTMF set users gain access to the features by hookflashing [pressing and releasing the hookswitch quickly, or pressing the **RECALL** button in Europe] and entering feature codes.

Single-Line Set Message Lamps

12.111 The optional Lamp Option Card (LOC) for the SLC or SLC-16 supports message waiting lamps. To enable a message waiting lamp on a specific single-line station, the station must have the Message Lamp station flag enabled (see page 6-212 in PROGRAMMING). In the default state, it is disabled.

12.112SLAs, T1C, and T1/E1 card circuits cannot support message waiting lamps.

Single-Line Ring Zones and Extended Ring Cadences

12.113 In some installations, where more than one single-line device is connected in series to an SLC or SLC-16 circuit, zoned ringing may be necessary. The system can support devices with up to 3 Ringer Equivalences (REs) per circuit. However, the system can only ring 8 REs per board. If the installation exceeds eight REs on any SLC or SLC-16, the system-wide SLC Ring Zones flag must be enabled (see page 6-261 in PROGRAMMING). Also, all circuits on the SLC or SLC-16 should have the Extended Ring Cadences station flag enabled to allow the longer cadence to be used instead of the short internal cadence (see page 6-211 in PROGRAMMING).

12.114 With ring zones enabled, devices 1-3 on the card will ring first, 4-6 ring next, and so on until all devices have rung.

12.115 The table below explains the difference between standard and extended ring cadences for the various types of single-line devices used on the Inter-Tel system in the United States and Europe.:

Table 5-30. *Single-Line Ring Zones and Extended Ring Cadences (U.S.)*

Single-Line Device	Cadence Type	Description
SLA, SLC, SLC-16, or T1 OPX	Standard IC Ring	This is a repeating 2-second cadence that rings twice. Each ring lasts 200 ms and is separated from the next ring by 400 ms of silence. The remaining 1.2 seconds is silent.
	Extended IC Ring	This is a repeating 6-second cadence that rings twice. Each ring lasts 800 ms and is separated from the next ring by 400 ms of silence. The remaining 4 seconds is silent
	Standard CO Ring	This is a repeating 6-second cadence that rings once for 2 seconds followed by 4 seconds of silence.
	Extended CO Ring	Same as Standard CO Ring cadence.
Executive, Professional, Standard, Associate, Model 8520, or Model 8560 Phone MDPM	Standard IC Ring	This cadence has two 250 ms rings every 2 seconds.
	Extended IC Ring	This is a repeating 3-second cadence that rings once for 1 second followed by 2 seconds of silence.
	Standard CO Ring	This is a repeating 6-second cadence that rings once for 2 seconds followed by 4 seconds of silence.
	Extended CO Ring	This is a repeating 6-second cadence that rings once for 4 seconds followed by 2 seconds of silence.

Table 5-30. *Single-Line Ring Zones and Extended Ring Cadences (U.S.) (Continued)*

Single-Line Device	Cadence Type	Description
Analog Phone Data Port Module	Standard IC Ring	This cadence rings for 1024 milliseconds followed by 2048 milliseconds of silence.
	Extended IC Ring	This is a repeating 2-second cadence that rings twice. Each ring lasts 200 ms and is separated from the next ring by 400 ms of silence. The remaining 1.2 seconds is silent.
	Standard CO Ring	This is a repeating 6-second cadence that rings once for 2 seconds followed by 4 seconds of silence.
	Extended CO Ring	Same as Standard CO Ring cadence.

Table 5-31. *Single-Line Ring Zones and Extended Ring Cadences (Europe)*

Single-Line Device	Cadence Type	Description
SLA, SLC, or SLC-16	Standard IC Ring	This is a repeating 6-second cadence that rings once for 2 seconds followed by 4 seconds of silence.
	Extended IC Ring	Same as Standard IC Ring cadence.
	Standard CO Ring	This is a repeating 2-second cadence that rings twice. Each ring lasts 200 ms and is separated from the next ring by 400 ms of silence. The remaining 1.2 seconds is silent.
	Extended CO Ring	This is a repeating 6-second cadence that rings twice. Each ring lasts 800 ms and is separated from the next ring by 400 ms of silence. The remaining 4 seconds is silent
Executive, Standard, Model 8520, or Model 8560 Phone MDPM	Standard IC Ring	This is a repeating 6-second cadence that rings once for 2 seconds followed by 4 seconds of silence.
	Extended IC Ring	This is a repeating 6-second cadence that rings once for 4 seconds followed by 2 seconds of silence.
	Standard CO Ring	This cadence has two 250 ms rings every 2 seconds.
	Extended CO Ring	This is a repeating 3-second cadence that rings once for 1 second followed by 2 seconds of silence.
Analog Phone Data Port Module	Standard IC Ring	This is a repeating 6-second cadence that rings once for 2 seconds followed by 4 seconds of silence.
	Extended IC Ring	Same as Standard CO Ring cadence.
	Standard CO Ring	This cadence rings for 1024 ms followed by 2048 ms of silence.
	Extended CO Ring	This is a repeating 2-second cadence that rings twice. Each ring lasts 200 ms and is separated from the next ring by 400 ms of silence. The remaining 1.2 seconds is silent.

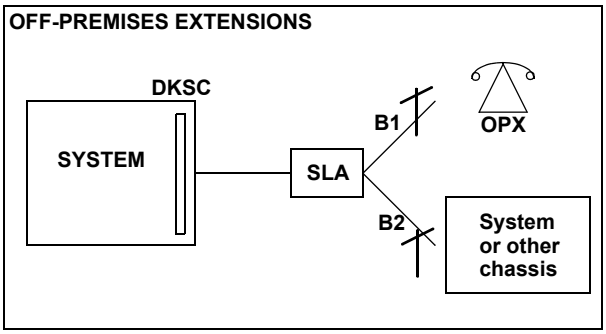
Off-Premises Extensions (OPX)

12.116 Off-premises extensions are remote single-line DTMF sets that are connected to the phone system through telephone company [network provider] OPX circuits (customer-provided circuits).

12.117 Off-premises station users gain access to the features by hookflashing, pressing and releasing the hookswitch quickly (pressing the **RECALL** button), and entering feature codes.

12.118 Some telephone company [network provider] off-premises extension (OPX) networks do not recognize certain single-line set ring cadences used in the phone system. This incompatibility results in an inability to provide certain ring signals to off-premises extensions. Because of this, the system is equipped with a station option (Extended Ring Cadences) which allows the ring cadences be altered for individual single-line circuits. See page 6-211 in PROGRAMMING for details.

12.119 Another programming option (the OPX Gain Levels flag) allows the receive and transmit gains to be automatically adjusted to improve OPX communication with the central office [local exchange]. See page 6-212 in PROGRAMMING.



NOTE: OPX circuits can connect to SLA, T1, or E1 circuits.

L. DIGITAL DIRECT STATION SELECTION/BUSY LAMP FIELD (DSS/BLF) UNITS

12.120 The Inter-Tel system supports three types of DSS/BLF Consoles/Units. The unit used depends on the phone.

- **With Executone IDS Phones** *(U.S. Only)*: IDS phones can be equipped with up to three 48-key DSS Consoles. There can be a total of 80 IDS 48-key DSS Consoles on the system. Each 48-key DSS Console requires an EKSC circuit.

  NOTE: For details about the 48-key DSS/BLF Unit, see a version 7.0 or earlier *Eclipse Installation and Maintenance Manual.*

- **With Executive Display, Standard Display, Professional Display, and Associate Display Phones**: PCDPM-equipped Executive Display and Standard Display Phones can connect to up to four 60-key, 6-row DSS/BLF Units. Professional Display and Associate Display Phones can connect to up to four 60-key, 12-row DSS/BLF Units. There can be a total of 20 of these 60-key DSS/BLF Units on the system. Because each unit is powered by an AC transformer and is connected to the phone's PCDPM, the 60-key DSS/BLF Units do not require a DKSC or DKSC-16/DKSC-16+ circuit that is separate from the phone.
- **With Model 8520 and Model 8560 Phones**: PCDPM-equipped Model 8520 and Model 8560 Phones may be connected to up to four 50-key DSS/BLF Units. There can be a total of 20 of these 50-key DSS/BLF Units on the system. Because each unit is powered by an AC transformer and is connected to the phone's PCDPM, the 50-key DSS/BLF Consoles do not require a DKSC or DKSC-16/DKSC-16+ circuit that is separate from the phone

NOTE: Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones with DSS/BLF Units need to be identified in DB Programming and equipped with PCDPMs (see PROGRAMMING, page 6-226).

12.121 Although the 60-key model is designed for Executive Display, Professional Display, Associate Display, or Standard Display Phones and the 50-key model is designed for Model 8520 or Model 8560 Phones, you can use either DSS/BLF Unit on any of these phones. Each unit provides one-button access to 50 or 60 extension numbers and/or other features/resources, while four units provide access to 200 or 240 extension numbers and/or other features/resources. (See page 3-205 for DSS/BLF Unit installation instructions.) Each keymap has four "pages" of layouts to accommodate four connected DSS/BLF Units. There can be up to 20 DSS/BLF Units in the system, up to four on a single phone.

NOTE: The presence of a Mini-DSS does not affect the number of DSS/BLF Units that you can attach to a phone.

12.122 Together, the lamps in the buttons create a busy lamp field that indicates the status of the resource (e.g., station, trunk, hunt group, or feature) assigned to the buttons (unless the resource is an off-node device). The LED indicator in the button indications are shown in the table on the next page.

12.123 You can program a DSS button for an off-node device, but not for "wildcard" off-node extensions (see Appendix A — Networking, for information on wildcard extensions). The buttons will show lamp status for stations on the other nodes.

12.124 In DB Programming, DSS/BLF buttons may also be programmed for direct access to other resources (such as paging zones), or they can be assigned feature codes. Busy lamp indication is provided to show the idle/busy status of the resource or feature. Note that programming a DSS/BLF button as a resource does not always provide a transfer capability. For example, a DSS button programmed for paging will not transfer a party to the paging system.

Table 5-32. *DSS/BLF Button Indications*

Indication	Meaning
Solidly Lit	The associated resource is busy or all stations in the associated hunt group are unavailable (in do-not-disturb, busy, or with hunt group remove enabled). Or, the associated feature is on.
Slow Flash	The station is in do-not-disturb.
Fast Flash	The station has a call ringing in or a call is camped on to the hunt group. (This option is programmable. If DSS ring flash is disabled, a call ringing in to a station will show a solidly lit LED.)
Continuous Flutter	The station is causing a STATION OFF-HOOK system alarm.

M. MINI-DSS UNITS

12.125 Version 7.0 and later software supports a Mini-DSS that has 16 user-programmable buttons. This unit attaches directly to the Model 8520 or 8560 Phone and does not require a PCDPM or AC transformer. Like the DSS/BLF Unit, these buttons contain lamps that light or flash depending on the status of a station or the button's programming. For details about the lamps, see the previous section.

NOTE: Version 7.0 or later software is required to use the Mini-DSS.

N. DESKTOP OAI, DSS UNITS, AND RS232

12.126 The desktop interface (DTI) protocol allows stations using a DTI connection to monitor different groups of system devices. Because this monitoring consumes system resources, there is a limit on the total number of devices that users can monitor with RS232 connections. (This does not apply to socket connections, which are virtually limitless.)

NOTE: One virtual DSS unit supports 60 devices. For example, to monitor all 300 hunt groups in a v7.0 or later system, you would need 5 DSS units (300/60 = 5).

12.127 Although the total number of virtual DSS units available (150) has not increased as of version 6.0, the number of devices has, resulting in a limited number of stations that can monitor all devices simultaneously. To compensate for the increased number of devices, versions 6.0 and later use a DTI monitor limiter scheme that bases each station's resource usage on the actual number of devices the station is monitoring. Any unused virtual DSS units are then available for use by other stations. For example, if a station is monitoring 120 hunt groups (instead of all 300), the remaining 3 DSS units become available for other stations, which increases the number of stations that can enable monitoring.

13. USING DSS/BLF BUTTONS (DSS/BLF UNIT OR PHONE DSS/BLF)

13.1 Direct Station Selection/Busy Lamp Field (DSS/BLF) buttons provide one-button access to extension numbers. It also shows the status of the station or hunt group (local or off-node) associated with the button. DSS/BLF buttons can be programmed on phones, DSS/BLF Units, or Mini-DSS Units.

13.2 The lamps in DSS/BLF buttons (phone or DSS/BLF unit) indicate the status of the associated station or hunt group, using the flash rates described on page 5-152.

13.3 DSS/BLF buttons that are programmed for feature code access can be used in place of the associated feature code or button in the procedures described in this FEATURES section.

NOTE: Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones with DSS/BLF Units need to be identified in DB Programming and equipped with PCDPMs (see PROGRAMMING, page 6-226).

13.4 To place an intercom call using DSS/BLF buttons:

1. *To use the handset,* lift the handset. The display shows ENTER EXTENSION NUMBER.

   To use the speakerphone, while on hook, press the Speaker button or the IC button. The Speaker button lights. The display shows ENTER EXTENSION NUMBER.

   To place a private (non-handsfree) call to a phone, press the pound [hash] (#) button. The display shows RING EXTENSION NUMBER.

2. Press the desired DSS/BLF button.

   If calling a phone with handsfree answering enabled, you are immediately connected.

   If placing a private call, calling a hunt group, calling a single-line set, or calling a phone with the Handsfree feature disabled, you hear repeating double tones until the call is answered.

   If the called OHVA-enabled station is busy and your station is enabled for immediate OHVA transmit, do not hang up when calling a busy phone. Press the DSS/BLF button again. If the phone's secondary voice path is available, you are immediately connected for an off-hook voice announce call and may speak. (For more information concerning the Off-Hook Voice Announce feature, see page 5-190.)

13.5 To transfer calls to stations:

1. While on a call, press the desired DSS/BLF button. The call is placed on transfer hold.

   — **Transfer to a station that is forwarded to voice mail**: You hear repeating double tones, and the display shows DEST FORWARDED TO VOICE MAIL.

   — **Transfer to voice mail**: The system waits for you to enter the mailbox number (display shows ENTER MAILBOX). Enter the desired mailbox number. *If you do not enter a mailbox number before you hang up,* the caller will be connected to the voice mail unit and must enter the mailbox number after listening to the introductory prompts.

   — **Transfer-to-park**: If transferring to a park location, hang up or press the Hold button to complete the transfer. Then page the desired party and announce the call. The party must reverse transfer the call. Note that if the call is not answered, it will recall your station if transferred directly, or it will recall the called transfer-to-park location's attendant if transferred to hold.

2. *If desired,* wait for an answer and announce the call. If you are calling a device that does not accept handsfree calls (such as an automated attendant), the display shows HANG UP TO FINISH TRANSFER.

3. You have the following options:

   To complete the transfer: Hang up or press another CALL button. The display shows CALL TRANSFERRED TO *(username)*.

   To transfer the call to hold: Press the Hold button, or press ∞ or SPCL and enter the Individual Hold feature code (336). The call will not ring or send call waiting signals until the Hold timer expires.

   To try another station: Press another DSS/BLF button.

   To return to the caller: Press the lit CALL button, trunk button, or IC button.

13.6 To reverse transfer (pick up) a call ringing or holding at another station:

EITHER, enter the Reverse Transfer feature code (4) and then press the DSS/BLF button of the station where the call is ringing.

OR, press the desired DSS/BLF button, and then press ∞ or SPCL and enter the Reverse Transfer feature code (4).

14. USER-PROGRAMMABLE FEATURE BUTTONS

NOTE: User-programmable feature buttons are available on Inter-Tel proprietary phones only.

14.1 Individual phone buttons can be designated as "user-programmable" in the keymaps during DB Programming. The installer can program the default feature codes (see PROGRAMMING, page 6-221), and the station user can reprogram the feature codes as desired. Default phone feature codes are shown on page 5-138. Up to 35 user-programmable phone buttons can be programmed to enter any of the general feature codes.

14.2 Some phones have a display menu and a defaulted FWD button. (This button, however, can be reprogrammed to use any of the Forward feature codes.) When the default FWD button is pressed, the display shows a menu of call forwarding options. The user must then select the desired option by pressing the corresponding button next to the display. If the FWD button is programmed to use any other Forward feature code (355-358), the forwarding options will not be displayed when the FWD button is pressed; the phones operate the same as other phones. Any phone user can make any user-programmable button a FWD button that uses any one of the forwarding feature codes (355-358).

14.3 Secondary extension buttons (described on page 5-139) can be assigned to user-programmable buttons during keymap programming. If so, the phone user can determine the primary station that is associated with the secondary extension button by using the procedures given below.

14.4 To display the current button entries:

1. While on hook, enter the Review Keys code (396). The display shows PRESS THE KEY TO REVIEW.
2. Press the feature or speed-dial button(s) to be displayed. The display shows the feature, user name, or speed dial location and KEY VALUE: *(feature code, extension number, or location code)*. For a secondary extension button, the display shows SECONDARY: *(station)*.

14.5 To return the buttons to default values:

While on or off hook, enter the Feature Key Default feature code (395). *If off hook,* hang up. The display shows FEATURE KEYS DEFAULTED.

14.6 To program a feature code in a user-programmable button or assign a primary station to a secondary extension button:

NOTE: Speed-dial buttons are programmed using the procedure shown on the next page. Entering an extension number, using these instructions, will create a DSS/BLF button for that extension number, not a speed-dial button (even if the button was previously programmed as a speed-dial button).

1. While on hook, enter the Program Keys feature code (397). Display phones show PRESS THE KEY TO PROGRAM.
2. Press the desired feature or secondary extension button. The display shows the current feature, station, or speed dial location name and KEY VALUE: *(feature, extension, or location code)*.

3. Enter the feature code or extension number to be stored under that button. (*If programming the FWD button,* only Call Forwarding feature codes 355-358 will be accepted.) You hear a single progress tone when the programming is completed. Display phones show the newly programmed feature. If you attempt to program a non-programmable button or enter an invalid code, you will hear repeating reorder tones, and the feature code assigned to the button will remain unchanged. Display phones show NON-PROGRAMMABLE KEY or FEATURE CODE INVALID.

NOTE: If you do not enter a feature code, the programming mode times out when the Long Interdigit timer expires; the feature code assigned to the button remains unchanged.

4. Repeat the procedure to program other buttons.

14.7 To create a *speed-dial location* in a user-programmable button:

1. While on hook, enter the Program Keys feature code (397). Display phones show PRESS THE KEY TO PROGRAM.
2. Press the desired feature button. The display shows the current feature, user, or speed dial location name and KEY VALUE: *(feature or name)*.
3. Enter the Station Speed-Dial feature code (382) or the System Speed-Dial feature code (381). The display shows SELECT STN SPEED or SELECT SYS SPEED.
4. Enter the desired speed-dial bin number (0-9 for station, 000-999 or 0000-4999 for system). The display shows the current contents of that bin.
5. Follow the instructions given on page 5-238 to program a new number in a station speed-dial bin. Administrators can use the procedures on page 5-297 to program system speed-dial locations.

15. AUTOMATIC CALL ACCESS

NOTE: Automatic call access is available on Inter-Tel proprietary phones only.

15.1 This feature allows a phone user to determine the way incoming calls are answered by entering a feature code to select the type of call access. The four variations are as follows:

- *No Automatic Call Access:* The user hears intercom dial tone when the handset is lifted and must press a CALL or individual trunk button (or the ANSWER button) to answer an outside call, or press the IC button (or CALL button, if there is no IC button) to answer an intercom call.
- *Automatic Intercom Call Access Only:* The user can answer ringing intercom calls by simply lifting the handset, but outside calls must be answered by pressing a CALL button, an individual trunk button, the ANSWER button, or the **ANSWER** menu option.
- *Automatic Outside Call Access Only:* The user can answer ringing outside calls by lifting the handset, but ringing intercom calls must be answered by pressing the IC button (or CALL button, if there is no IC button).
- *Automatic Access For All Calls:* The user can answer any ringing call by lifting the handset. If no call is ringing, the user hears intercom dial tone when the handset is lifted. (This is the way single-line sets work. When the system is in the default state, all phones have this type of call access.)

NOTE: In any of the above variations, phone users may still receive handsfree intercom calls (if enabled).

15.2 The selected option determines how all types of ringing intercom or outside calls (direct calls, transferred calls, recalls, etc.) are answered. If more than one call is ringing at the station, the first call received is the first answered.

15.3 When programmed for automatic outside call access, a station user with allowed answer, but without ring in, for a ringing trunk must always press an individual trunk button for that trunk, or enter the Automatic Trunk Answer feature code (350) to answer the incoming call. Transferred calls and recalls can be answered by lifting the handset.

15.4 Camped-on calls cannot be answered by simply lifting the handset or pressing the Speaker button. For example, a station is programmed to automatically answer ringing outside calls, but requires pressing the IC button to answer ringing intercom calls. If a private intercom call rings in and is immediately followed by an outside call ringing in, the display shows the intercom call message, and the outside call camps on. The intercom call also camps on when the handset is lifted. The user can then choose between the camped on calls by pressing either the IC button or the CALL or individual trunk button, ANSWER button, or **ANSWER** menu option.

15.5 The automatic call access options outlined below can be programmed at phones only. Single-line sets are designed to automatically answer ringing intercom and outside calls by lifting the handset and cannot be changed.

15.6 To program outside call access:

To change the current outside call access option to its opposite (toggle on or off), enter the Automatic Trunk Access On/Off feature code (360). You hear a confirmation tone. The display shows AUTO TRNK ACCESS ON (or OFF).

15.7 To program intercom call access:

To change the current intercom access option to its opposite (toggle on or off), enter the Automatic Intercom Access On/Off feature code (361). You hear a confirmation tone. The display shows AUTO IC ACCESS ON (or OFF).

16. MUSIC-ON-HOLD AND BACKGROUND MUSIC

NOTE: If you un-equip a card that is providing music, the music is terminated. Callers on hold or users listening to background music will hear nothing until the card is reprogrammed.

16.1 The Music-On-Hold feature not only makes waiting on hold as pleasant as possible, but it assures the holding party that the call is still connected.

16.2 Phone users can choose to listen to background music by entering the Background Music feature code (313).

16.3 The system can be equipped with one or more optional external music sources for the Music-On-Hold feature. The music source can be a customer-provided radio, tape player, or other device connected to the Call Processing Card (CPC), Central Processing Unit (CPU) card, and/or the Option Card(s) in the equipment chassis. When a music source is installed, music-on-hold is enabled or disabled in DB Programming. If music is not desired, the system can be programmed for tick tones, ringback, or silence on hold. (The Background Music feature uses the music source connected to the CPC or CPU card only).

NOTE: There are often broadcast restrictions associated with copyrighted music. Check with the music's original distributor and/or the broadcast source for restrictions and usage concerning background music and music-on-hold. Inter-Tel is not responsible for any illegal or improper use of copyrighted music connected to the system. With European systems, contact PPL (Phonographic Performance Ltd.) for licenses.

16.4 In an Inter-Tel system network, each node has its own music source(s). If a caller on Node 1 is holding for a user on Node 2, the caller hears the music on node 2. Note that if a caller is holding for a user on Node 1, and the call moves to Node 2 (due to a transfer, forward, or recall), the user will stop hearing the music on Node 1 and will begin hearing the music on Node 2.

16.5 As of 7.0, the system can be programmed to determine the music source a caller hears based on the device for which the caller is waiting. By default, the system determines the music source based on the trunk group on which the call resides. Throughout the manual, the term "music" refers to the selected option.

16.6 When a caller on a node *with* the v7.0 or later functionality waits for a device on a node *without* this functionality, the caller hears the music source programmed for **the node trunk group** on the destination node. When a caller on a node *without* the v7.0 or later functionality waits for a device on a node *with* this functionality, the caller hears the music source programmed for **the device** for which the caller waits.

16.7 Background music is interrupted for calls, pages, phone programming, and ringing.

16.8 To turn phone background music on or off:

While on or off hook, enter the Background Music On/Off feature code (313). *If off hook,* hang up. The display shows BACKGROUND MUSIC ON (or OFF).

NOTE: For information about analog phones and Executone phones, see a version 7.0 or earlier *Installation and Maintenance Manual*.

17. MULTILINGUAL CAPABILITY

NOTE: The Multilingual Support software license is required. It is installer-programmable to prevent unintentional language changes by users in an all-English system.

17.1 The phone system provides a choice between American English, British English, Spanish, and Japanese prompts and displays. Note that Spanish voice prompts use Mexico City Spanish Dialect for VPU and EVMC, and American Spanish Dialect for EM. The system selects the language to use for each call, as determined by the trunk, station, and Voice Processing programming, as described in the following sections. Japanese prompts can be viewed only on the Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones. Other phones do not support Japanese displays.

A. STATIONS

Language Selection

17.2 The system can be programmed to use a primary language and a secondary language. The available languages are American English, British English, Spanish, and Japanese. Note that Spanish voice prompts use Mexico City Spanish Dialect for VPU and EVMC, and American Spanish Dialect for EM. A station flag in DB Programming determines the language that will be used by each station. (See PROGRAMMING page 6-105 for information on programming the primary and secondary languages.)

17.3 For example, if the station is programmed for American English, all phone displays will appear in American English. Also, when the station is used for calling a Voice Processing application, the voice prompts will be in American English (unless overridden by a Call Routing Announcement). If programmed for Japanese, the displays will be in Japanese (Katakana) characters and the voice prompts will be in Japanese (unless overridden by a Call Routing Announcement). By default, all stations are set for the system's Primary Language.

NOTE: Only Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones can be used for viewing Japanese prompts. Other phones do not support Japanese displays.

17.4 All displays and default messages in the system are provided in both languages. Phone-programmed messages, including those programmed through the Administrator's phone, can include English/Spanish or Japanese characters, or a combination. In displayed lists, such as directories, the English/Spanish characters are alphabetized before the Japanese characters. Therefore, Japanese names will be at the end of the list.

17.5 When programming a feature that requires a time and date, an English/Spanish phone user will enter the date in the order "month, day, year" and the time in the order "time, AM/PM." The Japanese phone user will enter the date as "year, month, day" and the time as "AM/PM, time." The displays show the time and date differently, as well. For example, the English/Spanish display will show the time and date as "12:25 TUE NOV 28" and the Japanese display will show "TUE 11/28 12:25."

Secondary Language Selection

17.6 There is a new field added to each station in the system called "Secondary Language." This field corresponds to the Change Language feature (301). This feature used to toggle between the system primary language and the system secondary language. It now toggles between the system primary language and the station's secondary language, or can specify a language. This allows any station in the system to have its own secondary language or use the system secondary language, giving the system the ability to support more than just two languages.

17.7 If a station's secondary language field is programmed to be Use Primary Language the Change Language feature will do nothing because the station will toggle between the system primary language and the station's secondary language, which is the system primary language.

17.8 If a station's secondary language field is programmed to be Use Secondary Language the Change Language feature will act like version 5.2 and toggle between the system primary language and the station's secondary language, which is the system secondary language. This is the system default.

17.9 If a station's secondary language field is programmed to be Japanese the Change Language feature will toggle between the system primary language and the station's secondary language, which is Japanese.

17.10 The Language field for stations indicates what language the station is currently set to. This field used to toggle between the Use Primary Language and Use Secondary Language. It can be set to any specific language along with the Use Primary Language and Use Secondary Language. The side-effect of changing the Language field is that if you change it to Japanese and the station's secondary language field is set to Spanish, then the user will have no way to get back to Japanese if they enter the Change Language feature code. This is because the first time the user enters the feature code, the system will toggle the station to the Use Primary Language. The next time the user enters the feature code, the system will toggle the language to the station's secondary language, which is Spanish. To avoid this situation, simply change the station's secondary language to Japanese and the station's Language field to Japanese.

NOTE: Setting a station's Secondary Language field to a specific language will not work across nodes unless all nodes in the system are at least 5.2F6 or newer since the older systems do not understand specific languages and expect to receive only Primary Language or Secondary Language values.

B. TRUNKS

17.11 A flag in DB Programming determines the language that will be used by each trunk. This field can be set to any specific language so that the system can support more than two languages. The language choices are Use Primary Language, Use Secondary Language, American English, British English, Japanese, or Spanish (Mexico City Spanish Dialect for VPU and EVMC, American Spanish Dialect for EM). If the trunk is programmed for the primary language, callers who reach a Voice Processing application will hear the voice prompts in primary language (unless overridden). If programmed for the Secondary Language, the voice prompts will be in the secondary language (unless overridden). By default all trunks are set for the primary language.

NOTE: Setting a trunk or digit translation Language field to a specific language will not work across nodes unless all the nodes in the system are at least 5.2F6 or newer since the older systems do not understand specific languages and expect to receive only Primary Language or Secondary Language values.

C. DO-NOT-DISTURB AND REMINDER MESSAGES

17.12 The system has default do-not-disturb and reminder messages in American English, British English, Japanese, and Spanish. However, the messages that use Japanese characters can be reprogrammed only through an Administrator's station (see page 5-283).

17.13 The language of the messages seen by the station user, both when programming his or her station and when calling another station that is in do-not-disturb, are determined by the station's programmed language. That is, if a system's primary language is Japanese, and the station is programmed for the primary language, the user will see only Japanese messages when programing a message. If a Japanese-programmed station calls an American English-programmed station that has selected do-not-disturb message 02, the Japanese station's user will see the Japanese version of message 02.

NOTE: Only Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones can be used for viewing Japanese messages. Other phones do not support Japanese displays.

17.14 When do-not-disturb or reminder messages are reprogrammed, the programmer should attempt to keep the meanings for the messages in both lists the same. That is, if the Primary Language do-not-disturb message 02 is changed to "PAGE ME," a similar message should be programmed for the secondary language do-not-disturb message 02.

D. VOICE PROCESSING

17.15 When a Voice Processing application receives a call from a station or trunk, the phone system tells the application which language is programmed for that device. For example, if the Primary Language is set to American English and the Secondary Language is Japanese, then:

- If a trunk programmed for American English rings in to an application, the Voice Processing will play American English prompts.
- If a station programmed for Japanese calls voice mail, the station user will hear Japanese prompts.
- If a station programmed for American English receives a call on a trunk that is programmed for Japanese and then transfers the call to voice mail, the caller (using the trunk) will hear Japanese voice prompts.
- If a Japanese station is forwarded to voice mail, a caller on an American English station or trunk will hear the "American English" prompts (the user-recorded mailbox greeting will be heard in the language in which it was recorded) when the call is forwarded.

17.16 When using a Call Routing Announcement application with digit translation, the individual Voice Processing applications assigned to the digits can be programmed to override the device language and provide prompts in one language only. Using this method, you can program a Call Routing Announcement tree that offers callers a choice between languages. For example:

- The Call Routing Announcement application could have a greeting that says, "Thank you for calling. For English prompts, press 1. NIHONGO WA, 2 WO OSHITE KUDASAI (for Japanese, press 2)."
- The digit translation for digit 1 would lead to an application that overrides the calling device's programming and uses only American English prompts.
- The digit translation for digit 2 would lead to an application that overrides the calling device's programming and uses only Japanese prompts.

17.17 In the example above, the digit translations could be nodes that lead to various other language-specific applications. Or, the first level can give more choices, such as American English voice mail or automated attendant and Japanese voice mail and automated attendant. Either way, the individual applications or nodes can be programmed to play only one language, or they can use the device's language, as needed.

17.18 Multi-Language Feature on Network Operation: All nodes does not have to match the languages because the voice processing system installs any existing language prompt, not just the languages designated as primary or secondary. Therefore, for calls across network nodes, if the language does not match between the two nodes, the called voice processing system searches for the designated language in its system in the following order; Primary language, Secondary language, American English, British English, Japanese, and Spanish (Mexico City Spanish Dialect for VPU and EVMC, American Spanish Dialect for EM) and uses the designated language. If the designated language does not exist in the voice processing system, the system uses the default primary language instead.

E. DIRECTORIES

17.19 If Multilingual Support is enabled, directory users will be able to enter English/Spanish and Japanese (Katakana) characters, as needed. Procedures for using the directories are given on page 5-241.

17.20 In directories, the English/Spanish characters are alphabetized before the Japanese characters. Therefore, Japanese names will be at the end of the list, after the English/Spanish names.

F. CHANGE LANGUAGE FEATURE CODE

17.21 A station user can change the assigned language for the station by entering the Change Language feature code only when the station is idle.

17.22 To change the station's language:

Inter-Tel Phone: Enter the Change Language feature code (301) to change your station between the programmed primary and secondary languages, as desired. The display changes to the current language.

Single-Line Set: Lift the handset and enter the Change Language feature code (301) to change your station between the primary and secondary languages, as desired. (To verify the current language, you can call a Voice Processing application that does not override the device language.)

17.23 A phone user can assign the Change Language feature code to any user-programmable button.

17.24 Changing a phone's language automatically changes any display. For example, if a station changes from the primary language to the secondary language while in do-not-disturb mode with message 02, the display would change from primary language message 02 to secondary language message 02.

NOTE: Only Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones can be used for viewing Japanese displays. Other phones do not support Japanese displays.

18. AUTOMATIC DAYLIGHT SAVING TIME ADJUSTMENT

18.1 In previous versions, the system administrator had to reset the system time whenever Daylight Saving Time [British Summer Time in Europe] became effective. As of version 8.2, however, the system automatically adjusts the time whenever Daylight Saving Time is in effect. The system time is also modified when Daylight Saving Time ends.

18.2 You can program the system so that it automatically adjusts the system time whenever Daylight Saving Time is or is not in effect. If properly programmed, the system will adjust the time as follows:

- *(For the U.S.) On the appropriate day in spring,* the system time will "spring ahead" one hour at 1:59:59 AM, changing the time to 3:00:00 AM.
- *(For Europe) On the appropriate day in spring,* the system time will "spring ahead" one hour at 12:59:59 AM, changing the time to 2:00:00 AM.
- *On the appropriate day in fall,* the system time will "fall back" one hour at 1:59:59 AM, changing the time to 1:00:00 AM.

18.3 When the time changes, the system issues a message to Message Print. See for details.

18.4 Because this feature changes the system time, the scheduled delayed major reset may be affected as follows:

- *If the reset is scheduled to occur within four minutes of the time change,* the reset is delayed by five minutes. For example, if the reset is scheduled for 1:58, the reset will occur at 2:03, which is five minutes later.
- *If the reset is scheduled to occur between 2:00:00 AM and 3:00:00 AM in the spring,* the system does not reset. Instead, the system performs a delayed major at the next scheduled date and time.
- *If the reset is scheduled to occur between 1:00:00 AM and 2:00:00 AM in the fall,* the system resets twice: once before the time change and once after. For example, if the reset time is set for 1:30 AM on the day of the fall time change, the system resets at 1:30 AM. Then, at 2:00 AM (1:59:59), the time changes to 1:00 AM. When it is 1:30 AM, the system resets again.

NOTE: Inter-Tel recommends that you do ***not*** schedule automatic functions, such as backups and resets, to occur at 2:00 AM. If you do, the system may not perform the function when the time changes.

18.5 In previous versions, a scheduled major reset was required when various programming operations were completed. As of version 8.224, however, the following operations no longer require a major reset:

- Changing ACD agent IDs.
- Changing extension numbers (this includes any extension number, not just UCD/ACD, such as station, trunk group, etc.)

18.6 Because the major reset is not required for these actions, administrators are not prompted to perform an immediate or scheduled reset if performed from an administrator's phone. Instead, these database changes take effect immediately.

19. PRIVATE IP NETWORKING

19.1 Previously, the Internet Protocol (IP) Networking Module (3200N) was the primary method to connect Inter-Tel systems to IP networks (for details on the IP Networking Module, see *the IP Networking Module Manual,* document part no. 730.8001).

19.2 As of v7.0, however, a v1.x firmware networking Internet Protocol Resource Card (IPRC) or v8.x or later firmware IPRC can be programmed to provide private networking using an IP network.

A. COMPATIBILITY WITH EXISTING PRODUCTS

19.3 The telephone system supports several private networking configurations with various components. The compatibility of these components must be considered when setting up a private network. Between any two endpoints, there can be IPRCs, T1/PRI, E1/PRI or T1/E1/PRI cards, and IP Networking Modules. Each of these components can only communicate with like components.

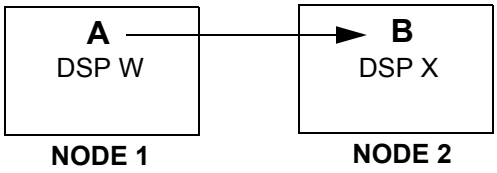
19.4 For example, an IPRC can communicate directly with another IPRC, but not with a T1/PRI, E1/PRI, or T1/E1/PRI card or with an IP Networking Module. While the IPRCs, T1/PRIs, T1/E1/PRIs, and IP Networking Modules cannot communicate directly, they can still be used in the same system multi-node network (see page A-2 for an illustration).

B. RESOURCE ALLOCATION

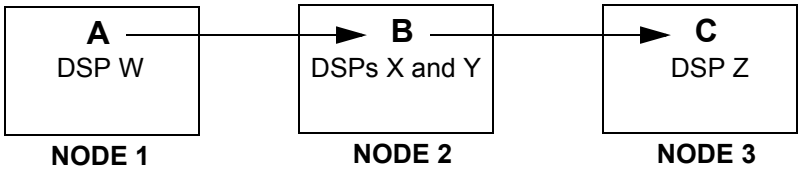
19.5 System resources for Private IP Networking are allocated in the following manner:

- **DSP Resources**: The system allocates DSPs upon call setup. If the DSPs necessary to complete a call are not available, the call camps-on to the DSPs and the system does not deliver the call to its destination until the resources are available.
- **Camp-Ons**: Similar to private networking using T1/E1/PRI, the system first attempts to route a call using every programmed route. If all routes fail, the system then camps-on to the first programmed route.
- **Transfers**: The system does not release DSP resources following a transfer, even if those resources are no longer needed for the current audio configuration.

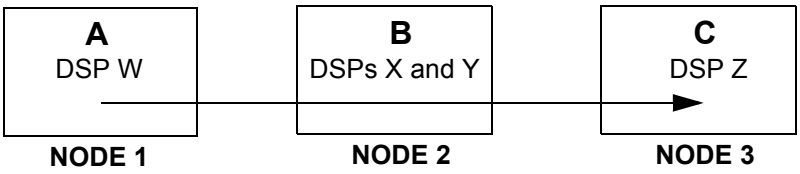
  For example, device A on node 1 calls device B on node 2 using DSP W on node 1 and DSP X on node 2.



Device B transfers the call to device C on node 3 using DSP Y on node 2 and DSP Z on node 3 for the transfer announcement call.



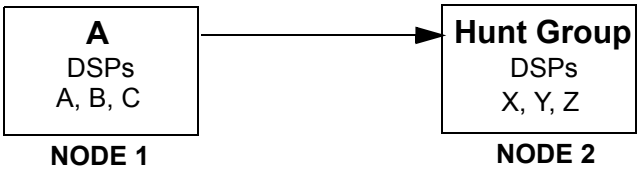
When device B hangs up to complete the transfer, the system may optimize the route such that the system uses only DSP W on node 1 and DSP Z on node 3 to make the audio connection. However, the system does not release DSP X and DSP Y on node 2 for use by other calls.



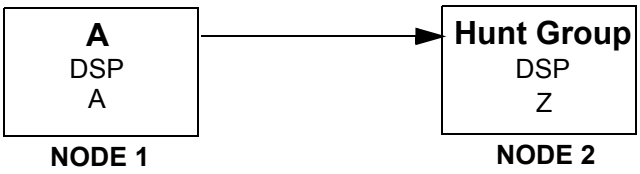
NOTE: Additional DSPs may be consumed if a call is transferred to multiple locations.

- **Multiple Ring-In Calls**: The system reserves DSP resources for each leg of a complex call. However, once the call is answered, the system releases all DSPs that are allocated for legs that no longer exist.

  For example, device A on node 1 calls an All-Ring Hunt Group with three members on Node 2. The system reserves three DSPs on each node allowing each member to receive an incoming call ring.



  If only two DSPs are available on one of the nodes, the system reserves the two available DSP pairs and rings only two members (camping onto the third DSP pair). After a member answers the call, the system releases the extra DSPs such that only one DSP on node 1 and one DSP on node 2 are used.



C. DATA CONNECTIONS

19.6 The following data connections are maintained by the Private IP Networking feature:

- **Data Channels**: The system maintains one data connection between each node programmed for Private IP Networking. The system uses this data connection for call control and datagram messaging. When a data connection fails, the system attempts to establish a new connection using alternative IPRCs if they are available. If all data connections fail, the system may drop calls in progress, and further call attempts using the connection are blocked.
- **Route Optimization**: Call control route optimization using IP works almost identically to call control route optimization using T1/E1/PRI. The system identifies instances where a call passes through the same node twice (a route loop) and eliminates the loop. Note that the system does release all resources within the loop, including DSPs.

D. AUDIO CONNECTIONS

19.7 The system monitors the following audio connections:

- **Audio Connection Status**: The system monitors the connectivity of all IPRCs configured for private networking to avoid connecting to an IPRC that is not connected to the network. To avoid network congestion, the system may take up to 30 seconds to identify an IPRC that has lost its connection. During this time, the system may fail to make audio connections for new calls.
- **Audio Connection Parameters**: The system uses the audio connection parameters specified by the node on which the call originates.
- **Audio Route Optimization**: The system attempts to find the audio route that requires the fewest conversions from IP to Pulse Code Modulation (PCM). It can only optimize the audio route between IPRCs. Also, audio route optimization occurs only when a party, such as a hunt group, multi-ring device, or transferring party, merges out of a call; not during call setup. If the call passes through a system node running version 6.X or earlier, the system aborts any attempt to optimize audio.
- **Insufficient Bandwidth**: The system monitors the in time audio packet thresholds, which are determined by the Average In Time Percentage Threshold and Average In Time Frame Timer fields (located in DB Programming under System\Devices and Feature Codes\Node IP Connection Groups\<*Extension*>\IP Call Configuration; see page 6-185 for more information on these settings). When the system detects that these thresholds have been violated on a call using a vocoder other than G.729B, the system will:
  - — Display a transient warning message that says INSUF BANDWIDTH FOR VOICE.
  - — Generate a message print warning.
  - — Leave the call active.

  If this occurs, you have the choice of continuing the call or terminating it. For details about bandwidth utilization, see page 2-94.

E. IP CONNECTIONS AND IP CONNECTION GROUPS

19.8 IP Connections: Contain the programming information for each IP connection in the network. The extensions associated with each connection are automatically added to this folder when local or remote IP connections are created. Programmable information for each connection includes IP addresses, gateway addresses, subnet masks, backplane configurations, etc. See page 6-172 for details.

19.9 Node IP Connection Groups: Correspond to an IP network connection between the remote node and the local node. Node IP Connection Groups are automatically added for each remote node created. Programmable information for each Node IP Connection Group includes IP call configurations, emergency outgoing access lists, outgoing access lists, etc. See page 6-185 for details.

F. QOS (QUALITY OF SERVICE)

19.10 The system sets the ToS (Type of Service) bits in the IP packet headers to aid the network in implementing QoS. The system uses the Audio RTP Type of Service and Data Type of Service flags (see page 6-174) to set the appropriate ToS bits in the header.

G. AUDIO CONNECTIONS TO IP DEVICES

19.11 The system recognizes that both IP phones and voice resources on IPRCs configured for private networking require different audio connection parameters than other devices in the system.

NOTE: When using the networking feature in systems running software version 7.0 and later, to avoid intermittent one-way audio, do not use two IPRCs on separate networks.

19.12 The affected connection parameters include:

- **Gain Values**: For the purpose of gain adjustment, the system now recognizes IP phones and IP connections as IP devices and adjusts the gain settings accordingly.
- **Enhanced Speakerphone Mode**: When a phone connects to an IP phone or to a voice resource on an IPRC, the system does not allow the phone to enter enhanced speakerphone mode. Enhanced speakerphone mode does not function properly when the audio travels over an IP network.

NOTE: Sometimes the system may not detect that a call is connected over an IP network. If this occurs, the system may allow users to enable enhanced speakerphone mode, which will affect the call quality.

H. DTMF CONFIGURATION

NOTE: To configure DTMF detection, IPRC firmware version 8.2.0 or later and Call Processing v8.224 or later is required.

19.13 You can program each IPRC to use a specific set of values that affect DTMF detection. This configuration allows you to make the system more or less sensitive to DTMF tones, which ultimately affects the amount of talkoff. (In general, the more sensitive the system is to DTMF tones, the more likely talkoff is to occur.) For details about configuring the DTMF setting, see page 6-154.

I. DIAGNOSTICS

19.14 The IP Networking application on the IPRC includes several tools for performing diagnostics when troubleshooting. These tools are separate from call processing and can display only information local to the IPRC. There are also programming displays to control how the call behaves. (See page 7-71 for details.)

19.15 The five groups of diagnostics tools are as follows:

- **Common Serial**: Contains text displays such as a thread registry, an on-line monitor, and network information.
- **IPRC-Specific Serial**: Contains text displays that show the current status of a call, how call processing has mapped the cross-point connections, and how call processing has mapped the sessions and channels.
- **Web/HTML**: Contains displays that have a graphical interface for ease of use.
- **Message Print**: Allows you to see any information, warning, or debug messages when a predetermined condition (e.g., a bus error) occurs.
- **Exception Log**: Is a collection of Exception Log entries that signify major violations, errors, or events. Major resets, minor resets, code-checker exceptions, bus errors, and unhandled exceptions are examples of Exception Log entries.

20. PEER-TO-PEER AUDIO

NOTICE

Passing real-time streaming data, such as audio, through encrypted virtual private networks (VPNs) may significantly impact network performance, router and firewall functionality, and audio quality.

20.1 The peer-to-peer (P2P) audio feature allows certain Internet Protocol (IP) and multi-protocol, SIP or Inter-Tel Protocol (ITP), devices to transmit and receive audio directly with each other. With this feature, the audio is not transmitted or received through the system chassis. This reduces delay and removes the audio stream from the Time Division Multiplex (TDM) highway.

20.2 P2P audio requires that you upgrade the IPRC and IP devices (IP phones and IP SLAs) to firmware version 8.2 or later. In addition, each node that will use P2P audio must be running software version 8.2 or later.

CAUTION

Inter-Tel recommends that you run the Network Qualifier (for at least 24 hours) to determine if the network meets the minimum requirements identified for P2P audio. These requirements are the same as those identified for IP private networking.

20.3 Devices that support P2P audio include:

- AudioCodes MP-114 or Quintum AFT 400 SIP gateway
- IP endpoints, except the IP SoftPhone
- SIP endpoints

A. USING P2P AUDIO

20.4 To use P2P audio, IP devices must be members of the same Network Group (see page 6-182). In addition:

- The IPRC and IP devices must be upgraded to version 8.2 or later firmware. (SIP endpoints do not require firmware upgrades.)
- The devices must terminate on an IPRC with v8.2 or later firmware.
- There cannot be a NAT or firewall between the communicating devices.
- For SIP endpoints, the latest version of SIP Server (v1.1 or later) must be installed.

NOTE: For details about SIP Server and the associated endpoints, see the latest version of the *SIP Server Installation and Configuration Manual*.

20.5 Because the devices are not using chassis resources, devices connected in a P2P call cannot use the following features:

- Agent Help
- Record-A-Call
- Station Monitor

NOTE: The IP SoftPhone does not support v8.x or later firmware and cannot use P2P audio.

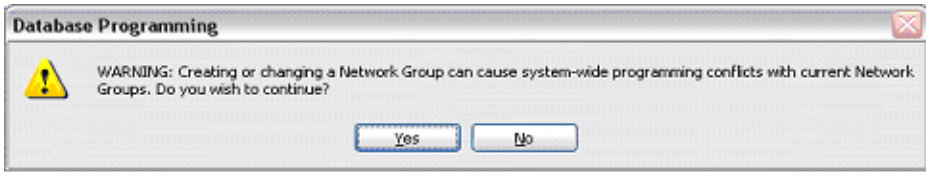
B. NETWORK GROUPS

20.6 Network Groups define the IP devices, MGCP/SIP gateways, SIP trunks, SIP endpoints, and MGCP endpoints that can connect to each other using P2P audio. Only devices within the same Network Group can talk to each other using P2P audio. Even devices that are on separate nodes must be in the same Network Group. For example, for devices on node 2 to communicate via P2P with devices on node 3, both nodes must have the same Network Group, with the same extension programmed. The associated devices on each node must then be members of that group. If two devices are in separate Network Groups, P2P is not used, and the call is routed through the chassis.

Network Broadcast of Network Group's Extension

20.7 In system software v9.0 and later, when you create a Network Group (see page 6-183) the creation is marked by a network broadcast of the Network Group's extension. When the broadcast is received by the remote node, the Network Group is automatically created if it does not already exist. The broadcast messages sends the Network Group's extension number and the value you select for the Use Peer-to-Peer Audio flag (**Yes** or **No**) to the other nodes. By default, the Use Peer-to-Peer Audio flag is set to **No**. Any time you change the Network Group's extension or the value set for the Use Peer-to-Peer Audio flag, the updated information is sent to the other nodes in the network.

20.8 Because creating or updating Network Groups has system-wide effects, DB Studio displays the following warning when you create or change a Network Group.



20.9 If there is a conflicting (duplicate) Network Group extension in a broadcast from another node, the receiving node ignores the new extension in favor of its existing extension number. If the value for the Use Peer-to-Peer Audio flag differs, the receiving node's flag is overwritten by the broadcasting node. **No broadcast message is sent when you delete a Network Group.**

NOTE: When you delete a Network Group, the IP devices in the deleted Network Group are moved to the Default Network Group. The Default Network Group cannot be deleted.

Export/Import Network Groups

20.10 Normally, when a node adds a new Network Group or changes an existing one, the node broadcasts the new extension information to the other nodes in the network. If another node does not receive the broadcast (e.g., DB Studio is active on the node, the node is down, etc.) the Export/Import function located on the Operations menu (see page 6-184) can be used to export/import Network Groups information to/from another node.

20.11 When you import or export Network Groups information and there is a conflicting (duplicate) Network Group extension in a broadcast from another node, the receiving node ignores the new extension in favor of its existing extension number. If there is a conflicting value for the Use Peer-to-Peer Audio flag between nodes (one node's flag is set to **No** and the other node's flag is set to **Yes**) the receiving node's flag is overwritten by the other node's flag.

20.12 For example, say Node 1 changes the Use Peer-to-Peer Audio flag for an existing Network Group (P7000) from **No** to **Yes** while Node 2 is down. When Node 2 comes online again the value for P7000's Use Peer-to-Peer Audio flag is different between nodes. (Node 1 is set to **Yes**, Node 2 is set to **No**.) If Node 2 imports Network Group information from Node 1, the Use Peer-to-Peer Audio flag for P7000 is overwritten and changes from **No** to **Yes** on Node 2.

20.13 For Network Groups troubleshooting information, see page 7-65.

21. SIGNALS AND TONES

21.1 There are several distinct signals and tones on the phone system.

A. RING SIGNALS

21.2 The system signals and tones generated by the ringer or phone speaker are as follows.

Table 5-33. *Ring Signals (U.S.)*

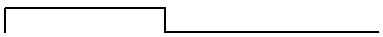
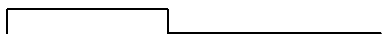
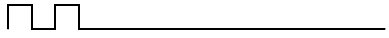
Type of Call	Signal
Outside call or recall	Single long tone every six seconds 
Intercom call, off-hook voice announce call, or queue callback	Two short tones and a pause (repeats for private calls and callbacks). **NOTE:** If the station option for different IC/CO alerting tones is disabled, this will be the same as the outside call signal shown above.
Reminder message	Eight short tones

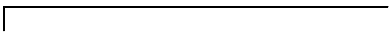
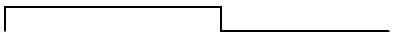
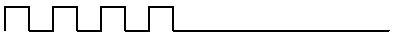
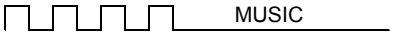
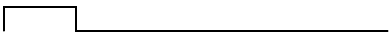
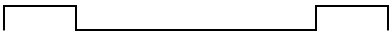
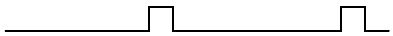
Table 5-34. *Ring Signals (Europe)*

Type of Call	Signal
Outside call or recall	Two short tones and a pause (repeats for private calls and callbacks). **NOTE:** If the station option for different IC/CO alerting tones is disabled, this will be the same as the outside call signal shown above.
Private intercom call, intercom recall, or queue callback	Single long tone every six seconds 
Handsfree intercom call or off-hook voice announce call	Two short tones and then connection 
Reminder message	Eight short tones

B. CALL PROCESSING SIGNALS

21.3 The following signals and tones are heard through the handset or phone speaker.

Table 5-35. *Call Processing Signals*

Signal	Action
Intercom dial tone 	Handset lifted or Speaker or IC button pressed.
Double tones (called station receives the same signal)	Intercom call—repeats for private call or call to non-handsfree station; one double tone for handsfree call.
(Europe Only) Long tone (called station receives the same signal) 	Private intercom call or call to non-handsfree station.
Intercom busy signal—slow repeating tones	Called station or selected trunk is busy.
A repeating signal of four fast tones and a pause	Called station is in do-not-disturb.
Reorder tones—fast tones (may be four or continuous) 	Entry was not accepted, number is toll restricted, or dialing timer expired.
System busy signal—reorder tones followed by music 	All resources are busy, station is camped on.
Single tone 	Confirmation that procedure was completed, or prompt that something needs to be entered.
Camp-on tone—single tone every 15 seconds (programmable) 	Another station has camped on to this single-line station. (This tone is heard through the phone speaker only.)
Single-line message waiting tone—six tones followed by intercom dial tone after lifting the handset or hookflashing (or pressing **RECALL**)	A message is waiting at this single-line station.
Single-line call forwarding/do-not-disturb tone—dial tone periodically interrupted by a single tone after lifting the handset or hookflashing (or pressing **RECALL**) 	This single-line station has call forwarding and/or do-not-disturb enabled.

22. INTERCOM CALLS

CAUTION

When the procedures tell you to hookflash (recall), quickly press and release the hookswitch [press **RECALL** in Europe]. If you press the hookswitch to hang up, hold it down until the SL Hookflash Maximum timer expires (default value is 1.2 seconds); otherwise, the system recognizes it as a hookflash.

22.1 The intercom can be used to place station-to-station calls that can be answered handsfree. Or, it can be used to place private (non-handsfree) calls. A station user that reaches a busy station can camp on, request a callback (queue), leave a message, or use the Off-Hook Voice Announce feature (if enabled). Other features that apply to both outside and intercom calls, as described later in this section of the manual, include placing calls on hold, call waiting, call transfer, reverse transfer, and call forwarding.

22.2 The network allows handsfree intercom calls when calling from one node to another, unless the call camps on to the node before being sent to the other node.

A. PROGRAMMING FOR PRIVATE INTERCOM CALLS

22.3 A station user can place private (non-handsfree) intercom calls by pressing the pound [hash] (#) button before dialing the extension number. Or, the station can be programmed to always send and/or receive private calls using the Ring Intercom Always or Handsfree Disable features.

Ring Intercom Always

22.4 A station user can always place private calls by programming the station with the Ring Intercom Always feature code. While this feature is enabled, the called party hears repeating double tones and must lift the handset (or press the Speaker button, ANSWER button, or IC button) to answer. (If the station does not have an IC button, the call will appear under a CALL button.)

22.5 In the default database, all single-line stations have this feature enabled.

22.6 **To turn ring intercom always on or off:**

Inter-Tel Phone: While on hook, enter the Ring Intercom Always On/Off feature code (377). The display shows RING IC ALWAYS ON (or OFF).

Single-Line Set: Lift the handset. Enter the Ring Intercom Always On/Off feature code (377).

Phone Handsfree Enable/Disable

22.7 A phone user can always receive private calls by disabling the phone Handsfree Answering feature using the Handsfree On/Off feature code, as described in the following paragraph. The phone user hears repeating double tones when receiving an intercom call and must lift the handset (or press the **SPKR**, **ANSWER**, or **IC** button) to answer while Handsfree Mode is disabled.

22.8 **To enable or disable handsfree mode:**

While on hook, enter the Handsfree On/Off feature code (319). The display shows HANDSFREE MODE ON (or OFF).

B. STATION-TO-STATION CALLING

Receiving Intercom Calls

22.9 To receive an intercom call:

- Inter-Tel Phone:
  - — *If you hear a double tone and a call announcement (handsfree call),* and the Speaker button is lit, respond handsfree or lift the handset. Display phones show IC TO *(username)* to identify the source of the call.
  - — *If you hear repeating double tones (private call),* lift the handset, or press the Speaker, ANSWER, or IC button to respond handsfree. Display phones show INTERCOM CALL FROM *(username)* to identify the source of the call.
- **Executive Display, Professional Display, or Model 8560 Phones**: Instead of answering the call, you may press one of the following menu buttons:
  - — **SEND TO V-MAIL**: Appears only if you have a voice mailbox. Pressing this button sends the call directly to your mailbox greeting.
  - — **IC CALL TO DND**: Sends do-not-disturb tones to the caller and allows you to program a do-not-disturb message. If using a display phone, the caller will see DO-NOT-DISTURB. (See page 5-257 for detailed instructions on using Do-Not-Disturb.)

22.10 To receive an intercom call on a single-line set:

When you hear repeating double tones, lift the handset.

22.11 To receive a transferred-to-hold intercom call:

> **NOTE:** If the station does not have an IC button, the call will appear under a CALL button.

- **Inter-Tel Phone**: When the IC button (or CALL button) is fluttering and the display shows *(calling station)* TFR FROM *(transferring station)*, lift the handset and press the IC button (or CALL button). Or, while on hook, press the IC button (or CALL button) to speak handsfree. You are connected to the caller. (Phones with menu buttons have the same menu button options as described for a ringing intercom call.)
- **Single-Line Set**: When you hear a page announcing that a call is holding at your station, lift the handset (at your station) and enter the Individual Hold feature code (336). You are connected to the caller.

Placing Intercom Calls

22.12 When placing an intercom call, begin dialing before the Dial Initiation timer expires. If the timer expires, the system drops the connection and sends reorder tones.

22.13 To place an intercom call:

NOTE: In the procedure below, if you want to place a private (non-handsfree) call to a phone, press the pound [hash] (#) button before dialing the extension number.

1. **To use the handset:** Lift the handset. The display shows ENTER EXTENSION NUMBER.

   To use the speakerphone: Skip to step 2 (begin dialing). The Speaker button lights. The display shows ENTER EXTENSION NUMBER.

   Phones with menu buttons: Lift the handset or begin dialing as described above. The display shows ENTER EXTENSION NUMBER. You may press one of the following menu buttons:

   — **IC DIRECTORY**: Press this button to use the Intercom Directory feature. See page 5-241 for instructions.

   — **OUTSIDE CALL**: If you would rather make an outside call, press this button and see the instructions on page 5-192.

   Single-Line Set: Lift the handset. (If the DTMF decoders are busy, you hear silence. Hang up and try again.)

2. Dial an extension number. The number can be 0 for the attendant, a station extension number, or a hunt group pilot number. Display phones show the number dialed. If you dial too slowly, the Interdigit timer may expire, and you will hear reorder tones. One of the following will occur:

   a. *If calling a handsfree phone,* speak after you hear a double tone. The display shows IC TO *(user name)*.

   b. *If placing a private call or calling a single-line station, a hunt group, or a phone with handsfree disabled,* you hear repeating double tones until the call is answered. The display shows CALL RINGING FOR *(username)*. When the call is answered, the display shows IC TO *(username)*. If your phone has menu buttons, you can press one of the following menu buttons while the call is ringing:

      – **LEAVE MESSAGE**: Appears only if you have called a station. Press this button to leave a message waiting indication at the station. See page 5-179 for details.

      – **LEAVE VOICE MAIL**: Appears only if the called station has a voice mailbox. Press this button to call the station's voice mailbox.

   c. *If you dial an invalid number,* you hear reorder tones. Display phones show INVALID NUMBER. If you do not have an assigned attendant and you dialed 0, the display shows NO ATTENDANT PROGRAMMED. If you dialed a hunt group number with no assigned stations, the display shows NO MEMBERS IN HUNT GROUP.

   d. *If the called station is busy,* the display shows *(username)* IS BUSY. You have the following options (in addition to the menu buttons described previously):

      1.) **Disconnect**: Hang up and try later. *If dialing handsfree (phones only)*, press the Speaker button to disconnect.

2.) **Camp on**: Do not hang up, your call camps on. When the called station is available, a private call is placed. See page 5-177 for details.

3.) **Request a callback (queue)**: This is similar to camping on except that you hang up and wait for the busy station to be available. For more information, see page 5-177. If your phone has menu buttons, you can press the **QUEUE** menu button.

e. *If you hear a repeating signal of four fast tones and a pause,* the station is in do-not-disturb. You can use the Queue Callback feature or, if enabled on your phone, use the Do-Not-Disturb Override feature to place a non-handsfree call to the station by pressing ∞ or SPCL and entering the Do-Not-Disturb Override feature code (373) or pressing the **OVERRIDE DND** menu button.

f. *If the called station is busy, in do-not-disturb, or there is no answer,* you can leave a message waiting indication with the station itself or with the station's message center. For details, see page 5-179. Executive Display, Professional Display, and Model 8560 Phone users can press the **LEAVE MESSAGE or LEAVE VOICE MAIL** menu button.

g. *If the called station is a busy phone, and if all the necessary conditions for OHVA are enabled,* you can complete the call using the Off-Hook Voice Announce feature. For a complete explanation of this feature, see page 5-190.

C. INTERCOM CAMP ON AND QUEUE CALLBACK

22.14 Camp-on and queue requests allow the caller to wait for an available resource. A station user waiting for a specific resource (a trunk or a station) will be served before a station user waiting for a group that contains that specific resource (trunk group or hunt group). This includes camped on Emergency Call feature 911 [999] calls – if the Emergency Call feature uses a trunk group, requests for specific trunks will be processed ahead of the emergency call.

Camp On

22.15 When a station user calls a busy station or hunt group, the system sends a busy signal. The caller can wait off hook to camp on (after the Camp-On timer expires) and hear music while waiting until the called station is available. The system periodically sends call waiting signals to the busy station(s).

22.16 A user can camp on to busy stations on other nodes and will hear the other node's music-on-hold while camped on.

22.17 If a station enables do-not-disturb while an intercom call is camped on, the camped-on caller is removed from the camped on state and that caller receives do-not-disturb indications (see page 5-205). Intercom callers cannot camp on to a station that is in do-not-disturb.

22.18 To camp on to an extension number:

- **All Phones**: If you hear a busy signal when calling an extension number and want to camp on, do not hang up. You hear music (if enabled) while camped on. When the station is available, a private call is automatically placed and you hear repeating double tones until the call is answered. The display shows WAITING FOR *(username)*.
- **Phones with menu buttons**: You may press one of the following menu buttons:
  - LEAVE MESSAGE: This button appears only if you have called a station. Press this button to leave a message waiting indication at the station. See page 5-179 for details.

- LEAVE VOICE MAIL: This button appears only if the called station has a voice mailbox. Press this button to call the station's voice mailbox.
- QUEUE: Press this button to request a callback when the resource is available, as described on the next page.

Busy Station Queue Callback

22.19 When a called station is busy or in do-not-disturb, the caller can request a callback (queue) and hang up until the station is available. This can be done even if the call camps on. Each station can have only one active queue callback request at a time.

22.20 The Queue Callback feature can also be used when calling a hunt group if all stations are unavailable.

22.21 As of v5.2, users can queue onto busy stations and trunks on other nodes.

22.22 Queue callbacks must be answered before the Queue Callback timer expires. If a callback is not answered, the queue is canceled. If the station is busy when called back, the queue request is placed behind any other waiting queue requests.

22.23 Queue callbacks do not follow forwarding and are not blocked by do-not-disturb mode.

NOTE: In v9.0 and later system software, two stations can no longer queue on to each other.

22.24 To request a callback from a station:

1. **Inter-Tel Phone**: When you hear a busy signal or do-not-disturb signal (a repeating signal of four fast tones and a pause) when calling an extension number, press the **QUEUE** menu button or press ∞ or SPCL and enter the Queue Callback feature code (6) and hang up. The display shows QUEUE REGISTERED FOR *(username)*.

   Single-Line Set: When you hear a busy signal or do-not-disturb signal when calling an extension number, hookflash (press **RECALL**) and enter the Queue Callback feature code (6). Then hang up.

2. When the queued station is available, your station will ring. Display phones show *(username)* IS NOW AVAILABLE. Lift the handset or press the ANSWER button. A private call is automatically placed to the queued station.

   Phones with menu buttons: You may press one of the following menu buttons:

   - CANCEL QUEUE: To cancel the queue request, press this button without lifting the handset.
   - ANSWER: You may press this button instead of lifting the handset to answer the queue callback using the speakerphone.

22.25 To cancel a queue request (before the queued station becomes available):

Inter-Tel Phone: While on hook, enter the Queue Callback feature code (6). The display shows QUEUE REQUEST CANCELED.

Single-Line Set: Lift the handset, enter the Queue Callback feature code (6) and hang up.

D. PEER-TO-PEER AUDIO FOR IP AND SIP DEVICES

NOTICE
Passing real-time streaming data, such as audio, through encrypted virtual private networks (VPNs) may significantly impact network performance, router and firewall functionality, and audio quality.

22.26 The peer-to-peer (P2P) audio feature allows certain Internet Protocol (IP) and multi-protocol (SIP or ITP) devices to transmit and receive audio directly with each other. With this feature, the audio is not transmitted or received through the system chassis. This reduces delay and removes the audio stream from the Time Division Multiplex (TDM) highway.

22.27 Devices that support P2P audio include:

- AudioCodes MP-114 or Quintum AFT 400 SIP gateway
- IP endpoints, except the IP SoftPhone
- SIP endpoints

22.28 To use P2P audio:

- The devices must be members of the same Network Group (see page 6-182).
- The IPRC and IP devices must be upgraded to version 8.1 firmware. (SIP endpoints do not require firmware upgrades.)
- The devices must terminate on an IPRC with v8.1 firmware.
- There cannot be a NAT or firewall between the communicating devices.
- For SIP endpoints, the latest version of SIP Server (v1.1 or later) must be installed.

NOTE: For details about SIP Server and the associated endpoints, see the latest version of the *SIP Server Installation and Configuration Manual*.

22.29 Because the devices are not using chassis resources, devices connected in a P2P call cannot use the following features:

- Agent Help
- Record-A-Call
- Station Monitor

NOTE: The IP SoftPhone does not support v8.1 or later firmware and cannot use P2P audio.

23. CALL LOGGING

23.1 The Call Logging feature allows digital/IP endpoints and IP softphone users to log the following calls on their endpoint (this feature is not applicable to Executone phones and single-line phones):

- Missed Calls
- Received Calls
- Dialed Calls

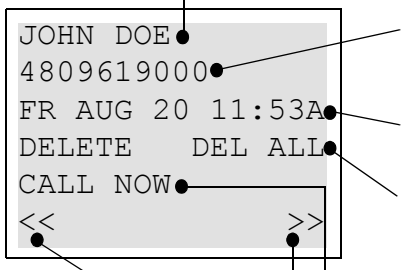
23.2 The Call Logging feature shows the user who called them when they were away, which makes it easy to redial those callers. Six-line display endpoints are recommended for field visibility and easier usage. Call Logging has many advantages including redial for intercom (IC) calls and outgoing central office (CO) calls, storage of caller ID from incoming calls, and missed call notification. The system provides the user with an interface similar to their personal cellular phone. The call lists are stored in Call Processing and are accessible through the user interface on endpoints.

23.3 The maximum number of entries that can be stored in a call logs list is 20 entries. It is programmed in DB Programming per endpoint. The valid range is 0 - 20 entries (the default value is *10*). See page 6-206 for programming details.

23.4 The following endpoints do not support Call Logging:

- IP endpoints in SIP mode
- Non-display
- Single line
- Executone
- Model 8524/8525/8526
- Model 8664/8665/8668

23.5 The following is an example of the Call Logging display on a six-line display endpoint. Two-line display endpoints show only the first two lines. Each call entry contains the following fields:



Name: Displays the name of the missed/received/dialed call. If not available, the first line is blank. If neither the Caller ID name nor number is available, UNKNOWN CALLER appears.

Number: Displays the number of the missed/received/dialed call. If not available, the second line is blank.

Date/Time: Displays the date and time that the call was missed/received/dialed.

DELETE/DELETE ALL: Allows the user to delete the entry indicated on the display or clear all entries.

CALL NOW: Allows the user to call the number indicated on the display.

ARROW MENU BUTTONS (<< and >>): Appear when there is more than one entry in the list on a six-line display endpoint.

23.6 The entries are displayed from newest to oldest. After the list is full, the newest call is moved to the front of the list and the oldest call entry is removed. The user can scroll through all of the entries in the list via the user interface (see page 5-182).

A. CALL LOGGING DISPLAYS ON A SIX-LINE DISPLAY ENDPOINT

NOTE: At any menu level, you can press the asterisk (*) button to cancel or return to the previous menu, or press the pound/hash (#) button to accept.

23.7 The following are examples of the call logging displays on a six-line endpoint:

- **Idle Display.** Six-line endpoints have an option on the idle menu to access the logs directly. Note that the six-line display administrator endpoints no longer show "ADMIN FEATURES" on the first line, but rather just "ADMIN" to make room for the "MISSED XX" (where XX can be from 1-99) calls menu option on the right side of the display (see page 5-182).

Non-Admin Endpoint

```
29999 JOHN DOE
 1:53 FRI AUG 20

OUTGOING      DND
DIRECTORY    LOGS
OTHER    FEATURES
```

Admin Endpoint

```
29999 JOHN DOE
 1:53 FRI AUG 20
ADMIN
OUTGOING      DND
DIRECTORY    LOGS
OTHER    FEATURES
```

- **The Main Menu.** When the LOGS menu option is selected, the following display appears.

```
LOG TYPE: MISS=1
RCV=2 DL=3 CLR=4
MISSED   CALLS
RECEIVED CALLS
DIALED   CALLS
CLEAR CALL LOGS
```

Or, "DIALLED CALLS" for European systems

- **Call Log Display.** When one of the call menu options (Missed=1, Received=2, or Dialed=3) is selected in the main menu, a display similar to the one below appears. The arrow (<< and >>) menu buttons at the bottom of the display only appear if there is more than one entry in the list on a six-line display endpoint.

```
JOHN DOE
4809619000
FR AUG 20 11:53A
DELETE   DEL ALL
CALL NOW
<<            >>
```

OR

```
JANE DOE
x29999
FR AUG 20 11:53A
DELETE   DEL ALL
CALL NOW
<<            >>
```

To scroll through the entries, press the >> menu button or the volume up button to scroll forward, or press << or the volume down button to scroll backward. If there is no previous page or entry, the << menu button is unavailable and the volume down button is unresponsive. Likewise, if there is no next page or match, the >> menu button is unavailable and the volume up key is unresponsive. For an analog display endpoint that does not have a volume soft button, press **9** to scroll forward and **7** to scroll backward.

To delete the entry, press **DELETE** or **0**. The display will then show the next entry in the list or will go back to the main menu if the list is empty.

To delete all entries, press **DEL ALL** or **4**.

To place a call, press **CALL NOW** or the pound/hash (**#**) button.

The Missed Calls Menu. *(For six-line display endpoints only)* If the user has any missed calls that have not yet been viewed, the "MISSED XX" (where XX can be from 1-99) calls menu option appears on the six-line display endpoint, as shown below.

Non-Admin Endpoint

```
29999 JOHN DOE
 1:53 FRI AUG 20
       MISSED 10
OUTGOING     DND
DIRECTORY   LOGS
OTHER   FEATURES
```

Admin Endpoint

```
29999 JOHN DOE
 1:53 FRI AUG 20
ADMIN  MISSED 10
OUTGOING     DND
DIRECTORY   LOGS
OTHER   FEATURES
```

When the user selects the "MISSED XX" calls menu option or enters the Call Logging feature code (333), the display changes to the Missed Calls menu that is similar to the Call Logs display on page 5-181. From there, they can navigate just as they could if they accessed the Missed Calls menu from the LOGS menu.

After the Missed Calls menu option is selected, the idle display will no longer show "MISSED XX" on the display whether the user looked at every one of the recently missed calls.

B. CALL LOGGING DISPLAYS ON A TWO-LINE DISPLAY ENDPOINT

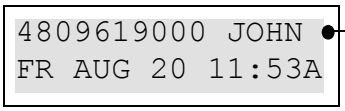
NOTE: At any menu level, you can press the asterisk (*) button to cancel or return to the previous menu, or press the pound/hash (#) button to accept.

23.8 The following are examples of the call logging displays on a two-line endpoint:

- **The Main Menu.** Two-line display users must use a feature code to access the Call Logs. When the user enters the Call Logging feature code (333), the following display appears prompting them to specify which Call Log they want to access or if they want to clear all call logs.

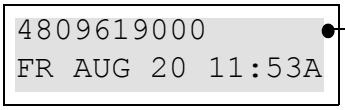
```
LOG TYPE: MISS=1
RCV=2 DL=3 CLR=4
```

- **Call Log Display.** When one of the call menu options (Missed=1, Received=2, or Dialed=3) is selected in the main menu, a display similar to the one below appears.



If both the calling number and Caller ID name are available, the first line contains the number and as much of the name as possible. If neither the calling number nor Caller ID name is available, UNKNOWN CALLER appears.

OR



If the calling number is available, but the Caller ID name is not available, the first line contains only the calling number.

To scroll through the entries, press the volume up button to scroll forward, or press the volume down button to scroll backward. For an analog display endpoint that does not have a volume soft button, press 9 to scroll forward and 7 to scroll backward.

To delete the entry, press 0. The display will then show the next entry in the list or will go back to the main menu if the list is empty.

To delete all entries, press 4.

To place a call, press the pound/hash (#) button.

C. CALL SCENARIOS

23.9 Keep in mind the following considerations regarding the Call Logging feature:

- All incoming calls are logged under the MISSED calls list as well as the RECEIVED calls list unless the user was in DND or forwarded. If the user was in DND or forwarded, the call would be logged as a MISSED call only.
- The following calls are logged under the RECEIVED calls list:
  - — Calls that answered a queue callback
  - — Calls that are recalled
- The following calls are *not* logged under the RECEIVED calls list:
  - — Reverse Transferred calls
  - — Calls that received a station message
- The following calls are logged under the DIALED calls list:
  - — Calls that queued onto an extension
  - — Conference calls (one entry per person)
  - — Calls that left a station message after dialing the called party
  - — Transfer to ring calls (both the transfer announcement call and the resulting transferred call)
  - — Transfer to hold calls are logged as only the transfer announcement call, however, if the Hold timer expires, it also logs the resulting transferred call

- The following calls are *not* logged under the DIALED calls list:
  - — Calls that left a station message (via the feature code) with the called party's message center
  - — Calls that requested Agent Help and/or Record-A-Call
  - — Calls that performed a page
- A busy endpoint that received a queue callback does not log that call until the queue callback is answered.
- Hunt group calls log one entry per call to that hunt group member. For example, if the No Answer Advance timer expires and it rings the same member again, it logs another entry. Also, if it is an all-ring hunt group, it logs an entry per each time it rings every member. Inter-Tel recommends that the Allow Duplicate Entries flag be disabled if the user is often involved in these scenarios (see page 6-206).
- If the endpoint is using system forward or station forward and the action for that endpoint is to forward immediately/all, the call will not be logged in any of the logs. The same holds true if the endpoint was being controlled by Unified Communicator and the Unified Communicator routing rule states that the call should be immediately forwarded, the call will not be logged in any list.

23.10 For information about programming the Call Logging feature, see page 6-206.

24. INTER-STATION MESSAGES

24.1 Intercom callers may leave a message waiting indication if a called station is busy, if there is no answer, if the station is in do-not-disturb, or when they are connected to or placed on hold by another station. Users are allowed to leave a message for and respond to messages from, users on other nodes. There are two messaging options:

- **Have the called party return your call.** Display phones show the source and time of the message. When the called party responds to the message indication, a call is automatically placed to your station.
- **Leave a message with the called party's message center.** Display phones show that a message was left with the phone user's assigned message center. When the called party responds to the message indication, a call is automatically placed to his or her message center.

24.2 To signal that a message is waiting, a called phone's MSG button flashes and the display shows the number of waiting messages. In addition, Model 8500, 8520, and 8560 Phones have a Message Lamp that flashes to indicate a waiting message. Each time the MSG button is pressed, the display shows the message source for each of the waiting messages in the order they were received. For messages from stations, the display shows MSG: *(station)* and the date and time of the message. For messages from voice mailboxes, the display shows XX MESSAGES FROM MBOX XXXX to indicate the number of waiting messages in each mailbox that left a message. (This is especially helpful at stations with multiple mailboxes.) For single-line sets, a system programming option can be enabled that sends six short message waiting tones when the user lifts the handset or presses the hookswitch.

NOTE: The Message Lamp, which is available only on the Model 8500, 8520, and 8560 Phones, is programmable. If the MSG button flashes but the Message Lamp does not, the Message Lamp may have been reprogrammed for another feature. Ask your System Administrator for details.

24.3 Any station, Voice Processing application, hunt group, or off-node device can be designated as the message center for a station. However, a station cannot be programmed as its own message center. (See page 6-203 for information about assigning a station's message center.)

24.4 If the designated message center is a voice mail hunt group, the voice mail hunt group is called after the Message Wait timer expires. When the voice mail unit answers the call, the called party's "mailbox" is automatically dialed. The caller can then leave a message in the mailbox. The called party's MSG button flashes, and the message display indicates that the message was left by the voice mail hunt group.

24.5 Each station user can leave message waiting indications at more than one station. If more than one message is received from one station, the message display will show only the first message, and all other requests will be ignored.

24.6 When a station is forwarded or has system forwarding, and a caller leaves a message waiting indication after calling the forwarded station, the message indication appears at the original station instead of the station that received the forwarded call.

A. LEAVING MESSAGES

24.7 To leave a message waiting indication:

1. When calling a station, if you hear a busy signal, do-not-disturb signal (repeating signal of four fast tones and a pause), or the call is not answered:

   Inter-Tel Phone: Press the MSG button, or press ∞ or SPCL and enter the Message feature code (365). The station will receive a message waiting indication. The display shows HANG UP OR WAIT FOR MESSAGE CENTER.

   Single-Line Set: Hookflash (or press **RECALL**) and enter the Message feature code (365).

2. **To have the called party call you:** Hang up or press a trunk button or the IC button to disconnect before the Message Wait timer expires. The display shows MESSAGE LEFT FOR *(username)*.

   To leave the message with the called party's message center: Do not hang up. When the Message Wait timer expires, a private call is automatically placed to the called party's message center. (If the message center is a voice mail hunt group, the called party's mailbox is automatically dialed after the voice mail unit answers the call.) Leave your message with the message center and hang up. The called station receives message waiting indications. If the message center does not answer the call (that is placed after the timer expires), the display will show that the message was from your station and you will be called when the station user responds to the message indications. The display shows MESSAGE LEFT FOR *(username)*.

NOTE: If the station you are calling does not have a designated message center, you will hear dial tone immediately after pressing the MSG button. The station will receive a message waiting indication.

Silent Messages

24.8 Silent messages can be placed while on or off hook without making an intercom call to the station. (This method should be used by analog voice mail units.)

24.9 To leave a silent message:

Phones with menu buttons: While on hook, press the MSG button. Then press the **LEAVE MESSAGE** menu button. The display shows LEAVE MESSAGE ON EXTENSION. Enter the desired extension number. The number is shown on the display. (If desired, you can press the **IC DIRECTORY** menu button to use the intercom directory to look up a number. See page 5-241 for instructions.)

Other Phones: While on hook, enter the Silent Message feature code (367). Then dial the extension number of the desired station.

Single-Line Set: Lift the handset and enter the Silent Message feature code (367). Then dial the extension number of the desired station and hang up.

Canceling a Message Waiting Indication That You Left

24.10 Occasionally, you may want to cancel a message waiting indication that you left, before the station user responds to the message. If the message was left with the station's message center, you cannot cancel the message waiting indication; only the message center can cancel that message. Use the procedure outlined below to cancel a message that you left.

24.11 To cancel a message you left:

1. **Phones with menu buttons**: While on hook, press the MSG button. Then press the **CANCEL MESSAGE** menu button. The display shows CANCEL MESSAGE ON EXT#. (If desired, you can press the **IC DIRECTORY** menu button to use the intercom directory to look up a number. See page 5-241 for instructions.)

   Other Phones: While on or off hook at your station, enter the Cancel Message feature code (366). The display shows CANCEL MESSAGE ON EXT #.

   Single-Line Sets: Lift the handset and enter the Cancel Message feature code (366).

2. Enter the extension number of the station that has the message indication that you want to cancel. You hear intercom dial tone if off hook or confirmation tone if on hook. The display shows MESSAGE CANCELED FOR *(username)*. (If a message was not present at the called station, you hear reorder tones, and the display shows NO MESSAGES TO CANCEL.)

B. RESPONDING TO MESSAGES

Viewing Waiting Messages

24.12 To view received messages at a display phone:

1. If your MSG button is flashing, one or more messages are waiting and the display shows YOU HAVE XX MESSAGES.

   Phones with menu buttons: While on hook, press the **VIEW MESSAGES** menu button. Then press **PREVIOUS** or **NEXT** to scroll to the desired message.

   Other Phones: While on hook, press the MSG button repeatedly to view waiting messages.

2. A displayed message will show MESSAGE RECEIVED FROM EXT XXXX. *If desired,* you can answer or cancel a message while it is displayed, as described below.

Answering Messages

24.13 To answer messages at a display phone:

1. When the desired message appeared (as described above), press the # button or the **REPLY** menu button while on or off hook. Depending on how the message was left, an intercom call is placed to either the station that left the message or your message center.

NOTE: If the called station is unavailable when you respond to the message (does not answer a private call, is busy, or is in do-not-disturb), your MSG button continues to flash (and the display indicates a waiting message) unless you immediately leave a message before hanging up or cancel the message after hanging up. The message waiting indications are canceled automatically only when you have actually connected with the called station. Message indications from voice mail will only be cleared if you save or delete the waiting voice mail message. The MSG button will re-light if you clear the message indication without saving or deleting the voice mail.

2. If the MSG button and/or Message Lamp continues to flash, there is another message (display shows the next message). Repeat step 1.

24.14 To answer messages at a non-display phone or single-line set:

1. **Inter-Tel Phone**: If you see a flashing MSG button, press the MSG button. Depending on how the message was left, an intercom call is placed to either the station that left the message or your message center.

   Single-Line Set: If you hear single-line message waiting tones (six tones) after lifting the handset or hookflashing (pressing **RECALL**) when the station is idle, enter the Message feature code (365). Depending on how the message was left, an intercom call is placed to either the station that left the message or your message center.

   If the called station does not answer, is busy, or is in do-not-disturb, you may hang up or leave a message at that station before hanging up. You will continue to receive message waiting indications (MSG button flash or tones), unless you left a message while attempting to retrieve the message or canceled the message after hanging up. The message waiting indications are canceled when you are actually connected with the called station. Message indications from voice mail will only be cleared if you save or delete the waiting voice mail message. The MSG button will re-light if you clear the message indication without saving or deleting the voice mail.

2. If message waiting indications continue, there is another message. Repeat this procedure.

Canceling a Waiting Message

NOTE: Message indications from voice mail will only be cleared if you save or delete the waiting voice mail message. The MSG button will re-light if you clear the message indication without saving or deleting the voice mail.

24.15 To cancel a displayed message without answering at a phone:

1. When the message you want to cancel appeared (as described on the previous page), remain on hook and press the asterisk (*) button or the **DELETE** menu button. The message is canceled.

2. If the MSG button continues to flash, there is another message. The display shows the next message. Either respond to it or cancel it.

24.16 To cancel a message without answering at a single-line set or without displaying messages at a phone:

1. If you receive message waiting indications, enter the Cancel Current Message feature code (368). The message is canceled.

2. If you continue to receive message waiting indications, there is another message. Either respond to it or cancel it.

Alternate Message Source

24.17 The Alternate Message Source feature was developed to enable a voice mail unit to leave messages through a station or hunt group that is designated as its alternate message source. When a message is left on the voice mail unit for a station user that has an assigned mailbox, the called station receives the message waiting indications. If it is a display phone, the display shows MESSAGE RECEIVED FROM *(alternate message source)* rather than from the voice mail circuit. When the station user retrieves the message, an intercom call is placed to the alternate message source.

24.18 To make efficient use of multi-port ***analog*** voice mail units (not Voice Processing Unit voice mail applications), the ports are placed in a voice mail hunt group and the voice mail hunt group's pilot number is assigned as the alternate message source for each of the individual voice mail ports. When a user responds to a message left by one of the voice mail ports, the pilot number is automatically dialed, and the call circulates through the hunt group until a voice mail port is available. Without the alternate message source hunt group, the call would return only to the port that left the message and would not circulate through the hunt group.

24.19 A station's message center or alternate message source does not need to be on the same node as the station. It can be a station, hunt group, Voice Processing application, or any off-node device. (See page 6-203 for information about assigning a station's alternate message source.)

25. OFF-HOOK VOICE ANNOUNCE (OHVA)

25.1 The Off-Hook Voice Announce (OHVA) feature allows an intercom caller (either single-line or phone) to establish voice contact with the called phone user, even though the user already has a call in progress on the handset. The called phone user can then talk freely on both connections, using the handset for the original call and the speakerphone for the OHVA intercom call. Note that the caller on the handset call *will* be able to hear the OHVA call.

25.2 In an Inter-Tel system network, intercom callers can establish OHVA calls to phones on other nodes.

25.3 A PC Data Port Module (PCDPM) must be installed on Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phones to provide the secondary voice path needed for the OHVA feature.

25.4 To enable the OHVA, the following must be completed:

- The OHVA system flag must be enabled. (When the system is in the default state, this flag is enabled.)
- Each OHVA-ready phone can be programmed for *placing* and/or *receiving* OHVA calls. (By default, OHVA is disabled at the station level.)
- Other phone and single-line stations can be programmed for *placing* OHVA calls, as desired. Phones without PC Data Port Modules or secondary voice paths and single-line sets cannot receive OHVA calls since they do not have secondary voice path circuitry. (By default, OHVA is disabled at the station level.)
- DSS/BLF-equipped phones can be programmed to override the Camp-On and OHVA Screening timers and make "immediate" OHVA calls. This is a useful option for attendant stations.

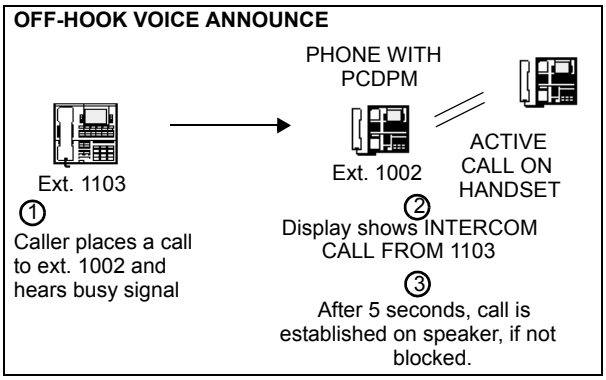
25.5 Any combination of OHVA-enabled and -disabled stations is available through DB Programming. For example, all attendants may be programmed to make, but not receive, OHVA calls, while other stations are allowed to receive, but not place, OHVA calls.

25.6 OHVA calls cannot be processed if the secondary voice path or the speakerphone of the called phone is not available. This occurs when the phone has a different OHVA call in progress, has an active data call in progress, is on an active handsfree intercom or outside call, has handsfree disabled, has a call being changed from handset to speakerphone, has a headset enabled, or is in do-not-disturb. Also, OHVA calls are not possible if the caller is placing a private intercom call or has the Ring Intercom Always feature enabled.

NOTE: To place an OHVA call using a single-line phone, the Ring Intercom Always feature must be disabled. To disable this feature, lift the handset and dial 377 (Ring Intercom Always ON/OFF feature code) on the single-line phone. By default, this is enabled.

25.7 To place an off-hook voice announce call:

1. When placing an intercom call to a busy OHVA-enabled phone, do not hang up. You hear busy signals until the Camp-On and OHVA Screening timers expire. (If you hear music after the Camp-On timer expires, conditions for an OHVA call were not met, and you are camped-on. See page 5-177.)



2. When the OHVA Screening timer expires (and if the phone's secondary voice path is available), you are automatically connected to the called party's speakerphone.

25.8 To place an immediate off-hook voice announce call (phones with DSS/BLF buttons only):

NOTE: This method can only be used if the database has been programmed to allow phones with DSS/BLF buttons to bypass the Camp-On and OHVA Screening timers. See page 6-212 in PROGRAMMING.

1. When an intercom call is placed using the DSS/BLF button to a busy OHVA-enabled phone, do not hang up.
2. Press the DSS/BLF button for that phone again. If the phone's secondary voice path is available, you are immediately connected and may speak.

25.9 To receive an off-hook voice announce call (OHVA-enabled phones only):

1. While on a call using the handset, you hear a camp-on tone. The display shows CALL ANNOUNCE FROM *(username)*. Do nothing. When the OHVA Screening timer expires, you hear a double tone. You are connected with the intercom caller via the speakerphone. Your original call remains connected on the handset.

   NOTE: If you press MUTE while speaking to the caller on the speakerphone, the handset microphone will be muted and the caller on the handset will not hear you or the OHVA caller. Press MUTE again to enable the handset.

2. **To terminate the OHVA call:** Press the lit Speaker button or have the OHVA caller hang up. If you terminate the original call by hanging up the handset, you remain connected to the OHVA call in the handsfree intercom mode.

25.10 To block an off-hook voice announce call (OHVA-enabled phones only):

1. While on a call using the handset, you hear a camp-on tone. The displays CALL ANNOUNCE FROM *(username)*. To cause the intercom call to camp on, press the Speaker button. The IC button flashes.
2. *If you want to send do-not-disturb signals to the camped-on call,* press the DND button.

26. OUTSIDE CALLS

26.1 When a trunk is selected for receiving or placing an outside call, the voice channel is seized and cannot be used by any other station (unless the Conference feature is used, see page 5-213). If the desired trunk is busy, the station user can camp on or request a callback (queue). Other features that apply to both outside and intercom calls are discussed later in this section of the manual. They include placing calls on hold, call waiting, call transfer, reverse transfer, conferencing, and call forwarding. See page 5-88 for an explanation of outgoing-access, allowed-answer, and ring-in assignments. Refer also to page 5-87 for an explanation of trunk groups and the use of the automatic trunk answer and selection feature codes.

A. PLACING OUTSIDE CALLS

26.2 Phones can be programmed to select a trunk, a trunk group, or ARS when a CALL button is pressed. In the default state, all CALL buttons are programmed for ARS access.

26.3 On display phones, the dialed number appears with hyphens separating the toll field, equal access field, area code, office code, hookflashes, pauses, asterisks, pounds/hashes, Centrex codes, and/or absorbed digits [or hyphens separating recalls, pauses, asterisks, hashes, and/or absorbed digits in Europe]. When the system absorbs local trunk digits, the digits are displayed even when they are not dialed. (For example, with a U.S. system, if 423 is absorbed and 6767 is dialed, 423-6767 appears; with a European system, if 8 is absorbed and 676767 is dialed, 8-676767 appears.)

NOTE: When placing a call, begin dialing before the Dial Initiation timer expires. If the timer expires, the system drops the trunk connection and sends repeating reorder tones. This prevents a trunk from being tied up accidentally.

26.4 To place an outside call:

1. Lift the handset. (Phone users can dial on hook by skipping this step.)
2. Select an outgoing trunk, using one of the methods below. The associated individual trunk or CALL button flashes slowly. (If on hook, the Speaker button is lit.)

   NOTE: You must have outgoing access to a trunk to seize it.

   Phones with menu buttons: Press the **OUTGOING** menu button.

   Any Phone: Select a trunk using one of the following methods:

   - Press an individual trunk button for direct access.
   - Enter a trunk group access code (92001-92208) or press a trunk group button. This selects a trunk in the chosen trunk group. (If the group has not been programmed, the display shows NO TRUNKS IN TRUNK GROUP.)
   - If your station is programmed to select a trunk group when a CALL button is pressed, press a CALL button for outgoing access.
   - See the Automatic Route Selection (ARS) procedure on page 5-194.
   - Enter the Outgoing Calls feature code (8). This automatically selects an outgoing trunk according to the programmed outgoing access mode for your station.

Single-Line Set: Select a trunk using one of the following methods:

- Enter the Outgoing Call feature code (8). This automatically selects an outgoing trunk according to the programmed outgoing access mode for your station.
- Enter a trunk group access code (92001-92208).
- See the Automatic Route Selection (ARS) procedure on the next page.

3. You hear one of the following signals:

   - **Outside Dial Tone**: Manually dial, speed dial, or redial the desired telephone number or use the outside directory. (Display phones show the numbers dialed. When dialing is ended and the call is determined to be valid, the display shows the elapsed time and call cost.) *If you hear repeating reorder tones,* you dialed a restricted number. Also, if you dial too slowly and the Interdigit timers expire, the call may be dropped, and you will hear repeating reorder tones. *If using a phone that is on hook,* you may lift the handset to speak privately. Go to the next step. On phones, an individual trunk or CALL button flashes slowly and the display shows the call cost and elapsed time.

     Menu button phone users may press one of the following menu buttons:

     – OUTSIDE DIR: Appears only if you are using ARS. Press the button to use the outside directory as described on page 5-241.

     – STN SPEED DIAL: Press this button to use a station speed-dial number as described on page 5-236.

     – SYS SPEED DIAL: Press this button to use a system speed-dial number as described on page 5-234.

     – REDIAL: Press this button to redial the last number dialed (or saved) as described on page 5-253.

   - **Single Progress Tone**: Enter a forced account code using the keypad buttons or a pre-programmed speed-dial button (see page 5-198). The display shows ENTER ACCOUNT CODE and the digits as they are entered. You hear outside dial tone when the code is accepted. On phones, an individual trunk or CALL button flashes slowly.

   - **Repeating Reorder Tones**: You are restricted from the trunk group chosen; choose another trunk group. Or, you are restricted to using ARS only; try again using ARS (see paragraph 26.5). The display shows OUTGOING ACCESS DENIED.

4. After dialing the call, you can place it on hold, transfer it, set up a conference, enter optional account codes, etc. as described in this manual. Some phones have menu buttons for easy use of outside call features.

   If you dialed an invalid or restricted telephone number, you hear reorder tones and the display shows INVALID OUTSIDE NUMBER or RESTRICTED OUTSIDE NUMBER.

5. **To use the speakerphone, if equipped:** *If off hook,* speak handsfree when the party answers. *If on hook,* press Speaker and hang up.

   To change from speakerphone to handset: Lift the handset and speak. The Speaker button goes off. You may later transfer back to the speakerphone by pressing the Speaker button and hanging up.

6. **To terminate the call:** *If on hook,* disconnect by pressing the Speaker button. *If off hook,* hang up.

26.5 To place a call using automatic route selection:

1. **Inter-Tel Phone**: While on or off hook, enter the Automatic Route Selection feature code (92000), or press the ARS button. You hear dial tone.

   Single-Line Set: Lift the handset and enter the Automatic Route Selection feature code (92000). You hear a single progress tone followed by dial tone.

   NOTE: If required, enter a forced account code using the keypad or a pre-programmed speed-dial button (see page 5-198). Display phones show ENTER ACCOUNT CODE and the digits as they are entered. You hear a progress tone when the code is accepted.

2. Dial the telephone number. Use the MUTE button to backspace if you make a mistake. Include the area code, if needed. You hear one of the following signals.

   - **Single Progress Tone**: Your call is being placed. The associated individual trunk or CALL button flashes slowly. Menu button phone users have the same options as described for outgoing calls on the previous page.
   - **Busy Signals, and then Music**: All of the trunks in the route group are busy. Do one of the following:
     - Camp-on and wait for the trunk.
     - Request a callback (queue). The system places your station in a queue for the trunk. When you hear ringing and the display shows TG XXXX *(or group name)* IS NOW AVAILABLE, lift the handset; phone users may also need to press the fast flashing CALL button or individual trunk button (or the ANSWER button). You hear dial tone and digits being dialed. The associated individual trunk or CALL button flashes slowly.
   - **Repeating Reorder Tones**: You are restricted from the number dialed or from using the route selected. The display shows INVALID OUTSIDE NUMBER or RESTRICTED OUTSIDE NUMBER. If the number that you dialed does not match any of the programmed ARS route group patterns, the display shows UNABLE TO ROUTE CALL.

B. RECEIVING OUTSIDE CALLS

26.6 To receive an outside call:

Phone

You will hear one of the following signals when receiving an outside call.

- **Repeating long tones and a CALL button or individual trunk button is flashing at the fast rate**: A call is ringing in. Lift the handset and/or press the flashing individual trunk button, flashing CALL button, or ANSWER button. (Executive Display, Professional Display, and Model 8560 Phones that have a programmed voice mail extension can press the **SEND TO V-MAIL** menu button to transfer the call to voice mail.) The individual trunk or CALL button flashes slowly during the call. Display phones show *(trunk name)* RINGING IN.
- **Intercom call or page announcing a call and a CALL button or individual trunk button is flashing at the fast rate**: A call has been transferred to your station. Lift the handset and/or press the flashing individual trunk button, flashing CALL button, or ANSWER button. (Phones with menu buttons that have a programmed voice mail extension have a **SEND TO V-MAIL** menu button that transfers the call to voice mail.) The individual trunk or CALL button flashes slowly during the call. Display phones show TG XXXX *(or group name)* TFER FROM EX XXXX *(or username)*.
- **An individual trunk button is flashing at the fast rate, there is no ring signal, and you have allowed answer for the trunk. You may hear ringing on another phone**: Press the fast-flashing individual trunk button as described above or enter the Automatic Trunk Answer feature code (350). (The ANSWER button cannot be used to answer these calls.)

Single-Line Set

- **If you hear a page announcing a transfer and/or repeating double tones**: A call is ringing in. Lift the handset; you are connected to the caller.
- **If you hear a call ringing in on another station and you have allowed answer**: Lift the handset and enter the Automatic Trunk Answer feature code (350). You are connected to the lowest-numbered trunk ringing in for which you have allowed answer.

26.7 To receive a call that was transferred to hold:

When you hear a page announcing that an outside call is holding at your station (a CALL button or individual trunk button is fluttering):

- **Phone**: Lift the handset and/or press the fluttering CALL button or individual trunk button, or press the ANSWER button. You are connected to the caller.
- **Single-Line Set**: Lift the handset and press the HOLD button or enter the Individual Hold feature code (336). You are connected to the caller.

C. PHONE ON-HOOK MONITORING

26.8 A phone user with a speakerphone can monitor the call (listen to a recorded message, wait for the call to be answered, or wait on hold), and then speak handsfree when answered.

26.9 To use on-hook monitor:

1. While on a call, press the Speaker button and hang up. *If you do not want to be heard and if the* MUTE *button is not lit,* press the MUTE button or **MUTE** menu button, or press ∞ or SPCL and enter the Microphone Mute On/Off feature code (314). You can hear the other party (if the microphone is muted, you cannot be heard). The Speaker button is lit (and the MUTE button is lit if the microphone was muted).
2. **To return to the conversation:** You may lift the handset or speak handsfree. (If the microphone was muted, press the lit MUTE button or the **UNMUTE** menu button to speak. The MUTE button goes off.)

D. PHONE GROUP LISTEN FEATURE

26.10 The Group Listen feature allows a phone user to transmit a conversation over the phone speaker while in handset or headset mode. This allows other people in the room to listen to the conversation. However, the phone microphone remains disabled so that only the headset or handset user can speak.

26.11 This feature cannot be used on a handsfree call. The user must be on a call using the handset or a headset before entering the feature code. Group Listen cannot be used on single-line sets.

26.12 To turn the group listen feature on or off during a call:

While on a call using the handset or a headset, press ∞ or SPCL and enter the Group Listen feature code (312). You hear a confirmation tone, and the display shows GROUP LISTEN ON or GROUP LISTEN OFF. (The other party will not hear the confirmation tone.)

26.13 When the Group Listen feature is active in handset mode, the Speaker lamp will remain unlit. This allows the user to place the call into handsfree mode at any time during the call by pressing the Speaker button. When the feature is active in headset mode, the Speaker lamp is lit. Pressing the button will disconnect the call.

NOTE: When using Group Listen on a Model 8500, 8520, or 8560 endpoint, the volume control adjusts the volume level of the headset speaker or handset speaker, not the external speaker.

26.14 The Group Listen feature code can be programmed under a feature button or DSS/BLF button just like any other feature. The button functions as a toggle. If the button has a lamp, it is lit when the feature is activated and unlit when the feature is turned off.

26.15 If the Group Listen feature is active when an OHVA call is received by the phone, the OHVA call will camp on (because the speaker is busy).

E. TRUNK CAMP ON AND BUSY TRUNK CALLBACK (QUEUE)

26.16 Camp-on and queue requests allow the caller to wait for an available resource. A station user waiting for a specific resource (a trunk or a station) will be served before a station user waiting for a group that contains that specific resource (trunk group or hunt group). This includes camped on Emergency Call feature 911 [999] calls.

Camp On

26.17 When a station user attempts to select a busy outgoing trunk or trunk group, the system sends a busy signal. The station user can wait off hook to camp on until the trunk is available.

26.18 A user can camp on to busy resources on other nodes and will hear the other node's music-on-hold while camped on. When the system has several routing options for a call, it tries to use each route, in order, until it finds a free route. If it is unable to find a free route, the system tries once again to use the first route. If the first route is still busy, the system camps on to the first route.

26.19 To camp on to an outgoing trunk group:

1. When you hear a busy signal after selecting an outgoing trunk, do not hang up. You camp on and hear music until a trunk in the trunk group is available. The display shows WAITING FOR *(trunk name)*.
2. When you hear a single progress tone and outside dial tone, dial the desired telephone number. The associated CALL button or individual trunk button flashes slowly.

Busy Trunk Callback (Queue)

26.20 If you attempt to select a busy outgoing trunk group and hear a busy signal or if you camp on, you can request a callback (queue) and hang up until the system signals your station that a trunk in the trunk group is available. Each station can place only one queue request at a time. If a second request is made, the first request is canceled and replaced by the second request.

26.21 A user cannot queue onto busy resources on another node.

26.22 Queue callbacks must be answered before the Queue Callback timer expires. If a callback is not answered, the queue is canceled. If the station is busy when a trunk in the queued trunk group becomes available, the queue request is placed at the end of the queue list.

26.23 To queue a busy outgoing trunk:

1. **Phone**: When you hear a busy signal while selecting a trunk or if you are camped on, press the **QUEUE** menu button or press ∞ or SPCL and enter the Queue Callback feature code (6) and hang up. The display shows QUEUE REGISTERED FOR *(trunk name)*.

   Single-Line Set: When you hear a busy signal while selecting a trunk or if you are camped on, hookflash (press **RECALL**) and enter the Queue Callback feature code (6). Hang up.
2. Your station rings when the queued trunk is available. Display phones show TG XXXX *(or group name)* IS NOW AVAILABLE.
3. **Phone**: Lift the handset and/or press the fast-flashing CALL button, individual trunk button, or the ANSWER button. Then dial the desired telephone number. The associated CALL button or individual trunk button flashes slowly. If ARS was used when the call was placed originally, the number is dialed automatically.

Single-Line Set: Lift the handset and dial the desired number. If all DTMF decoders are busy when your station is called back, the system sends repeating reorder tones instead of dial tone and the queue is canceled. If ARS was used when the call was placed originally, the number is dialed automatically.

26.24 To cancel a queue before the callback:

Phone: While on hook, press the **CANCEL QUEUE** button or **QUEUE** button. Or, enter the Queue Callback feature code (6). The display shows QUEUE REQUEST CANCELED.

Single-Line Set: Lift the handset and enter the cancel Queue Callback feature code (6). Hang up.

F. ACCOUNT CODES

26.25 Account codes are 3- to 12-digit codes that can be used in conjunction with the Station Message Detail Recording (SMDR) feature to aid record keeping. Account codes can be assigned to measure telephone use and/or to identify calls for customer billing. There are Standard, Forced, and Optional types of account codes. The account code, when used, is recorded in the SMDR report as soon as the call is completed. If more than one account code is entered during a call, the last account code that was entered is recorded.

Database-Programmed Account Codes

26.26 Any station can be assigned a standard account code or one of six [four in Europe] types of forced account codes, four [two in Europe] are ARS dependent. Or, if desired, the station can have no associated account code.

26.27 There can be up to 256 standard and 256 forced account codes in the database (512 of each in a triple- or quad-chassis system).

26.28 The account code types for the U.S. system are:

- **Standard**: Automatically appears in the SMDR printout to identify calls from this station. The station user is not required to enter any digits during the call.
- **Forced – All Calls Validated**: Requires users to enter a forced account code before an outside call can be placed. If the code matches one of the pre-programmed forced account codes in the database, the call is allowed. If the code does not match, the call is blocked. ***Stations with this account code type cannot forward calls to outside telephone numbers.***
- **Forced – All Calls Non-Validated**: Requires users to enter an account code before placing an outside call. The code is not checked against any lists and the call is allowed as soon as the code is entered. ***Stations with this account code type cannot forward calls to outside telephone numbers.***
- **Forced – Local Toll Calls Validated**: Requires users to enter an account code after dialing a local toll call number (a toll call within the local area codes) or a long distance call when using ARS. If the code matches one of the pre-programmed forced account codes in the database, the call is allowed. If the code does not match, the call is blocked. ***Stations with this account code type cannot forward calls to outside telephone numbers.***
- **Forced – Local Toll Calls Non-Validated**: Requires users to enter an account code after dialing a local toll call number (a toll call within the local area codes) or a long distance call when using ARS. The code is not checked against any lists and the call is allowed as soon as the code is entered. ***Stations with this account code type cannot forward calls to outside telephone numbers.***

- **Forced – Long-Distance Toll Calls Validated**: Requires users to enter only an account code if the system detects that a long distance call (a toll call outside of the local area codes) has been dialed when ARS is used. If the code matches one of the pre-programmed forced account codes in the database, the call is allowed. If the code does not match, the call is blocked. ***Stations with this account code type cannot forward calls to outside telephone numbers.***
- **Forced – Long-Distance Toll Calls Non-Validated**: Requires users to enter only an account code if the system detects that a long distance call (a toll call outside of the local area codes) has been dialed when the ARS is used. The code is not checked against any lists and the call is allowed as soon as the code is entered. ***Stations with this account code type cannot forward calls to outside telephone numbers.***
- **None**: Does not require users to enter an account code. The user may still use optional account codes or enable the Account Code For All Calls Following feature, as desired.

26.29 The account code types for Europe are:

- **Standard**: Automatically appears in the SMDR printout to identify calls from this station. The station user is not required to enter any digits during the call.
- **Forced – All Calls Validated**: Requires users to enter a forced account code before an outside call can be placed. If the code matches one of the pre-programmed forced account codes in the database, the call is allowed. If the code does not match, the call is blocked. ***Stations with this account code type cannot forward calls to outside telephone numbers.***
- **Forced – All Calls Non-Validated**: Requires users to enter an account code before placing an outside call. The code is not checked against any lists, and the call is allowed as soon as the code is entered. ***Stations with this account code type cannot forward calls to outside telephone numbers.***
- **Forced – Toll (National) Calls Validated**: Requires users to enter an account code only if the system detects that a toll call has been dialed when ARS is used. If the code matches one of the pre-programmed forced account codes in the database, the call is allowed. If the code does not match, the call is blocked. ***Stations with this account code type cannot forward calls to outside telephone numbers.***
- **Forced – Toll (National) Calls Non-Validated**: Requires users to enter an account code only if the system detects that a toll call has been dialed when the ARS is used. The code is not checked against any lists, and the call is allowed as soon as the code is entered. ***Stations with this account code type cannot forward calls to outside telephone numbers.***
- **None**: Does not require users to enter an account code. The user may still use optional account codes or enable the Account Code For All Calls Following feature, as desired.

26.30 Station account code programming is explained on page 6-301 in PROGRAMMING.

26.31 If the Inter-Tel system is installed in a network, the network validates account codes against the account code table on the same node as the caller. The account code follows the call as it moves from node to node and appears on every SMDR record associated with the call. However, if an optional account code is entered during the call, that account code will be used in the SMDR report.

26.32 If ARS Dial Rule #5 – Echo Account Code [Dial Rule #4 in Europe] is enabled, the account code must be entered ***before*** the end of dialing for it to be included in the dialed digits. Forced, standard, and "all calls following" account codes are automatically entered before the digits. If using an optional account code, it must be entered after seizing a trunk, but before dialing the number, to be included in the ARS digits.

Optional Account Codes

26.33 Optional account codes can be entered at any time during a call. These user-defined codes are not pre-programmed, but must be within the maximum length set in programming. If entered, optional account codes are printed in the SMDR report for that call in place of standard or forced account codes that may have been used.

26.34 To enter an optional account code after selecting a trunk or while on an outside call:

1. **Phone**: Press ∞ or SPCL and enter the Optional Account Code feature code (390). Display phones show ENTER ACCT CODE. The outside party, if any, does not hear you enter the code.

   Single-Line Set: Hookflash (press **RECALL**) and enter the Optional Account Code feature code (390). The outside party, if any, hears music until step 2 is completed.

2. Enter the account code using the keypad; phone users may press a speed-dial button to enter an account code number that has been stored in that speed-dial number location. Display phones show numbers. You hear a single progress tone when the code is accepted. Single-line users are reconnected when the system accepts the account code. (If an account code is entered that has fewer than the programmed maximum number of digits, press # to return to the call. Pressing * will cancel the account code entry.)

Enabling the Account Code for All Calls Following Feature

26.35 You can use the Account Code For All Calls Following feature to enter an account code once and apply it to all calls placed from that phone until the feature is canceled. The account code is stored in system memory and is used for all calls made by that station. It affects other account code programming as follows:

- **Forced**: The "all calls following" account code is used for all calls and the station user will not be prompted to enter an account code until the "all calls following" code is canceled.

  NOTE: If account codes are validated, and the Account Code For All Calls Following is an invalid code, calls will not be allowed at the station until the code is removed or reprogrammed.

- **Standard**: The "all calls following" account code overrides the standard code.
- **None**: The "all calls following" code will be used as an optional account code.
- **Optional**: In the event that this feature is in effect and an optional account code is also entered, the optional account code will override the "all calls following" account code for that call only. All subsequent calls will be associated with the "all calls following" account code.

26.36 To cancel this feature, the feature code is entered without an account code (just press # to terminate programming).

26.37 To enter an account code and use it for all following calls:

1. **Phone**: Enter the Account Code For All Calls Following feature code (391). Display phones show ENTER ACCT CODE.

   Single-Line Set: Lift the handset and enter the Account Code For All Calls Following feature code (391).

2. Enter the account code using the keypad; phone users may press a speed-dial button to enter an account code number that has been stored in that speed-dial number location. *If an account code is entered that has fewer than the programmed maximum number of digits,* press # to terminate the code. Display phones show numbers. You hear a single progress tone when the code is accepted.

26.38 To cancel the account code for all calls following feature:

1. **Phone**: Enter the Account Code For All Calls Following feature code (391). Display phones show ENTER ACCT CODE.

   Single-Line Set: Lift the handset and enter the Account Code For All Calls Following feature code (391).

2. *If on hook,* lift and replace the handset or press # to exit. *If off hook,* hang up.

27. PLACING CALLS ON HOLD

27.1 There are several ways to place intercom and outside calls on hold. While on hold, the caller hears music (if equipped). There are three hold applications in the phone system:

- **Individual Hold**: Places the call on hold at one station. It can then be picked up directly at that station or it can be picked up at another station using the Reverse Transfer feature.
- **System Hold**: Places the call on hold so that it can then be picked up directly at any phone station that has an individual trunk button and has allowed-answer and/or outgoing access for the associated trunk, or at the station that placed it on hold. Attempting to place a conference on system hold will place the conference on individual hold. Intercom calls cannot be placed on system hold. Single-line sets cannot place calls on system hold; attempting to do so at a single-line station will place the call on individual hold. In a network setting, when a user puts a trunk on system hold, only that user and users on the same node as the trunk can access the call.
- **Consultation Hold**: Allows a single-line set user to pause during a call, use other system features, and then return to the caller by hookflashing (pressing **RECALL**). If a single-line set user attempts to hang up after placing a call on consultation hold, the call recalls the station.

27.2 If a call remains on hold until the Hold timer expires, it recalls the station where it is on hold, and the Recall timer is started. If it is still unanswered when the Recall timer expires, it recalls the station's attendant, and the Abandoned Call timer is started. (If the station does not have an attendant, the call continues to recall at the station that placed it on hold.) If the call is not answered before the Abandoned Call timer expires, the call is disconnected by the system.

27.3 **For users' convenience, the system has two Hold timers**: *Hold* and *Hold - Alternate*. In the default state, the Alternate timer is set for a longer time period than the Hold timer. However, both timers are programmable. The "Alternate Hold Timer" station flag determines which timer each station will use. If the flag is disabled, the station uses the Hold timer. If it is enabled, the station uses the Alternate timer.

NOTE: Phone users can avoid the Hold timer by muting the microphone during a call instead of placing the call on hold. If this is done, the caller will not hear music-on-hold and will not recall the station.

A. INDIVIDUAL HOLD

27.4 To place an intercom or outside call on individual hold:

1. While on the call:

   Inter-Tel Phone: Press the Hold button, or press ∞ or SPCL and enter the Individual Hold feature code (336). You hear intercom dial tone and the CALL button, individual trunk button, or IC button flutters.

   Single-Line Sets: Hookflash (press **RECALL**) and enter the Individual Hold feature code (336). You hear intercom dial tone.

2. Hang up or place another call.

27.5 To return to the caller on individual hold:

Inter-Tel Phone: Lift the handset and/or press the fluttering IC button for intercom calls, or the CALL button or individual trunk button (or the ANSWER button) for outside calls.

Single-Line Set: Lift the handset and enter the Individual Hold feature code (336).

B. SYSTEM HOLD (INTER-TEL PHONES ONLY)

NOTE: Intercom calls and conferences cannot be placed on system hold. If you attempt to do so, the call or conference will actually be placed on individual hold.

27.6 To place an outside call on system hold (Inter-Tel phones only):

1. While on a call, press ∞ or SPCL and enter the System Hold feature code (335). You hear intercom dial tone and the associated CALL button or individual trunk button flutters.
2. Hang up or place another call.

27.7 To return to a call on system hold (Inter-Tel phones only):

At any phone that shows a fluttering individual trunk button for the call and has allowed-answer and/or outgoing access for that trunk, lift the handset and/or press the fluttering button.

At the phone that placed the call on hold, press the fluttering individual trunk or CALL button.

C. CONSULTATION HOLD

27.8 To use consultation hold (single-line sets only):

1. While on a call, hookflash (press **RECALL**) to place the call on consultation hold. You hear three short tones followed by dial tone.
2. Place a call or enter a feature code.

   NOTE: If you hang up while the call is on consultation hold, the call will immediately recall your station. If you hang up after dialing an extension number, the call will transfer to that station.

3. Return to the caller on hold by hookflashing (pressing **RECALL**).

D. HOLD RECALL

27.9 To answer a hold recall:

When the recall rings at your station. The display shows *(station or trunk)* RECALLING), lift the handset or press the ANSWER button. Menu button phone users can press the ANSWER menu button. Single-line sets are automatically connected; phone users may need to press the flashing CALL or individual trunk button.

E. MICROPHONE MUTE

27.10 Whether handsfree or using the handset, you can temporarily turn off your microphone while on a call. The call is still connected; you can hear the other party, but they cannot hear you. Because the call is not placed on hold, no timer is activated. The MUTE button is lit when the microphone is muted; the light will go off when you press the MUTE button to re-enable the microphone.

27.11 If you receive an off-hook voice announce call (see page 5-190), you can press the MUTE button while speaking to the off-hook voice announce caller to mute the handset microphone. The caller on the handset will not hear you or the OHVA caller. When you press MUTE again, the handset microphone is re-enabled.

27.12 You cannot mute the microphone on your phone while your call is on hold at another station.

27.13 Muting the phone microphone:

1. To temporarily turn off your microphone while on a call, press the MUTE button or **MUTE** menu button, or press ∞ or SPCL and enter the Microphone Mute On/Off feature code (314). The MUTE button is lit, and the display shows MICROPHONE MUTE ON. *If off hook,* do not hang up (unless you press the Speaker button first).
2. Press the lit MUTE button or **UNMUTE** menu button, or press ∞ or SPCL and enter the feature code (314) to turn the microphone on. The MUTE button goes off, and the display shows MICROPHONE MUTE OFF.

28. CALL WAITING

28.1 While a station is in use, incoming intercom and outside calls camp on until the busy station is available. The busy party hears a single camp-on tone every 15 seconds (unless the Camp-on Tone timer is changed or camp-on tones are disabled at the station).

28.2 A station programming flag (Camp-On Indications) allows camp-on tones to be enabled or disabled for incoming intercom and outside calls. This flag is programmed on a station-by-station basis (see page 6-210 in PROGRAMMING).

28.3 The DID/E&M [DDI in Europe] Receive Busy Instead Of Camp-On station flag determines whether E&M and DID [DDI in Europe] callers will receive busy signal or receive ringback and camp on when calling a busy station. In the default state, busy tones are disabled, and the callers will hear ringback while camped on to the called station. This flag is programmed on a station-by-station basis (see page 6-210 in PROGRAMMING).

28.4 To respond to call waiting using a phone:

1. If, while on a call, you hear a single camp-on tone and/or see the IC button, a CALL button, or an individual trunk button flashing rapidly, you have a call waiting. The display indicates the type of call that is waiting (intercom, outside, recall, etc.). If you are a hunt group member, the display shows the number of calls camped on to the hunt group. Do one of the following:

   a. *If you want to end the current call,* hang up. A waiting outside call rings as an incoming call; an intercom call rings as a private call. Answer as usual.

   b. *If you want to place the current call on hold,* press the Hold button, or press ∞ or SPCL and enter the Individual Hold feature code (336). The CALL button, individual trunk button, or IC button flutters (unless an intercom call is camped on, in which case the IC button flashes rapidly until the camped on call is picked up, then the IC button flutters). Answer the waiting call by pressing the flashing CALL button, individual trunk button, or IC button.

      Phones with menu buttons: You may press one of the following menu buttons:

      – ANSWER: Press this button to end the current call and answer the ringing call.

      – SEND TO V-MAIL: Appears only if you have a voice mailbox. Press this button to transfer the call to your mailbox.

      – IC CALL TO DND: Press this button to send do-not-disturb tones to the caller. If using a display phone, the caller will see DO-NOT-DISTURB.

2. **To return to the holding call:** Press the fluttering individual trunk button, CALL button, or IC button.

28.5 To respond to call waiting using a single-line set:

When you hear a single camp-on tone every 15 seconds while you are on a call, another call is waiting. Do one of the following.

1. *If you want to end the current call,* hang up. The call rings at your station. Lift the handset to speak to the waiting caller.

   If you want to place the current call on hold, hookflash (press **RECALL**) and then enter the Individual Hold feature code (336). You are connected with the waiting caller.

2. To return to the first call on hold:

   If you want to end the current call, hang up. Return to the holding call by lifting the handset and entering the Individual Hold feature code (336).

   If you want to place the current call on hold and connect with the original call: Hook-flash (press **RECALL**) and then enter the Individual Hold feature code (336) twice.

28.6 To move a camped-on intercom call to do-not-disturb:

While on a call, you hear call waiting signals:

Phones with menu buttons: Press the **DO-NOT-DISTURB or IC CALL TO DND** menu button.

Other Phone: Press the DND button, or enter a Do-Not-Disturb feature code (370 or 372). The DND button lights. The intercom caller is no longer camped on and receives do-not-disturb indications.

Single-Line Set: Hookflash (press **RECALL**) and enter a Do-Not-Disturb feature code (370 or 372). The intercom caller is no longer camped on and receives do-not-disturb indications. hookflash (press **RECALL**) twice to return to the current call.

29. CALL TRANSFER

29.1 There are two feature codes for transferring intercom and outside calls to other stations, Voice Processing applications, hunt groups, off-node devices, or outside telephone numbers. The call transfer options are as follows:

- **Transfer to Ring**: You can transfer intercom or outside calls to another station, a Voice Processing application, a hunt group, or an outside telephone number.
- **Transfer to Hold**: Either intercom or outside calls can be transferred to another station and placed on hold using this feature.

NOTE: For information on screened and announced transfers from automated attendants, see page 5-211 in SYSTEM FEATURES and page 4-16 in VOICE PROCESSING FEATURES.

A. TRANSFERRING CONFERENCE CALLS

29.2 An established conference can be transferred to a station. While the transfer is taking place, the parties in the conference remain connected to each other and may converse. The transfer will appear at the destination station in the same manner as any other transferred call along with a CONFERENCE TFR FROM *(username)* display and may be answered by the station user.

B. TRANSFER TO A SYSTEM FORWARD

29.3 When a station user transfers a call to the principal station of a system forwarding path, the Handsfree Announce System Forward system-wide flag determines what the caller will hear. If enabled, the caller will place a handsfree call to the principal station and can announce the transfer (the call does not enter the forwarding path). If the option is disabled, the transferring station user will hear ringing and can announce the call only if it is answered by the principal station or one of the forwarding points. See page 6-260 for system flag programming.

C. TRANSFERS AND OUTSIDE CALLS

29.4 If a station user answers another call while a party is waiting on transfer hold, the waiting party will be moved to individual hold at that station. If the Hold timer expires while a call is on transfer hold, the call will recall the station. The station user must return to the call on hold and begin the transfer process again.

29.5 There may be some reduction in voice volume when an outside call is transferred to an outside telephone number depending on the quality of the trunks being used.

D. TRANSFER TIMERS

29.6 There are five timers that affect the Call Transfer feature. They are as follows:

- **Transfer Attendant**: When an attendant transfers a call, this timer limits the time a transferred call rings unanswered before it recalls the attendant. The default value of this timer is 30 seconds. It can be reprogrammed to any value between 10-255 seconds.
- **Transfer Available**: This timer limits the amount of time a call that is transferred by a station user will ring, unanswered, before it recalls the transferring station. (Does not apply to calls transferred by attendant stations or Voice Processing.) The default value of this timer is 20 seconds. It can be reprogrammed to any value between 10-255 seconds.
- **Transfer Busy**: This timer limits the amount of time a call that is transferred by a station user will wait at a busy station before recalling the transferring station. (Does not apply to calls transferred by attendant stations or Voice Processing.) The default value of this timer is 24 seconds. It can be reprogrammed to any value between 10-255 seconds.
- **Transfer Voice Processor**: When a Voice Processing port transfers a call, this timer limits the time a transferred call rings unanswered before it recalls the port. The default value of this timer is 20 seconds. It can be reprogrammed to any value between 10-255 seconds. (Outside calls will not recall to Voice Processing Unit ports if the phone system is unable to communicate with the Voice Processing Unit. The calls will ring at the destination stations until they are answered or the Abandoned Call timer expires.)
- **Unsupervised CO**: The duration of an outside call that is transferred to an outside telephone number is limited by this timer. When it expires, the unsupervised call recalls the primary attendant. If the attendant does not answer the recall before the Abandoned Call timer expires, the call is disconnected by the system. The default value of this timer is 5 minutes. It can be reprogrammed to any value between 1-255 minutes.

E. TRANSFER-TO-CONNECT STATION FLAG

29.7 If the Transfer-To-Connect Allowed station flag is enabled, an announced transfer call (including an announced transfer from voice mail) connects immediately with the destination once the transferring party hangs up. (If the flag is disabled, the recipient must press a CALL button or the ANSWER button to answer the transferred call).

29.8 If the user has a headset enabled, or if the station user picks up the handset for the initial call and the transferring party disconnects, the transferred party will hear a single tone and then be immediately connected without any further action from the station user. (Calls will not automatically connect in handsfree speakerphone mode.) This flag only applies to phones and is disabled by default.

F. TRANSFER TO RING

29.9 To transfer a call or conference:

1. **Inter-Tel Phone**: While on a call, press the TRANSFER button, or press ∞ or SPCL and enter the Transfer to Ring feature code (345). You hear intercom dial tone. The display shows TRANSFER CALL TO EXTENSION.

   Single-Line Set: While on a call, hookflash (press **RECALL**). You hear tones followed by intercom dial tone.

2. **EITHER,** Select an outgoing trunk group or press the **OUTSIDE PHONE** menu button, and dial the desired telephone number. *If all trunks in the trunk group are busy,* select another trunk group or camp on and wait for a trunk. *If you hear repeating reorder tones,* you dialed a restricted telephone number. To return to the caller on transfer hold, press the lit CALL button, trunk button, or IC button (phone) or hookflash (press **RECALL**) twice (single-line set).

   OR, Dial the desired extension number or press a speed-dial button, the **VOICE MAIL** menu button, or the **MESSAGE CENTER** menu button. Executive Display, Professional Display, and Model 8560 Phone users can use the Directory feature by pressing the **IC DIRECTORY** menu button. The number can be a station extension number, 0 for the attendant, a hunt group pilot number, or a voice mail extension number.

   a. **Transfer to a station that is forwarded to voice mail**: You hear repeating double tones, and the display shows DEST FORWARDED TO VOICE MAIL.

   b. **Transfer to voice mail**: The system waits for you to enter the mailbox number. The display shows ENTER MAILBOX. Enter the desired mailbox number. If your phone has menu buttons, you can use the Directory feature by pressing the **IC DIRECTORY** menu button, if the system is validating mailbox numbers. *If you do not enter a mailbox number before you hang up,* the caller will be connected to the voice mail unit and must enter the mailbox number after listening to the introductory voice prompts.

3. *If desired,* wait for an answer and announce the call. If you are calling a device (such as an automated attendant) that does not accept call announcements, the display shows HANG UP TO FINISH TRANSFER.

4. You have the following options:

   — **To complete the transfer:** Hang up or press another CALL button. The display shows CALL TRANSFERRED TO *(username)*.

   — **To return to the caller:** Press the fluttering CALL button, trunk button, or **IC** button (phone) or hookflash (press **RECALL**) twice (single-line set).

   — **To try another station:** Press the TRANSFER button, or press ∞ or SPCL and enter the Transfer to Ring feature code (345) and dial another extension number.

   — **To transfer the call to hold when calling a station** *(Inter-Tel phones only)***:** Press the Hold button or the **TRANSFER-TO-HOLD** menu button. Or, press ∞ or SPCL and enter the Individual Hold feature code (336). The call will not ring or send call waiting signals until the Hold timer expires.

G. TRANSFER TO HOLD

29.10 A call transferred to hold at a station does not ring or send a display message while holding. After the Hold timer expires, the station rings or sends call waiting signals. Also, calls transferred to hold do not recall the transferring party; they recall the receiving party's attendant if unanswered after the Hold and Recall timers expire. Besides the procedure listed below, phone users have an alternative method for transferring calls to hold; see step 4 in the previous procedure.

29.11 To transfer a call to hold:

1. **Phones with Menu Buttons**: While on a call, press the **TRANSFER-TO-HOLD** menu button. The display shows TRANSFER CALL TO EXTENSION.

   Other Phones: While on a call, press ∞ or **SPCL** and enter the Transfer To Hold feature code (346). The display shows TRANSFER CALL TO EXTENSION.

   Single-Line Sets: While on a call, hookflash (press **RECALL**) and enter the Transfer To Hold feature code (346).

2. Dial the desired extension number. Menu button phone users can use the Directory feature by pressing the **IC DIRECTORY** menu button.

3. Hang up to complete the transfer.

29.12 To pick up a call that was transferred to hold:

If a call was transferred to hold at your station:

Inter-Tel Phone: When your **IC** button, a CALL button, or individual trunk button is fluttering, lift the handset and/or press the fluttering button (or the **ANSWER** button for an outside call). You are connected to the caller.

Single-Line Set: Lift the handset and enter the Individual Hold feature code (336). You are connected to the call.

H. TRANSFER RECALLS

29.13 If a call is transferred to another station and is not answered before the appropriate Transfer timer expires (and if it was not transferred to hold), the call recalls the transferring station's recall destination (defaults to the station) and the Recall timer is started. The call rings until the Recall timer expires. If unanswered, it recalls the transferring party's attendant and the Abandoned Call timer is started. (If the transferring station has no attendant, the call continues to recall at the transferring station.) If the call is not answered before the Abandoned Call timer expires, the call is disconnected by the system.

29.14 In a network, a station's transfer recall destination can be an off-node device.

29.15 To answer a transfer recall:

Inter-Tel Phone: When the recall rings at your station, the display shows *(username)* RECALLING FROM *(username)*. Lift the handset and/or press the **ANSWER** button. Press the medium-flashing CALL button or individual trunk button, or press the **ANSWER** button if the call is not automatically connected.

Single-Line Set: If you hear a repeating signal of four tones and a pause, lift the handset. You are immediately connected.

30. CALL SCREENING

30.1 Calls transferred from the Automated Attendant or a Call Routing Announcement application can be screened, announced, or unannounced. Separate programming flags in the database determine the methods used for transferring calls to stations with mailboxes, stations with extension IDs, and extensions without mailboxes or IDs.

30.2 When a call is received by an automated attendant or Call Routing Announcement application and the caller enters an extension number, the programmed Transfer Method determines how the call will be transferred. The Transfer Method flags can be programmed in the database. If allowed in mailbox programming, they can also be programmed by the mailbox user. (See page 4-51 in VOICE PROCESSING FEATURES and pages 6-355, 6-374, and 6-392 in PROGRAMMING.) The available Transfer Methods are as follows:

- *Announce Only:* The caller is asked to state his or her name. Then the call is transferred to the associated extension number. When the station user answers the transfer, the voice processing system plays the caller's name and completes the transfer.
- *Screened:* The caller is asked to state his or her name. Then the call is transferred to the associated extension number. When the station user answers the transfer, the voice processing system plays the caller's name. The station user has the options of replaying the name, sending the call to voice mail (if the extension has a mailbox), transferring the call to another extension, accepting the call, or rejecting the call.
- *Unannounced:* The call is transferred to the associated extension number without announcing the call. This is the default method.

30.3 In a network setting, the Voice Processing Unit can provide call screening for a destination extension on another node. However, the node where the Voice Processing Unit is connected must have an off-node device programmed for the destination extension and access to the remote node.

30.4 Receiving An Announced Or Screened Transferred Call: Depending on the Transfer Method programmed for your mailbox or extension ID, one of the following will occur when you receive a call transfer from the automated attendant or a Call Routing Announcement application:

- **"Announce Only" Calls**: When the caller enters your extension number, the system asks the caller to record his or her name. The Voice Processing Unit then places a private call to your station. When you answer, it plays *"You have a call from (caller's name)."* It then completes the transfer, and the transferred call rings at your station. Answer as usual.
- **"Screened" Calls**: When the caller enters your extension number, the system asks the caller to record his or her name. The Voice Processing Unit then places a private call to your station. When you answer, it plays *"You have a call from (caller's name),"* and gives you the following options:
  - — *To accept the call,* press **#** or the **ACCEPT** menu button. The call will ring at your station.
  - — *To replay the announcement,* press **1** or the **REPLAY** menu button.
  - — *To send the call to voice mail,* press **2** or the **SEND TO V-MAIL** menu button. The caller is transferred to your mailbox and hears your personal greeting.

— *To forward the call to another extension,* press 3 or the **FORWARD** menu button. Then enter the extension number. The call is sent to the other station. If that station has screened or announced transfers, the caller's name will be played again for that station.

— *To refuse the call,* press * or the **REFUSE** menu button, or simply hang up. The caller receives a recording that says you are not available and offers them the option of leaving a message.

- **"Unannounced" Calls**: This is the default method for transferring a call. The call is transferred to your station without any kind of announcement.

31. REVERSE TRANSFER

31.1 Calls ringing or recalling at a station, or a hunt group can be picked up at any other station, using this feature. Calls on hold can also be picked up from stations. A user can reverse transfer a call from a station or a hunt group on another node.

31.2 If more than one call is ringing or holding at the station or hunt group, a priority list determines which call is reverse transferred. Calls are selected in the following order, and if more than one call of the same type is at the station, the calls are picked up in the order they were received:

1. Ringing calls
2. Camped-on calls
3. Holding outside calls
4. Holding intercom calls

31.3 If a call is reverse transferred from a hunt group announcement or overflow station, and a station in the hunt group becomes available, the call will be disconnected from your station immediately when answered by the hunt group station.

31.4 **Group Call Pick-Up**: A call can be reverse transferred from a station within a hunt group, using the hunt group's extension number, even if the call was not a hunt group call. To reverse transfer the call, use the station's extension number or the extension number of its hunt group. This feature is called "Group Call Pick-up."

NOTE: Group Call Pick-up does not affect DSS lamp status any differently than standard hand group calls or individual calls.

31.5 There is a programmable station flag called "Transient Call Indication On Call Answer" that determines if the station user will see a call display when reverse transferring a call (or using a secondary extension button). If the flag is enabled, the display will indicate whether the call was ringing, recalling, transferred, or holding at the other station.

31.6 **To reverse transfer a call:**

EITHER, Lift the handset (phones can remain on hook) and enter the Reverse Transfer (Call Pick-Up) feature code (4), then dial the extension or pilot number where the call is ringing or holding. You are automatically connected to the caller.

OR, Lift the handset (phones can remain on hook) and dial the extension or pilot number where the call is ringing or holding, then press ∞ or SPCL, or perform a hookflash (press **RECALL**) and enter the Reverse Transfer (Call Pick-Up) feature code (4). You are automatically connected to the caller.

32. CONFERENCE CALLS

32.1 Station users can establish multi-party conference calls without operator assistance. There can be up to four parties in a conference. In addition to the initiating station, the conference can include any combination of up to three intercom and/or outside calls. The CPC or CPU provides 12 conference channels (24 on a triple- or quad-chassis system). Additional channels can be added, via DSP programming on the Options card, to support conferencing.

NOTE: During a conference, some reduction in voice volume may be noticed, depending on CO trunk quality.

32.2 During a conference, if any station user presses a keypad button, the DTMF tones will be heard by all other parties in the conference. This allows conference callers access to DTMF-controlled devices.

32.3 An established conference can be transferred to another station, using the call transfer feature described on page 5-207. While the transfer is taking place, the parties in the conference remain connected to each other and may converse. The transfer will appear at the destination station in the same manner as any other transferred call along with a CONFERENCE TFR FROM *(username)* display and may be answered by the party.

NOTE: While this system is designed to be reasonably secure against CO trunk misuse by outside callers, there is no implied warranty that it is not vulnerable to unauthorized intrusions. If the central office [local exchange] does not provide supervision, and does not disconnect the call when one party hangs up, making it possible for a caller to remain connected to a CO trunk circuit. If this happens, and the caller begins dialing, the call could be placed through the phone system and would then be billed to the system's owner. The system cannot check this type of call for toll restriction and may not register the call in SMDR. This problem could arise when a call is connected to a station or when a call is in an unsupervised conference.

32.4 In a network setting, a user can build a conference with party members (stations and/or trunks) on other nodes. The conference circuits used to build the conference will be allocated from the originating party's node. Any party, not just the originating party, can add a party member to the conference.

- If the party who adds the new member is on the originating party's node, the party will be added to the conference circuit already allocated. This conference is still restricted to the four party limit.
- If the party who adds the new member is not on the originating party's node, a new conference circuit is allocated from the second node. Because this allows more than four parties in a conference, users should be warned that voice levels can be degraded as more parties are added to the conference.

A. PLACING A CONFERENCE CALL

32.5 To place a conference call:

1. While on a call:

   Inter-Tel Phone: Press the CNF button, or press ∞ or SPCL and enter the Conference feature code (5). The party is on conference wait hold (hears music), and the CNF button flutters. The display shows CALL NEXT PARTY TO CNF.

   Single-Line Set: Hookflash (press **RECALL**) and enter the Conference feature code (5). The party is on conference wait hold (hears music).

2. Place an intercom or outside call to another party (or return to a call on hold) to be included in the conference. If your phone has menu buttons, you can press the **IC DIRECTORY** or **OUTSIDE DIR** menu button to look up numbers, if desired.

   NOTE: An existing conference is not a valid "party" for another conference.

3. Place the party on conference wait hold:

   Inter-Tel Phone: Press the CNF button again. Or, press ∞ or SPCL and enter 5 again.

   Single-Line Set: Hookflash (press **RECALL**) and enter the Conference feature code (5). The party is on conference wait hold (hears music).

4. Repeat steps 2 and 3 until all desired parties are on conference wait hold (or the maximum allowed number has been reached).

5. Connect all waiting parties in the conference:

   Phones with Menu Buttons: Press the **CONNECT TO CONFERENCE** menu button. All parties are connected. The display shows CNF IN PROGRESS. The CNF button and a CALL button flash slowly. You can use the following menu buttons:

   — *HOLD:* Press this button to place the conference on hold.

   — *DISCONNECT:* Press this button to remove your station from the conference. If there are no other stations in the conference, all parties are disconnected. If there is still another station in the conference, the parties remain connected.

   — *ADD PARTY:* Appears only if you have not reached the maximum number of allowed parties. Press this button to leave the conference parties connected while you place another call to be added to the conference.

   — *MUTE:* Press this button to turn the phone microphone on or off during the conference.

   — *TRANSFER CNF:* Press this button and enter an extension number to transfer the conference to another station or the Voice Mail application.

   Other Phones: Press the CNF button again. Or, press ∞ or SPCL and enter 5 again. All parties are connected. Display phones show CNF IN PROGRESS. The CNF button and a CALL button flash slowly.

   Single-Line Set: When all parties are on conference wait hold, hookflash (press **RECALL**) and enter the Conference feature code (5). All parties are connected.

NOTE: If the number of parties in your conference exceeds the capacities of the system resources, you will hear reorder tone and see a display that tells you MAX NUMBER OF PARTIES EXCEEDED. To complete the conference setup, return to the parties on conference wait hold one by one (phone users press the associated CALL or IC button, single-line set users enter the Individual Hold feature code twice) and release parties until there are fewer than the maximum. Then repeat step 5 to connect the conference.

B. ADDING PARTIES TO A CONFERENCE

32.6 To add parties to an existing conference (any inside party):

NOTE: This procedure cannot be performed if the maximum number of conference participants has been reached.

Inter-Tel Phone:

1. During the conference, press the CNF button. The other parties remain connected. The CALL button and CNF button flutter.
2. Place an intercom or outside call to the other party (or return to a call on hold) to be included in the conference and press the CNF button.
3. **To connect all parties in a conference:** Press the CNF button again.
4. **To add another party:** Repeat steps 1-3 again.

Single-Line Set:

1. During the conference, hookflash (press **RECALL**). The other parties remain connected.
2. Place an intercom or outside call to the other party to be included in the conference (or return to a call on hold). Hookflash (press **RECALL**) and enter the Conference feature code (5).
3. **To connect all parties in a conference:** Hookflash (press **RECALL**) and enter the Conference feature code (5) again.
4. **To add another party:** Repeat steps 1-3 again.

C. EXITING A CONFERENCE

32.7 There are several options for leaving the conference:

- Exit the conference by hanging up. The other parties remain connected only if at least one is a station (inside party). You cannot return to the conference. If there are no stations remaining in the conference, it is disconnected.
- Leave the conference and allow the outside parties to remain connected by pressing the CNF button or entering the Conference feature code. You can return to the conference at any time.
- End the conference and place all of the parties on individual hold. You can return to the parties one at a time.
- Place the entire conference on hold. The conferenced parties remain connected.

NOTE: If a user attempts to place a conference call on system hold, the conference will be placed on individual hold.

Exiting and Re-Entering the Conference

32.8 To exit the conference and leave the other parties connected (inside party):

32.9 Inter-Tel Phone:

During the conference, press the CNF button and hang up. The CNF button and a CALL button flutters.

NOTE: If all remaining parties are outside parties and the Hold timer expires while you are out of the conference, the conference recalls your station. The recall rings at your station, and the CNF button flashes at the medium rate (display shows CONFERENCE RECALLING). Press the CNF button to return to the conference. If the recall is not answered before the Recall timer expires, it will recall the attendant.

To return to the conference: Lift the handset and/or press the CNF button again.

32.10 Single-Line Set:

During the conference, hookflash (press **RECALL**) and hang up.

NOTE: If the Hold timer expires while you are out of the conference, the parties recall your station. When the recall rings at your station, lift the handset to answer the recall. If the recall is not answered before the Recall timer expires, it will recall the attendant.

To return to the conference: Lift the handset and/or enter the Conference feature code (5).

Ending the Conference and Placing All Parties on Hold

32.11 To end the conference and put all parties on hold (any inside party):

32.12 Inter-Tel Phone:

During the conference, press the CNF button and hang up. The CNF button and a CALL button flutters. The parties are still connected in a conference.

To place the parties on individual hold: Press the Hold button or enter the Individual Hold feature code (336). You can then return to each party individually by pressing the associated CALL or IC button.

32.13 Single-Line Set:

During the conference, hookflash (press **RECALL**) and hang up. The parties are still connected in a conference.

To place the parties on individual hold: Lift the handset and enter the Individual Hold feature code (336). You can then return to each party individually by entering the Individual Hold feature code again.

Placing the Entire Conference on Hold

32.14 Any inside party involved in a conference can place the conference on hold at his or her station. The other conference parties remain connected and can converse. The station user that placed the conference on hold can then place calls or use system features while the conference is on hold. Then the user can return to the conference on hold. Note, however, that a conference may not be terminated while it is on hold.

32.15 After a station user has moved a conference to hold, the user may, if desired, establish another conference. Because each conference is established under a separate CALL button, it is possible for the user to have access to any individual conference for further call processing. If the user elects to use the CNF button for conference access, the CNF button will rotate through the conferences in priority order. The CNF button will always reflect the status of the top-priority conference. The priority order for the CNF button when multiple conferences are in effect is:

- Conference or call on transfer hold
- Ringing (ring in, recall, or camp on) conference
- Conference on hold
- Active conference

32.16 To place the conference on hold (any inside party):

1. During a conference call:

   Inter-Tel Phone: Press the Hold button, or press ∞ or SPCL and enter the Individual Hold feature code (336). The conferenced parties remain connected. The call and CNF buttons flutter.

   Single-Line Set: Hookflash (press **RECALL**) and then enter the Individual Hold feature code (336). The conferenced parties remain connected.

   NOTE: If you do not return to the conference before the Hold timer expires, the conference will recall your station. If the recall is not answered, the conference will recall to your attendant's station. At the attendant, the call will be terminated automatically after the Abandoned Recall timer if no inside parties are still in the conference and if it is not answered by the attendant. (If the attendant answers the call, a conference is re-established with the attendant station as one of the conference parties.)

2. To return to the conference:

   Inter-Tel Phone: Press the Hold button, CNF button, or the fluttering CALL button.

   Single-Line Set: Lift the handset and enter the Conference feature code (5).

33. RECORD-A-CALL

NOTE: The Record-A-Call software license is required for this feature.

33.1 If your system is programmed with a Record-A-Call application, the stations can be programmed to use the Record-A-Call feature. It allows users to enter a feature code whenever they want to record an ongoing call in their designated Record-A-Call mailbox. Users can retrieve the recorded messages later, just as they would any other mailbox messages.

33.2 When a user requests the Record-A-Call feature, the system establishes a conference call with the current call parties and a mailbox. If there are no conference circuits available when the user requests Record-A-Call, or if there are already four parties on the call, the user hears reorder tones and cannot use the feature.

33.3 The Record-A-Call mailbox records the call as a voice mail message. All parties will be included in the recording. If desired, the Record-A-Call mailbox can be programmed to play a message announcing that the Record-A-Call feature is in progress (see page 6-369 in PROGRAMMING). There can be separate messages for day and night modes.

33.4 To signal to the other parties that the Record-A-Call feature is in use, a system-wide Record-A-Call Tone flag can be enabled during system programming. If the flag is disabled, there will be no alerting tone. If enabled, the beep will occur periodically throughout the call. The Record-A-Call Tone Interval timer determines how often this tone is generated. If the timer is set to 0, the tone is generated only when the feature is first activated. (See page 6-336 in PROGRAMMING.)

33.5 There are two ways a station can be programmed to use this feature:

- The station can be programmed to use its personal mailbox or another mailbox, as the assigned Record-A-Call mailbox. Only this assigned mailbox can be selected. (This is the default programming for all stations.)
- The station can be programmed with a default mailbox but with the option of selecting a different mailbox. If the user chooses to not enter a mailbox number, the system automatically selects the default mailbox.

33.6 In a network setting, the Record-A-Call destination does not have to be on the same node as the station, but the Voice Processing Unit must be programmed with a mailbox for that station.

33.7 When the requesting station user hangs up, all parties are automatically disconnected. If all parties on the call hang up except the requesting station user, an intercom call will remain connected between the requesting station and the Record-A-Call application. This allows the user to make additional comments before ending the recording.

33.8 When the station user turns off Record-A-Call or ends the call being recorded, the system delivers the message to the mailbox. The associated station will receive message waiting indications as usual.

33.9 Record-A-Call can be used during Agent Help and three-party conference calls. In addition, software versions 7.004 and later or 8.002 and later allow a monitored station to initiate a Record-A-Call session without terminating call monitoring. Hunt group supervisors, however, cannot initiate a Record-A-Call while monitoring a station, and a station using Record-A-Call cannot be monitored. In other words, to use Record-A-Call and station monitoring simultaneously, the supervisor must first be monitoring the station, and the station, not the supervisor, must initiate the recording.

NOTE: ATM systems do not support the simultaneous use of Record-A-Call and station monitoring.

33.10 The Record-A-Call feature code can be assigned to a user-programmable feature button. If the feature button has a lamp, it flashes while the Record-A-Call conference is being set up and is lit while the feature is active.

33.11 The programmer can set a maximum length for Record-A-Call messages. The Record-A-Call Maximum Message Length timer can be set at 0-600 minutes. A 0 setting allows messages of any length, limited only by the available disk space. The default is 30 minutes. (This overrides the mailbox's maximum message length setting. However, if a Record-A-Call message exceeds the maximum message length, the mailbox will be considered full until that Record-A-Call message is deleted.) See page 6-423 in PROGRAMMING.

33.12 As with any other conference call, any inside party involved in a Record-A-Call conference can use the Hold, Transfer, Agent Help, or other features. However, if any inside party has enhanced speakerphone enabled, the enhanced mode will be disabled when the conference begins and must be re-enabled if desired.

NOTE: Pressing **Hold** during a record-a-call terminates the record-a-call.

33.13 To use record-a-call:

1. **Inter-Tel Phones**: Press ∞ or **SPCL** and dial the Record-A-Call feature code (385), or press the **RECORD-A-CALL** menu button. The display shows REQUESTING RECORD-A-CALL.

   Single-Line Sets: During a call, hookflash (press **RECALL**) and dial the Record-A-Call feature code (385).

2. *If required or allowed,* enter the desired mailbox number. The display shows ENTER MAILBOX NUMBER. (If you do not enter a mailbox number, and you have a default Record-A-Call mailbox, the system will automatically select your default mailbox. If you do not have a default mailbox, the feature will time out or can be canceled by pressing # or **ACCEPT**.)

3. You hear a confirmation tone when the Record-A-Call feature is activated. The display shows RECORD-A-CALL IN PROGRESS.

4. **To turn off Record-A-Call,** repeat step 1 or end the call. The display shows RECORD-A-CALL TERMINATED.

34. AGENT HELP

NOTE: This feature requires the Agent Help software license.

34.1 The Agent Help feature allows a station user to request help from a designated "Agent Help Extension" during a two- or three-party call. When the request-for-help call rings, the Agent Help Extension can choose to join the call or reject the request. The Agent Help Extension can be a supervisor or other station, an extension list, or a hunt group.

34.2 In DB Programming, there are two fields for each station that affect how this feature operates. (See page 6-203 in PROGRAMMING.)

- **Agent Help Extension**: Determines the station that is called when the Agent Help feature code is entered at the station. This can be set to any valid extension number for a station, extension list, or hunt group, or it can be set to "None."
- **User-Keyed Extension**: Allows the user to select the station that will receive the Agent Help request, even if a default Agent Help Extension is programmed as described above.

Table 5-36. *User-Keyed Extension*

Agent Help Ext	User-Keyed	Result
Ext. number	Disabled	The Agent Help Extension is called when the feature code is entered.
Ext. number	Enabled	The user is prompted to enter the desired number after the feature code is entered. If a number is not entered before the dialing timer expires, the programmed Agent Help Extension number is dialed automatically.
None	Enabled	The user is prompted to enter the desired number after the feature code is entered.
None	Disabled	The user hears reorder tone if the feature code is entered.

34.3 When a station user enters the Agent Help feature code, a private call is placed to the station's Agent Help Extension. While the call is ringing at the supervisor's extension, neither the requesting user nor any other parties on the call can hear the private call ringing. If the supervisor answers the call, the system creates a conference to include the supervisor in the requesting user's original call.

34.4 If the Agent Help Extension is a phone, the microphone is muted, and the supervisor cannot be heard unless the MUTE button is pressed. If the Agent Help Extension is a single-line set, the supervisor ***can*** be heard as soon as the conference is established. In either case, the supervisor can hear all other parties on the call.

34.5 In a network setting, the Agent Help Extension does not need to reside on the same node as the user requesting Agent Help.

34.6 The type of station instrument determines how the Agent Help Extension is alerted:

- *Display Phone:* If the Agent Help Extension is a display phone, the display shows that the incoming private call is an Agent Help request and identifies the username of the requesting station. The Agent Help Extension can choose to reject the call by not answering it or by entering the Agent Help Reject feature code (376). The call appears under the IC (or a CALL button if there is no IC button).
- *Non-Display Phones and Single-Line Sets:* The Agent Help Request private call rings just as any private intercom call would; there is no way to distinguish it.

34.7 If the Agent Help Extension is an extension list or hunt group, the private call circulates as usual until it is answered. If an Agent Help request is not answered before the Forward No Answer timer expires, the request is considered rejected.

34.8 To signal to the other parties that the Agent Help Extension has joined the call, a system-wide Agent Help Tone flag can be enabled during DB Programming. If the flag is disabled, there will be no alerting tone. A timer, called the Agent Help Tone Interval timer, determines how often this tone is generated. If the timer is set to 0, the tone is generated only when the Agent Help Extension enters the call, a party is added to the call, or the call is placed on hold and retrieved.

34.9 If the Agent Help feature code is assigned to a phone feature button with a lamp, the lamp status shows the following:

- The lamp flashes when the Agent Help Extension is being called.
- The lamp goes off if the Agent Help request is rejected or the feature is terminated.
- The lamp is lit solidly when the Agent Help Extension is in the conference.

34.10 When the requesting station user hangs up, all parties are automatically disconnected. The Agent Help Extension can leave the call at any time, without affecting the other parties, by hanging up. If the other party (or parties) hangs up first, the requesting station and the Agent Help Extension remain connected in an intercom call. The requesting station can cancel the request (or remove the Agent Help Extension from the call) by re-entering the Agent Help feature code, thereby terminating the feature.

34.11 If necessary, the requesting station in an Agent Help call can use the Hold, Transfer, Record-A-Call, or other features during the call, while the Agent Help Extension is connected. However, if any inside party has enhanced speakerphones enabled, the enhanced mode will be disabled when the Agent Help conference begins and must be re-enabled if still desired.

34.12 The Agent Help feature will not function in the following cases:

- If the feature is not available, any user attempting to enter the Agent Help feature code will hear reorder tones and, if at a display phone, see a RESERVED FEATURE display.
- The Agent Help Extension cannot be in do-not-disturb mode or have call forwarding enabled. If so, the requesting user will hear reorder tones after entering the Agent Help feature code or entering the desired extension number. The Agent Help request will not follow the forward or any programmed system forwarding.
- If the station user who enters the Agent Help feature code is on a four-party conference call, the system will send reorder tones to signal that the Agent Help Extension cannot be added. There can be a maximum of four parties in a conference; attempting to add the Agent Help Extension would exceed the maximum.
- If there are no conference circuits available when the station user enters the Agent Help feature code, the user will hear reorder tones and must try again later when circuits are available.

34.13 If the call is being monitored by a hunt group supervisor, the monitoring feature is terminated when the Agent Help Extension joins the call; an Agent Help conference call cannot be monitored.

34.14 To use the agent help feature:

1. **Inter-Tel Phones**: While on a call, press ∞ or **SPCL** and then enter the Agent Help feature code (375). If your phone has menu buttons, you can press the **AGENT HELP** menu button.

   Single-Line Sets: While on a call, hookflash (press **RECALL**) and then enter the Agent Help feature code (375).

   If you hear reorder tone, the Agent Help feature is not available at your station, you already have four parties in your call, no conference circuit is currently available, or the Agent Help Extension is in do-not-disturb.

2. Depending on programming, you may be prompted to enter an extension number. If so, enter the extension number of the station you want to call for help. *If you do not enter a valid extension number,* you hear reorder tones and must start over. The display shows REQUESTING AGENT HELP.

3. *If the Agent Help Extension accepts the call,* you will hear the Agent Help tone, if enabled. Display phones show AGENT HELP IN PROGRESS. When the Agent Help Extension hangs up, your display shows AGENT HELP TERMINATED.

   If the Agent Help Extension rejects the call, you will hear a confirmation tone. Display phones will show AGENT HELP REJECTED.

34.15 To respond to an agent help request at a display phone:

When you receive a private call from an Agent Help request, you will see a display that shows *(name)* REQUESTS HELP. You can do one of the following:

To accept the call: Answer as usual or press the **ANSWER** menu button. You hear the Agent Help Tone, if enabled. Your display shows AGENT HELP IN PROGRESS.

To reject the call: Enter the Agent Help Reject feature code (376) or press the **REJECT HELP** menu button. You hear confirmation tone.

To place your station in do-not-disturb to block the call: Enter the Do-Not-Disturb feature code (370) or press the **DND** button or the **IC CALL TO DND** menu button.

35. SYSTEM FORWARDING

NOTE: This feature requires the System Forwarding software license. See page 6-97 in PROGRAMMING.

35.1 System forwarding provides the ability to program the system so that calls ringing at a station will follow a database-programmed "forwarding path" that routes the call based on the type of call and the status of the intended station. (The Call Forwarding station feature can be used to override System Forwarding.) Several terms will be used to describe this feature, including:

- **Principal Station**: Original station being called.
- **Forwarding Point**: A destination (station, hunt group, or off-node device) to which the system-forwarded call is directed. Each forwarding path can contain up to four forwarding points.
- **Forwarding Path**: The sequence of forwarding points to which the call will be sent. Forwarding paths are programmed in the database. Up to 200 unique paths can be programmed. Each station can have up to three forwarding paths.

35.2 Two system timers are used with this feature:

- **System Forwarding Initiate**: Determines how long a call will ring (unanswered) at the principal station before moving to the first forwarding point. The default value is 15 seconds and the range is 2-255 seconds.
- **System Forwarding Advance**: Determines how long the call will ring (unanswered) before moving to the next forwarding point. The default value is 15 seconds and the range is 2-255 seconds.

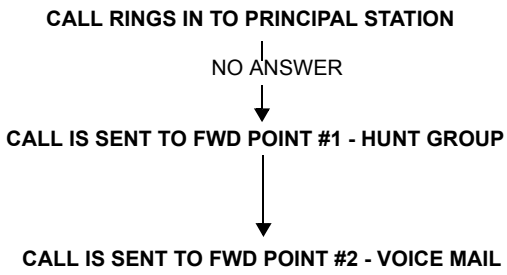
35.3 For each forwarding path assigned to the station, the system checks the following three criteria to determine if and when a call should be forwarded:

- **Type of Incoming Call**: Up to six different types of calls can be programmed to be sent to the forwarding path. They are:
  - — Outside calls received through a call routing table, including DID and E&M [DDI in Europe] calls, but not including DISA calls
  - — Ringing outside calls
  - — Transferred outside calls (including automated attendant and voice mail transfers)
  - — Recalling outside calls
  - — DISA calls (including DISA calls received through a call routing table)
  - — Intercom calls
- **Station Status**: The system recognizes four different types of station status.
  - — *No Answer:* If the call is not answered at the principal station before the System Forwarding Initiate timer expires, the system sends the call to the forwarding path.
  - — *Busy*: If the principal station is busy, the system immediately sends the call to the forwarding path. Both "No Answer" and "Busy" can be selected together to form a "Not Available" status.
  - — *Do-Not-Disturb*: If the principal station is in do-not-disturb, the system immediately sends the call to the forwarding path.
  - — *Immediate*: The system immediately sends the call to the forwarding path. The principal station user cannot answer an immediately forwarded call, but can reverse transfer it from the forwarding point.

- **Day-Mode/Night-Mode Status**: The system day/night mode is used to determine if a forwarding path is taken. Either day mode, night mode, or both may be specified.

35.4 For an example, see the diagram below. In the example, the "principal" station user wants direct ring-in or transferred calls to ring at his or her station during day mode. The principal station has a forwarding path with two forwarding points: a hunt group and voice mail. If the principal does not answer the call, it follows the forwarding path to forwarding point #1, a hunt group. If the hunt group does not answer the call, it would continue on the forwarding path and go to the principal station's voice mailbox (forwarding point #2).

35.5 If the principal station is a phone, the "ring principal once" option can be set that will signal the principal station when a call begins to follow the forwarding path. The signal to the principal station consists of a display (CALL SENT TO FORWARD PATH) and a single burst of ring tone. The call cannot be answered at the principal station, but can be reverse transferred from the system forward point.



35.6 If a call rings in to multiple stations, and one or more of those stations has system forwarding, the call will not follow any of the forwarding paths.

35.7 Agent Help requests will not follow any forwarding path.

35.8 If a principal station or a station forwarding point is a member of a hunt group, calls placed to the hunt group's pilot number are unaffected by system forwarding. The hunt group calls will be received at the station as usual and will not enter the system forwarding path.

35.9 A call follows only the forwarding path of the principal, even if a forwarding point has a forwarding path of its own. The call that originated at the principal station will follow only the principal station's forwarding path.

A. INDIVIDUAL STATION FORWARDING POINTS

35.10 The ringing duration at each station forwarding point is determined by the System Forwarding Advance timer (defaults to 15 seconds). If the station forwarding point is busy, the call will camp-on until the System Forwarding Advance timer expires and then it moves to the next forwarding point.

35.11 If the station forwarding point is in do-not-disturb, the forwarding path will bypass that station and immediately send the call to the next forwarding point.

35.12 If a forwarding point is a Voice Processing application, and the system is unable to communicate with the Voice Processing PC, the call will bypass the forwarding point.

35.13 A station forwarding point can place calls or transfer calls to the principal.

B. HUNT GROUP FORWARDING POINTS

35.14 If a forwarding point is a hunt group, a call following the forwarding path will be sent to the announcement station or overflow station as usual; it will advance to the next forwarding point if not answered before the System Forwarding Advance timer expires.

35.15 The call will ring at the hunt group until the System Forwarding Advance timer expires. It will then move to the next forwarding point. The No Answer Advance timer determines how long the call will ring at each hunt group station, as usual.

35.16 If all stations in a hunt group forwarding point have do-not-disturb or hunt group remove enabled, the call will camp on until the System Forwarding Advance timer expires. The call will then be sent to the next forwarding point.

C. HANDSFREE ANNOUNCE SYSTEM FORWARD OPTION

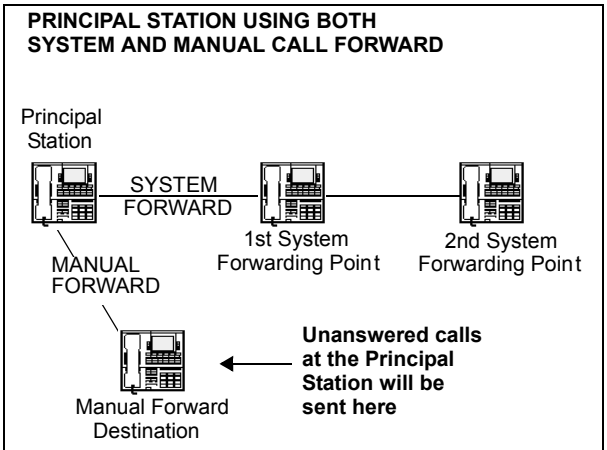
35.17 When a call is transferred to the principal station by another station user ***and the principal station's system forward is set for the "no answer" station status***, a system-wide option determines what the caller hears, as shown in the table below.

Table 5-37. *Handsfree Announce System Forward Option*

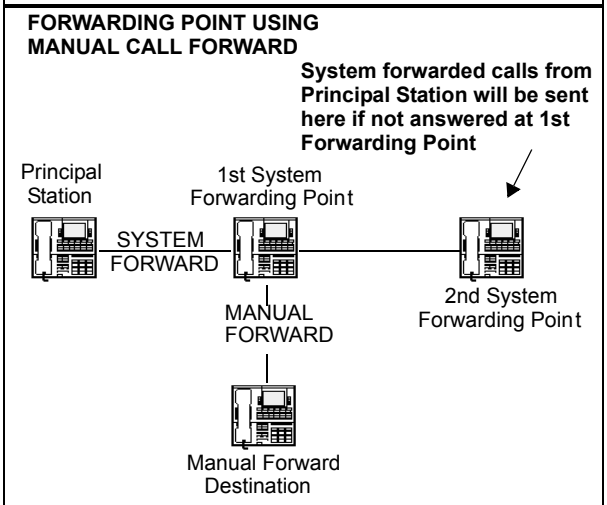
Handsfree Ann. Option	Sys. Forward Option	Result
Disabled	Intercom and/or outside calls forwarded	The intercom call from the transferring station enters the forwarding path after the System Forwarding Initiate timer expires. The transferring station user hears ringing and can announce the call only if it is answered.
Enabled	Outside calls forwarded	The transferring station user will place a handsfree call to the principal station and can announce the transfer; the *System Forwarding Initiate* timer does not start until the transfer is completed.
Enabled	Intercom calls forwarded	The handsfree announce option is overridden and the intercom call from the transferring station enters the forwarding path after the System Forwarding Initiate timer expires. The transferring station user hears ringing and can announce the call only if it is answered.
Enabled or Disabled	Calls forwarded to voice mail	The transfer is completed when the transferring station hangs up. The principal station's mailbox number is automatically dialed.

D. MANUAL CALL FORWARDING AND SYSTEM FORWARDING

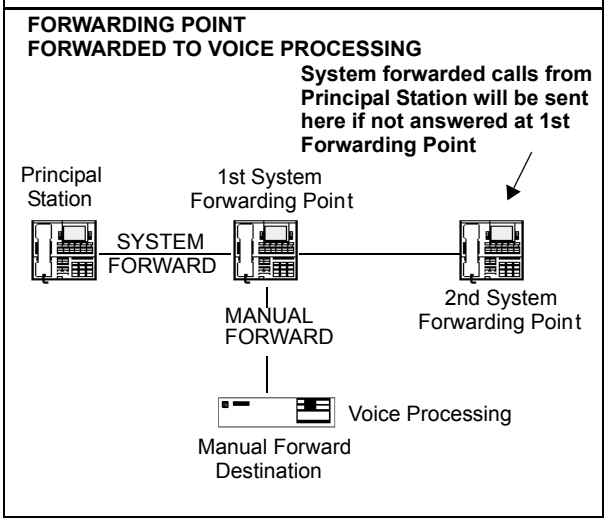
35.18 If the principal station has manual call forward enabled (using the FWD button or one of the Call Forwarding feature codes), the Call Forward feature overrides System Forwarding.



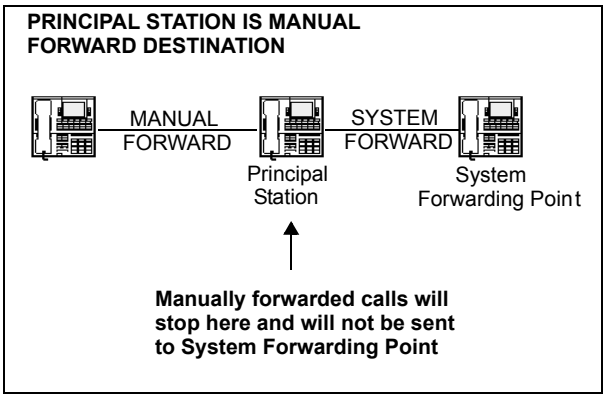
35.19 If a forwarding point has the Call Forward feature enabled, a system-forwarded call will ring at the forward destination until it is answered or the System Forward Advance timer expires; then the call moves on to the next forwarding point.



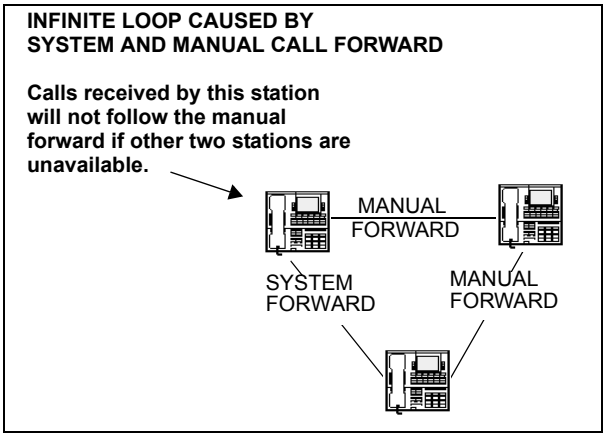
35.20 If a station forwarding point is manually forwarded to voice mail, the call will not follow the manual forward to Voice Processing; it will ring at the station forwarding point.



35.21 If the principal station *receives* a manually forwarded call (not a system forward), that call will ***not*** follow the principal station's system forwarding path.



If an infinite forward loop results from the combination of manual forwards and system forwarding paths, the station that was originally intended to receive the call will ring, even if the station is in do-not-disturb.



E. SYSTEM FORWARD ENABLE/DISABLE

35.22 The station user has the option of entering feature codes that will disable or enable the use of system forwarding at that station. Note that this flag does not affect the station's appearance on any other station's call forwarding path. The flag merely determines if the station will have its own calls system forwarded.

35.23 To enable or disable system forwarding:

Inter-Tel Phone: Enter the System Forward On (352), System Forward Off (353), or System Forward On/Off (354) feature code. The display shows SYSTEM FORWARD ON (or OFF). If disabled, the system will ignore any system forwarding paths programmed for this phone until you re-enable system forwarding.

Single-Line Set: Lift the handset and enter the System Forward On (352), System Forward Off (353), or System Forward On/Off (354) feature code. If disabled, the system will ignore any system forwarding paths programmed for this phone until you re-enable system forwarding.

F. UNANSWERED SYSTEM FORWARD CALLS

35.24 If a call that is following a system forwarding path is not answered or all forwarding points are in do-not-disturb or busy, its final destination depends on the type of call as shown below.

Table 5-38. *Unanswered System Forward Calls*

Type of Call	All Fwd Points are Busy or Do Not Answer	All Fwd Points are in Do-Not-Disturb
Intercom	Rings at the last forwarding point until it is answered or the caller hangs up.	Call never leaves the principal station.
Call routing ring-in (except DISA)	Rings at last forwarding point until it is answered or caller hangs up.	Rings at the principal station until it is answered or the caller hangs up.
Direct ring-in	Rings at last forwarding point until it is answered or caller hangs up.	Rings at the principal station until it is answered or the caller hangs up.
Transfer from a station, automated attendant or voice mail	Recalls the transferring station's transfer recall destination when the System Forwarding Advance timer expires at the last forwarding point.	*If immediate forwarding is enabled,* rings the principal station until the applicable Transfer timer expires. Then it recalls the transferring station's transfer recall destination. *If a conditional forwarding is enabled,* rings principal station until the System Forward Initiate timer expires. Then it recalls the transferring station's transfer recall destination.
DISA (including Call Routing to DISA)	Rings at last forwarding point until the Abandoned Call timer expires.	Rings at the principal station until the Abandoned Call timer expires.
Recalls	Recalls the principal station's attendant when the System Forwarding Advance timer expires at the last forwarding point. If the principal station does not have an attendant, the call continues to ring at the last forwarding point until the Abandoned Call timer expires.	*If immediate forwarding is enabled,* rings principal station until the Recall timer expires. Then it recalls the principal station's attendant. *If a conditional forwarding is enabled,* rings principal station until the System Forward Initiate timer expires. Then it recalls the principal station's attendant. If the principal station does not have an attendant, the call continues to ring at the principal station until the Abandoned Call timer expires.

36. CALL FORWARDING

36.1 With call forwarding, a station user can route incoming intercom and outside calls (including direct ring-in calls) to another station, to a hunt group, or to an outside telephone number, if allowed by toll and trunk restrictions. In a network setting, the network allows the user to forward calls to stations or hunt groups on another node. (See also page 5-264) The four forwarding options are:

- **Forward All Calls**: All incoming calls are immediately forwarded.
- **Forward If No Answer**: Incoming calls are forwarded if they are not answered before the Forward No Answer timer expires.
- **Forward If Busy**: Incoming calls are immediately forwarded if the station is busy.
- **Forward If No Answer or Busy**: Incoming calls are forwarded immediately if the station is busy, or forwarded if calls are not answered before the Forward No Answer timer expires.

36.2 If forward all calls is enabled, display phones show the call forwarding status and destination until the request is canceled. If one of the conditional forwards is enabled (no answer, busy, or no answer/busy), display phones show the forward status for five seconds and then return to normal. If the station receiving the forward is a display phone, it shows EXT XXXX FWD FROM EXT XXXX for each forwarded call received.

36.3 Station users can chain forwards from station to station providing the forwards do not form a loop and the chain does not include more than 10 stations. The conditional forward features (if busy, if no answer, if busy/no answer) may form a loop that the system cannot detect until a call is placed to the forwarding station. For example, if two station users forward their calls to each other using the Forward If Busy feature, the system accepts the requests. However, if a call rings in while both stations are busy, the forwards create an illegal loop. In this case, the call camps on to the called station and that station's display shows INVALID FORWARD PATH.

36.4 If more than one station has ring in for a trunk group, direct ring-in calls on that trunk group will forward to extension numbers, but not outside numbers or voice mail ports. The display of the phone receiving the forwarded call will show it as a forwarded call, and the CALL button or individual trunk button flashes to show ring in. (The individual trunk button also flashes on the phone that is being forwarded.)

36.5 Calls cannot be forwarded to restricted outside telephone numbers or stations in do-not-disturb. If the station that is programmed to receive your forwarded calls is later placed in do-not-disturb and you receive a call, you will momentarily see a display showing that the destination station is in do-not-disturb; the call will remain at your station.

NOTE: Direct ring-in calls that are forwarded to a station in DND will ring the DND station per the rules of DND.

36.6 If your station is in do-not-disturb and you have call forwarding programmed, the call is still forwarded, unless you enabled Forward No Answer. With Forward No Answer, intercom callers will see the do-not-disturb display that you programmed, and the call will not be forwarded.

36.7 If calls are forwarded to a Voice Processing application, and the system is unable to communicate with the Voice Processing PC, the call will not be forwarded. It will remain at your station.

36.8 Call forwarding overrides system call forwarding at the principal station.

36.9 If a station is programmed for Forward No Answer or Forward No Answer/Busy and receives a transferred call, the Forward No Answer timer is started when the intercom call from the transferring station begins ringing. The Forward No Answer timer is restarted if the transferring station completes the transfer before the timer expires (the transferred call will then be forwarded when the timer expires).

36.10 Agent Help Request calls, queue callbacks, and recalls do not forward, except that a recall at an attendant's station will forward to another station.

A. FWD BUTTON AND MENU BUTTONS

36.11 Some phones use a combination of the default FWD button and forwarding menu buttons. When the default FWD button is pressed, the display shows a menu of call forwarding options. The user must then select the desired option by pressing the corresponding button next to the display. If the FWD button is programmed to use any other Forward feature code (355-358) the forwarding options will not be displayed when the FWD button is pressed; the phone will operate the same as any other phone.

36.12 Any phone user can make any user-programmable button (including the FWD button) a FWD button that uses one of the forwarding feature codes (355-358). See the procedures on page 5-156 for programming user-programmable buttons.

36.13 A user-programmed FWD button is lit only when the station is programmed for the call forwarding condition enabled by that button. For example, if a user-programmed FWD button is set to forward calls when the station is busy, the button will be lit when the Forward If Busy feature is enabled, but not if the Forward If No Answer feature is enabled.

36.14 If a phone user has both a fixed FWD button and a user-programmable FWD button, the fixed button will always light when the station is forwarded. However, the user-programmable FWD button will be lit only when the forwarding option activated by that button has been selected. For example, if the user has the Forward All Calls feature programmed under a user-programmable button, that button will light if either the fixed or programmable FWD button is used to select that feature. However, if the fixed button or a feature code is used to set the Forward If Busy feature, only the fixed button will light.

B. FORWARD TO AN OUTSIDE NUMBER

36.15 When programming a station for call forward to an outside telephone number, a trunk access code is programmed before the telephone number. If the station is called while the selected trunk group is busy, the call will not be forwarded. Calls will ring at the forwarded station until the call is answered or the caller hangs up. The forwarded station's display shows FORWARD TO BUSY TRUNK.

36.16 The forwarded station's (not the intercom caller's) trunk and toll restrictions are checked when an intercom call is forwarded to an outside number.

36.17 When an outside call is forwarded to an outside number, the Unsupervised CO timer is activated. When the timer expires, the call recalls the attendant. If the attendant does not answer the recall before the Abandoned Call timer expires, the call is disconnected.

36.18 Stations with Forced Local Toll Call and Forced Long-Distance Toll Call account codes or Toll (National) account code in Europe cannot forward calls to outside numbers.

36.19 If a trunk group is assigned direct ring in to multiple stations, and one of those stations is forwarded to an outside number, incoming calls on that trunk group are not forwarded to the outside number.

NOTE: There *may* be some reduction in voice volume when an outside call is forwarded to an outside telephone number, depending on central office [local exchange] trunk quality.

36.20 Manual Forwarding to Public Network: There is a station flag called Manual Forwarding to Public Network. When the Manual Forwarding to Public Network station flag is disabled, it prevents the station from manually forwarding calls to the public network. This prevents users from calling in to a local station and reaching toll numbers via forwarding. This station flag is located in System\Devices and Feature Codes\Stations\Local\<*station*>\Flags.

36.21 Screened Transfer Method: When a subscriber programs manual call forwarding to their cell phone (outside public number), they cannot use the Screened Transfer Method. This is not a supported configuration.

C. FORWARD TO THE MESSAGE CENTER

36.22 Station users can forward calls to their message centers. Phone users can forward calls to their designated message center by pressing the FWD button and then the MSG button or **MESSAGE CENTER** menu button. Single-line users must use the call forwarding procedure and dial the extension number of the message center.

D. FORWARD TO AN ATTENDANT

36.23 Station users can forward calls to their attendant by pressing the FWD button and then 0, or by entering a Call Forwarding feature code and dialing 0.

E. FORWARD TO VOICE MAIL

36.24 If a station is forwarded to voice mail or if the message center is voice mail, the user's mailbox number is automatically dialed when the voice mail unit answers the forwarded call.

36.25 If a chain of forwarded stations ends in voice mail, the mailbox number of the first station in the chain will be selected when the voice mail unit answers the call.

36.26 If a trunk group is assigned direct ring in to multiple stations, and one of those stations is forwarded to a voice mail unit, incoming calls on that trunk group are not forwarded to the voice mail unit.

F. HOW TO FORWARD CALLS TO AN EXTENSION OR OUTSIDE NUMBER

36.27 Station users can also forward calls using the Remote Programming feature described on page 5-262.

36.28 To forward calls:

1. **Phones with Menu Buttons**: Press the FWD button. The display shows SELECT TYPE OF FORWARD. Press the menu button for the desired forwarding option (ALL, NO ANSWER, BUSY, or NO ANSWER/BUSY). The display shows ENTER FORWARD DEST.

   Other Phones: While on or off hook, press the FWD button or enter one of the following feature codes. (Display phones show ENTER FORWARD DEST.)

   - Call Forward All Calls (355)
   - Call Forward If No Answer (356)

- Call Forward If Busy (357)
- Call Forward If No Answer Or Busy (358)

Single-Line Set: Lift the handset and enter one of the feature codes listed above.

2. *If forwarding to an extension number or pilot number,* do one of the following:

   - Press a DSS/BLF button.
   - Dial the extension number or pilot number to receive the calls.
   - Dial 0 to forward to your attendant.
   - *(Inter-Tel phones only)* Press the MSG button or **MESSAGE CENTER** menu button to forward calls to your message center.
   - *(Phones with menu buttons only)* Press the **VOICE MAIL** button to forward to your mailbox.
   - *(Phones with menu buttons only)* Press the **IC DIRECTORY** menu button to look up and select a number.

   NOTE: If you dialed an invalid extension number, your display shows FORWARD TO INVALID NUMBER, you hear reorder tones, and must try again. If you dialed a station that is in do-not-disturb, the display shows DESTINATION IN DND, you hear reorder tones and must enter a new destination.

   If forwarding to an outside telephone number, do one of the following:

   - Dial a trunk access code (8, 92001-92208). The display shows ENTER PHONE #. Dial the desired telephone number.
   - *(Inter-Tel phones only)* Press a trunk access button (OUTGOING, ARS, trunk group button, or trunk button). The display shows ENTER PHONE #. Dial the desired telephone number.
   - *(Phones with menu buttons only)* Press the **OUTSIDE PHONE** menu button. The display shows ENTER PHONE #. Dial, speed dial, or redial the desired telephone number or press the **OUTSIDE DIR** menu button to look up and select a number.

   NOTE: If the forward destination is invalid because of toll restriction, ARS-Only restriction, no outgoing access for the trunk, no ARS or ARS route group for the dialed number, or an invalid dialed number, you hear reorder tones and must enter a new destination. (The display explains why the call was denied.)

3. **Phone**: *If off hook,* hang up. You hear a progress tone and the FWD button is lit. The display shows FWD *(condition)* TO XXXX. If you attempted to forward calls to a station that is unconditionally forwarded to your station or if the forwarding path already has 10 stations, your display shows INVALID FORWARD PATH, and you hear reorder tones and must enter a new destination.

   Single-Line Set: Hang up.

36.29 To cancel any call forward request:

Phones with Menu Buttons: Press the FWD button and then press the **FWD OFF** menu button. You hear a progress tone and the FWD button is unlit. The display shows ANY CALL FORWARD CANCELED.

Other Phones: Press the lit FWD button (the Speaker button lights) then press the SPKR button (both buttons go off). You hear a progress tone. Display phones show ANY CALL FORWARD CANCELED.

Single-Line Set: Lift the handset, enter any Call Forwarding feature code (355-358), and hang up.

37. REDIRECT CALL

37.1 The Redirect Call feature allows the phone user to route ringing outside, intercom, and camped on calls to another station, hunt group, or outside number. This is in addition to the currently available option of redirecting calls to voice mail or Do-Not-Disturb. Routing redirected calls is subject to toll and trunk restrictions and does not require any software licenses.

37.2 To redirect a call:

Phones with menu buttons: Select the **SEND TO DEST** menu option or enter the Redirect Ringing Call feature code (331) followed by the destination number. Then hang up.

Other Phones: Enter the Redirect Ringing Call feature code (331) followed by the destination number. Then hang up.

If the station receiving the redirect call has a display phone, the display shows EXT XXXX FWD FROM EXT XXXX. If the system does not redirect a call to a destination, it rings back at the original station.

37.3 The call types listed below do not follow call forwarding and cannot be redirected:

- Agent Help Request calls
- Queue Callback calls
- Recalls

37.4 In addition, calls may not be redirected to the following destinations:

- Restricted outside numbers
- Stations in Do-Not-Disturb mode
- Stations that are forwarded
- Another station's mailbox
- The original station (display phones show CALL FORWARDED TO SOURCE)
- An unreachable destination

37.5 If an ARS, trunk group, or trunk number is entered, the phone system prompts the user to enter the destination telephone number. Calls may not be redirected to a node number, but may be redirected to a station on another node by entering the correct extension number.

37.6 If the applicable Forwarding timer expires before the user completes the redirection process, the system terminates the call.

38. SPEED DIALING

NOTE: SIP trunks do not support the speed dial features.

38.1 Three phone system features provide speed-dialing. They are as follows:

- **System Speed Dialing**: Up to 1000 (5000 with a CP Server), 48-digit system speed-dial numbers can be stored in the system.
- **Station Speed Dialing**: Each station user can program up to 10, 16-digit station speed-dial numbers. Display phone users can program a 16-character name for each station speed-dial number.
- **System Directory (Intercom and Outside)**: The intercom directory enables display phone users to "look up" intercom extension numbers and user names. The outside directory enables display phone users to "look up" system speed-dial numbers and associated names.

NOTE: This feature requires the Directories software license. See page 6-97 in PROGRAMMING.

A. SYSTEM SPEED DIALING

38.2 Speed dialing allows station users to dial stored telephone numbers quickly. Up to 1000 (5000 with a CP Server), 48-digit system speed-dial numbers can be stored in system memory. If desired, an identifying name can also be stored with each number. See also the System Directory feature on page 5-241 and Caller ID [CLIP] on page 5-72.

38.3 Each node in a network has its own system speed dial numbers. System speed dial numbers can be used only on the node where they are programmed.

38.4 To keep system speed-dial numbers confidential, some or all can be programmed as non-display numbers (see PROGRAMMING, page 6-317). Non-display numbers can be used by any station user, but are displayed only on the programming station's phone. (Non-display numbers cannot be redialed or saved as station speed-dial numbers at a display phone.) Non-display numbers *will* appear in the SMDR report.

38.5 System speed-dial names can be programmed by an Administrator using English/Spanish or Japanese characters, or a combination. The programmed language for the phone does not affect the characters that can be viewed. That is, no matter what language the phone uses or which characters are in the name, the phone user will be able to see the speed-dial names exactly the way they were programmed. In lists, such as speed-dial directories, the English/Spanish characters are alphabetized before the Japanese characters (Japanese names will be at the end of the list).

38.6 System speed-dial numbers are subject to toll restriction unless a system-wide option has been enabled that allows any station to dial any system speed-dial number regardless of toll restriction.

38.7 The system speed-dial numbers are stored using location codes (0000-4999). When dialed, the numbers appear on a display phone unless they have been programmed as non-display numbers. Display phone users can also view system speed-dial numbers and names without dialing; however, with non-display numbers, only the name appears.

38.8 System speed dial codes 0000-4999 can be stored in user-programmable phone buttons by following the directions given on page 5-156. These buttons allow one-button dialing of system speed-dial numbers.

38.9 The system speed-dial numbers and names are stored in battery-backed RAM and will not be erased in the event of a power failure.

38.10 The system speed-dial numbers and names are programmed by the installer or at any Administrator station. (See page 5-297 for Administrator programming instructions.)

Viewing System Speed-Dial Numbers

38.11 To view system speed-dial numbers and names:

NOTE: If you make a mistake, use the MUTE button to backspace.

1. While on hook, press SYS SPDL or enter the System-Speed Dial feature code (381). The display shows REVIEW SYS SPEED # (0000-4999).
2. Enter the desired location code (0000-4999) or press the **SEARCH** menu button. The display shows the programmed name and number.
3. *If desired,* scroll forward through the speed-dial locations in alphabetical order by pressing the high end of the **VOLUME** button or the **NEXT** button, or scroll backward by pressing the low end of the **VOLUME** button or the **PREVIOUS** button.
4. Repeat steps 2 and 3 for each speed-dial location to be viewed. Or, press * or CANCEL to terminate this feature.
5. **To automatically select a trunk and dial the displayed number,** press #.

Dialing System Speed-Dial Numbers

38.12 To dial system speed-dial numbers at a phone:

1. *If desired,* lift the handset and select an outgoing trunk. (You may skip this step.)
2. Press SYS SPDL or enter the System Speed-Dial feature code (381). The display shows REVIEW SYS SPEED # (0000-4999). Non-display phone users hear a confirmation tone.
3. Enter the location code (0000-4999) for the desired number. *If you have not selected a trunk,* press # or the **ACCEPT** menu button to accept the number and automatically select a trunk. Or, press the ARS button, a trunk button, a trunk group button, or the OUTGOING button to select a trunk. The number is automatically dialed. (Display phones show the number dialed unless it is a non-display number.) If you hear reorder tones, your station is restricted from dialing the selected number. If the display shows INVALID BIN NUMBER, you entered an invalid speed-dial location code.

38.13 To dial system speed-dial numbers at a single-line set:

1. Lift the handset and select an outgoing trunk.
2. Hookflash (press **RECALL**) and enter the System Speed-Dial feature code (381). You hear a confirmation tone.
3. Enter the location code (0000-4999) for the desired number. The number is automatically dialed. If you hear reorder tones, your station is restricted from dialing the selected number.

B. STATION SPEED DIALING

38.14 For convenience, each station user can program up to 10, 16-digit station speed-dial numbers. Single-line sets use speed-dial location codes (0-9). Phones use speed-dial buttons, if programmed in the phone's keymap, or location codes 0-9. Together, the lamps in the phone speed-dial buttons create a busy lamp field that indicates the status of the stations programmed under the buttons (see the lamp status chart on page 5-141). Speed-dial buttons can contain outside telephone numbers, feature codes, station extension numbers, or hunt group pilot numbers.

38.15 Station speed-dial codes 0-9 can be stored in user-programmable buttons to create speed-dial buttons.

Programming Station Speed-Dial Numbers and Names

38.16 When entering the outside telephone numbers, use one of the following methods:

- Use the keypad to manually dial the number.
- On a phone, press the REDIAL button to enter the last number (up to the first 16 digits) dialed or saved at the station.
- On a phone, press the SYS SPDL button and enter the system speed-dial location code (000-999 or 0000-4999) to store one of the system speed-dial numbers in a station speed-dial location. (If the number is over 16 digits, only the first 16 digits are stored.) Non-display system speed-dial numbers cannot be stored in phone station speed-dial locations.

38.17 If desired, outside telephone numbers can be preceded with a trunk access code to allow one-button dialing of outside telephone numbers. For example, a button programmed with "89619000" would select a trunk using the Outgoing Calls feature code (8) then dial 961-9000.

38.18 A station extension number can be preceded with a pound [hash] (#) to always speed dial private intercom calls to the station. Or, a "4" may be entered before a station extension number or hunt group pilot number to quickly reverse transfer (pick up) calls from that station or hunt group. If either of these options is used, normal handsfree intercom calls cannot be placed using that station speed-dial location or speed-dial button, and the speed-dial button will not show the station's status.

38.19 An outside telephone number can be preceded by a trunk access code for easier trunk selection and number dialing. Station users can also program pauses and/or hookflashes (recalls) into the stored outside telephone numbers. For example, with a U.S. system, the number can contain an SCC local number, a pause, and an access code. When programming speed-dial numbers, each hookflash (press **RECALL**) and each pause is considered one digit. The durations of the hookflash (recall) and the pause are determined by the programmable LS/GS CO Hookflash [LS CO Hookflash in Europe] and Pause Dialing Digit Length timers.

38.20 Display phone users can program an associated name for each station speed-dial number. Speed-dial names can contain up to 16 characters. To program speed-dial names, keypad buttons are used to enter the desired letters, numbers, and punctuation in English/Spanish and/or Japanese (Katakana) as shown in the chart on the next page. (Only Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones can display Japanese characters.) When programming speed-dial names, you can press the MSG button or **USE ALPHA MODE/USE NUMERIC MODE** menu button to switch back and forth between alphanumeric and numeric mode.

- **In Numeric Mode**: the keypad buttons are used to enter numbers 0-9, the pound [hash] (#) button is used for entering a hyphen (-), and the asterisk (*) button is used for entering a colon (:). For example, 1*00 would enter "1:00" in numeric mode.

- **In Alphanumeric Mode**: the keypad buttons are used to enter the desired letters, numbers, and punctuation. The number of times a button is pressed determines which character is entered. For example, 533266 would enter "JEAN." When adjoining characters are located under the same button, press the FWD button to advance to the next character. For example, 66 FWD 6667776 would enter "NORM." See the chart below to program information in alphanumeric mode. (Note that letters correspond to the letters printed on the keypad buttons.)

BTN	Number of Times Button is Pressed										
	1	2	3	4	5	6	7	8	9	10	11
	English/Spanish Characters					Katakana Characters**					
1	-	&	(	)	1	A	I	U	E	O	a
2	A	B	C	’	2	KA	KI	KU	KE	KO	i
3	D	E	F	!	3	SA	SHI	SU	SE	SO	u
4	G	H	I	*	4	TA	CHI	TSU	TE	TO	e
5	J	K	L	# or /*	5	NA	NI	NU	NE	NO	o
6	M	N	O	Ñ or #*	6	HA	HI	FU	HE	HO	tsu
7	P	Q	R	S	7	MA	MI	MU	ME	MO	ya
8	T	U	V	?	8	YA	YU	YO	.	,	yu
9	W	X	Y	Z	9	RA	RI	RU	RE	RO	yo
0	@	:	.	,	0	WA	WO	N	pa	ba	long

* The character available depends on the software version.

** The Japanese characters are only available if the Multilingual feature is enabled and Japanese is installed as the secondary language.

38.21 To program station speed-dial numbers at a display phone:

NOTE: If you make a mistake, press * or **CANCEL**; then start over. The number in memory remains unchanged. Or, use the MUTE button to backspace.

1. While on hook, enter the Program Station Speed Dial feature code (383). The display shows PROGRAM STN SPD (0-9) OR SCROLL. Do one of the following:
   - **Use a Speed-Dial Button**: Press the speed-dial button to be programmed.
   - **Use a Location Code**: Dial the desired speed-dial location code (0-9).
   - **Scroll through the Displayed Numbers**: Scroll to the desired number by pressing the **VOLUME**, **NEXT**, or **PREVIOUS** button.
   - **Use the Menu Button**: Use the menu buttons to select the speed-dial location.
2. When the display shows ENTER NAME do one of the following:
   - **To change or program the name:** Enter the desired name for the speed-dial number using your keypad as described on page 5-236. See the chart above. Then press # or **ACCEPT**. *If you make a mistake,* use the MUTE button to back up and erase existing characters or the **CLEAR** button to erase the whole entry.
   - **To erase the current name and leave it blank:** Press **CLEAR** or press MUTE repeatedly until the name is erased. Then press # or **ACCEPT**.
   - **To leave the name the same:** Press # or **ACCEPT**.

3. When the display shows ENTER NUMBER, do one of the following:

   - **To enter or change a number:** Dial the extension number or telephone number (up to 16 digits, including pauses and hookflashes/recalls) to be stored. *To include *, #, a hookflash/recall, or a pause in the number,* press ∞ or SPCL once for an asterisk (*), twice for a pound [hash] (#), three times for a hookflash (recall), or four times for a pause. You can enter more than one *, #, or hookflash (recall) in a row by pressing the FWD button to move to the next character (not necessary after pauses). *If you make a mistake,* press MUTE to back up and erase, or press * or **CANCEL** to leave the number unchanged; then start over at step 1.

   - **To erase an existing number and leave it blank:** Press MUTE until the number is erased.

4. Press # to exit. Repeat these steps for each additional number to be stored.

38.22 To program station speed-dial numbers at a non-display phone:

NOTE: If you make a mistake, press * or **CANCEL**; then start over. The number in memory remains unchanged. Or, use the **MUTE** button to backspace.

1. While on hook, enter the Program Station Speed Dial feature code (383).
2. Press the speed-dial button to be programmed or dial the desired speed-dial location code (0-9). If you hear reorder tones, you entered an invalid speed dial location code; try again.
3. Do one of the following:

   To enter or change a number: Dial the extension number or telephone number (up to 16 digits, including pauses and hookflashes/recalls) to be stored. *To include *, #, a hookflash (recall), or a pause in the number,* press ∞ or SPCL once for an asterisk (*), twice for a pound [hash] (#), three times for a hookflash (recall), or four times for a pause. You can enter more than one *, #, or hookflash (recall) in a row by pressing the FWD button to move to the next character (not necessary after pauses). *If you make a mistake,* press MUTE to back up and erase, or press * or **CANCEL** to leave the number unchanged; then start over at step 1.

   To erase an existing number and leave it blank: press MUTE until the number is erased.

4. Press # to exit. Repeat these steps for each additional number to be stored.

38.23 To program station speed-dial numbers at a single-line set:

NOTE: If you make a mistake, press *; then start over. The number in memory remains unchanged.

1. Lift the handset and enter the Program Station Speed-Dial feature code (383). You hear a confirmation tone.
2. Dial the desired speed-dial location code (0-9). If you hear reorder tones, you entered an invalid speed dial location code; try again.

3. Do one of the following:

   To enter or change a number: Dial the extension number or telephone number (up to 16 digits, including pauses and hookflashes/recalls) to be stored. *To include *, #, a hookflash (recall), or a pause in the number,* hookflash (recall) once for an asterisk (*), twice for a pound [hash] (#), three times for a hookflash (recall), or four times for a pause. *If you make a mistake,* press ✳ to leave the number unchanged; then start over at step 1.

   To erase an existing number and leave it blank: Overwrite the existing number by entering only a hookflash (recall) as the new number.

4. Hang up to exit. Repeat these steps for each additional number to be stored.

Viewing Station Speed-Dial Numbers

38.24 To view the contents of a speed-dial button (on a display phone):

1. While on hook, enter the Review Keys feature code (396). The display shows PRESS THE KEY TO REVIEW. Press the desired speed-dial button. The display shows the currently stored number and name.
2. After viewing the number, press the Speaker button, ✳, or # to exit, or wait for the display to return to date and time.

38.25 To view the stored numbers and names (on a display phone):

1. While on hook, enter the Program Station Speed Dial feature code (383). The display shows PROGRAM STN SPD # (0-9) OR SCROLL. Do one of the following:
   - **Use a Location Code**: Dial the desired speed-dial location code (0-9). When the name appeared, press # or **ACCEPT** to view the number.
   - **Scroll through the Displayed Numbers**: Scroll to the desired number by pressing the **VOLUME**, **NEXT**, or **PREVIOUS** button. The name and number are displayed for each location.
   - **Use the Menu Buttons**: Use the menu buttons to select the desired speed-dial location. When the name appeared, press # or **ACCEPT** to view the number.
2. After viewing the number, press the Speaker button, ✳, or # to exit, or wait for the display to return to date and time.

Dialing Station Speed-Dial Numbers

38.26 To dial a station speed-dial number at a single-line set:

1. Lift the handset. (Phones can remain on hook.)
2. *If placing an outside call,* select an outgoing trunk or use ARS.

   If placing an intercom call or if the outside number contains a trunk access code, skip this step.
3. Hookflash (press **RECALL**) and enter the Station Speed-Dial feature code (382). You hear a confirmation tone. Then dial the location code (0-9). The number is dialed.

38.27 To dial a station speed-dial number at a non-display phone:

1. Lift the handset or remain on hook.
2. *If placing an outside call,* select an outgoing trunk or use ARS.

   If placing an intercom call or if the outside number contains a trunk access code, skip this step.
3. **Phone with a speed-dial button**: Press the speed-dial button of the desired number. The number is dialed.

   Phone without a speed-dial button: Enter the Station Speed-Dial feature code (382). You hear a confirmation tone. Then enter the location code (0-9). The number is dialed.

38.28 To dial a station speed-dial number at a display phone:

1. Lift the handset or remain on hook.
2. *If placing an outside call,* select an outgoing trunk or use ARS.

   If placing an intercom call or if the outside number contains a trunk access code, skip this step.
3. **Phone with a speed-dial button**: Press the speed-dial button of the desired number. The number is dialed. Display phones show the number.

   Phone without a speed-dial button: Enter the Station Speed-Dial feature code (382). Then do one of the following to dial the number:

   - **Use a Location Code**: Dial the desired speed-dial location code (0-9).
   - **Scroll through the Displayed Numbers**: Scroll to the desired number by pressing the **VOLUME**, **NEXT**, or **PREVIOUS** button.
   - **Use the Menu Buttons**: Use the menu buttons to select the desired speed-dial location.

39. INTERCOM, SPEED-DIAL, AND FEATURE CODE DIRECTORY

NOTE: These features are only available on Inter-Tel phones. In addition, the Directories feature is required. See page 6-97 in PROGRAMMING.

39.1 The Directory feature enables display phone users to "look up" intercom extension numbers and user names, system speed-dial numbers and names, or system feature codes. After the desired information has been displayed, the user may automatically dial the number, store the number in a station speed-dial location, select a different number/name, terminate the directory routine, or switch to the other directory. A directory search can be performed when a call is waiting on conference or transfer hold at the station.

39.2 In a network setting, a user can see intercom directory names and extension numbers for all devices that are programmed as "off-node devices" on the local node. Devices that are represented by wildcard off-node extensions will ***not*** show up in the intercom directory. (See page 6-198 for off-node device programming information.)

39.3 To use the directory, the user enters a letter, a string of letters, a valid extension number, or a valid feature code. If searching for a name, the full name need not be entered. The system will find the closest match and show the number and its associated name on the phone display. Or, the user can scroll alphabetically through the stored list of names. (It is not possible to scroll through the extension numbers or feature codes numerically; extensions and features scroll alphabetically.)

39.4 Directory names can include English/Spanish or Japanese characters, or a combination. In the directories, English/Spanish characters are alphabetized before the Japanese characters. Therefore, Japanese names will be at the end of the list.

NOTE: When using Japanese as the Primary or Secondary language, you can only search for IC and CO directory names based on the last name. You cannot search based on the first name.

39.5 Keypad buttons are used to enter the desired English/Spanish or Japanese (Katakana) letters, numbers, and punctuation. See the chart on the following page. (Only Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones can be used for viewing Japanese prompts. Other phones do not support Japanese displays.) The station user can switch between numeric and alphanumeric modes. (Numeric mode = MSG button unlit; alphanumeric mode = MSG button lit.) In alphanumeric mode, the number of times a button is pressed determines which character is entered. For example, 77776444844 would enter "SMITH." When adjoining characters are located under the same button, press the FWD button once to advance to the next character. For example, 5666 FWD 66337777 would enter "JONES." The letters correspond to the letters printed on the keypad buttons.

39.6 The intercom directory is automatically updated whenever user names and/or extension number information is reprogrammed. The outside directory is updated whenever a system speed-dial number or name is reprogrammed.

39.7 Only Administrator stations will be able to view Administrator feature codes using the directory. If the Diagnostics Mode feature is enabled, the Administrator will also be able to see the diagnostic feature codes.

BTN	Number of Times Button is Pressed										
	1	2	3	4	5	6	7	8	9	10	11
	English/Spanish Characters					Katakana Characters**					
1	-	&	(	)	1	A	I	U	E	O	a
2	A	B	C	’	2	KA	KI	KU	KE	KO	i
3	D	E	F	!	3	SA	SHI	SU	SE	SO	u
4	G	H	I	*	4	TA	CHI	TSU	TE	TO	e
5	J	K	L	# or /*	5	NA	NI	NU	NE	NO	o
6	M	N	O	Ñ or #*	6	HA	HI	FU	HE	HO	tsu
7	P	Q	R	S	7	MA	MI	MU	ME	MO	ya
8	T	U	V	?	8	YA	YU	YO	.	,	yu
9	W	X	Y	Z	9	RA	RI	RU	RE	RO	yo
0	@	:	.	,	0	WA	WO	N	pa	ba	long

* The character available depends on the software version.

** The Japanese characters are only available if the Multilingual feature is enabled and Japanese is installed as the secondary language.

39.8 To perform a directory search:

1. While on or off hook, enter the Directory feature code (307). The MSG button lights (and the Speaker button lights if on hook) and the display shows DIR TYPE: (IC=1 OUT=2 FEAT=3).

2. Press one of the following:

   For Intercom Directory: Press 1 or the **IC DIRECTORY** menu button to select the intercom directory. The display shows INTERCOM NAME.

   For Outside Directory: Press 2 or the **OUTSIDE DIR** menu button to select the outside number directory. The display shows OUTSIDE NAME.

   For Feature Directory: Press 3 or the **FEATURE DIR** menu button to select the feature code directory. The display shows FEATURE NAME.

3. Enter alpha and/or numeric characters for the number, code, or user/feature name (up to 10 characters for intercom directory, 16 for outside or feature directory). If searching for a name, the whole name does not have to be entered. You may switch between modes by pressing the MSG button or the **USE NUMERIC MODE/USE ALPHA MODE** menu button. If necessary, press * to cancel the search.

   a. *In numeric mode (MSG button and/or Message Lamp unlit):* Press the keypad buttons to enter a valid feature, extension, or speed-dial location number. Press MUTE to backspace, if necessary. Press the **SEARCH** menu button or press MSG to change to alphanumeric mode and then press # to begin the search.

b. *In alphanumeric mode (*MSG *button and/or Message Lamp lit):* Press the keypad buttons to enter the desired characters. See the chart above. Press FWD once to advance or twice to leave a space. Press MUTE to backspace. Press # or the **SEARCH** menu button to begin the search. The closest match will be displayed. Menu button phone users can return to the initial entry by pressing the **START OVER** menu button.

4. *If desired,* press the high end of the Volume button or the **NEXT** button to scroll forward alphabetically through the directory or press the low end of the Volume button or the **PREVIOUS** button to scroll backward.

5. Do one of the following:

   To store the selected number in a station speed-dial location: Press the desired speed-dial button. You hear confirmation tone when the number is accepted.

   To automatically dial a displayed extension number: Press #, the IC button, or the **ACCEPT** button. (If an intercom call is camped on and the IC button is lit, pressing the **IC** button answers the waiting call and terminates the Directory feature.)

   To dial a displayed outside number: Press #, the **CALL** menu button, or a CALL button. The system automatically dials the number using your programmed outgoing access code.

   To dial a displayed feature code: Press # or the **ACCEPT** button. The system automatically dials the feature code.

   To search for another name or number: Repeat the procedure beginning at step 3.

40. INTELLIGENT DIRECTORY SEARCH

40.1 When using an Axxess system, the user may encounter several situations where inputting text is necessary. Examples of these situations may include searching an IC, Outside, or Feature directory for a particular number, name, or feature. Of such situations, searching the directory is likely the most common scenario for entering text.

40.2 The Intelligent Directory Search (IDS) feature improves searching for names in the directory by allowing the user to press one number for each letter rather than several. Each button represents several letters. For example, dialpad button 5 represents "5," "J," "K," "L," "j," "k," or "l." To enter JONES, the user presses 5 6 6 3 7. As buttons are pressed, several possible matches appear on the display endpoint. This is similar to the "text on nine keys (T9)" feature on many cellular phones. No additional license or DB Programming is required to support IDS.

40.3 A display endpoint or an Inter-Tel Model 8602 IP softphone application is required for using the IDS feature. Also, an IP endpoint must be in ITP mode.

Dialpad Buttons for Display Endpoints (except for the 8602 IP Softphone)

40.4 System v9.1 or later supports new dialpad button characters. When needing to search for a number, name, or feature in a directory, users can press the following dialpad buttons to enter characters or perform an action. The buttons apply to buttons pressed on the endpoint.

Table 5-39. *Dialpad Buttons for Display Endpoints*

Dialpad Button	Represented Characters	Comment
0	0	
1	1	
2	2 A B C a b c Ç â ä à å ç Ä Å á	
3	3 D E F d e f é ê ë è É	
4	4 G H I g h i ï î ì í	
5	5 J K L j k l	
6	6 M N O m n o ô ö ò Ö ó ñ Ñ	
7	7 Q P R S q p r s	
8	8 T U V t u v ü û ù Ü ú	
9	9 W X Y Z w x y z ÿ	
Dialpad Button	**Action Characters**	**Comment**
*	Cancel	Returns endpoint to state before entering IDS.
#	Dial or perform	Dials or performs the first entry.
MUTE	Backspace	
Menu Buttons	Dial or perform the entry or scroll a page	The Menu Buttons are located on the right or left side of the six-line LCD. Two-line display endpoints do not have Menu Buttons.

Table 5-39. *Dialpad Buttons for Display Endpoints (Continued)*

Dialpad Button	Represented Characters	Comment
Volume Down	Previous match	Ignored if there is no previous match.
Volume Up	Next match	Ignored if there is no next match.

NOTE: The IDS feature works differently on the Executone display endpoints because the display is a different size than the traditional Inter-Tel endpoints. On the display, you may not see the >> arrow menu button to scroll forward or the << arrow menu button to scroll backward, instead the Volume Up and Volume Down buttons can be used to scroll through entries. Also, the IDS information may be displayed differently. There is no Mute button (unless programmed on one of the programmable buttons) for backspacing. Cancel the feature and start over if an error is made.

Dialpad Buttons for the 8602 IP Softphone

40.5 With the 8602 IP softphone, users have access to a keyboard and can type a name directly into a field. IDS handles the following input for those users.

Table 5-40. *Dialpad Buttons for the 8602 IP Softphone*

Dialpad Button	Represented Characters	Comment
A-Z and a-z	The matching letter, with or without diacritical/accent marks (e.g. the “~” on “ñ” in Spanish), and without attention to case	
0-9	The matching number	These numbers differ from those pressed on the endpoint, which match letters associated with that number as well.
Dialpad Button	**Action Characters**	**Comment**
Backspace	Backspace	
Escape	Cancel	Cancels the entry.
Enter	Dial or perform	Dials or performs the first entry.
Up Arrow	Previous match	Ignored if there is no previous match.
Down Arrow	Next match	Ignored if there is no next match.
Pg Up	Scroll to previous page	Ignored if there is no previous page.
Pg Down	Scroll to next page	Ignored if there is no next page.

40.6 The following pages provide information about how IDS is used in the IC, Outside, and Feature directory. For information about the Outside and Feature directories, see page 5-250.

A. IC DIRECTORY

Finding a User in the IC Directory

40.7 The traditional IC directory allows the user to find another user by entering the display name, which corresponds to the Username field or the extension number in DB Programming (under System\Devices and Feature Codes\Stations).

40.8 **Non-IDS Limitation**: The limitation of using only the Username field is that it holds only 10 characters, which is inadequate for holding both first and last names. Therefore, the names in the current IC directory may not reflect the criterion that a user may apply to find another user (for example, the administrator might have all usernames programmed to the first name of the endpoint's owner, but a particular user knows only the last name of the person he or she is seeking).

40.9 **IDS Functionality**: IDS seeks to improve that aspect by using the Description field (under System\Devices and Feature Codes\Stations). Currently, for Voice Mail, the administrator is advised to program the description to "last_name, first_name" because Voice Mail has a similar searching algorithm. The Description field can hold up to 20 characters, which will be adequate to hold most users' first and last names. This requires the administrator to program "last_name, first_name" (as much as possible) in the Description field. Note that the field is case-sensitive, therefore, IDS will display "Bond, James" rather than "BOND, JAMES" if that is what the administrator programmed into the database. When using this method, the algorithm considers everything before the comma as the last name and everything after the comma as the first name. For example, a user named "Billy Bob Via Linda" should have his description programmed as "Via Linda, Billy Bob." Be careful with descriptions like "Smith, George IP" to indicate that this extension is George Smith's IP phone because IDS will interpret the name is "George IP Smith."

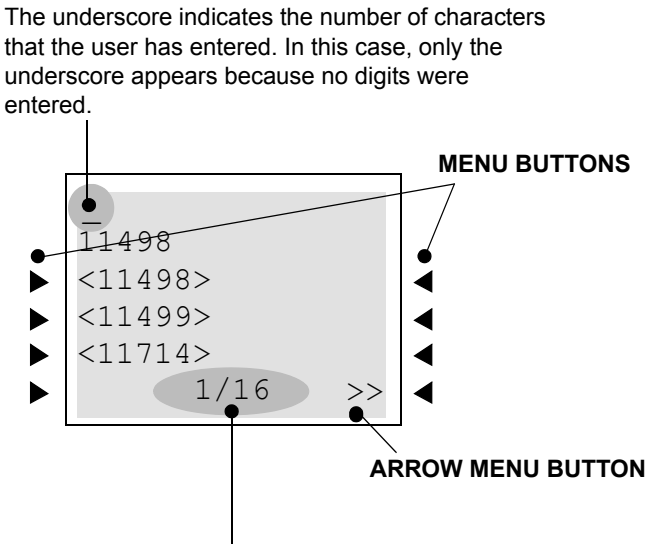
40.10 To allow the user to search by using either the first or last name, the directory effectively has two entries for each user. Although the existing IC directory allows searching by the extension number, which is essentially a reverse lookup, IDS does not currently support this function because there would be three entries for each user and reverse lookup is not a very useful feature in the IC directory.

40.11 Note that the user does not have to enter spaces or punctuation marks. For example, if the Description field for a user is "BOND, JAMES," the algorithm will match both the IDS equivalent of "BONDJAMES" (266352637) and the IDS equivalent of "JAMESBOND" (526372663). Because what the user is entering does not match what the user sees on the display (there may be spaces and punctuation marks), the cursor will always be immediately after the last matched character except in the case of secondary extension, which will always have an asterisk in the first column to indicate a secondary extension (see page 5-249).

IC Directory Examples

40.12 If users have not yet entered any digits in the IC directory, users will see the entire directory. The empty matches are displayed first. Instead of displaying a blank line, IDS shows the extension in brackets for any match that is empty. The following is an example of the IC directory that shows extensions that have no descriptions on a six-line display endpoint.

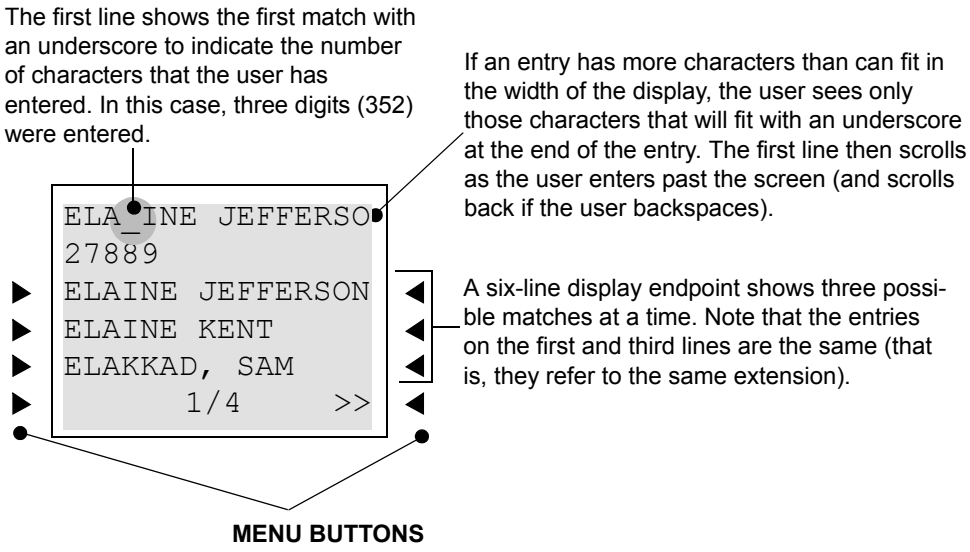
NOTE: Two-line display endpoints show only the first two lines.



The bottom line displays two numbers separated by a forward slash. The first number indicates which entry in the list of matches is the current entry, and the second number is the total number of matches (exact and partial) for the current input. For example, “1/16” indicates that the first entry refers to the first match out of sixteen). In the event that the screen cannot hold this information, IDS will display no information (for example, “10000/15200” will not fit on the display with the arrow (<< and >>) menu buttons).

40.13 To scroll through the entries, press the >> menu button or the volume up button to scroll forward, or press the << menu button or the volume down button to scroll backward. The arrow menu button scrolls per screen with a screenful of names, while the volume up/down button scrolls per entry. The arrow (<< and >>) menu buttons at the bottom of the display only appear when there are more than three entries on a six-line display endpoint. Also, if there is no previous page or entry, the << menu button is unavailable and the volume down button is unresponsive. Likewise, if there is no next page or match, the >> menu button is unavailable and the volume up button is unresponsive. For an analog display endpoint that does not have a volume soft button, press 9 to scroll forward and 7 to scroll backward.

40.14 When users enter 3 5 2 in the IC directory, a display similar to the one below appears.



40.15 The list of matches is in alphabetical order except that an alphabetical order of exact matches shows first. An exact match occurs when users enter a string and matches a name exactly (that is, the user has entered all typeable characters in the name).

40.16 In general, the algorithm shows exact matches and then shows partial matches. Both lists of matches will be in alphabetical order with respect to the first letter. If more than one exact match exists, these will be in alphabetical order (with respect to the first letter) as well and before the list of partial matches. If a user presses an invalid button that would make the list of matches empty, the display does not change, and a beep occurs to indicate an error.

40.17 For any number string, there will be a number of exact and partial matches initially, but the list will become smaller as the user enters more numbers. For example, the user can search for "MONICA SALES" by entering 6 6 6 4 2 2 7 2 5 3 7 and stopping at any time when the list contains "MONICA SALES." If the database is programmed correctly, the user can search for the same person by entering 7 2 5 3 7 6 6 6 4 2 2 to search for "SALES, MONICA" as an alternative. Most likely, users will not need to enter the entire number string in either case.

40.18 To dial the user in the first line, whose extension appears in the second line, press the pound/hash (#) button or the menu button next to the name. The menu buttons are only available on a six-line display endpoint.

Hidden Entries in the Voice Mail Directory

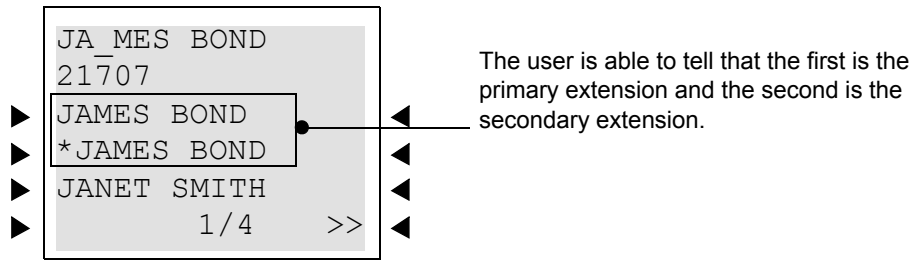
40.19 There is a Voice Mail feature that allows the administrator to program a description for a user in DB Programming but hides the description in the Voice Mail directory. For example, a user may have a primary extension (for example, at the office) programmed as "BOND, JAMES." However, the same user may have a secondary extension (for example, at home) that he does not want to be published from the Voice Mail directory.

40.20 To hide an entry in the Voice Mail directory:

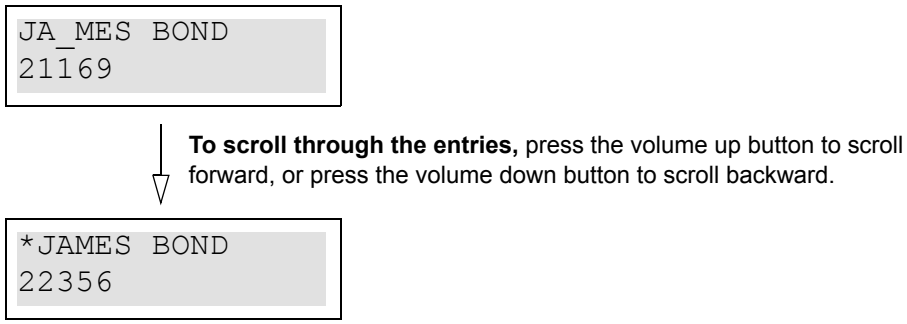
40.21 Enter "~BOND, JAMES" in the Description field under System\Devices and Feature Codes\Stations. The tilde character (~) makes the entry unaccessible in the Voice Mail directory. Note that if someone knew the number of the secondary extension, they could dial that extension directly. Someone entering "BOND" (2 6 6 3) at the Voice Mail prompt, however, would find only the primary extension.

40.22 The IC directory is intended for internal use, therefore, both entries will be in the IC directory. There may be any number of primary (no tilde in description) and secondary extensions (first character of description is a tilde). To differentiate between these two, an asterisk appears immediately before the entry as shown below. The following is an example of what an IDS user may see after pressing "JA" (5 2).

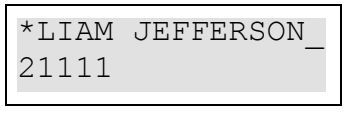
Six-Line Display Endpoint:



Two-Line Display Endpoint:

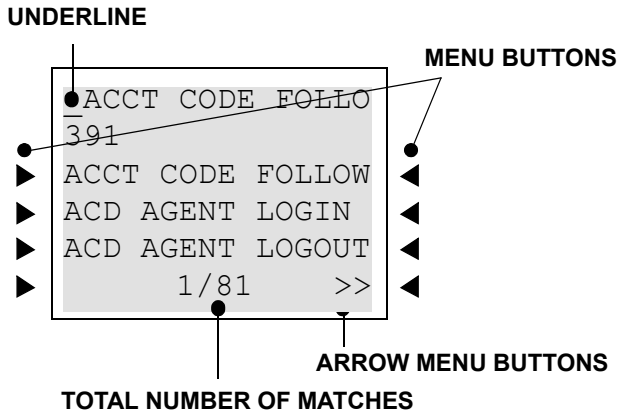


40.23 The asterisk remains in the entry even if the user enters digits beyond what the display can hold. In the following example, the username is "WILLIAM JEFFERSON," but the user has already entered "WILLIAMJEFFERSON" (9 4 5 5 4 2 6 5 3 3 3 3 7 7 6 6).

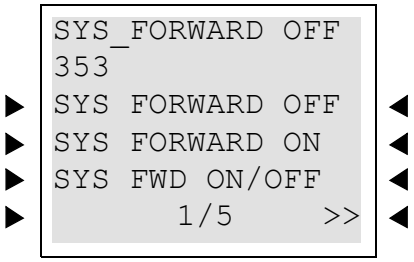


B. FEATURE DIRECTORY

40.24 The following are examples of the Feature directory displays. Upon accessing the Feature directory, the user will see the entire directory. Just like the IC directory, a six-line display endpoint shows three possible matches at a time. The entries on the first and third lines see the same extension. The menu buttons, arrow menu buttons, the total number of matches, underline, and dialpad buttons in the Feature directory function the same as the IC directory.

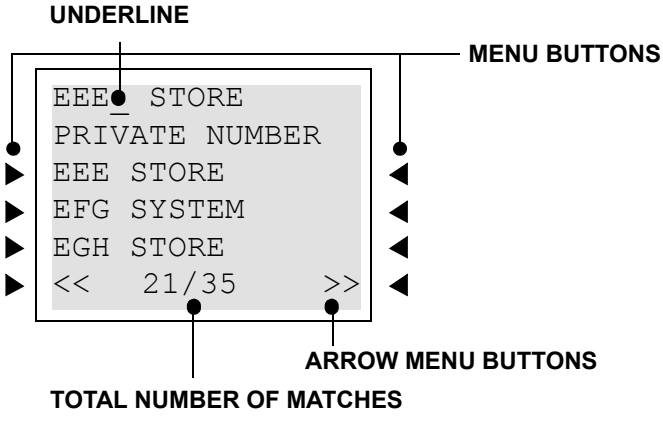


40.25 In the following example, the user entered "SYS" (7 9 7) to search one of the System Forward features.



C. OUTSIDE DIRECTORY

40.26 Outside directory also functions just like the Feature directory. Upon accessing the Outside directory, the user will see the entire directory. If the number is programmed as "Private" in DB Programming, that number appears as "PRIVATE NUMBER" in the display, as shown below.



41. HOUSE PHONE

41.1 This feature provides users with the ability to place a pre-designated intercom or outside call simply by lifting the handset (or pressing the Speaker button, if using a phone) on a designated house phone. In a network, the house phone can be programmed to dial an off-node device. There are several applications for this feature, such as:

- **Courtesy Paging Phone**: Visitors hear pages instructing them to pick up the house phone (such as the paging phones used in airport terminals). When they lift the handset, they are connected to a pre-programmed station user who can give them a message or connect them to a call.
- **Emergency Phone**: The house phone can be programmed to automatically dial the Emergency Call feature code, default is 911 [999]. This could save time in an emergency.
- **Service Phone**: Customers can use the house phone(s) to place orders or receive special services from the lobby. For example, the house phone would automatically dial the extension number of a service representative (or hunt group number of the service department).
- **Intercom Network**: House phones could be placed in specific locations throughout a building (such as examination rooms in a doctor's office) and could be programmed to call a specific station or group of stations (such as a nurses' station).

41.2 Any station may be designated as a house phone in DB Programming (see PROGRAMMING, page 6-219), however this feature is usually assigned to single-line sets. The primary attendant should not be a house phone.

41.3 The number dialed by the house phone is determined by the station's speed-dial programming. The number programmed through the database or in station speed-dial location 0 is automatically dialed during day mode, and the number in location 1 is dialed during night mode. This number can be either an extension number or an outside telephone number. If it is an outside number, it must be preceded with a trunk access code (and a pause if necessary).

41.4 After the house phone status has been programmed, the speed-dial number can only be changed while on hook (if it is a phone) or through individual station information (special purpose station) programming because lifting the handset will cause the station to dial the designated number.

41.5 Incoming calls take precedence over outgoing calls. If using a single-line set or a phone that is programmed for automatic trunk access (see page 5-158), any ringing call is automatically answered when the handset is lifted or the Speaker button is pressed.

41.6 There is a flag, called "House Phone Mode," that affects single-line house phones. The house phone can be programmed to return to dial tone after a call is disconnected (normal mode). This allows the user to enter a feature code or place a call. Or, the dial tone can be disabled (restricted mode) to prevent the user from performing any operation other than placing a house-phone call. In the default state, this is *normal.* See page 6-260 for programming information. The interaction of the House Phone Mode flag with the programmed speed dial number and system feature is shown below.

Table 5-41. *House Phone Mode*

Situation	Day/Night #	Normal Mode	Restricted Mode
House phone user lifts the handset	Complete extension	Dials the extension number	Dials the extension number
House phone user lifts the handset	Blank	User receives intercom dial tone	User receives reorder tone
House phone user lifts the handset	Incomplete extension number	System dials the partial number and waits for further digits	System dials partial number, then times out after Long Interdigit timer expires, and sends reorder tones
House phone is connected to ARS and the user performs a hookflash (recall)	Dials an outside number using ARS	System inserts a hookflash in the number and registers end-of-dialing	System restarts ARS by clearing the number and returns outside dial tone to the user
House phone is connected to an outside call and the user performs a hookflash (recall)	N/A	The call is placed on consultation hold and the user hears intercom dial tone	The trunk dials a hookflash (recall) and toll restriction is restarted
Party the house phone is connected to hangs up before the house phone user	N/A	User receives intercom dial tone and can use another feature code or dial a number	System redials the house phone digits after the SL Wait for Disconnect timer expires

42. REDIALING

42.1 The Redial feature can store one telephone number in redial memory at the station (manually dialed or speed-dialed numbers up to 32 digits). If the station user reaches a busy number, is disconnected, or if there is no answer, the number can be redialed easily. The station user simply lifts the handset and presses the REDIAL button or enters the Redial feature code. A trunk access code is automatically entered and the telephone number is redialed. (If redialing while still connected to an outside call, the connection is dropped and the trunk is reseized before the number is dialed.)

42.2 Only one telephone number can be stored in the station's redial memory at a time. This number can be stored in one of two ways, depending on station programming:

- **Last Number Saved**: The desired number is manually stored in redial memory by the station user. Dialing other numbers does not change the stored number. It only changes when a new number is stored.
- **Last Number Dialed**: The last number manually dialed or speed dialed is automatically stored. It changes every time the user dials a telephone number. (This is the default value of the Redial feature.)

42.3 The Last Number Saved and Last Number Dialed features work differently depending on the station status when it is used, as shown in the table below. Individual station programming determines the mode of the Redial feature (Last Number Dialed or Last Number Saved). See page 6-213 in PROGRAMMING for more information. The Redial feature code (380) performs the redial function (programmed under the REDIAL button).

42.4 Non-display system speed-dial numbers cannot be redialed at a display phone station.

Table 5-42. *Redial Feature*

Station Status When Redial Feature is Used	Station Has "Last Number Saved"	Station Has "Last Number Dialed"
Idle station	Saves the last trunk access code and telephone number that was dialed.	A trunk is selected using the same trunk access code as used to place last outside call, and the last telephone number is dialed.
Intercom dial tone (on or off hook)	Saves the last trunk access code and telephone number that was dialed.	A trunk is selected using the same trunk access code as used to place last outside call, and the last telephone number is dialed.
On an intercom call	Releases the current call, seizes a trunk using the saved trunk access code, and dials the saved telephone number.	Releases the current call, seizes a trunk using the access code used on the last outside call, and dials the last telephone number dialed.
After selecting a trunk but ***before*** dialing	Redials the saved telephone number on the currently selected trunk. (Note that the current trunk is used instead of the saved trunk access.)	Redials the last telephone number dialed, using the currently selected trunk.
After selecting a trunk ***and dialing*** one or more digits	Releases the current call, seizes a trunk using the saved trunk access code, and dials the saved telephone number.	Releases the current call, reseizes the trunk (using the access code used on that call), and redials the digits that were dialed.
On an incoming outside call	Releases the current call, seizes a trunk using the saved trunk access code, and dials the saved telephone number.	Releases the current trunk, seizes a trunk using the same trunk access code used on the last outgoing call, and dials the last number dialed.

A. USING THE LAST NUMBER SAVED FEATURE

42.5 Use the following procedures to save and redial numbers at stations that are programmed to use the Last Number ***Saved*** feature.

42.6 To save a number on a station programmed for last number saved:

NOTE: The saved number is replaced each time you repeat this procedure.

Inter-Tel Phone: While the phone is idle or while listening to intercom dial tone, press the **REDIAL** button, or press ∞ or SPCL and enter the Redial feature code (380). You hear a confirmation tone and the display shows LAST OUTSIDE NUMBER SAVED.

Single-Line Set: Lift the handset and enter the Redial feature code (380). When you hear the confirmation tone, hang up.

42.7 To redial a number (last number saved):

Phone: While on a call or after selecting a trunk, press the REDIAL button, or press ∞ or SPCL and enter the Redial feature code (380). See the chart on the previous page to determine which trunk is selected. The saved telephone number is dialed.

Single-Line Set: While on a call or after selecting a trunk, hookflash (press **RECALL**) and then enter the Redial feature code (380). See the chart on the previous page to determine which trunk is selected. The saved telephone number is dialed.

B. USING THE LAST NUMBER DIALED FEATURE

42.8 Use the following procedure to redial numbers at stations that are programmed to use the Last Number ***Dialed*** feature.

42.9 To redial a number (last number dialed):

Phone: While on or off hook, press the REDIAL button, or press ∞ or SPCL and enter the Redial feature code (380). See the chart on the previous page to determine which trunk is selected and which digits are dialed.

Single-Line Set: Lift the handset or hookflash (press **RECALL**). Then enter the Redial feature code (380). See the chart on the previous page to determine which trunk is selected and which digits are dialed.

43. REDIRECT CALLS

43.1 The Redirect Call Feature (feature code 331) allows the phone user to route ringing outside, intercom, and camped on calls to another station, hunt group, or outside number. Routing of the redirected call is still subject to trunk and toll restrictions. This feature provides these options in addition to the currently available options which allow the station user to redirect calls to Voice Mail or Do-Not-Disturb.

43.2 The Redirect Call feature does not require any software licenses. Some specific aspects of this feature are described below.

- If the user has an Executive Display, Professional Display, or Model 8560 Phone, the display will show a call menu option which reads SEND TO DEST. The user may choose this option or enter feature code 331. The display then prompts the user to enter the destination number directly or by scrolling through the IC directory.
- For non-display phones, calls are redirected using the feature code 331. This code is entered, followed by the desired destination number.
- If the phone receiving the redirected call is a display phone, it will show EXT XXXX FWD FROM EXT XXXX just as it does for a manually forwarded call.
- Calls may not be redirected back to the originating station. This will result in a message on the display saying CALL FORWARDED TO SOURCE. For non-display phones, the call will simply ring back at the station.
- Calls may not be redirected to restricted outside numbers, to stations in Do-Not-Disturb mode, or to stations that are forwarded. Redirected calls will not follow call forwarding.
- Agent Help Request calls, queue callbacks, and recalls cannot be redirected (these types of calls do not follow call forwarding).
- Calls cannot be redirected to another station's mailbox, because redirected calls will not follow forwarding.
- If the call is redirected to an unreachable destination, it will not be redirected and will ring back at the original station.
- If an ARS, trunk group, or trunk number is entered, the system will prompt the user to enter the destination telephone number. Calls may not be redirected to a node number, but may be directed to a station on another node by entering the correct extension number.
- If the applicable Forwarding timer expires before the user completes the redirection process, the system terminates the call.

44. PAGING

44.1 The Paging feature allows announcements to be made through phone internal speakers. Optional external paging equipment (amplifiers and paging speakers) may also be installed three ways:

- Using the external paging port on the CPC or CPU card (see INSTALLATION on page 3-242)
- Using internal speaker leads in the phone (see INSTALLATION on page 3-242)
- Using non-T1, analog loop start or ground start trunk circuits (see INSTALLATION on page 3-242).

44.2 There can be up to 10 paging zones. When the system is in the default state, all phones are assigned to paging zone 1 to provide an all-page zone. Phones, trunks, and the external paging port(s) can be assigned to any, all, or none of the paging zones, as desired.

NOTE: Placing a large number of phones in a paging zone may affect system performance. If system operation is affected when a page is placed to a particular page zone, remove some phones from that zone or change to external paging for the area served by that page zone.

44.3 In a network, all phones and trunks within a page zone must reside on the same node as the page zone. However, a page zone can contain external page ports on other nodes.

NOTE: Because the external paging ports have intercom numbers – 91000 - 91003 [99000 - 99003 in Europe], stations can place an intercom call to the external paging ports, instead of using the Page feature, to make a page over the external paging speakers.

44.4 Pages are not heard on phones that have been removed from paging (using the feature code as described below), are in do-not-disturb, are ringing, or are in use. Also, background music on phones and on external speakers connected to phone speaker leads is interrupted for pages. If a trunk in a paging zone is unplugged, the page cannot be completed.

44.5 If desired, phone users can program feature buttons to select the paging zones 0-9 or 0-49 using paging access codes 9600-9609 or 9600-9649 (respectively).

44.6 The Page timer limits the length of pages. If it is set to 0, pages are unlimited in length.

44.7 To make a page:

1. Lift the handset.
2. **Phone**: Press the PAGE button, or enter the Page feature code (7). The display shows SELECT PAGE ZONE # (0-9).

   Single-Line Set: Enter the Page feature code (7).
3. Enter the zone code (0-9 or 0-49) for the desired zone.
4. After the single progress tone, make your page before the Page timer expires. If you hear reorder tones and the display shows PAGE ZONE NOT PROGRAMMED, there are no stations in the selected page zone. If the display shows PAGE ZONE BUSY TRY AGAIN LATER, the necessary system resources are busy or the paging zone is being used by another station. Users cannot camp on to or queue the paging system.
5. Hang up.

45. REMOVE FROM PAGING

45.1 A phone user can prevent the phone from receiving pages or allow it to receive pages again using these feature codes. If the phone is assigned to more than one page zone, all zones are removed or replaced; the user cannot toggle individual zones.

45.2 To halt or enable zone pages:

1. While on or off hook, enter the Page Receive On/Off feature code (325) to prevent the phone from receiving pages. You hear a confirmation tone, and the display shows PAGE RECEIVE ON (or OFF).
2. Hang up if off hook.

46. DO-NOT-DISTURB

46.1 Placing a station in do-not-disturb halts all pages, incoming intercom calls, camped-on calls, and transferred calls to that station. Queue callbacks, recalls, and direct ring-in calls are not blocked. Another user calling the station while it is in do-not-disturb hears a repeating signal of four fast tones and a pause (display phones show the do-not-disturb message); the user cannot camp on, but can queue or leave a message at the station.

NOTE: Direct ring-in calls that are forwarded to a station in DND will ring the DND station per the rules of DND.

46.2 If desired, individual stations can be prevented from using do-not-disturb by disabling the do-not-disturb option in the database (see PROGRAMMING, page 6-211.)

46.3 If a hunt group station is in do-not-disturb, calls to the user's hunt group do not cause the phone to ring, but the individual trunk button will flash if all other stations in the hunt group are busy, forwarded, have hunt group remove enabled, or are in do-not-disturb. Hunt group announcement stations and overflow stations can use do-not-disturb to block hunt group calls by using do-not-disturb.

46.4 Phones may be given do-not-disturb override permission. These stations, when reaching a station in do-not-disturb, can enter the Do-Not-Disturb Override feature code (373) to place a non-handsfree intercom call to the station. Single-line stations cannot be enabled to use the Do-Not-Disturb Override feature.

46.5 When a station is placed in do-not-disturb, the user may select one of 20 system-stored messages that will appear on the top line of the display (unless do-not-disturb is enabled while the user is on a call or off hook, in which case message 1 is automatically selected). When a station in do-not-disturb is called by a display phone user, the caller sees the selected message.

46.6 Do-not-disturb messages can be programmed in the database or using an Administrator station. Default do-not-disturb messages are programmed as follows:

01 DO-NOT-DISTURB	11 OUT OF TOWN 'TIL
02 LEAVE A MESSAGE	12 OUT OF OFFICE
03 IN MEETING UNTIL	13 OUT UNTIL
04 IN MEETING	14 WITH A CLIENT
05 ON VACATION 'TIL [ON HOLIDAY 'TIL in Europe]	15 WITH A GUEST
06 ON VACATION [ON HOLIDAY in Europe]	16 UNAVAILABLE
07 CALL ME AT	17 IN CONFERENCE
08 AT THE DOCTOR	18 AWAY FROM DESK
09 ON A TRIP	19 GONE HOME
10 ON BREAK	20 OUT TO LUNCH

46.7 Do-not-disturb works as follows in a network setting:

- Each node has its own list of do-not-disturb messages that can be used only on that node.
- An intercom caller will see another user's do-not-disturb display when calling across nodes.
- The network allows DND override across nodes.

46.8 The system has default do-not-disturb messages in both the primary and secondary languages. However, messages that use Japanese characters can be reprogrammed only through an Administrator's station (see page 5-283).

46.9 The language of the messages seen by the station user, both when programming his or her station and when calling another station that is in do-not-disturb, is determined by the station's programmed language. That is, if a station is programmed for Japanese, the user will see only Japanese messages when programing a do-not-disturb message. If that station calls an American English-programmed station that has selected do-not-disturb message 02, the Japanese station's user will see the Japanese version of message 02.

NOTE: Only Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones can be used for viewing Japanese prompts. Other phones do not support Japanese displays.

46.10 The second line of the message can be customized with a numeric/alphanumeric message of up to 16 characters (English/Spanish and/or Japanese characters). The customized message is entered as described below. When entering a customized do-not-disturb message, you can press the MSG button or **USE ALPHA MODE/USE NUMERIC MODE** menu button to switch back and forth between alphanumeric and numeric mode.

- **In Numeric Mode**: The keypad buttons are used to enter numbers 0-9, the pound [hash] (#) button is used for entering a hyphen (-), and the asterisk (*) button is used for entering a colon (:). For example, 1*00 would enter "1:00" in numeric mode. When programming a customized do-not-disturb message, the station is automatically in numeric mode.

- **In Alphanumeric Mode**: The keypad buttons are used to enter the desired letters, numbers, and punctuation. The number of times a button is pressed determines which character is entered. For example, 33377744432999 would enter "FRIDAY." When adjoining characters are located under the same button, press the FWD button once to advance to the next character. For example, 6 FWD 666 FWD 6632999 would enter "MONDAY." See the chart below to program information in alphanumeric mode. (Note that letters correspond to the letters printed on the keypad buttons.)

BTN	Number of Times Button is Pressed										
	1	2	3	4	5	6	7	8	9	10	11
	English/Spanish Characters					Katakana Characters**					
1	-	&	(	)	1	A	I	U	E	O	a
2	A	B	C	'	2	KA	KI	KU	KE	KO	i
3	D	E	F	!	3	SA	SHI	SU	SE	SO	u
4	G	H	I	*	4	TA	CHI	TSU	TE	TO	e
5	J	K	L	# or /*	5	NA	NI	NU	NE	NO	o
6	M	N	O	Ñ or #*	6	HA	HI	FU	HE	HO	tsu
7	P	Q	R	S	7	MA	MI	MU	ME	MO	ya
8	T	U	V	?	8	YA	YU	YO	.	,	yu
9	W	X	Y	Z	9	RA	RI	RU	RE	RO	yo
0	@	:	.	,	0	WA	WO	N	pa	ba	long

* The character available depends on the software version.

** The Japanese characters are only available if the Multilingual feature is enabled and Japanese is installed as the secondary language.

46.11 When using either mode, phone users may use the speed-dial buttons and/or the REDIAL button to enter stored numbers or messages. Speed-dial numbers can be chained together when entering messages that require more that 16 keystrokes to create the desired message. When programming a message in speed-dial memory, you can use ∞ or SPCL to enter a hook-flash/recall (F) in place of the FWD button to advance or insert spaces, or to enter a pause (P) to change numeric/alphanumeric modes within the message. ∞ or SPCL can also be used to enter * or # in the speed-dial number.

46.12 Station users can also place their stations in do-not-disturb using the Remote Programming feature described on page 5-262.

46.13 To enable do-not-disturb mode:

1. **Phone**: While on hook, press the DND button or enter the Do-Not-Disturb feature code (370) or Do-Not-Disturb On/Off feature code (372). Display phones show SELECT DND MESSAGE # (01-20).

   Single-Line Set: Lift the handset and enter the Do-Not-Disturb feature code (370) or the Do-Not-Disturb On/Off feature code (372). You hear a progress tone.

2. To select message 01 (DO-NOT-DISTURB):

   Phone: Press the Speaker button, or lift and replace the handset. The DND button lights.

   Single-Line Set: Hang up.

   To select any of the 20 pre-programmed messages: Enter the desired number (01-20) or press the **SCROLL** menu button to begin scrolling. Display phone users may scroll through the messages by pressing the high end of the Volume button or the **NEXT** button to go forward or the low end of the Volume button or the **PREVIOUS** button to go backward (the display shows the selected message). If you enter an invalid message number, you will hear reorder tones, and the display shows INVALID DND MESSAGE NUMBER; try again.

3. **If desired, customize the second line of the message**: Use any combination of the following methods.

   - *Remain in numeric mode:* Press the keypad buttons to enter the desired number. Use the pound (hash) button (#) for a hyphen (-) and the asterisk button (*) for a colon (:). Phone users can press the FWD button once to leave a space, or press the MUTE button to backspace.

   - *Change to alphanumeric mode (phones only):* Press the MSG button (the button lights) or the **USE ALPHA MODE** menu button, then enter the desired characters. See the chart above. Phone users can press the FWD button once to advance or twice to leave a space, or press the MUTE button to backspace.

   - *Use speed-dial and/or redial numbers (phones only):* In either numeric or alphanumeric mode, you can speed dial a number (using the speed-dial button or feature code). In numeric mode you can press the REDIAL button to enter the stored characters. You may chain speed-dial and/or redial numbers together.

4. Terminate programming:

   Phone: Press the Speaker button or **ACCEPT**. Or, lift and replace the handset. The DND button lights. (Display phones show selected message. Date and time appear on the second line if there is no customized message.)

   Single-Line Set: Hang up.

46.14 To cancel do-not-disturb:

Phones with Menu Buttons: While on hook, press the lit DND button. The button goes off and the display shows DO NOT DISTURB OFF.

Other Phones: While on hook, press the lit DND button or enter the Cancel Do-Not-Disturb feature code (371) or Do-Not-Disturb On/Off feature code (372). The button goes off and the display shows DO NOT DISTURB OFF.

Single-Line Set: Lift the handset, enter the Cancel Do-Not-Disturb feature code (371) or Do-Not-Disturb On/Off feature code (372), and hang up.

46.15 To place your busy station in do-not-disturb while a call is camped on:

If, while on a call, you hear call waiting signals and you want to let the caller know you do not want to be disturbed:

Phone: Press the DND button or press ∞ or SPCL and then enter a Do-Not-Disturb feature code (370 or 372). The button lights. The display shows DO-NOT-DISTURB ON. The intercom caller is no longer camped on and receives do-not-disturb indications.

Single-Line Set: *(This procedure will interrupt your call in progress.)* Hookflash (press **RECALL**) and enter the Do-Not-Disturb feature code (370 or 372). The intercom caller is no longer camped on and receives do-not-disturb indications. Hookflash (press **RECALL**) twice to return to the current call.

46.16 Do-Not-Disturb Override

46.17 If your phone is enabled for do-not-disturb override and you hear do-not-disturb tones when placing an intercom call, you may use the following procedure to break through do-not-disturb and complete the call.

46.18 Do-not-disturb override (phones with this feature enabled only):

When you hear do-not-disturb tones while placing an intercom call, press ∞ or SPCL and enter the Do-Not-Disturb Override feature code (373). If the called station is idle, the call rings as a private intercom call. If the called station is busy, your call camps on.

47. REMOTE PROGRAMMING FEATURE

47.1 The Remote Programming feature (359) allows a user to forward a station's calls, change the station's DND status, and program the station's password from another station or through DISA. Each station has a password to limit access to this feature. (See page 5-118 for information about DISA.)

47.2 In a network, Remote Programming can be performed on one node for any phone that is programmed as an off-node device on that node (see page 6-198 for off-node device programming information). You must go through DISA for remote programming of any phones that are not programmed as off-node devices. The features that can be programmed remotely are as follows:

- Call Forward All Calls (feature code 355)
- Call Forward If No Answer (356)
- Call Forward If Busy (357)
- Call Forward If No Answer/Busy (358)
- Do-Not-Disturb (370)
- Do-Not-Disturb Cancel (371)
- Program Station Password (392)

A. STATION PASSWORD

47.3 The station password is used only for the Remote Programming feature.

NOTE: To prevent unauthorized use of the call forward feature, all stations using Remote Programming should have a password. To make the passwords difficult to guess, they should not match the extension number or consist of one digit repeated several times.

47.4 The station password can be up to eight digits in length. The default password is the extension number of the station. The password can be changed by entering the Program Station Password feature code at the station or when using the Remote Programming feature. It can also be programmed through Individual Station programming.

47.5 If the password is changed from a station, the user will be prompted for the old password, then the new password, and will be asked to verify the new password. If it is programmed through Remote Programming, the user will already have entered a correct password. So, the user is only prompted for the new password and will be asked to verify the password.

47.6 If the verified password and new password not match, the old password will be retained and the programming session will be canceled.

47.7 To change the station password from your station:

1. Enter the Program Station Password feature code (392). You hear a confirmation tone. If you have a display, it shows ENTER PASSWORD.
2. Enter your current password, followed by #. (At default, it is your extension number.) If you have a display, it shows NEW PASSWORD.

NOTE: *If you enter an incorrect password,* the display will show INVALID PASSWORD, and your station will return to the idle state.

3. Enter the new password followed by #. You hear a confirmation tone.
4. Enter the new password again for verification followed by #. You hear a confirmation tone. If you have a display, it shows PASSWORD SAVED. Your station returns to the idle state.

47.8 To change the station password using remote programming:

1. *If calling from an outside telephone,* access the system through a DISA line. Enter the DISA security code (if applicable). You hear a confirmation tone.

   If programming from another station, lift the handset or press the Speaker button.
2. Enter the Remote Programming feature code (359). You hear a confirmation tone. If you have a display, it shows ENTER EXTENSION. Menu button phone users can press the IC DIR button to use the intercom directory to look up a number.
3. Enter the extension to be programmed. You hear a confirmation tone. If you have a display, it shows ENTER PASSWORD. *If you entered an invalid extension number,* you will hear reorder tones and can try again.
4. Enter the extension's password followed by #. You hear a confirmation tone.

   If you enter an incorrect password, the display will show INVALID PASSWORD. Outside callers will be returned to dial tone, and stations will return to the idle state.
5. Enter the Program Station Password feature code (392). You hear a confirmation tone. If you have a display, it shows NEW PASSWORD.
6. Enter the new password, followed by #. You hear a confirmation tone.
7. Enter the new password again for verification, followed by #. You hear a confirmation tone. If you have a display, it shows PASSWORD SAVED. Outside callers will hear dial tone, and stations return to the idle state.

B. REMOTE DO-NOT-DISTURB PROGRAMMING

47.9 To place a station in do-not-disturb mode using the remote programming feature:

NOTE: After each entry, non-display phone users will hear a confirmation tone. Outside callers using a DISA line will hear confirmation tone followed by DISA dial tone. Display phone users will not hear confirmation tones but will see a prompt asking for the next entry.

1. *If calling from an outside telephone,* access the system through a DISA line. Enter the DISA security code (if applicable). You hear a confirmation tone followed by DISA dial tone.

   If programming from another station, lift the handset or press the Speaker button.

2. Enter the Remote Programming feature code (359). If you have a display, it shows ENTER EXTENSION. Menu button phone users can press the IC DIR button to use the intercom directory to look up a number.

3. Enter the extension to be placed in DND. If you have a display, it shows ENTER PASSWORD. *If you entered an invalid extension number,* you will hear reorder tones and can try again.

4. Enter the extension's password, followed by #. If you have a display, it shows ENTER FEATURE CODE.

   NOTE: *If you enter an incorrect password,* the display will show INVALID PASSWORD, and the call is disconnected. Station users will hear reorder tones.

5. Turn on Do-Not-Disturb:

   a. Enter the Do-Not-Disturb On feature code (370). (The On/Off feature code cannot be used in remote programming.) *If you entered an invalid feature code,* you will hear reorder tones and can try again.

      NOTE: *If the station you are programming is not allowed to use DND,* you will hear reorder tones, and the display will show CANNOT ACCESS RESERVED FEATURE. Outside callers will be returned to dial tone, and stations will return to the idle state.

   b. Enter the desired do-not-disturb message number (01-20). *If you entered an invalid message number,* you will hear reorder tones and can try again.

   c. Enter the optional second-line message text, if desired, using any combination of the following methods:

      Remain in numeric mode: Press the keypad buttons to enter the desired number. Use the pound [hash] button (#) for a hyphen (-) and the asterisk button (*) for a colon (:). Phone users can press the FWD button once to leave a space, or press the MUTE button to backspace.

      Change to alphanumeric mode (Inter-Tel phones only): Press the MSG button (the button lights) or the **USE ALPHA MODE** menu button, then enter the desired characters. (See the chart and instructions in your user guide.) Phone users can press the FWD button once to advance or twice to leave a space, or press the MUTE button to backspace.

Use speed-dial and/or redial numbers (Inter-Tel phones only): In either numeric or alphanumeric mode, you can speed dial a number (using the speed-dial button or feature code). In numeric mode you can press the REDIAL button to enter the stored characters. You may chain speed-dial and/or redial numbers together.

a. Hang up to complete the programming.

6. **To turn off Do-Not-Disturb:** Enter the Do-Not-Disturb Off feature code (371). (The On/Off feature code cannot be used in remote programming.) The call is automatically disconnected when the feature code is accepted. *If you entered an invalid feature code,* you will hear reorder tones and can try again.

C. REMOTE FORWARD PROGRAMMING

47.10 To forward a station using the remote programming feature:

NOTE: After each entry, non-display phone users will hear a confirmation tone. Outside callers using a DISA line will hear confirmation tone followed by DISA dial tone. Display phone users will not hear confirmation tones but will see a prompt asking for the next entry.

1. *If calling from an outside telephone,* access the system through a DISA line. Enter the DISA security code (if applicable). You hear a confirmation tone followed by DISA dial tone.

   If programming from another station, lift the handset or press the Speaker button.

2. Enter the Remote Programming feature code (359). If you have a display, it shows ENTER EXTENSION. Menu button phone users can press the IC DIR button to use the intercom directory to look up a number.

3. Enter the extension to be forwarded. If you have a display, it shows ENTER PASSWORD. *If you entered an invalid extension number,* you will hear reorder tones and can try again.

4. Enter the extension's password, followed by #. If you have a display, it shows ENTER FEATURE CODE.

   NOTE: *If you enter an incorrect password,* the display will show INVALID PASSWORD and the call is disconnected. Station users will hear reorder tones.

5. To turn on Call Forward:

   a. Enter one of the following Call Forward feature codes. *If you entered an invalid feature code,* you will hear reorder tones and can try again.

      Call Forward All Calls355
      Call Forward If No Answer356
      Call Forward If Busy 357
      Call Forward If No Answer/Busy358

   a. Enter the forwarding destination. This can be an extension number or a trunk access code followed by an outside telephone number.

      If the forward destination is an extension number, programming is complete, and the call is disconnected automatically.

      If the forward destination is an outside number, hang up to complete the programming.

If the station is not permitted to forward to the destination entered, you will hear reorder tones and can try again.

If you hang up before entering a forward destination, the programmed forward is canceled.

6. **To turn off Call Forward:** Enter 355 and then hang up instead of entering a destination. You hear a confirmation tone.

48. DEFAULT STATION

48.1 A station user can enter one feature code that will return the following features to their default states. Returning to default:

- Cancels the following features, if in effect:
  - — Do-Not-Disturb
  - — Manual Call Forwarding
  - — Queue Request
  - — Account Code For All Calls Following
  - — Background Music
  - — Ring Intercom Always
  - — Headset Mode
- Restores the following features, if disabled:
  - — Handsfree
  - — Page Receive
  - — Hunt Group Replace
  - — System Forwarding
- Returns phone volumes to default levels
- Returns phone to standard keymap
- If the phone is a member of one or more ACD hunt groups, logs into the hunt group(s)
- Does not affect PCDPM bit rate

48.2 This feature is especially useful for installers and troubleshooters who need to know exactly how a given station is programmed.

48.3 To return the station to default:

Phone: While on or off hook, enter the Default Station feature code (394). You hear a confirmation tone and the display shows STATION DEFAULTED. *If off hook,* hang up.

Single-Line Set: Lift the handset and enter the Default Station feature code (394). You hear a confirmation tone followed by dial tone. Hang up.

49. HOOKFLASH [RECALL IN EUROPE]

49.1 A timed hookflash (recall) may be required for phone and single-line users to use certain telephone company or PBX features. The CO Hookflash (Recall) feature code sends a timed hookflash (recall) over the trunk when entered. A hookflash (recall) will restart the call cost display and toll restriction, plus it will start a new line in the SMDR printout. However, the call will remain under the same CALL button.

49.2 The Hookflash (Recall) feature can be enabled or disabled on a trunk group-by-trunk group basis. It can be used on any outside call, including conference calls.

49.3 The duration of the hookflash (recall) can be changed by reprogramming the LS/GS CO Hookflash timer (default value is 0.6 seconds) or LS CO Hookflash in Europe (default value is 0.1 seconds). If the system is not installed behind a PBX, the Hookflash (Recall) timer must be programmed to be compatible with the system's trunks.

49.4 To send a timed hookflash (recall) while on a trunk:

Phone: Press ∞ or SPCL and enter the CO Hookflash feature code (330).

Single-Line Set: Hookflash (press **RECALL**) and then enter the CO Hookflash feature code (330).

50. REMINDER MESSAGES

NOTE: Reminder messages are available on Inter-Tel proprietary phones only.

50.1 Reminder messages are set, like an alarm clock, to signal a phone at a specified time. The user can select the message and time up to 24 hours in advance. Twenty different reminder messages are available. These messages can be reprogrammed by the installer or using an Administrator's station.

50.2 At the programmed time, the reminder message signals the station with eight short tones. A display phone shows the message until it is canceled; a non-display phone receives tones only. If the station is busy, the user still hears the tones and the message displays for 10 seconds during the call, then the display returns after the user hangs up. (Reminder displays interrupt, but do not affect, programming.)

50.3 Messages can be changed by the installer or Administrators station, if desired. They are limited to 16 characters. The default messages are:

01 MEETING	11 CALL ENGINEERING
02 STAFF MEETING	12 CALL MARKETING
03 SALES MEETING	13 CALL ACCOUNTING
04 CANCEL MEETING	14 CANCEL DND
05 APPOINTMENT	15 CANCEL CALL FWD
06 PLACE CALL	16 TAKE MEDICATION
07 CALL CLIENT	17 MAKE RESERVATION
08 CALL CUSTOMER	18 REVIEW SCHEDULE
09 CALL HOME	19 LUNCH
10 CALL CORPORATE	20 REMINDER

50.4 In a network, each node has its own list of reminder messages that can be used only on that node.

50.5 The system has default reminder messages in both the primary and secondary languages. However, messages using Japanese characters can be reprogrammed only through an Administrator's station (see page 5-283).

50.6 The language of the messages seen by the station user is determined by the station's programmed language. That is, if a station is programmed for Japanese, the user will see only Japanese messages when programing a reminder message.

NOTE: Only Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, and Model 8560 Phones can be used for viewing Japanese prompts. Other phones do not support Japanese displays.

50.7 To request reminder messages:

NOTE: Lift and replace the handset to stop the process without selecting a message.

1. While on hook, enter the Reminder Message feature code (305). Display phones show SELECT REMINDER MSG # (01-20).
2. **To select a specific message:** Enter the two-digit message code (01-20). Display phones show the selected message and SCROLL OR ACCEPT.

   To scroll through the messages: View the available messages by pressing the high end of the Volume button or the **NEXT** button to scroll forward or pressing the low end of the Volume button or the **PREVIOUS** button to scroll backward. Each message displays for 0.5 seconds before another can be selected. When the desired message appeared, press **#** or **ACCEPT**.
3. Enter the time that you want to receive the message. (If using a station programmed for Japanese, these prompts will be reversed, and you will set the AM/PM before the hour and minutes.)
   a. Enter the hour and minutes using four digits (e.g., 0900 for 9:00).
   b. *If the system is set for 12-hour display format,* the display shows SELECT AM OR PM. Press **1** or the **AM** button for AM or press **2** or the **PM** button for PM. You hear a progress tone when it is accepted, and the display momentarily shows the selected reminder message and time.

      NOTE: If an invalid time is entered, you hear reorder tones and must enter the time again. Display phones show INVALID TIME.

50.8 To receive a reminder message:

1. At the selected time, you hear eight short tones and the message displays. Do one of the following:

   If your station is idle, go to the next step.

   If you are on a call, the message will display for 10 seconds. After you hang up, the reminder message returns and must be cleared.

   If you are programming your phone, the programming function is interrupted. The message will display, but will not interrupt programming; you can continue. After programming is completed, the reminder message will return and must be cleared.
2. **To clear the message:** Remain on hook and press *****. The display shows EXPIRED REMINDER MESSAGE CLEARED.

50.9 To cancel all of your reminder message requests:

While on hook, enter the Cancel Reminder Message feature code (306). The display shows REMINDER MSGS CANCELED. You cannot view or cancel individual messages.

51. OPTIONAL DATA DEVICE ATTACHMENTS

51.1 A customer-provided, modem-equipped data device (such as a data terminal or a personal computer) may be attached to a phone that has an optional data port module installed (see page 2-76 for more information). Special programming is required to enable the phone circuit to support the data port function. (See page 5-146 for information about Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone data port modules. Also see page 6-226 in PROGRAMMING for more details.)

51.2 The data port module circuit has an individual extension number and complete station programming setup including its own trunk access and toll restriction information. In a network setting, a station's data port extension can reside on another node.

51.3 Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone users can adjust the bit rate (bps) of the PCDPM connected to the phone. The rate can be set to 300, 600, 1200, 2400, or 4800 (1200, 2400, or 4800 if it is a system serial port). To adjust the bit rate of your PCDPM, see the instructions on page 5-146.

51.4 The circuit with the data port module can be programmed to be associated with one or more other phones. The associated phone(s) should have a DATA button that will show the status of the data device and will allow easy transfers between the phone and the data device.

51.5 Data calls are transmitted over the data port voice path. This leaves the attached phone's voice path available for normal phone operation. However, receiving off-hook voice announce (OHVA) calls at a Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone attached to an MDPM is not possible during a data call since the data call is using the secondary voice path.

51.6 Calls can be transferred back and forth between an associated phone's primary voice path and data port as often as needed provided that the data port is on hook at the time of the transfer attempt. For example, a call can be placed using the data device, then transferred to the phone's primary voice path to announce the call, and then transferred back to the data device so that it can communicate with the called device. (Note that transferring to the phone's primary voice path disconnects the data device from the call. During that time, the call cost display for the data call is shown on the phone.)

51.7 Operation of the data device begins with the initiation of a call in one of four ways:

- The user may initiate a call using the phone, while on or off hook, and then transfer it to the data device.
- The user may lift the handset to answer a ringing call and then transfer it to the data device.
- The user may initiate a call using the data device, without touching the phone.
- The data device may receive a call directly, without using the phone.

A. INITIATING OR ANSWERING A CALL USING THE PHONE

51.8 The phone user can place or receive a data call using the phone and then transfer it to the data device. It can be an intercom or outside call, and the phone can be on or off hook. The call is placed to or received from another modem-equipped data device.

51.9 After the connection is established, the phone user presses the DATA button, or presses ∞ or SPCL and enters the Data feature code (340) to transfer the call to the data device. If the data device does not answer before the Transfer Available timer expires, the call recalls the phone station.

51.10 Transferring a call to the data device using the DATA button places the call without providing ring voltage to the data device. The user must manually cause the data device to answer the call. (Note that if the phone user wants to cause ring voltage to be applied to the data device, the Transfer feature must be used to send the call to the data device.)

51.11 While the user is establishing the call using the phone, the DATA button is unlit. From the time that the user presses the DATA button, until the data device answers, the DATA button will flash. While the data device is on an active call, the DATA button flutters on the associated phone(s) (indicating that the data call may be retrieved by the phone). If the phone is off hook, the user will hear dial tone.

51.12 To place a data call using the phone:

1. Place or receive an intercom or outside call.
2. When you hear modem tone, press the DATA button, or press ∞ or SPCL and enter the Data feature code (340). The DATA button flashes slowly.
3. Hang up or press the Speaker button. The DATA button begins to flutter when the data device answers, and the display returns to date and time. The phone can now be used as usual.
4. Operate the data device according to the manufacturer's instructions. The device must be able to come off hook and answer the incoming call.

   If you want to return the active data call to your primary voice path, press the fluttering DATA button. The call cost and elapsed time of the data call are displayed.

B. INITIATING A CALL USING THE DATA DEVICE

51.13 The data device can be used for initiating a data call, without the use of a phone. The user causes the modem to select a trunk or intercom channel and dial the desired number.

51.14 The data device can dial an extension number or a trunk access code followed by a pause and then a telephone number. As soon as the data device comes off hook, the DATA button is lit at the associated phone(s) to indicate that the data port is busy. The button remains lit until the data device finishes dialing (outside call) or the station user answers (intercom call). This protects the call during dialing to ensure that the dialing is not interrupted. A phone user who attempts to transfer another data call to the data device or pick up the data call while the button is lit will hear a burst of reorder tone.

51.15 When a data device places an intercom call, the call is placed in the non-handsfree mode. (This can be changed in DB Programming, if desired, by disabling the Ring Intercom Always flag. See page 6-213 in PROGRAMMING.) The incoming intercom call to the destination station from the data device appears as a normal intercom call with the usual displays and tones.

51.16 If an extension number is dialed and that station is in do-not-disturb, the user will hear do-not-disturb tone through the data device's speaker (if it has one). Whenever continuous busy, reorder, or do-not-disturb tone is sent by the system, the Inactivity Alarm timer is registered. The DATA button is lit while the tones are present. If the timer expires before the data device hangs up, off-hook alarm tone is sent, and the DATA button flutters fast until the device hangs up. Note that once the DATA button begins fluttering, the voice path on the phone attached to the data port module is released and can be used for other call processing applications. (Note that the absence of a lamp, or of a DATA button, will not interfere with these timing parameters, but the user will not have a visual indication of them.)

51.17 If the data port places an outside call, the system performs toll restriction using the data port's class-of-service. If an illegal extension number or access code is dialed, the user will hear reorder tone through the data device's speaker. (Call cost information for a data call appears on an associated phone only when the data call is connected to a phone's primary voice path.)

51.18 If an attempt is made to initiate a data call from the data device when the required voice path is not available (the secondary voice path of the attached phone is busy), the data device will not receive dial tone. The activity on the phone's secondary voice path continues uninterrupted.

51.19 To place a data call using the data device:

1. Following the procedures of your data device's communications software, instruct the device to come off hook and dial the extension number, or a trunk access code and the telephone number, of the data device to be called. The call will be processed and will be automatically connected when answered. The DATA button is lit until dialing is completed, then it flutters.
2. Operate the data device according to the manufacturer's instructions.

C. ANSWERING A CALL USING THE DATA DEVICE

51.20 The data device can be called directly (intercom or outside calls) and receive transferred calls via its extension number.

51.21 When a call is placed directly or transferred to the device's intercom number, the data device will receive ring signal. If the data device does not have auto-answer capability, the user must manually answer the call using the data device. (Transferring a call to the data device using the DATA button places the call without providing ring voltage to the data device. The user must manually cause the data device to answer the call.)

51.22 While ring signal is being sent to the data device, the DATA button flashes. As soon as the data device answers, the DATA button flutters. The button continues to flutter for the duration of the call, indicating that the data path is in use but may be picked up by associated phones.

51.23 To answer a data call using the data device:

1. When the data device receives a call, the DATA button flashes on the associated phone(s). If the data device does not have an auto-answer modem, follow the procedures for the data device's communications software and instruct the device to answer the incoming call. The DATA button flutters while the call is in progress.
2. Operate the data device according to the manufacturer's instructions.

D. TERMINATING A DATA CALL

51.24 Termination of a data call may be accomplished in one of three ways:

- Your data device can hang up. This causes the trunk or intercom channel to be released, terminating the call to the other data device.
- The other data device may hang up. When the call is disconnected, the DATA button is unlit. Depending on the type of device used, the data device may detect the loss of carrier and automatically terminate the entire connection. If the data device does not have this capability, you must manually cause the data device to hang up.
- The data call can be transferred to an associated phone's primary voice path (releasing the data device) and then disconnected by the phone. Note however, that the DATA button will flutter rapidly if the data device does not hang up before the Inactivity timer expires.

51.25 To terminate data device connection:

To have the data device disconnect the call: Use the appropriate commands for the data device to instruct it to hang up. When the call is disconnected, the DATA button is unlit.

To transfer the data call back to an associated phone before disconnecting the call: While on or off hook:

a. Press the fluttering DATA button. The DATA button goes off. *If desired,* speak to the party at the site of the remote data device. This can only be done if the modem at the site of the remote data device can be turned off without disconnecting the trunk.

b. **To disconnect:** *If off hook*, hang up. *If on hook,* press the Speaker button. The DATA button will flutter rapidly until the data device hangs up.

52. ADMINISTRATOR FEATURES

52.1 Any display phone station (attendant or non-attendant) can be assigned as a system administrator. Administrator stations provide the following services:

- Place the local node or other nodes in the network in night mode or day mode
- Set the date and time of the local node
- Set the network date and time and re-synchronize clocks in the network.
- Make database changes (see page 5-276 for a list of programming areas)
- Program system speed-dial numbers on the local node
- Receive and clear displayed system and network alarms
- Use diagnostic mode features:
  - — Freeze and unfreeze the database history for the local node or any node in the network using programmed freeze zones
  - — Print error logs
  - — Seize specific devices for troubleshooting purposes

52.2 Any phone station can be programmed to be an Administrator station by the database programmer or by another Administrator station.

52.3 If a non-Administrator station user attempts to use the Administrator features, the user will hear reorder tones, and the display will show CANNOT ACCESS RESERVED FEATURE.

A. PLACING THE SYSTEM IN NIGHT MODE

52.4 An Administrator station can place the local node in day or night mode. The day/night mode determines which lists the system will use for trunk access, toll restriction, etc.

52.5 Night mode also affects the night transfer relays on the Options Card (OPC). The relays are activated when the system is placed in night mode. See page 2-67 in SPECIFICATIONS for details.

52.6 To turn night mode on or off:

While on hook, enter the Night Ring On/Off feature code (9860). You hear a single confirmation tone. The display shows NIGHT MODE IS ON (or OFF). Then, if night mode was turned on, the display shows THE SYSTEM IS IN NIGHT MODE until day mode is turned on.

B. PLACING NODES IN DAY OR NIGHT MODE

52.7 An Administrator station can place one or more nodes in day or night mode. The day/night mode determines which lists the system will use for trunk access, toll restriction, etc.

52.8 The network determines the day/night mode status of a call based on the day/night mode status of the node where the trunk resides.

52.9 To turn night mode on:

1. While on hook, enter the Enable Network Night feature code (9861).
2. You are prompted to enter a node number. Enter the desired node number. (For individual nodes, enter a number between 01 and 63. For all nodes, enter 00.)
3. You hear a single confirmation tone. The display shows NIGHT MODE IS ON. Then the display shows NODE X IS IN NIGHT MODE until day mode is turned on.

52.10 To turn day mode on:

1. While on hook, enter the Enable Network Day feature code (9862).
2. You are prompted to enter a node number. Enter the desired node number.
3. You hear a single confirmation tone. The display shows DAY MODE IS ON.

C. SETTING SYSTEM DATE AND TIME

52.11 Occasionally, the system time or date needs to be reset (for example, when the system is defaulted or for daylight-saving time). Any Administrator can change the date and time that appears on all display phones and in the SMDR reports. The day of the week is automatically calculated and set by the system when the date is entered.

52.12 To set the system date and time:

NOTE: If you make a mistake, press the MUTE button to backspace or press * or **CANCEL** to leave it unchanged and start over.

1. While on hook, enter the Set Date/Time feature code (9800). Your display shows DATE *(current date)*.

   If you do not need to change the date, press # or **ACCEPT** to skip to the TIME XX:XX prompt.

2. Use the keypad buttons to enter the month, day, and year. For example, press 020305 for February 3, 2005. When finished, the display shows TIME *(current time)*. If you entered the date incorrectly, the display shows INVALID DATE and you are prompted to enter a new date.

   NOTE: If using a station programmed for Japanese, enter the date as year, month, date. For example, 050203 for February 3, 2005.

   If you do not need to change the time, press # or **ACCEPT** twice to exit. The display shows SYSTEM DATE AND TIME UPDATED.

3. Use the keypad buttons to enter the time in hours and minutes. (For example, enter 0900 for 9:00.) If you entered the time incorrectly, the display shows INVALID TIME and you are prompted to enter a new time.

4. *If the system is set for 12-hour display format,* the display shows SELECT AM OR PM (AM=1 PM=2). Press 1 (or the **AM** menu button) for AM or press 2 (or the **PM** menu button) for PM. The display shows SYSTEM DATE AND TIME UPDATED. If you press any button other than 1 or 2, the display shows INVALID TIME and you are prompted to enter a new time.

   NOTE: If using a station programmed for Japanese, the prompts will be reversed, and you will set the AM/PM before the hour and minutes.

D. SETTING NETWORK DATE AND TIME

52.13 Occasionally, the network time or date needs to be reset (for example, when the system is defaulted or for daylight-saving time). Any Administrator can change the date and time that appears on all display phones and in the SMDR reports in the network. The day of the week is automatically calculated and set by the system when the date is entered.

52.14 To set the network date and time:

NOTE: If you make a mistake, press the MUTE button to backspace or press * or **CANCEL** to leave it unchanged and start over.

1. While on hook, enter the Set Network Date and Time feature code (9810). Your display shows DATE *(current date)*.

   If you do not need to change the date, press # or **ACCEPT** to skip to the TIME XX:XX prompt.

2. Use the keypad buttons to enter the month, day, and year. For example, press 020305 for February 3, 2005. When finished, the display shows TIME *(current time)*. If you entered the date incorrectly, the display shows INVALID DATE and you are prompted to enter a new date.

   NOTE: If using a station programmed for Japanese, enter the date as year, month, date. For example, 050203 for February 3, 2005.

   If you entered the date incorrectly, the display shows INVALID DATE and you are prompted to enter a new date.

   If you do not need to change the time, press # or **ACCEPT** twice to exit. The display shows SYSTEM DATE AND TIME UPDATED.

3. Use the keypad buttons to enter the time in hours and minutes. (For example, enter 0900 for 9:00.)

   NOTE: If using a station programmed for Japanese, the prompts will be reversed and you will set the AM/PM before the hour and minutes.

   If you entered the time incorrectly, the display shows INVALID TIME and you are prompted to enter a new time.

4. *If the node is set for 12-hour display format,* the display shows SELECT AM OR PM (AM=1 PM=2). Press 1 (or the **AM** menu button) for AM or press 2 (or the **PM** menu button) for PM. The display shows SYSTEM DATE AND TIME UPDATED. If you press any button other than 1 or 2, the display shows INVALID TIME and you are prompted to enter a new time.

E. SYNCHRONIZE NETWORK TIME

52.15 Administrators can synchronize the minutes past the hour across the network without changing the hour. This is useful when the nodes are in different time zones.

52.16 If the node is an ATM-based system, the individual chassis-pairs cannot span different time zones because there is no way to synchronize the time of the dual chassis connected to the ATM Switch.

NOTE: If a node's time is off by more than 30 minutes, synchronizing the minutes may cause the hour to change. Also, network time is automatically synchronized every day at 12:30 AM (00:30), using the time setting on the node with the lowest number.

52.17 An Administrator can synchronize the clocks in all nodes in the network using the following procedure.

52.18 To synchronize network time:

While on hook, enter the Synchronize Network Time feature code (9811). You hear a confirmation tone, and the display shows NETWORK TIME SYNCHRONIZED.

F. DB PROGRAMMING

52.19 Any Administrator station can perform DB Programming using the phone. However, it requires a ***display*** phone, and an Executive Display, Professional Display, or Model 8560 Phone is strongly recommended.

NOTE: If using an analog phone, the Administrator will need to use the **PREVIOUS/NEXT** or **UP/DOWN** buttons in place of the Volume button in the instructions in this section.

52.20 The database areas that can be programmed by an Administrator station include the following:

52.21 Station Programming:

- Create or delete Administrator stations
- Create or delete Attendant stations
- Create or delete House Phones
- Assign stations to Attendants
- Program usernames
- Program station toll restriction

52.22 System Option Programming:

- Program do-not-disturb messages
- Program a password for the DB Programming feature
- Program station extensions
- Swap extensions

52.23 Trunk Programming:

- Individual trunk answer supervision type, caller information, hybrid balance, signaling type (DTMF or pulse), and trunk group assignment
- Trunk group answer access, ring-in, toll restrictions, and trunk lists

52.24 Entry to the DB Programming feature at the Administrator stations can be protected using a password. A password would prevent unauthorized users from altering the system database.

NOTE: Passwords are very important to system security. Without sufficient password protection, the system database is vulnerable to unauthorized access.

52.25 Depending on the database changes made by the Administrator, the system may require a reset after programming. If so, the system will prompt the Administrator for a reset and ask if it should be done immediately or delayed. Delaying the reset would prevent interruption in service. However, if a reset is required, it should be done as soon as possible to permit proper system operation.

NOTE: A system reset will drop all calls in progress.

52.26 When entering alphanumeric information, such as a username, reminder message, or do-not-disturb message, press the MSG button or **USE ALPHA MODE/USE NUMERIC MODE** menu button to switch back and forth between alphanumeric and numeric mode.

- **In Numeric Mode**: The keypad buttons are used to enter numbers 0-9, the pound [hash] (#) button is used for entering a hyphen (-), and the asterisk (*) button is used for entering a colon (:). For example, 1*00 would enter "1:00" in numeric mode.
- **In Alphanumeric Mode**: The keypad buttons are used to enter the desired letters, numbers, and punctuation. The number of times a button is pressed determines which character is entered. For example, 533266 would enter "JEAN" in English. When adjoining characters are located under the same button, press the FWD button to advance to the next character. For example, 66 FWD 6667776 would enter "NORM." See the chart below to program information in alphanumeric mode. (Note that letters correspond to the letters printed on the keypad buttons.) The Japanese characters will be available only if the Multilingual Support premium feature is licensed and the Administrator's Executive Display, Professional Display, Standard Display, Associate Display, Model 8520, or Model 8560 Phone is set for Japanese.

BTN	Number of Times Button is Pressed										
	1	2	3	4	5	6	7	8	9	10	11
	English/Spanish Characters					Katakana Characters**					
1	-	&	(	)	1	A	I	U	E	O	a
2	A	B	C	'	2	KA	KI	KU	KE	KO	i
3	D	E	F	!	3	SA	SHI	SU	SE	SO	u
4	G	H	I	*	4	TA	CHI	TSU	TE	TO	e
5	J	K	L	# or /*	5	NA	NI	NU	NE	NO	o
6	M	N	O	Ñ or #*	6	HA	HI	FU	HE	HO	tsu
7	P	Q	R	S	7	MA	MI	MU	ME	MO	ya
8	T	U	V	?	8	YA	YU	YO	.	,	yu
9	W	X	Y	Z	9	RA	RI	RU	RE	RO	yo
0	@	:	.	,	0	WA	WO	N	pa	ba	long

* The character available depends on the software version.

** The Japanese characters are available only if the Multilingual feature is enabled and Japanese is installed as the secondary language.

Station Programming Using an Administrator's Station

52.27 An Administrator station can be used for programming the following station information:

- **Create or delete Administrator stations**: An Administrator can program any other phone station to be an additional Administrator station or delete Administrators. (You cannot program this for your own station or a single-line station.)
- **Create or delete Attendant stations**: An Administrator can program any station to be an Attendant station or delete Attendants.
- **Create or delete House Phones**: An Administrator can program any station to be a House Phone, or delete House Phones.
- **Assign stations to Attendants**: An Administrator can change the assigned Attendant for each station.
- **Program usernames**: The Administrator can program or change the username for any station.
- **Program station toll restrictions**: The Administrator can program toll restriction classes of service for the stations. Station toll restrictions are described in detail on page 5-103.

52.28 The Station DB Programming process is summarized in the flowchart shown on page 5-282. Program planning sheets are included on the system software CD-ROM Documentation folder. You can also find all documentation on the Inter-Tel *edGe* Web site. For readers located in Europe, contact the Inter-Tel Europe office for technical documentation.

NOTE: If necessary, you can press the asterisk (✳) or Speaker button to cancel programming and discard any unsaved changes at any time during the following procedure.

52.29 To program stations:

1. While on hook, enter the Program Database feature code (9932 or 9132).
2. *If a password is required,* the display shows ENTER PASSWORD. Use the keypad to enter your 1-8 digit password and press #. (If you enter an incorrect password or do not press #, the display shows INVALID PASSWORD.)

   If a password is not required, skip this step.
3. The display shows ENTER DATABASE OPTION. (Executive Display, Professional Display, and Model 8560 Phones also show the options: STATION, SYSTEM, and TRUNK.) Press 1 or the **STATION** menu button.
4. The display shows ENTER STATION EXTENSION. Enter the extension number of the station to be programmed using one of the following methods. (If you enter an invalid extension number, you hear reorder tones and must try again.)

   Enter a complete number: Enter the extension number using your keypad. When the circuit information is displayed, press # again to continue.

   Enter a partial number: Enter a partial extension number, then press the # button, high end of the Volume button, or the **ACCEPT** menu button. The display shows the extension number, username and circuit number of the station that most closely matches the partial extension number. Press # or **ACCEPT** to program the displayed station, or scroll to another station as described below.

Scroll through the numbers: To scroll through the extension number list, press the high end of the Volume button or the **NEXT** button to scroll forward, or press the low end of the Volume button or the **PREVIOUS** button to scroll backward. When the desired extension number is displayed, press # or **ACCEPT** to continue.

5. The display shows ENTER STATION OPTION. Select one of the following:

   a. *Station Flags:* Allows you to set the Administrator Station, Attendant, and House Phone flags. To select it, press 1 or the **STATION FLAGS** menu button. There are three flags that can be programmed: Administrator, Attendant, and House Phone. (However, you cannot program the Administrator flag for your own station or for a single-line station.) To program the flags, do the following:

      1.) *To scroll to the desired flag:* Press the high end of the Volume button or the **NEXT** button to scroll forward, or press the low end of the Volume button or the **PREVIOUS** button to scroll backward.

      2.) *To enable or disable a displayed flag:* Press 1 or the **ON** menu button to enable the flag. Or, press 2 or the **OFF** menu button to disable the flag.

      3.) *To save your programming when all flags are set correctly:* Press # or **ACCEPT** to save the new flag settings. The display shows DATABASE UPDATED and then returns to the ENTER STATION OPTION prompt.

      4.) *To exit without saving your changes:* Press * or **CANCEL** to exit. The display shows NO UPDATE PERFORMED and then returns to the ENTER STATION OPTION prompt.

   b. *Station Information:* Allows you to select an attendant for the station or program the station's username. To select it, press 2 or the **STATION INFO** menu button. The display shows ENTER STATION INFO OPTION. Select one of the following:

      1.) *Attendant:* To assign an attendant to serve this station, press 1 or the **ATTENDANT** menu button. The display shows ENTER ATTENDANT EXTENSION. Enter the desired extension number. When the circuit information is displayed, press # again to return to the ENTER STATION INFO OPTION prompt. The display shows DATABASE UPDATED. (Or, to cancel your entry, press * or CANCEL. The display shows NO UPDATE PERFORMED.)

      2.) *Username:* To change the username of the station, press 2 or the **USERNAME** menu button. The display shows ENTER USERNAME. Enter the new name as described in paragraph 52.26 on page 5-277. (Or, to cancel your entry, press * or **CANCEL**. The display shows NO UPDATE PERFORMED.)

c. *Toll Restriction:* Allows you to set the station class of service for day and/or night modes. To select it, press 3 or the **TOLL RESTRICTION** menu button. Then do the following:

1.) The display shows TOLL RESTRICTION OPTION. Press 1 or the **COS DAY** menu button to program day mode toll restriction. Or, press 2 or the **COS NIGHT** menu button to program night mode toll restriction.

The display shows SET DAY (or NIGHT) COS XX. If this is not the COS you want to program, scroll to the correct COS by pressing the high end of the Volume button or **NEXT** menu button to scroll forward, or press the low end of the Volume button or **PREVIOUS** menu button to scroll backward. The default COS numbers are as follows. See page 5-103 for definitions.

For U.S. Systems:

COS 01 – ARS Only
COS 02 – Deny Area/Office
COS 03 – Deny Operator
COS 04 – Deny Toll Access
COS 05 – Deny International
COS 06 – Deny Equal Access
COS 07 – Deny Local Calls
COS 08 – Denied Numbers
COS 09 – Allowed Numbers

For European Systems:

COS 01 – ARS Only
COS 02 – Deny Operator
COS 03 – Deny Toll Access
COS 04 – Deny International
COS 05 – Deny Local Calls
COS 06 – Denied Numbers
COS 07 – Allowed Numbers

NOTE: Numbers listed in the "Emergency Number" table are never restricted, no matter how the COS shown above is programmed.

2.) When the correct COS is displayed, press 1 or the **ON** menu button to enable the toll restriction. Or, press 2 or the **OFF** menu button to disable it.

3.) If desired, repeat steps c2 and c3 to program additional COS toll restrictions.

4.) Press # or **ACCEPT** to save the COS programming. The display shows DATABASE UPDATED and then returns to the TOLL RESTRICTION OPTION prompt.

5.) Press # again to exit to the ENTER STATION OPTION prompt.

6. When the display shows ENTER STATION OPTION, press # again to exit to the ENTER STATION EXTENSION prompt. You can then program another station by repeating these steps or press # or **ACCEPT** once more to exit to the ENTER DATABASE OPTION prompt.

7. When finished with all programming, press # while the ENTER DATABASE OPTION prompt is displayed. This ends the programming session.

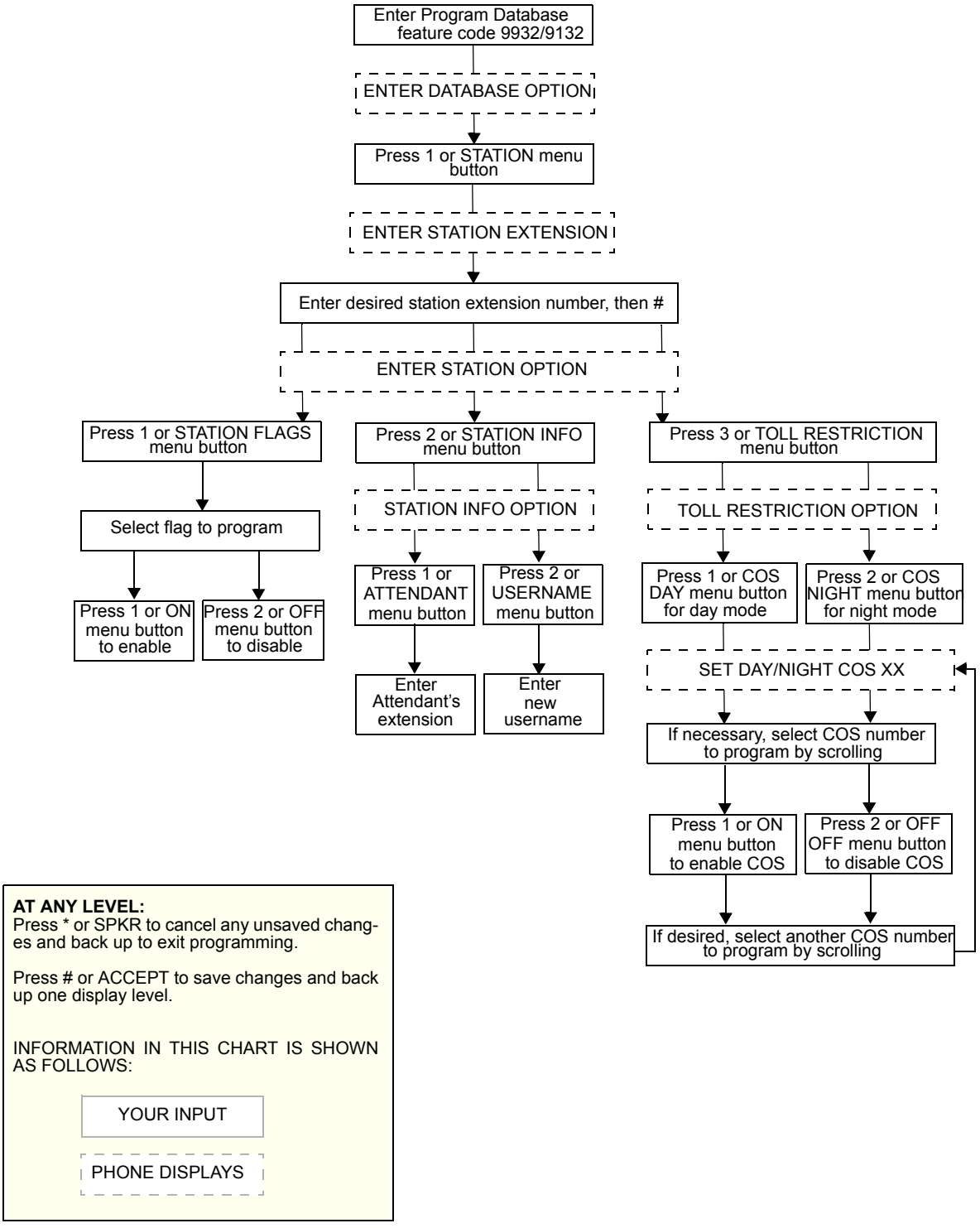
8. *If a system reset is required,* the display shows ENTER SYS RESET OPTION. Do one of the following:

   — *Delayed Reset:* Press **1** or the **DELAYED** menu button to delay the reset. The display shows DELAYED RESET SCHEDULED. The system will be reset at the pre-programmed time (see page 6-271).

   — *Immediate Reset:* Press **2** or the **IMMEDIATE** menu button to reset the system now.

NOTE: A system reset will drop all calls in progress.

Figure 5-9. *Station DB Programming Flowchart*



System Programming Using an Administrator's Station

52.30 The system-wide information that can be programmed by an Administrator station includes the following:

- **Define Do-Not-Disturb Messages**: The messages for the Do-Not-Disturb feature can be reprogrammed through an Administrator's station. (See page 5-26 for information concerning their use.) Administrators can delete or change messages 01-20 to any value (up to 16 characters). When the system has a programmed primary and secondary language, the system has default do-not-disturb messages in both languages. (Available languages are American English, British English, Spanish, and Japanese.) The current language of the programming phone determines which list is programmed. (See page 5-160 for a description of the Change Language feature.) The secondary language translation has the same meaning as the primary language message. The default messages are:

01 DO-NOT-DISTURB	11 OUT OF TOWN 'TIL
02 LEAVE A MESSAGE	12 OUT OF OFFICE
03 IN MEETING UNTIL	13 OUT UNTIL
04 IN MEETING	14 WITH A CLIENT
05 ON VACATION 'TIL [ON HOLIDAY 'TIL in Europe]	15 WITH A GUEST
06 ON VACATION [ON HOLIDAY in Europe]	16 UNAVAILABLE
07 CALL ME AT	17 IN CONFERENCE
08 AT THE DOCTOR	18 AWAY FROM DESK
09 ON A TRIP	19 GONE HOME
10 ON BREAK	20 OUT TO LUNCH

  When two languages are enabled and do-not-disturb messages are changed, the programmer should attempt to keep the meanings for the messages in both lists the same. That is, if the primary language message 02 is changed to "PAGE ME," a similar message should be programmed for the secondary language message 02.
- **Select an Administrator DB Programming Password**: Entry to the DB Programming feature at the Administrator stations can be protected using a password. A password would prevent unauthorized users from altering the system database.
- **Define Reminder Messages**: System reminder messages can be changed using an Administrator's station. (See page 5-267 for information on using reminders.) The messages can have up to 16 characters each. When primary and secondary Languages are enabled, the system has default reminder messages in both languages. The current language of the programming phone determines which list is programmed. (See page 5-160 for a description of the Change Language feature.) Each secondary language translation has the same meaning as the primary language message. Default messages are:

01 MEETING	11 CALL ENGINEERING
02 STAFF MEETING	12 CALL MARKETING
03 SALES MEETING	13 CALL ACCOUNTING
04 CANCEL MEETING	14 CANCEL DND
05 APPOINTMENT	15 CANCEL CALL FWD
06 PLACE CALL	16 TAKE MEDICATION
07 CALL CLIENT	17 MAKE RESERVATION
08 CALL CUSTOMER	18 REVIEW SCHEDULE
09 CALL HOME	19 LUNCH
10 CALL CORPORATE	20 REMINDER

When two languages are enabled and do-not-disturb messages are changed, the programmer should attempt to keep the meanings for the messages in both lists the same. That is, if the primary language message 02 is changed to "GO TO AIRPORT," a similar message should be programmed for the secondary language message 02.

- **Program New Extension Numbers for Stations**: The extension number for any station can be changed by an Administrator. The new extension number cannot conflict with an existing number.
- **Swap Extensions**: An extension number can be relocated (swapped) to another station. To swap extensions, the two affected stations must meet the following criteria:
  - — Both devices must reside on the same node as the Administrator performing the swap.
  - — Both devices must be the same type (i.e., both phones or both single-line sets).
  - — Neither device can be the administrator phone performing the swap.

NOTE: You can also swap extensions using the Attendant Console.

52.31 The System-Wide DB Programming process is summarized in the flowchart shown on page 5-288. Program planning sheets are included on the system software CD-ROM, Documentation folder. You can also find all documentation on the Inter-Tel *edGe* Web site. For readers located in Europe, contact the Inter-Tel Europe office for technical documentation.

52.32 If necessary, you can press ✱ or the Speaker button to cancel programming and discard any unsaved changes at any time during the following procedure.

52.33 To program the system database:

NOTE: If you want to change the do-not-disturb or reminder message for the Primary and Secondary Language sets, make sure your station is set in the proper mode. See page 5-163 for an explanation of the Change Language feature.

1. While on hook, enter the Program Database feature code (9932 or 9132).
2. *If a password is required,* the display shows ENTER PASSWORD. Use the keypad to enter your 1-8 digit password and press #. (If you enter an incorrect password, the display shows INVALID PASSWORD.)

   If a password is not required, skip this step.
3. The display shows ENTER DATABASE OPTION. (Executive Display, Professional Display, and Model 8560 Phones also show the options: STATION, SYSTEM, and TRUNK.) Press 2 or the **SYSTEM** menu button.
4. The display shows ENTER SYSTEM OPTION. Select one of the following:
   a. ***Do-Not-Disturb Messages:*** Allows you to program the do-not-disturb messages used by the stations. To select it, press 1 or the **DND MESSAGES** menu button. Then do the following:
      1.) The display shows SELECT DND MESSAGE #. Enter a message number or scroll to the desired message. (To scroll to the correct message, press the Volume button or the **SCROLL** plus **NEXT** and **PREVIOUS** menu buttons.)
      2.) When the display shows the desired DND message, enter the new message as described in paragraph 52.26 on page 5-277.

3.) Press # or **ACCEPT** to save the new message. The display shows DATABASE UPDATED and then returns to the SELECT DND MESSAGE prompt. (Or, to cancel your entry, press * or **CANCEL**. The display shows NO UPDATE PERFORMED.)

4.) To program another message, scroll to the desired message and repeat these steps.

5.) Press # again to exit to the ENTER SYSTEM OPTION prompt.

b. ***Password:*** Allows you to set a password that limits access to the Administrator programming feature. To select it, press 2 or the **PASSWORD** menu button. Then do the following:

1.) The display shows CHANGE PASSWORD TO. Enter a password of up to eight digits, then press #. (Or, to erase the password and leave it blank, just press #.)

2.) The display shows VERIFY PASSWORD. Enter the password exactly as you did in the step above, followed by #. The display returns to the ENTER SYSTEM OPTION prompt. (If you hear reorder tones and see an error message, the passwords did not match and you must start over at the CHANGE PASSWORD prompt.)

c. ***Reminder Messages:*** Allows you to program the reminder messages used by the stations. To select it, press 3 or the **REMINDER MSGS** menu button. Then do the following:

1.) The display shows SELECT REMINDER MSG #. Enter a message number or scroll to the desired message. (To scroll to the correct message press the Volume button or the **SCROLL** plus **NEXT** and **PREVIOUS** menu buttons.)

2.) When the display shows the desired message, enter the new message as described in paragraph 52.26 on page 5-277.

3.) Press # or **ACCEPT** to save the new message. The display shows DATABASE UPDATED and then returns to the SELECT REMINDER MSG prompt. (Or, to cancel your entry, press * or CANCEL. The display shows NO UPDATE PERFORMED.)

4.) To program another message, scroll to the desired message and repeat these steps.

5.) Press # again to exit to the ENTER SYSTEM OPTION prompt.

d. ***Station Extensions:*** Allows you to assign new extension numbers to stations. To select it, press 4 or the **STN EXTENSION** menu button.

If programming a station:

1.) Press 1 or **CHANGE EXT**. The display shows ENTER STATION EXTENSION. Enter the extension number of the station to be programmed using one of the following methods. (If you enter an invalid extension number, you hear reorder tones and must try again.)

Enter a complete number: Enter the extension number using your keypad. When a valid number is entered, the circuit information appears. Press # again to continue.

Enter a partial number: Enter a partial extension number, then press the # button, high end of the Volume button, or the **ACCEPT** menu button. The display shows the extension number, user name and circuit number of the station that most closely matches the partial extension number. Press # or **ACCEPT** to program the displayed station, or scroll to another station as described below.

Scroll through the numbers: To scroll through the extension number list, press the high end of the Volume button or the **NEXT** menu button to scroll forward, or press the low end of the Volume button or the **PREVIOUS** menu button to scroll backward. When the desired extension number is displayed, press # or **ACCEPT** to continue.

2.) The display shows CHANGE X (*number)* TO EXTENSION. Enter the new extension number for the station. If you enter an invalid number, you hear reorder tones and must try again. *If you entered an extension number that is already assigned,* the display shows CONFLICTING EXTENSION and you hear reorder tones. When the display returns to ENTER STATION EXTENSION, you must start over and select a new extension number.

3.) Press # or **ACCEPT** to exit to the ENTER STATION EXTENSION prompt. The display shows UPDATING DATABASE for four seconds and then DATABASE UPDATED. (Or, to cancel your entry, press * or CANCEL. The display shows NO UPDATE PERFORMED.)

NOTE: Although the system begins to update the database when you press # or **ACCEPT**, it may take longer than the four-second display to change the extension in the system – especially in a large or busy system.

4.) Press # or **ACCEPT** again to exit to the ENTER SYSTEM OPTION prompt. (If an extension number is displayed, press * to exit, instead of # or **ACCEPT**.)

If swapping stations:

1.) Press 2 or **SWAP EXTS**. The display shows ENTER STATION EXTENSION. Enter the first extension number to be swapped using one of the following methods. (If you enter an invalid extension number, you hear reorder tones and must try again.)

Enter a complete number: Enter the extension number using your keypad. When a valid number is entered, the circuit information appears. Press # or **ACCEPT** to continue.

Enter a partial number: Enter a partial extension number, then press the # button, high end of the Volume button, or the **ACCEPT** menu button. The display shows the extension number, user name and circuit number of the station that most closely matches the partial extension number. Press # or **ACCEPT** to continue, or scroll to another station as described below.

Scroll through the numbers: To scroll through the extension number list, press the high end of the Volume button or the **NEXT** menu button to scroll forward, or press the low end of the Volume button or the **PREVIOUS** menu button to scroll backward. When the desired extension number is displayed, press # or **ACCEPT** to continue.

NOTE: If the feature times out while entering an extension number, the display shows INCOMPLETE EXTENSION.

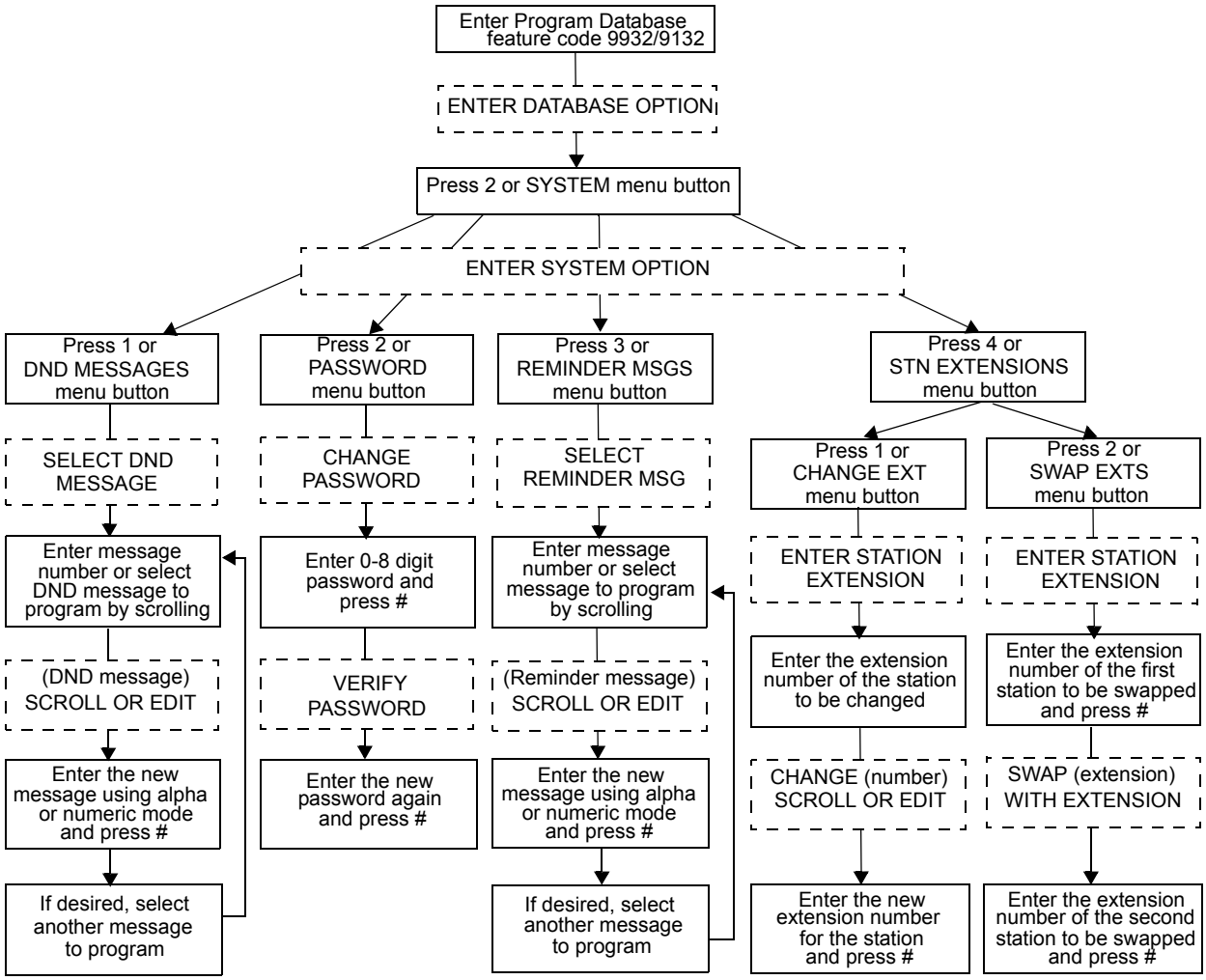
2.) The display shows SWAP *(extension)* WITH EXTENSION. Enter the second extension number to be swapped using the same methods previously described. The display shows DATABASE UPDATED.

NOTE: If you cancel or terminate the feature without entering the second extension number, the system discards any changes that have not yet been saved, and the display shows NO UPDATE PERFORMED. If the entered extension is the administrator station performing the swap, the same extension number as the target station, or the wrong device type, the display shows CONFLICTING EXTENSION.

5. When the display shows ENTER SYSTEM OPTION, press # again to exit to the ENTER DATABASE OPTION prompt.
6. When finished with all programming, press # while the ENTER DATABASE OPTION prompt is displayed. This ends the programming session.
7. *If a system reset is required,* the display shows ENTER SYS RESET OPTION. Do one of the following:
   - *Delayed Reset:* Press 1 or the **DELAYED** menu button to delay the reset. The display shows DELAYED RESET SCHEDULED. The system will be reset at the pre-programmed time (see page 6-271).
   - *Immediate Reset:* Press 2 or the **IMMEDIATE** menu button to reset the system now.

NOTE: A system reset will drop all calls in progress.

Figure 5-10. *System DB Programming Flowchart*



AT ANY LEVEL:
Press * or the Speaker button to cancel any unsaved changes and back up to exit programming.

Press # or ACCEPT to save changes and back up one display level.

INFORMATION IN THIS CHART IS SHOWN AS FOLLOWS:

YOUR INPUT

PHONE DISPLAYS

Trunk Programming Using an Administrator's Station

52.34 The trunk information that can be programmed by an Administrator station includes the following:

Individual trunks

- **Answer Supervision Type**: Answer supervision determines whether the system should consider a call valid when it receives polarity reversal from the central office [network provider] or wait for the Valid Call timer to expire. Any outside call put on hold becomes valid immediately, regardless of Answer Supervision or Valid Call timer status. There are three options available under answer supervision type:
  - **Polarity Reversal**: A loop reversal must be received to consider the call valid. When the first loop reversal is received, the call is made valid immediately, and the phone display will begin call cost. When a second loop reversal is received, the system will terminate the call.

    NOTE: If a second loop reversal is not received, the system will not terminate the call unless the inside party hangs up or loss-of-loop is received from the CO [network provider].

  - **Valid Call Timer**: After the Valid Call Timer expires, the call is validated. All polarity reversals received before and after the Valid Call Timer are ignored.
  - **Valid Call Timer with Polarity Reversal**: If a loop reversal is received before the Valid Call Timer expires, the call will be made valid immediately, and the phone display will begin call cost. When a second loop reversal is received, the system will terminate the call. If a loop reversal is not received before the Valid Call Timer expires, the call will be made valid by the timer. If a loop reversal is received after the timer expires, the loop reversal is ignored, but the call cost is reset. If a second loop reversal is then received, the system will terminate the call.

    NOTE: If a second loop reversal is not received, the system will not terminate the call unless the inside party hangs up or loss-of-loop is received from the CO.

- **Caller ID [CLIP]**: When selected, the Caller ID [CLIP] option indicates that the associated trunk will provide caller identification signals. This cannot be programmed unless the Advanced CO Interfaces premium feature is licensed. See page 6-97. Caller information features require DSP programming, described on page 6-117. Caller ID [CLIP] uses Caller ID [CLIP] receivers.
- **Hybrid Balance**: Hybrid balance can be set to Normal or Short to allow the installer to adjust the hybrid impedance of the trunk based on local central office conditions.
  - *Short:* The short balance is used when trunks are supplied by other equipment located within the same building, using private, copper wire termination, e.g., an OPX supplied by another PBX that does not use the public network [a multiplexor or host PBX in Europe], or when the site is located close to the central office [local exchange].
  - *Normal:* The normal balance can be used by most trunks.

  Version 10.0 or later system allows you to test the trunk interface hybrid balance manually. Hybrid Balance Test is a tool that provides data and a recommendation you can use to fine-tune the hybrid balance of individual analog loop start trunks connected to loop/ground start trunk ports. See page 6-246 for details.

- **DTMF or Dial-Pulse Signaling**: Trunks can be set for DTMF or dial pulse signals.
- **Trunk Group Assignment**: Trunks can be assigned to trunk groups on an individual basis.
- **Busy Trunk**: This feature allows the administrator to temporarily remove a trunk from service. While the trunk is out of service, it cannot be used for placing outgoing calls. However, incoming calls on analog trunks will ring in and can be answered. Incoming calls on ISDN trunks cannot be answered. The trunk will remain out of service until the Trunk Busy Out timer expires or the administrator places the trunk back in service.

Trunk Groups

- **Day/Night Answer Access**: The Administrator can program separate lists for stations with allowed-answer access in day and night modes.
- **Day/Night Ring-In**: The Administrator can program separate lists of stations with ring-in for day and night modes.
- **Change Toll Restrictions**: The Administrator can program toll restriction classes of service for the trunk groups. Classes of service are described in detail on page 5-103.
- **Trunk List**: Each trunk group has a programmed list of individual trunks.

52.35 The Trunk DB Programming process is summarized in the flowchart shown on page 5-295. Program planning sheets are included on the system software CD-ROM, Documentation folder. You can also find all documentation on the Inter-Tel *edGe* Web site. For readers located in Europe, contact the Inter-Tel Europe office for technical documentation.

NOTE: If necessary, you can press the asterisk (*) or Speaker button to cancel programming and discard any unsaved changes at any time during the following procedure.

52.36 When the procedure asks you to enter a station, trunk, or trunk group extension number, you can use any of the following methods. (If you enter an invalid extension number, you hear reorder tones and must try again.)

- **Enter a Complete Number**: Enter the extension number using your keypad. When the circuit information is displayed, press # again to continue.
- **Enter a Partial Number**: Enter a partial extension number then press the # button, high end of the Volume button, or the **ACCEPT** menu button. The display shows the extension number, username, and circuit number of the station, trunk, or trunk group that most closely matches the partial extension number. Press # or **ACCEPT** to accept the displayed extension, or scroll to another station as described below.
- **Scroll through the Numbers**: To scroll through the extension number list, press the high end of the Volume button or the **NEXT** menu button to scroll forward, or press the low end of the Volume button or the **PREVIOUS** menu button to scroll backward. When the desired extension number is displayed, press # or **ACCEPT** to continue.

52.37 To program trunk information:

1. While on hook, enter the Program Database feature code (9932 or 9132).
2. *If a password is required,* the display shows ENTER PASSWORD. Use the keypad to enter your 1-8 digit password and press #. (If you enter an incorrect password, the display shows INVALID PASSWORD.)

   If a password is not required, skip this step.

3. The display shows ENTER DATABASE OPTION. (Executive Display, Professional Display, and Model 8560 Phones also show the options: STATION, SYSTEM, and TRUNK.) Press **3** or the **TRUNK** menu button. The display shows ENTER TRUNK OPTION.

4. **Program an Individual Trunk**: Press **1** or the **INDIVIDUAL TRUNK** menu button. Then program trunk options as follows:

   a. The display shows ENTER TRUNK EXTENSION. Enter the extension number of the trunk to be programmed using one of the methods described in paragraph 52.36.

   b. The display shows INDIVIDUAL TRUNK OPTION. Enter one of the following options. To view the last two options on an Executive Display, Professional Display, or Model 8560 Phone, press the arrow (<< or >>) menu button at the bottom of the display.

      Answer Supervision: To program the answer supervision for the trunk, press **1** or the **ANS SUPERVISION** menu button. When the display shows ANS SUPERVISION OPTION, press **1** or the **POLARITY REVERSE** menu button for the polarity-reversal supervision type. Or, press **2** or the **VALID CALL** menu button for Valid Call timer supervision. Or, press **3** or the **VALID + POLARITY** menu button for Valid Call timer with Polarity Reversal supervision. Then press **#** or **ACCEPT** to continue.

      Caller ID [CLIP]: To enable or disable Caller ID [CLIP], press **2** or the **CALLER ID** menu button. When the display shows SET CALLER ID, press **1** or ON to enable the feature, or press **2** or OFF to disable it. Then press **#** or **ACCEPT** to continue.

      Hybrid Balance: To set the hybrid balance of the trunk to normal or short, press **3** or the **HYBRID BALANCE** menu button. When the display shows HYBRID BALANCE OPTION, press **1** or the **NORMAL** menu button, or press **2** or the **SHORT** menu button. Then press **#** or **ACCEPT** to continue.

      DTMF/Dial-Pulse Signaling: To set the trunk for DTMF or dial-pulse signaling, press **4** or the **SIGNALING** menu button. When the display shows ENTER SIGNALING OPTION, press **1** or the **DTMF** menu button, or press **2** or the **PULSE** menu button. Then press **#** or **ACCEPT** to continue.

      Trunk Group Number: To assign the trunk to a trunk group, press **5** or the **TRUNK GROUP** menu button. When the display shows ENTER TRUNK GRP EXTENSION, enter the desired trunk group number using one of the methods described in paragraph 52.36.

      Busy Trunk: To temporarily remove the trunk from service, press **6** or the **BUSY TRUNK** menu button. When the display shows MARK TRUNK BUSY, press **1** or the **ON** menu button, or press **2** or the **OFF** menu button. Then press **#** or **ACCEPT** to continue.

   a. To exit, press **#** or **ACCEPT** to move backward through the prompts until you reach the ENTER DATABASE OPTION prompt.

5. **To Program A Trunk Group:** Press 2 or the **TRUNK GROUP** menu button. Then program the trunk group, as follows:

   a. The display shows ENTER TRUNK GRP EXTENSION. Enter the extension number of the trunk group to be programmed using one of the methods described in paragraph 52.36.

   b. The display shows ENTER TRUNK GROUP OPTION. Select one of the following options. To view the last three options on an Executive Display, Professional Display, or Model 8560 Phone, press the arrow (<< or >>) menu button at the bottom of the display.

      Answer Access For Day Or Night Mode: To program the station(s) that will have answer access for this trunk group for day mode, press 1 or the **ANS-ACCESS DAY** menu button. To program for night mode, press 2 or the **ANS-ACCESS NIGHT** menu button. When the display shows ENTER STATION EXTENSION, do the following:

      1.) Enter a station extension using one of the methods described in paragraph 52.36.

      2.) The display shows the extension number and a Yes/No prompt. Press 1 or the **YES** menu button, followed by #, to include the station in the list. Or, press 2 or the **NO** menu button, followed by #, to exclude the station from the list.

      3.) When the display returns to ENTER STATION EXTENSION, select another extension number. Or, if finished programming the station list, press # or **ACCEPT** to continue.

      Ring-In For Day Or Night Mode: To program the station that will have ring in for this trunk group for day mode, press 3 or the **RING-IN DAY** menu button. To program for night mode, press 4 or the **RING-IN NIGHT** menu button. When the display shows ENTER STATION EXTENSION, enter stations into the list one at a time as described above for Answer Access.

      Toll Restriction: To program the toll restrictions for the trunk group, press 5 or the **TOLL RESTRICTION** menu button. When the display shows TOLL RESTRICTION OPTION, press the numbers that correspond to the toll restrictions you want to program, as follows:

      1.) *COS for day or night mode:* To program the toll restriction class of service for day mode, press 1 or the **COS DAY** menu button. To program class of service for night mode, press 2 or the **COS NIGHT** menu button. The display shows SET DAY (or NIGHT) COS XX. If this is not the COS you want to program, scroll to the correct COS by pressing the high end of the Volume button or the **NEXT** menu button to scroll forward, or press the low end of the Volume button or the **PREVIOUS** menu button to scroll backward. The default COS numbers are listed below. (See page 5-103 for definitions. COS 01 is not used for trunks.) When the correct COS is displayed, press 1 or the **ON** menu button to enable the toll restriction. Or, press 2 or the **OFF** menu button to disable it. If desired, repeat this step to program additional COS toll restrictions. Then press # or **ACCEPT** to continue.

For U.S. Systems:

COS 02 – Deny Area/Office
COS 03 – Deny Operator
COS 04 – Deny Toll Access
COS 05 – Deny International
COS 06 – Deny Equal Access
COS 07 – Deny Local Calls
COS 08 – Denied Numbers
COS 09 – Allowed Numbers

For European Systems:

COS 02 – Deny Operator
COS 03 – Deny Toll Access
COS 04 – Deny International
COS 05 – Deny Local Calls
COS 06 – Denied Numbers
COS 07 – Allowed Numbers

2.) *Exempt from ARS:* To program the Exempt from ARS option, press 3 or the **EXEMPT FROM ARS** menu button. When the display shows EXEMPT FROM ARS, press 1 or the **YES** menu button to enable the option. Or, press 2 or the **NO** menu button to disable it. Then press # or **ACCEPT** to continue.

3.) *Subject to toll restriction:* To program the trunk group Subject to Toll Restriction option, press 4 or the **SUBJECT TO TOLL** menu button. When the display shows SUBJECT TO TOLL, press 1 or the **YES** menu button to enable the option. Or, press 2 or the **NO** menu button to disable it. Then press # or **ACCEPT** to continue.

Trunk List: To program the list of trunks for the trunk group, press 6 or the **TRUNK LIST** menu button. When the display shows ENTER TRUNK EXTENSION, enter trunks into the list as follows:

1.) Enter a trunk extension using one of the methods described in paragraph 52.36.

2.) The display shows the extension number and a Yes/No prompt. Press 1 or the **YES** menu button, followed by #, to include the trunk in the list. Or, press 2 or the **NO** menu button, followed by #, to exclude the trunk from the list.

3.) When the display returns to ENTER TRUNK EXTENSION, select another extension number. Or, if finished programming the trunk list, press # or **ACCEPT** to continue.

a. To exit, press # or **ACCEPT** to move backward through the prompts until you reach the ENTER DATABASE OPTION prompt.

6. When finished with all programming, press # while the ENTER DATABASE OPTION prompt is displayed. This ends the programming session.

7. *If a system reset is required,* the display shows ENTER SYS RESET OPTION. Do one of the following:

   — *Delayed Reset:* Press **1** or the **DELAYED** menu button to delay the reset. The display shows DELAYED RESET SCHEDULED. The system will be reset at the pre-programmed time (see page 6-271).

   — *Immediate Reset:* Press **2** or the **IMMEDIATE** menu button to reset the system now.

NOTE: A system reset will drop all calls in progress.

Figure 5-11. *Trunk DB Programming Flowchart*

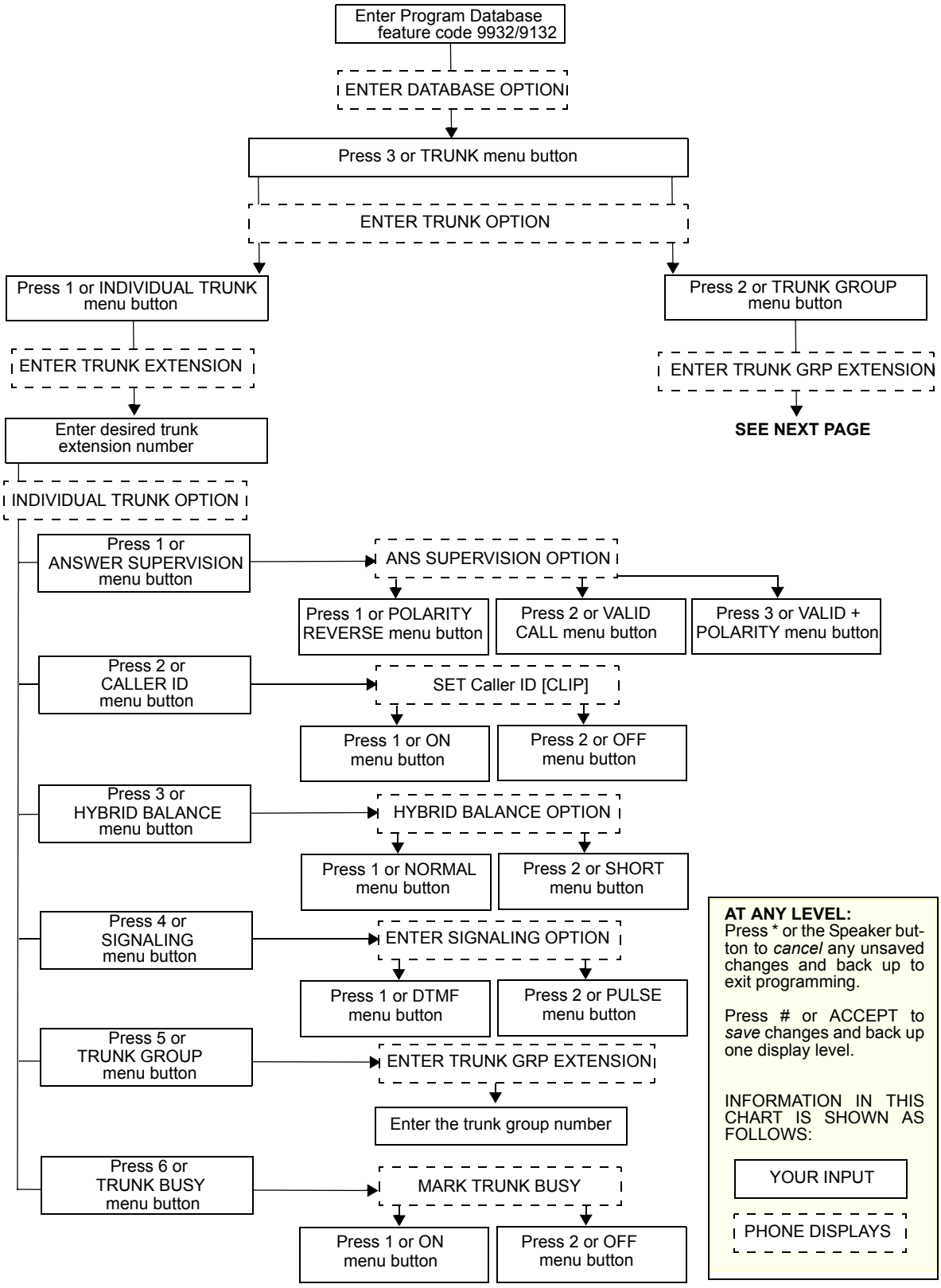
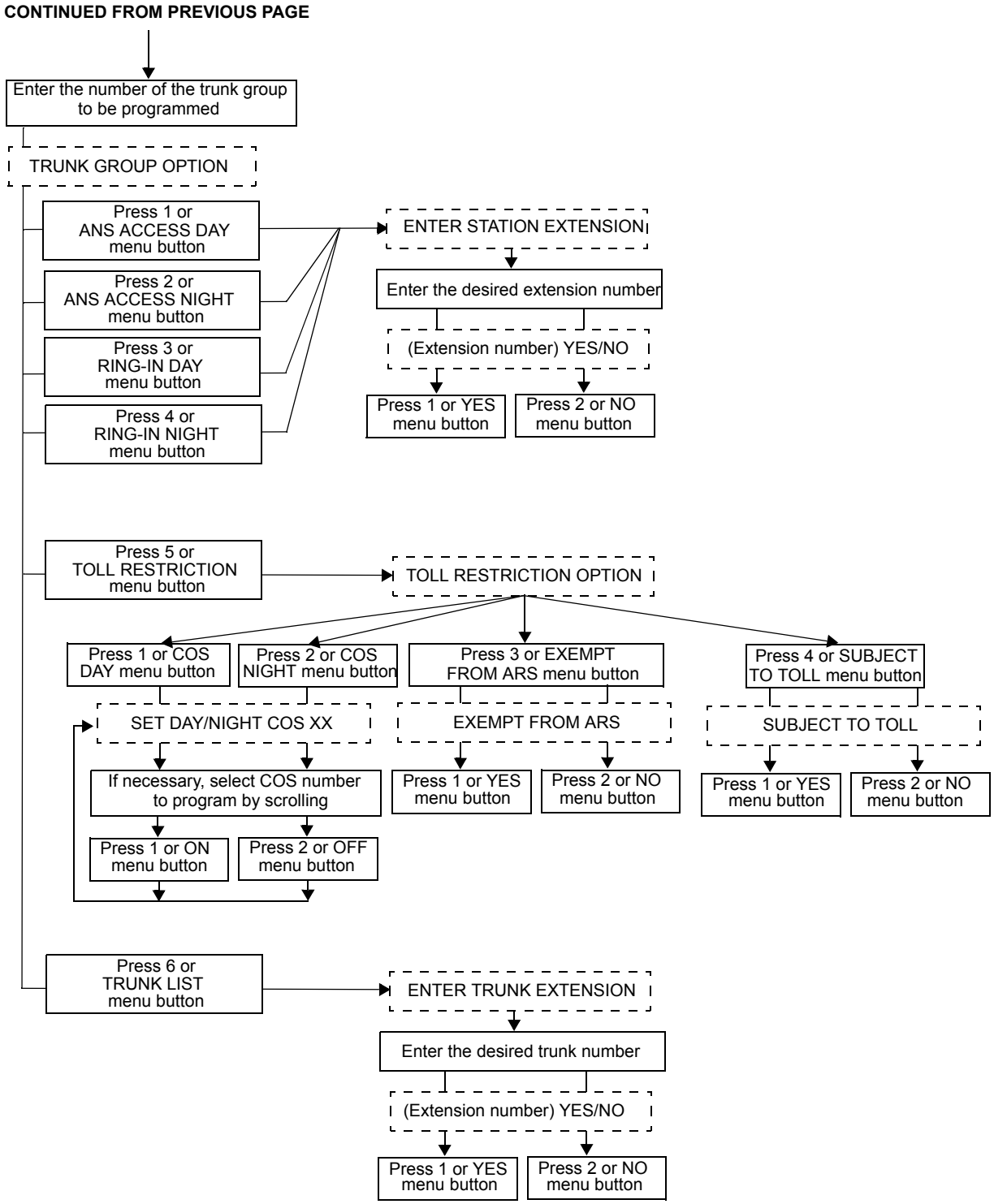


Figure 5-11. *Trunk DB Programming Flowchart (Continued)*



G. PROGRAMMING SYSTEM SPEED-DIAL NUMBERS

52.38 System speed-dial numbers and names can be programmed by the installer or at any Administrator station.

52.39 The system speed-dial numbers can contain up to 48 digits each and can include hookflashes/recalls and/or pauses for dialing a series of numbers. For example, the speed-dial number can contain an SCC local number (U.S. only), a pause, an access code, and the telephone number. For using PBX trunks, the other system's trunk access codes, followed by a pause, may be included in speed-dial numbers. The lengths of the hookflash (recall) and the pause are determined by the programmable LS/GS CO Hookflash [LS CO Hookflash in Europe] and Pause Dialing Digit Length timers.

52.40 To program system speed-dial numbers, use one of the following methods:

- Use the keypad to manually dial the number. If you make a mistake, press the MUTE button to backspace.
- Press the REDIAL button to enter the last number dialed or saved at the station (up to 48 digits).
- Press one of the speed-dial buttons on the phone to enter the outside telephone number (up to 16 digits) programmed under that button.

52.41 Speed-dial names can contain up to 16 characters. To program speed-dial names, keypad buttons are used to enter the desired letters, numbers, and punctuation in English/Spanish, or Japanese (Katakana). When programming speed-dial names, you can press the MSG button or **USE ALPHA MODE/USE NUMERIC MODE** menu button to switch back and forth between alphanumeric and numeric mode.

- **In Numeric Mode**: The keypad buttons are used to enter numbers 0-9, the pound [hash] (#) button is used for entering a hyphen (-), and the asterisk (*) button is used for entering a colon (:). For example, 1*00 would enter "1:00" in numeric mode.
- **In Alphanumeric Mode**: The keypad buttons are used to enter the desired letters, numbers, and punctuation. The number of times a button is pressed determines which character is entered. For example, 533266 would enter "JEAN." When adjoining characters are located under the same button, press the FWD button to advance to the next character. For example, 66 FWD 6667776 would enter "NORM." See the chart below to program information in alphanumeric mode. (Note that letters correspond to the letters printed on the keypad buttons.) The Japanese characters will be available only if the Multilingual Support premium feature is licensed.

BTN	Number of Times Button is Pressed										
	1	2	3	4	5	6	7	8	9	10	11
	English/Spanish Characters					Katakana Characters**					
1	-	&	(	)	1	A	I	U	E	O	a
2	A	B	C	’	2	KA	KI	KU	KE	KO	i
3	D	E	F	!	3	SA	SHI	SU	SE	SO	u
4	G	H	I	*	4	TA	CHI	TSU	TE	TO	e
5	J	K	L	# or /*	5	NA	NI	NU	NE	NO	o
6	M	N	O	Ñ or #*	6	HA	HI	FU	HE	HO	tsu
7	P	Q	R	S	7	MA	MI	MU	ME	MO	ya
8	T	U	V	?	8	YA	YU	YO	.	,	yu
9	W	X	Y	Z	9	RA	RI	RU	RE	RO	yo
0	@	:	.	,	0	WA	WO	N	pa	ba	long

* The character available depends on the software version.

** The Japanese characters are only available if the Multilingual feature is enabled and Japanese is installed as the secondary language.

52.42 To program or change system speed-dial numbers and names:

NOTE: If you make a mistake, use the **MUTE** button to backspace. Or, press ✳ or **CANCEL** at any time to cancel the operation (the name and number in memory remains unchanged and the display shows FEATURE CANCELED); then start over.

1. While on hook, enter the Program System Speed-Dial feature code (9801). Display phones show PROGRAM SYS SPD # (000-999 or 0000-4999).
2. Enter the speed-dial location code (000-999 or 0000-4999). The display shows ENTER NAME: *(current name)*. If you do not continue to the next step, the display times out after the Long Interdigit timer expires and you must start over.
3. **To change or program the name:** Enter the desired name for the speed-dial number using your keypad as described on page 5-297, then press **#** or **ACCEPT**. See the chart on the previous page. *If you make a mistake,* use the **MUTE** button to back up and erase existing characters or the **CLEAR** button to erase the whole entry.

   To erase the current name and leave it blank: Press **CLEAR** or press **MUTE** repeatedly until the name is erased. Then press **#** or **ACCEPT**.

   To leave the name the same: Press **#** or **ACCEPT**.
4. When the display shows ENTER NUMBER, enter the number (up to 48 digits) to be stored using the keypad, one of the speed-dial buttons, or the **REDIAL** button. *If necessary,* use the **MUTE** button to back up and erase existing numbers. (Display phones show the number as it is entered.) *To include *, #, pause, or hookflash (recall) in the number,* press ∞ or **SPCL** once for an asterisk (*), twice for a pound [hash] (#), three time for a hookflash/recall (F), or four times for a pause (P). Menu button phone users can press the *, #, **FLASH**, or **PAUSE** menu buttons to enter special characters. You cannot exceed 48 digits.
5. Press **#** or **ACCEPT** to save the number. You hear a single progress tone when the number is accepted. Your display shows SYS SPD BIN #XXX UPDATED.

52.43 To erase system speed-dial names or numbers:

1. While on hook, enter the Program System Speed-Dial feature code (9801). Display phones show PROGRAM SYS SPD # (000-999 or 0000-4999).
2. Enter the speed-dial location code (000-999 or 0000-4999). The display shows ENTER NAME: *(current name)*.
3. You may erase the name and/or number:

   To erase the name: Press the MUTE button repeatedly until the name is erased or press **CLEAR**.

   To erase the number: Press # or **ACCEPT** to display the ENTER NUMBER prompt, then press the MUTE button repeatedly until the number is erased or press **CLEAR**.
4. Press # or **ACCEPT** to exit. You hear a single progress tone when accepted. Your display shows SYS SPD BIN #XXX UPDATED.

H. WORKING WITH THE MODEM

52.44 The Call Processing Card (CPC) has a built-in modem. This modem allows remote users to access DB Programming. It also functions as a serial port, used for SMDR, Message Print, etc. If necessary, System Administrators can enable, disable, or reset the modem and program a DSS/BLF button to reflect the modem status. (For information about connecting to the modem for DB Programming, see page 2-100.)

Enabling/Disabling the Modem

52.45 To enable/disable a modem:

1. Enter the Modem Enable (9866) or Modem Disable (9867) feature code. The display shows ENABLE (DISABLE) MODEM EXT #.
2. Enter the extension of the modem (local or off-node) to enable/disable. The display shows MODEM ENABLED (DISABLED).

52.46 If disabled, the modem is placed into DND. In the disabled state, the modem rejects all calls and does not allow camp ons.

NOTE: If an invalid modem extension is entered, the display shows INVALID MODEM EXTENSION.

Resetting the Modem

CAUTION

Do not reset the modem if it is connected to a DB Programming session. If you do, the programming session is dropped, potentially corrupting the database. For this reason, Inter-Tel strongly recommends that the System Administrator assign the modem extension to a DSS/BLF button.

52.47 To reset the modem:

NOTE: A modem reset drops any call that is currently connected to the modem.

1. Enter the Modem Reset (9869) feature code. The display shows RESET MODEM EXT #.
2. Enter the extension of the modem (local or off-node) to reset. The display shows MODEM RESET.

NOTE: If an invalid modem extension is entered, the display shows INVALID MODEM EXTENSION.

Assigning the Modem to a DSS/BLF Button

52.48 If desired, you can program a Direct Station Selection/Busy Lamp Field (DSS/BLF) button to reflect the state of the CPC modem. If programmed, the DSS/BLF button indicates the following states:

- **Unlit**: Modem is not active.
- **Fast Flash**: Modem is ringing.
- **Slow Flash**: Modem is disabled.
- **Solid**: Modem is connected to a call.

I. SYSTEM AND NETWORK ALARM REPORTING

52.49 To allow one administrator to monitor multiple nodes, the system provides both system alarms and network-wide alarms.

System Alarms

52.50 The system's Alarm Reporting feature detects equipment failures. If there is a system failure that affects service, a major alarm appears at all affected phones. If all phones in the system are inoperative, any alarm messages are sent to the system's primary serial port (on the CPC or CPU).

52.51 When a minor equipment failure occurs, a minor alarm is generated and appears on the primary attendant's display and, if enabled, at Administrators' stations. If enabled in the Message Print programming, the alarm also appears in the error/message report.

52.52 Alarms 1-99 indicate problems that can be corrected without calling service personnel. Alarms 100-199 (phone system) and 200-299 (Voice Processing) require attention from service personnel. Note that when a Voice Processing alarm is in effect, the phone system may be functioning properly, but the Voice Processing Unit may be inoperative. See page 7-12 in TROUBLESHOOTING for a listing of the possible alarms and their meanings.

Network Alarms

52.53 Each node has two flags (see page 6-261) which determine whether the node broadcasts or receives network-wide alarms.

- The **Send Network Alarms** flag determines if a node will broadcast alarms that occur on that node to the rest of the network.
- The **Receive Network Alarms** flag determines if the node will receive and display alarms sent by other nodes in the network.

52.54 To differentiate between network-wide and local alarms, network-wide alarms appear on administrator phones preceded by "NET ALARM," and local system alarms are preceded with "SYS ALARM."

52.55 On remote nodes, network-wide alarms will indicate the name of the node on which the alarm occurred. The node name is obtained from the username.

52.56 An administrator can clear a network-wide alarm on the local node only or on every node in the network.

- **Clear Network Alarm (9851)**: Clears network-wide alarms on every node in the network, but will not affect system alarms. The Clear Network Alarm feature code may be entered on any node in the network, but the Send Network Alarms flag must be set for the administrator to clear alarms on other nodes in the network.
- **Clear System Alarm (9850)**: Clears all local and network-wide system alarm displays on your node.

52.57 Network-wide alarms will override system alarms on an administrator's phone display.

52.58 To respond to a minor alarm from any alarm display station:

1. When a minor alarm indication appears, write down the alarm information.
2. While on hook, clear the alarm by entering the Clear System Alarm feature code (9850) or the Clear Network Alarm feature code (9851). The display shows SYSTEM (or NETWORK) ALARM CLEARED.
3. Take action on the alarm as follows:
   - *#10 XXXXX OFF-HOOK:* A station remained off hook and inactive until the Inactivity Alarm timer expired. The display and the SMDR indicate which station is off hook. The station's button on DSS/BLF Units and on other stations' speed-dial buttons flutters continuously. Locate the station and replace the handset in the cradle. The station's display shows STATION OFF HOOK.

     NOTE: Calls being transmitted over the secondary voice path are not affected or interrupted by an off-hook alarm condition.

   - *#11 XXXXX EMERGENCY:* A user has dialed the Emergency Call feature code 911 [999] and the system has selected a trunk and dialed the programmed emergency number.
   - *#12-13 & 20 CHECK PRINTER:* The printer is not functioning properly. Check that the cable and the power cord are connected and that it has paper and ribbon.
   - *#14 SET DATE/TIME:* The system real-time clock has not been initialized or has been corrupted. To set the system date and time see page 5-274.

- *#21 V-MAIL 80% FULL/#22 V-MAIL 100% FULL:* The Voice Mail disk space is either 80% or completely full. If it reaches 100%, Voice Mail cannot accept any messages until disk space is made available.
- *#23 SMDR 80% FULL/#24 SMDR 100% FULL:* The SMDR disk space is either 80% or completely full. If it reaches 100% before the SMDR information is cleared, no further SMDR recording will be possible.
- *#100-199 & 200-299 CALL TECHNICIAN:* These alarms indicate that an equipment failure has occurred that requires the attention of service personnel. When calling for service be prepared to provide them with the alarm information as well as what was taking place when the alarm occurred, if anything.

52.59 A major alarm message, MAJOR ALARM, appears on all display phones in the event of a system-wide failure. If the major alarm appears on a group of phones, the associated keyset card may have failed. The warning might also appear on a single phone if the phone or its cabling may be defective. Because they indicate that all or part of the system is inoperative, major alarms require immediate attention from service personnel.

J. FREEZING/UNFREEZING THE SYSTEM HISTORY

52.60 When certain system failures occur, service personnel might request a "system history freeze." This allows them access to the system database so that they can analyze system activity for the time period preceding the alarm.

52.61 To freeze or unfreeze system history:

NOTE: Do not perform this procedure unless directed to do so by service personnel.

1. While on hook, enter the Diagnostics Mode feature code – 9900 [9100 in Europe] to turn on diagnostics mode. The display shows DIAGNOSTICS ON.
2. Enter the Freeze System History feature code (9993 or 9193) or Unfreeze System History feature code (9998 or 9198). The display shows HISTORY FROZEN (or UNFROZEN). *If you attempt change the state of the system history to its current status,* the display shows HISTORY ALREADY FROZEN (or UNFROZEN).
3. Enter the Diagnostics Mode feature code (9900 or 9100) to turn off diagnostics mode. The display shows DIAGNOSTICS OFF.

K. FREEZING/UNFREEZING THE NETWORK HISTORY

52.62 Administrators can freeze or unfreeze the fault history for any node in the network. Service personnel can then pull the information from each node for troubleshooting purposes. Nodes are assigned to freeze zones by the database programmer to allow selective freezes. The Administrator must know which zone to freeze/unfreeze and then enter the freeze feature code followed by the zone number.

52.63 A node can be in more than one freeze zone. If the Administrator freezes a zone that contains a node that is already frozen, the display will show "SOME NODES ALREADY FROZEN" and the node will remain frozen. If the Administrator unfreezes a zone with unfrozen nodes in it, the display shows SOME NODES ALREADY UNFROZEN. To determine the individual freeze status of the nodes, check the message print output.

NOTE: Do not perform this procedure unless directed to do so by service personnel.

52.64 To freeze or unfreeze network history:

1. While on hook, enter the Diagnostics Mode feature code (9900 or 9100) to turn on diagnostics mode. The display shows DIAGNOSTICS ON.
2. Enter the Network Freeze Zone System Histories feature code (9939 or 9139) or Network Unfreeze Zone System Histories feature code (9989 or 9189). Display shows SELECT FRZ ZONE (0-9) #.
3. Enter the number of the zone you want to freeze or unfreeze. The display shows FREEZE ZONE XX IS FROZEN (or UNFROZEN). *If you attempt change the state of the system history to its current status,* the display shows FREEZE ZONE XX ALREADY FROZEN (or UNFROZEN).
4. Enter the Diagnostics Mode feature code (9900 or 9100) to turn off diagnostics mode. The display shows DIAGNOSTICS OFF.

L. SEIZING A DEVICE

52.65 For troubleshooting purposes, the Seize Device feature code allows an Administrator to enable diagnostics mode and then seize a specific trunk or extension by entering the board, circuit, and device number.

52.66 The Seize Device is affected by trunk restriction. The Administrator's station must have outgoing access permission for the trunk to seize it. Toll restriction is applied to any calls placed after the seizure of the trunk.

52.67 To seize a device:

1. While on hook, enter the Diagnostics Mode feature code (9900 or 9100) to turn on diagnostics mode. The display shows DIAGNOSTICS ON.
2. Press ∞ or SPCL and enter the Seize Device feature code (9973 or 9173). The display shows ENTER BOARD.
3. Enter the board number on which the device to be seized is located (if you enter a single digit, press # after the digit to continue). (If you enter an invalid board number, the display shows INVALID BOARD NUMBER.) Do not enter the CPC or CPU board number (0). The display shows ENTER PORT.
4. Enter the port (circuit) number of the device to be seized (if you enter a single digit, press # after the digit to continue). Valid numbers depend on the type of board. (If you enter an invalid board number, the display shows INVALID BOARD NUMBER.)

If there is only one device on the selected port, the selected device is seized as if the user had directly entered the extension or trunk number.

If there is more than one device on the selected port, the display shows ENTER DEVICE. Enter the number of the device to be seized. Valid numbers depend on the type of devices installed. (If you enter an invalid board number, the display shows INVALID BOARD NUMBER.) When the device has been selected, it is seized as if the user had directly entered the extension or trunk access number.

5. Enter the Diagnostics Mode feature code (9900 or 9100) to turn off diagnostics mode. The display shows DIAGNOSTICS OFF.

52.68 For example, if the Administrator wanted to seize the T1 or E1 trunk on the tenth circuit of the T1C or E1C in card slot 3 (circuit 3.10.0), the entries would be as follows:

ENTER BOARD: ***3***
ENTER PORT: ***10***
(Device number not required.)

However, to select the first single-line device connected to the second circuit of the KSC in card slot 5 (circuit 5.2.1), the entries would be as follows:

ENTER BOARD: ***5***
ENTER PORT: ***2***
ENTER DEVICE: ***1***

53. RECORD KEEPING AND MAINTENANCE FEATURES

A. CALL COST ACCOUNTING

NOTE: The phone system Call Cost Accounting feature is intended to provide a *cost estimate* that is applied to the various classes of calls. Due to the wide variation in charges among network carriers, the system's call cost calculation cannot be used as a prediction of actual charges. This feature can only be used as a management tool to *estimate* call cost.

ALSO: Call cost or elapsed call time cannot be displayed on Executone phones.

53.1 The Call Cost Accounting feature estimates the cost of outgoing and incoming calls, displays it on the phones, and prints it in the SMDR reports. The cost is based on the type of call, telephone number dialed, the elapsed time of the call, the day of the week, and the time of day. A table in the database supplies the rates for all types of calls, including multiplicative factors for evening and weekend rate changes on outgoing calls. The equation for calculating call cost is:

Daytime Rate X Multiplicative Factor X Connect Time.

53.2 *For U.S. Systems:* The multiplicative factor adjusts the daytime per-minute call cost for evening and weekend rates of outgoing calls. For example, the evening call cost multiplier is 0.65 if calls are 35% less expensive after 5:00PM. The daytime rate (D), evening (E) multiplicative factor, and night/weekend (N/W) multiplicative factors are used on the following schedule:

Table 5-43. *Multiplicative Factors (U.S.)*

	SUN	MON	TUE	WED	THU	FRI	SAT
8 AM to 5 PM	N/W	D	D	D	D	D	N/W
5 PM to 11 PM	E	E	E	E	E	E	N/W
11 PM to 8 AM	N/W	N/W	N/W	N/W	N/W	N/W	N/W

53.3 *For European Systems:* The multiplicative factor adjusts the peak per-minute call cost for standard and cheap rates of outgoing calls. For example, the call cost multiplier is 0.65 if calls are 35% less expensive after 6:00PM. The peak (P), standard (S), and cheap (C) multiplicative factors are used on the following schedule:

Table 5-44. *Multiplicative Factors (Europe)*

	SUN	MON	TUE	WED	THU	FRI	SAT
9 AM to 1 PM	C	P	P	P	P	P	C
1 PM to 6 PM	C	S	S	S	S	S	C
6 PM to 9 AM	C	C	C	C	C	C	C

53.4 The following two programmable factors are based on the non-programmable Daytime (Peak) factor of 1.00. The system is always in the Peak time period, using the Daytime (Peak) multiplicative factor of 1.00, unless the system falls into an Evening (Standard) or Night/Weekend (Cheap) time period at which time it follows the corresponding designated factor:

Table 5-45. *Programmable Factors*

Description	Multiplicative Factor
Daytime (Peak)	1.00 (See Note below)
Evening (Standard)	1.00
Night/Weekend (Cheap)	1.00

NOTE: Multiplicative Factor is applied to a non-programmable Peak rate of 1.00. Because the Peak rate is non-programmable, it does not appear in the Call Cost screen/window under Multiplicative Factors in DB Programming.

53.5 If call cost is set to zero, call cost will not display during the call, and the SMDR printout will show 00.00.

53.6 In a network setting, the call cost shown on the display and SMDR output use the factors and rates for the node on which the trunk resides. In other words, if a station on Node 1 dials what is considered a long-distance number on Node 1, but the call is routed to Node 2 where the number is considered local, the phone will use the local call cost rate from Node 2. The call rate used for calls between nodes will be based on the Network call rate. (See page 6-268 for programming information.)

53.7 All outgoing calls using a trunk that is not subject to toll restriction are classified as one of the following call types for call cost calculation (call cost type is programmed in the database).

For U.S. Systems:

- Local
- Toll Local
- Toll Long Distance
- Incoming
- Free
- Network
- Operator
- International

For European Systems:

- Free
- Local
- Incoming
- Toll (National)
- Network
- Operator
- International

53.8 When a trunk that is subject to toll restriction is used, call cost type is determined according to the digits dialed, as follows:

For U.S. Systems:

- **Local**: The following calls are classified as local calls:
  - Calls to N11 or 1N11 (where N is 2-9), except 411 or 1411
  - Any call to a toll-free area code
  - All 7-digit or 10-digit calls within a local area code to office codes that are allowed in User Group 1
  - Any call that does not begin with 1, and does not fall into any of the other call cost categories
- **Toll Local**: The following calls are classified as toll local calls:
  - Any calls to 411 or 1411
  - All 7-digit or 10-digit calls within a local area code to office codes that are restricted in User Group 1
  - Any call that begins with 1, that does not fall into any of the other call cost categories
- **Toll Long Distance**: Any call to an area code other than a local area code is classified as a toll long distance call.
- **Incoming**: Any call, except DID/DNIS calls, that rings into the phone system and is answered is classified as an incoming call.
- **Network**: Any call placed to or received from a network node is classified as a network call.
- **Free**: Any call within a PBX (a trunk access code is not dialed) is a free call.
- **Operator**: Any call starting with 0 or containing only 0 is classified as operator (0, 0+).
- **International**: Any call starting with 01 is classified as international (01+, 011+).

For European Systems:

- **Free**: 0800, 0500, 999, 112, and any call within a PBX (a trunk access code is not dialed).
- **Local**: Calls to N (where N is 2-9), 0345, and 0645.

NOTE: Local Calls may vary in length from 6 to 8 digits.

- **Incoming**: Any call, except DDI calls, that ring into the phone system and is answered.
- **Toll (National)**: Any calls to 01 or 0 except for free, local, and international calls.
- Network: Any call placed to or received from a network node.
- **International**: Any call starting with 00.
- **Operator**: Any call in the ranges 100-111 and 113-199.

B. STATION MESSAGE DETAIL RECORDING (SMDR)

53.9 Station Message Detail Recording (SMDR) is a system feature that provides a detailed record of outgoing and incoming calls. The system records only valid calls. Outgoing calls become valid when the Valid Call timer expires or polarity reversal is detected (depending on system programming). Outgoing calls become valid immediately if placed on hold or transferred. Incoming calls are always valid immediately.

53.10 Station call data can be recorded on a customer-provided printer or an alternate device, such as a magnetic tape or floppy disk. This output device is connected to the CPC, CPU, or Options card. You can also send SMDR information to the System Manager server. See SPECIFICATIONS, page 2-99, for details. Only one output port for SMDR can be active at any time.

53.11 The Voice Processing PC can be used to capture Station Message Detail Recording (SMDR) information via its RS232 connection with the phone system. The SMDR information is stored in a file on the Voice Processing PC hard disk for later retrieval.

53.12 In a network, each node has its own SMDR programming, SMDR output port, and SMDR output port backup. (See page 6-276 for programming information.) You can enable or disable network call records on each node. When network calls are enabled, the following applies:

- The system generates SMDR records for outgoing calls on the node where each trunk used resides. For example, if a caller on Node 1 places a call using a trunk group on Node 2, the SMDR report for Node 1 will show the outgoing call to Node 2 and the SMDR reports on Node 2 will show the incoming call from Node 1 and the outgoing call on the trunk group.
- The call record type for network SMDR records is NET.
- If a node's SMDR output port is a node, the network sends SMDR records to the specified node.
- You cannot select a node as the SMDR output port backup.
- If SMDR output programming forms a loop, the system will send the SMDR output to the node's backup SMDR port. For example, if the SMDR port on Node 1 routes to Node 2 and the SMDR port on Node 2 routes to Node 1, the configuration will cause an infinite loop of SMDR routing. SMDR reports for Node 1 would be printed to the backup serial port on Node 2 and vice versa.
- The Voice Processing Unit will buffer SMDR records that are sent from other nodes in the network.

53.13 Selectable SMDR options include any combination of the following. (See page 5-305 for a definition of call cost types. See page 6-276 in PROGRAMMING for programming information.)

- **Record All Incoming Calls**: Records all incoming calls that are answered, except DID/DNIS calls (not applicable to Europe).
- **Record All Local Calls**: Records all calls that use the "local" call cost.
- **Record All Free Calls**: Records all calls that use the "free" call cost.
- **Record All Ring-in Diagnostics**: A ring-in message is recorded for every incoming call (whether answered or unanswered) to indicate how long it rang.
- **Record All Toll Local Calls** *(U.S. Only)*: Records all calls that use the "toll local" call cost.
- **Record All Toll Long Distance Calls** *(U.S. Only)*: Records all calls that use the "toll long distance" call cost.
- **Record All Toll (National) Calls** *(Europe Only)*: Records all calls that use the "national" call cost.
- **Record All Operator Calls**: Records all calls that use the "operator" call cost.
- **Record All International Calls**: Records all calls that use the "international" call cost.
- **Record All DISA Calls**: Records all DISA calls.
- **Record All Conference Calls**: Records all conference calls. If all conference parties exit except one station and one outside party, the call is no longer considered a conference and when completed it will be recorded using the appropriate call cost for that outside call.
- **Record All DID/DNIS Calls** *(U.S. Only)*: Records all calls received through DID and DNIS.
- **Record All Trunk To Trunk Calls**: Records all calls made from one outside caller to another.
- **Record All Network Calls**: Records all calls placed to and received from network nodes.
- **Record All Off-Node Devices**: Records all calls placed to and received from off-node devices. (This flag has no effect on the Record All Network Calls flag described above.)

53.14 With a U.S. system, the programmer can determine whether absorbed digits, equal access digits, and/or toll field digits appear in the SMDR printout. As an example, assume the following number was dialed: 89 (other system's trunk access code) - 10288 (equal access code) - 1 (toll field) - 602 (area code) - 961-9000 (seven-digit number). The following programming options can be used:

- **Suppress Absorbed Digits**: Absorbed digits (on local or PBX lines) will not appear in the report if this option is selected. In the sample above, the absorbed digits (the other system's trunk access code) would be suppressed so that only 10288- 1-602-961-9000 appears.

  NOTE: If absorbed digits are repeatable on a local line, the absorbed digits will not appear in the SMDR report even when repeated.

- **Suppress Equal Access Digits**: Equal access digits will not appear in the report if this option is selected. In the sample above, the equal access code field would be suppressed to print only 89-1-602-961- 9000.

- **Suppress Toll Digits**: When this option is selected, toll digits will not appear in the report. In the sample above, the toll field would be suppressed so that only 89-10288-602-961-9000 prints.

53.15 With a European system, the programmer can determine whether absorbed digits and/or toll field digits appear in the SMDR printout. As an example, assume the following number was dialed: 89 (other system's trunk access code) - 01 (toll field) - 933 (STD code) - 402332 (number). The following programming options can be used:

- **Suppress Absorbed Digits**: Absorbed digits (on local or PBX lines) will not appear in the report if this option is selected. In the sample above, the absorbed digits (the other system's trunk access code) would be suppressed so that only 01933402332 appears.

NOTE: If absorbed digits are repeatable on a local line, the absorbed digits will not appear in the SMDR report even when repeated.

- **Suppress Toll Digits**: When this option is selected, toll digits will not appear in the report. In the sample above, the toll field would be suppressed so that only 89-933402332 prints.

53.16 Any combination of the above can be used. If all three fields are suppressed, only 602-961-9000 (or 933402332) will print.

53.17 The programmer can also suppress or allow call information in the SMDR report "Dialed Digits" field. The following options are available:

- **Suppress Outside Party Number**: Caller information that is received through ANI or Caller ID [CLIP] will not appear if this option is selected.
- **Suppress Trunk Number**: Information received through DID or DNIS [DDI in Europe] will not be included in the report if this option is selected.

53.18 The programmer can determine which equipped station(s) and/or trunks will be included in the report, and whether off-node devices will be included in reports. However, for incoming calls, DISA calls, conference calls, and/or ring-in diagnostics, all calls are recorded even when they involve stations not in the programmed station list.

53.19 SMDR can be programmed to record the elapsed time of calls in seconds (S=XXXXXX) or hours and minutes (HH:MM). If programmed to record elapsed time in seconds, the ELAPSED TIME field will show "S=XXXXXX" (XXXXXX represents the number of seconds) for calls up to 999999 seconds long. For calls lasting longer than 999999 seconds, ELAPSED TIME will show "HH:MM" (hours and minutes rounded up to the nearest minute).

53.20 The SMDR output report is printed as shown in Figure 5-12 on the following page. A page heading (with the day of the week, date, month, year and column headings) is generated just after midnight to show the change in date. A header is also printed after output from another system source (alarm, informative message, etc.), using the same output device, interrupted the SMDR output.

Figure 5-12. *SMDR Report Format*

```
TYP   EXT# TRUNK      DIALED DIGITS          START    ELAPSED   COST     ACCOUNT CODE

XX                    XXXX       XXXXXXX                                          <28
   characters>                                                                     HH:MM
   HH:MM:SS                                          $XX.XX
   XXXXXXXXXXXX*
```

TYP Call type abbreviations for:

For U.S. Systems:	
Free	000
Local	LOC
Toll Local	TLC
Toll Long Distance	TLD
Operator and International	O/I*
DISA	DSA
Conference	CNF
DID/DNIS (non-DISA)	DID
Incoming	IN
Network	NET

For European Systems:	
Free	FOC
Local	LOC
Toll (National)	NAT
International	INT
Operator	OP
DISA	DSA
Conference	CNF
Incoming	IN
Network	NET

* If the Display "O/I" for Operator and International Calls flag is disabled, operator calls are displayed under "OP" and international calls are displayed under "INT." See page 6-277 for details about this flag.

EXT The extension number (XXXXX) of the last station to handle the call is shown. For a CO-to-CO call, this field shows the extension number of the second trunk involved. For a conference call, it shows the station that initiated or answered the call. For an unanswered ring-in, it shows *****. An outgoing call that has been initiated by another trunk will show the initiating trunk's number.

TRUNK The extension number of the trunk used during the call is shown. For an IP network call (using the IPRC), this field shows the extension number of the IP connection used for the call.

DIALED DIGITS **For an Outgoing Call**: The first 28 digits of the telephone number are shown (if ARS was used to place the call, the modified number, not the dialed digits, are shown). A ">" at the end of the number indicates that more than 28 digits were dialed. Some digits may be suppressed (see the previous page for an explanation). For a conference call, this field shows the station that brought the trunk into the conference unless the conference ends as a call with only one station and one trunk or if all conference parties are put on individual hold (in which case the field shows the last party to handle the call).
For an Incoming Call: This field is determined by the service type of the trunk that was used for the call and whether the digits are being suppressed. If the information is not suppressed, it is included in the report as follows.

Figure 5-12. *SMDR Report Format (Continued)*

For U.S. Systems:

Service Type	Call Record Shows:	Ring-in Record* Shows:
DID, DNIS, or *DNIS* (not DISA ring-in)	Trunk DID or DNIS Information	Ring... Trunk *DID or DNIS Information*
DID, DNIS, or *DNIS* with DISA ring-in	Blank if answered; DISA digits if unanswered	DISA calls do not generate a ring-in record
Caller ID [CLIP], ANI, or *ANI*	Outside Party ANI or Caller ID [CLIP]	Ring...*Outside Party ANI or Caller ID [CLIP]*
*ANI*DNIS*	*Outside Party ANI or Caller ID [CLIP]* **and** *Trunk DID or DNIS Information*	Ring...*Outside Party ANI or Caller ID [CLIP]* **and** *Trunk DID or DNIS Information*
None	Blank	Blank

* Ring-in records appear only if ring-in diagnostic records are enabled as described on page 5-308.

For European Systems:

Service Type	Call Record Shows:	Ring-in Record* Shows:
DDI (not DISA ring-in)	Trunk DDI Information	RG... Trunk *DDI Information*
DDI with DISA ring-in	Blank if answered; DISA digits if unanswered	DISA calls do not generate a ring-in record
Caller ID [CLIP]	Outside Party Caller ID	RG...*Outside Party Caller ID*
DDI and Caller ID	*Outside Party Caller ID* **and** *Trunk DDI Information*	RG...*Outside Party Caller ID* **and** *Trunk DDI Information*
None	Blank	RG.....

* Ring-in records appear only if ring-in diagnostic records are enabled as described on page 5-308.

START Shows the time that the call became valid (see paragraph 53.9 on page 5-308 for an explanation). For an unanswered ring-in, it shows the time the call began ringing. It is shown in 24-hour time (00:00 - 23:59).

ELAPSED Shows the call length from the START time (above) until disconnect or the length of time an unanswered call was ringing. If the option is enabled that shows call duration in seconds, calls up to 999999 seconds long appears as S=XXXXXX (XXXXXX represents the number of seconds) and calls lasting longer than 999999 seconds appear as HH:MM (hours and minutes).

COST *For U.S. Systems:* Displays an estimated cost of the call ($XX.XX), based on the database information. If cost exceeds $99.99, it is printed without the decimal point ($XXXXX). If it exceeds $99999, $$$$$$ is shown instead. If there is no cost, the field is blank.

For European Systems: Displays an estimated cost of the call (XX.XX), based on the database information. If cost exceeds 99.99, it is printed without the decimal point (XXXXX). If it exceeds 99999, XXXXX is shown instead. If there is no cost, the field is blank.

ACCOUNT CODE Shows the standard, forced, or optional account code (up to 16 digits). An optional account code overrides standard or forced account codes. The field is blank if no account code was used.

NOTE: If an asterisk (*) appears at the end of the record, as shown in the sample, it indicates that the call was terminated because the system detected a disconnect signal from the trunk (that is, the caller hung up before the station user on the call).

C. SYSTEM ERROR/MESSAGE PRINTING

53.21 The system has a self-diagnostic feature that monitors the system. When a system fault or other noteworthy condition is detected, the software determines the impact of the condition and classifies it as an alarm (major or minor) or other message type. If the Message Print option is enabled, as described on page 6-274 in PROGRAMMING, the messages are sent to the designated output port. For a complete list of Diagnostics codes, refer to the *Inter-Tel Message Print Diagnostics Manual*.

NOTE: As of version 8.0, a utility, called Diagnostics Monitor, is automatically installed with DB Programming. This application allows you to view Message Print, SMDR, and other information. For details about Diagnostics Monitor, see page C-30.

Message Print Types

53.22 The programmer may enable any combination of the error message types. The available message types are:

- **Information Messages**: Give information concerning system operation.
- **Warning Messages**: Indicate that a condition exists which may affect system performance.
- **Alarm Messages**: Indicate that a minor alarm has occurred, but that general system operation was not affected.
- **Severe Messages**: Indicate that a severe error has occurred in the system.

53.23 The messages can be programmed to be sent to any output device attached to the system. This includes devices attached to the CPC, CPU card, Options cards, and phone data ports. Messages can be printed through only one system port at a time. The print width is adjustable (64, 80, or 132 characters) to accommodate different output devices. When a line of information is longer than the programmed width, the information is printed on subsequent lines and preceded with "+" to indicate the continuation.

53.24 The system database also has a reserved area for recording the history of messages to allow service personnel to request a report at any time (see page 6-274 in PROGRAMMING). The message history area has enough memory to maintain a record of at least 50 of the most recently generated messages. When the reserved area is full, the more recent messages begin to replace the older ones so that the most recent messages are available.

53.25 In a network setting, each node has its own message print programming, message print output port, and message print output port backup. There should be a message print terminal at each node to monitor node and network performance and aid in troubleshooting.

- If a node's Message Print output port is a node, the network sends Message Print records to the specified node.
- You cannot select a node as the message print output port backup.
- If Message Print output programming forms a loop, the system will send the output to the node's backup Message Print port. For example, if the Message Print port on Node 1 routes to Node 2 and the Message Print port on Node 2 routes to Node 1, the configuration will cause an infinite loop. Message Print reports for Node 1 would be printed to the backup serial port on Node 2 and vice versa.

- A two digit node number is inserted in front of the sequence number to indicate the node where the message originated. The following is an example of the new message print string format for a message the originated on node 3:

  -03:001- 15:30 03-20 M6009 WRN EG CP
  Corrupted Queue Was Cleared: HISTORY QUEUE

- Message Print will ***not*** route diagnostic messages that apply to another node, except during manual Message Print requests. Those messages will only be printed locally.
- Message Print will not route messages during a startup or reset. The system will wait one minute before starting to send messages. Startup messages will only be printed locally.
- Messages may not be printed in time sequence order on remote nodes. The order of an incoming message print depends on how the message print was routed to the destination node. Therefore, a message print could possibly arrive at the destination message print node out of order.

53.26 In addition to the messages, the following error indications will occur in the event of a system alarm:

- If a major system alarm occurs, MAJOR ALARM appears on all display phones. If the major alarm appears on a group of phones, the associated keyset card has failed. The warning might also appear on a single phone if the phone is defective. If all phones on the system are inoperative, the alarm message is sent to the system's primary serial port on the CPC or CPU.
- A minor alarm sends a message to the primary attendant and to administrators, if programmed to receive alarms.

53.27 Each message record includes the following fields:

- **Sequence Number**: Shows the order in which the faults occurred. If there is a skip in a series of sequence numbers, it indicates that a message was logged that is not programmed to be printed.
- **Time**: Shows the time that the fault occurred. This field shows the hours and minutes in 24-hour format as HH:MM.
- **Date**: Shows the date that the indicated fault occurred. The date shows the month and date as MM-DD.
- **Fault Code**: Holds the associated number of the indicated fault. This field is four characters in length and the first character is an "M" for messages or "A" for minor alarms. Each of the different system faults that the system is capable of detecting is assigned a unique number (M100-M999 or A001-A299) in the system database. Fault codes are listed on page 7-12 in TROUBLESHOOTING.
- **Type**: Indicates the following:
  - — *Alarms:* Shows "ALARM" for system faults that have not stopped system operation.
  - — *Information Messages:* Shows "INF." The condition requires no attention.
  - — *Warning Messages:* Shows "WRN." The condition requires attention because it may affect system performance.
  - — *Severe Messages:* Shows "SVR." The condition requires attention because it has affected or will affect system performance.

- **Realm**: *(Does not appear for Alarm messages.)* Indicates the realm responsible for the message. The field will contain one of the following abbreviations:
  - — *DB* indicates that the message applies to the system database. These problems can often be fixed with appropriate DB Programming.
  - — *HW* indicates the messages concerns the system hardware. Typically these problems are corrected by adjusting or replacing the affected hardware.
  - — *EG* is indicative of a software error which will, most likely, require the intervention or attention of an Inter-Tel software engineer.
- **Source**: *(Does not appear for Alarm messages.)* Indicates the software source responsible for the message.
  - — *CP* indicates that Call Processing is the source.
  - — *OS* refers to the Operating System.
  - — *SS* refers to the System Software
  - — *AL* refers to Voice Processing Unit link.
- **Text**: Holds a text message that identifies the particular fault indicated by the record. This field is at least 40 characters in length and may extend onto subsequent lines if necessary to present adequate information about the fault. The information printed on an additional line is preceded with the character "+" to indicate the continuation.
- **Auxiliary**: Includes additional information that may be printed as part of a message on separate lines. This additional information is referred to as "auxiliary text." To identify this text as pertaining to the previous message, each line is preceded with the "<"character.

History and Message Print Queues

53.28 The new CPC contains more RAM than the current CPU cards, which increases the capacity for the history and Message Print queues. If using a CPC, the history queue can store up to 4 MB of information, as opposed to 260 KB on an older 512CPU. In addition, the Message Print queue can store up to 300 KB, instead of 130 KB.

53.29 Because downloading 4 MB of history queue information over a modem would take too long, the 4 MB is separated into 500 KB blocks. Each of these blocks contains start and stop time/date stamps, allowing you to determine which block contains the required information for reviewing site issues.

53.30 Each block is given a unique name for identification purposes. The format for the block filenames is **`N<NodeNumber>_<Date>_<StartTime>_<StopTime>_<BlockNumber>.cph`** where:

- <*NodeNumber*> is the number of the node.
- <*Date*> is the date the block information was recorded. This is in the format *MMMDD* (e.g., Nov18).
- <*StartTime*> is the start time for the 500K block. This is in 24-hour format and is displayed as HHMM (e.g., 1310).
- <*StopTime*> is the stop time for the 500K block. This is in 24-hour format and is displayed as HHMM.
- <*BlockNumber*> is the file number for the block of information. Block numbers begin at "1" whenever a system freeze is initiated.

NOTE: Because the Message Print queue stores only 300 KB of information, only one Message Print **.log** file is saved. This file is always assigned "1" as the <*BlockNumber*>.

53.31 For example, the filename **N1_Apr14_0845_0850_1.cph** represents the first file in the history queue for Node 1. This file includes freeze information for April 14 between the hours of 8:45AM and 8:50AM. Subsequent files for this history queue would be **N1_Apr14_0850_0900_2.cph**, **N1_Apr14_0900_0910_3.cph**, etc.

53.32 Because the blocks are based on a limited amount of memory (i.e., 500K), the time intervals for each file may not be the same. For example, if the system is extremely busy, two files may reflect the same time interval. If this occurs, the only difference between the file names would be the <*BlockNumber*> (e.g., **N1_Apr14_0950_0950_1.cph** and **N1_Apr14_0950_0950_2.cph**).

53.33 Also, if the time changes (e.g., for daylight savings time) during a system freeze, one block may appear to have the start and stop times reversed. For example, you may see **N1_Apr14_0100_0010_2.cph** followed by **N1_Apr214_0010_0020_3.cph**. This shows that the time changed at 1:00AM and that the second block of information was recording during this time. For information about freezing the system, see page 6-52.

D. POWER FAILURE CAPABILITIES

53.34 The Call Processing Card (CPC) and Central Processor (CPU) card contain relay contacts that can be used for power failure transfer which allows one or more single-line sets to select predetermined trunks in the event of a processor board failure or a power failure. Additional equipment is required. See SPECIFICATIONS, page 2-103, for more information.

E. CUSTOMIZED SYSTEM PROGRAMMING REPORTS

53.35 Customized system programming reports can be created through the database (see pages 6-82 and 6-339 in PROGRAMMING). The programmer can select the programming area to be reported and then select the information that will be included. When extension numbers are listed in a report, they can be sorted by name, extension number, or circuit number. Reports can also be titled.

54. DIAGNOSTIC FEATURES

54.1 System software v9.0 and later includes several diagnostics features used to diagnose call processing and audio problems and improve system reliability. This section describes the diagnostics features available for systems running software v9.0 and later.

A. ADMINISTRATOR ENDPOINT SUPPORT

54.2 For troubleshooting purposes, diagnostics information is available from an administrator's endpoint. You can enable and disable diagnostics functions and also dump system diagnostics information to Message Print from an administrator's endpoint.

NOTE: Inter-Tel recommends that you contact Technical Support for assistance with diagnostics issues.

B. SYSTEM DIAGNOSTICS

54.3 The following diagnostics features are available for systems running software v9.0 and later. These features are included in the system features troubleshooting table on page 7-27.

54.4 **Resource Manager CPH Diagnostics flag**: The Log Resource Manager Output flag exists in the System\Maintenance\History Queue folder to enable or disable the resource manager Call Processing History (CPH) diagnostics feature. The Call Processing history is a collection of inputs/outputs used to diagnose problems that occur on the system. This flag can only be accessed in On-Line Monitor (OLM) mode. By default, this flag is set to **No**.

NOTE: Do not use OLM mode unless you are instructed to do so by support personnel.

54.5 **Call Processing Server (CPS) Call Processing History (CPH) Freeze File Compression**: When a system freeze is performed on a CPS, a snapshot of the memory-mapped files is taken and copied into a freeze directory. The directory contains a new Freeze_00 file which is always the most current. The CPS CPH Freeze File Compression feature zips the freeze files into a compressed file that includes the node number, start and end freeze date and time, and CPS capacity in the zip file name. For example, "n12_512_06-02-04_123010_06-02-04_160023_freeze.zip" represents a node 12 system freeze occurred on a CPS 512 on June 2nd, between 12:30 pm and 10 seconds and 4:00 pm and 23 seconds.

54.6 The time stamp includes hours, minutes, and seconds to handle situations where the CPS freezes multiple times in the same minute. The zipped freeze file resides in the C:\ICP\FREEZE directory on the CPS. When a freeze occurs on the CPS, the compression utility appears in a console window. After one to two seconds, the console window minimizes automatically (high system utilization may increase the delay before the system minimizes the window).

NOTE: When Automatic Diagnostics Delivery (ADD) is enabled on the CPS, the system sends the single zipped file instead of the collection of freeze_00 files that were sent prior to v9.0. For ADD setup information, see Knowledge Base answer ID 219.

54.7 For CPS systems upgrading to v9.0 or later, be aware that the compression utility has an upper limit of 2 GB. Because of this limitation, Inter-Tel recommends that pre-9.0 CPS sites backup or remove the "Freeze" file contents before installing the 9.0 CPS software.

54.8 System Diagnostics Dumps: Feature codes used to dump system device information have been added for v9.0 and later. The default feature codes used to dump system diagnostics information are 9933 (for the U.S.), and 9133 (for the U.K.). These diagnostics feature codes allow a system administrator to dump system diagnostic information to Message Print. For more information about diagnostics features for v9.0 or later, see page 5-317.

54.9 Clearing CPC History Queues/Log Files: Version 9.0 and later system software allows you to clear a CPC or CPS's history queue to baseline a site for monitoring purposes. To clear the CPS history queues and log files, move/delete the Memory Map Files (**`.mmf`**) on a CPS, but do not delete the static_**`.mmf`** file. To clear the CPC history queues and log files, use the Clear Error Information option in DB Studio (in On-Line Monitor mode). When this operation is performed, the system generates a new Message Print "M6099 Corrupted Queue was Cleared" that confirms when the queue or log file has been cleared.

NOTE: Do not use OLM mode unless you are instructed to do so by support personnel.

54.10 Traceroute Functionality Between an IPRC and CP Card (CPC): Version 9.0 and later system software supports the traceroute utility to test the communication path to any IP address in the network. For example, a technician could perform a traceroute from the IPRC to an IP endpoint to confirm the IPRC can communicate with it. The technician could also run a traceroute to any IP device in the network (a PC, some public IP address, etc.). The traceroute commands lists the IP addresses of each router that exist along the path from the local host to the given IP address (destination host on the network).

54.11 The method for acquiring this information relies on ICMP (Internet Control Message Protocol) Echo and Time Exceeded messages which some routers are programmed to not send or ignore. The traceroute will list a "no reply" when it encounters a router that ignores the request, or when the IPRC is not communicating with the CPC. A traceroute command is available from the IPRC with v9.0 or later firmware and CPC On-Line Monitor consoles.

54.12 To run the traceroute command: Select the "Debug Session" option from an RS-232 or telnet console. At the prompt, invoke the command by entering "tracert" followed by an IP address in numeric form at the prompt.

C. AUDIO DIAGNOSTICS

54.13 The Audio Diagnostics feature (default feature code 320) included with system software v9.0 and later allows a user to generate diagnostics information for audio problems. After the Audio Diagnostics feature is initiated, the user is prompted to answer questions about the audio problems by pressing the associated buttons on his or her endpoint. Based on the user's selections, the system generates alarm 128 (see page 7-17) which appears on the administrator's endpoint. The collected data is then sent to Inter-Tel Technical Support via Automatic Diagnostics Delivery (ADD) if enabled.

NOTE: By default, ADD is not enabled.

54.14 The Audio Diagnostics feature can be accessed when the endpoint is idle or when the user is on an active call. The amount of diagnostics information that the endpoint provides to the system depends on the state of the endpoint when the feature is accessed. For example, an active call produces more diagnostic information than a endpoint in an idle state. If users do not want to access the Audio Diagnostics feature while on a call, they can access the feature immediately after they hang up.

NOTE: The Audio Diagnostics feature is not available with System OAI Display Control because the external applications have control of the menu buttons and dialpad.

54.15 To enable audio diagnostics for a station, set the Audio Diagnostics flag (System\Devices and Feature Codes\Stations\Local\<*extension*>\Flags to **Yes**. If the user accesses the Audio Diagnostics feature but does not respond to the prompts on the display, the feature times out after 30 seconds. If the user or the other party terminates the call before completing the diagnostics, the feature is terminated. When the feature times out or is terminated, the diagnostics information is not captured.

54.16 To program audio diagnostics for the system, see page 6-304. For troubleshooting information, see page 7-27.

Audio Problem

54.17 When the Audio Diagnostics feature (default feature code 320) is used, the system prompts the user to select a possible audio problem. Six-line display endpoint users can either press the menu button that corresponds to the displayed audio problem or enter the numeric codes. The 2-line display and non-LCD display endpoint users must enter the numeric codes because they do not have menu buttons.

54.18 The following table shows 12 audio problems and their assigned numbers. Although there are 12 pre-defined audio problems as shown in the table below, only four can be enabled in DB Studio. In v9.115 or later system, the first four audio problem numbers are selected by default.

Table 5-46. *Audio Diagnostics*

AUDIO PROBLEM	NUMBER	AUDIO PROBLEM	NUMBER
Echo	01*	Too Loud	07
Static	02*	Distorted	08
One-Way Audio	03*	Delayed Audio	09
No Audio	04*	Clipping	10
Choppy Audio	05	Cross-Talk	11
Too Soft	06	Other	12

* Enabled by default.

54.19 For instructions on how to use the Audio Diagnostics feature, see page 5-320.

Audio Direction

54.20 After selecting the audio problem, the system prompts the user to choose the direction of the audio problem. Options include:

- Only I hear it (2-line display and non-display users press **1**)
- Only the outside (2-line display and non-display users press **2**)
- We both hear it (2-line display and non-display users press **3**)

Record-A-Call

54.21 If the Record-A-Call feature has been programmed for the station, the user can record the call while using the Audio Diagnostics feature (see page 5-320).

NOTE: The Record-A-Call feature cannot be used on certain calls such as Agent Help, Station Monitor, Conferences, Paging, etc.

Data Collection

54.22 To collect the diagnostics data, simply retrieve the Freeze information or send the data to Technical Support via ADD. The diagnostics data that is collected consists of:

- Source endpoint (extension and board number)
- Destination endpoint (extension and board number)
- Phone number
- Echo canceller settings (IPRC)
- Resource manager dump
- Crosspoint/voice channel information
- Current volume levels (near end)
- Hybrid balance values
- De-coupling values
- IPRC channel used
- IPRC session used
- Network group

Using the Audio Diagnostics Feature

54.23 Only stations with the Audio **Diagnostics station flag** enabled (System\Devices and Feature Codes\Stations\Local\<*extension*>\Flags) can use the Audio Diagnostics feature.

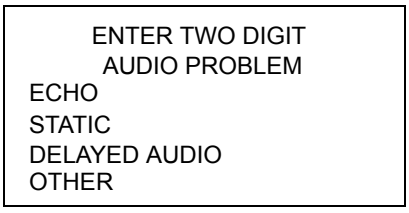
54.24 To use the Audio Diagnostics feature:

NOTE: Only 6-line display endpoints have the menu buttons mentioned in the following instructions. The 2-line display and non-display endpoint users require a list of problems and descriptions to use the Audio Diagnostics feature. Phantom devices and modems cannot use the Audio Diagnostics feature.

1. ***If the endpoint is idle,*** while on hook enter the Audio Diagnostics feature code (320).

   If you are on an active call, press ∞ then enter the Audio Diagnostics feature code (320).

   The display shows ENTER TWO DIGIT AUDIO PROBLEM.



2. Do one of the following to select an audio problem:

   EITHER, Press the desired menu button.

   OR, enter the two-digit numeric code (01-12) that corresponds to the audio problem (see page 5-319).

NOTE: Consult your system administrator for the list of Audio Diagnostics codes that are used on your system.

The display shows PLEASE SELECT THE AUDIO DIRECTION.

3. Select the direction of the audio problem.

   EITHER, Press the desired menu button.

   OR, press the dialpad button that corresponds to one of the following responses:

   - Press **1** for ONLY I HEAR IT.
   - Press **2** for ONLY OTHER PARTY.
   - Press **3** for WE BOTH HEAR IT.

   PLEASE SELECT
   AUDIO DIRECTION

   ONLY I HEAR IT
   ONLY OTHER PARTY
   WE BOTH HEAR IT

   If the Record-A-Call feature is enabled for your endpoint, the display shows WOULD YOU LIKE TO RECORD CALL?

4. If the Record-A-Call feature is available, you have the option of recording the call.

   EITHER, Press the desired menu button.

   OR, press the dialpad button that corresponds to one of the following responses:

   - Press **1** to record the call.
   - Press **2** to continue without recording.

   WOULD YOU LIKE
   TO RECORD CALL?

   YES
   NO

NOTE: If the system cannot accurately record the call (e.g., because different cross-point connections are used) the display shows CALL CANNOT BE COMPLETED.

5. Hang up to complete the Audio Diagnostics feature. System Alarm 128 is generated and displayed on the System Administrator's endpoint.

Responding to the Audio Diagnostics Feature

54.25 After the user completes the Audio Diagnostics feature, the system generates Alarm 128 and the administrator station displays SYS ALARM #128 <EXT> AUDIO FRZ.This alarm indicates the extension that generated the Audio Diagnostics feature.

SYS ALARM #128
<EXT> AUDIO FRZ
CLEAR ALARM
CLEAR ALL ALARMS

IDLE MENU

54.26 To respond to Alarm 128:

54.27 Alarm 128 indicates that someone has completed the Audio Diagnostics feature, and you need to collect the freeze that contains Message Print entry. Review the freeze diagnostics data. If you need further assistance, submit the data to Technical Support for analysis.

54.28 For more information about collecting diagnostics data, contact Inter-Tel Technical Support.

D. RELIABILITY AND ROBUSTNESS

54.29 The following features are included with system software v9.0 and later. These features are included in the system features troubleshooting table on page 7-27.

54.30 Periodic Diagnostics: Monitors and reconciles system resources and checks for and fixes static database corruptions. This option in DB Studio is available in On-Line Monitor (OLM) mode only. Periodic Diagnostics is responsible for two areas of the system: system resources and the static database. Periodic Diagnostics checks/reconciles/repairs the following:

- DTMF receivers
- Caller ID receivers and transmitters
- Speakerphone resources
- Connection IDs
- Voice channels
- VoIP resources
- System hardware
- Static database
- Off-node device information
- Unassociated mailbox information
- Temporary extensions

NOTE: Do not use OLM mode unless you are instructed to do so by support personnel.

54.31 Enhanced Exception Handling: For v9.0 and later, critical areas of the system have enhanced exception handling to prevent the system from performing a major reset. Any input message to the hardware system that causes an exception to occur is now deleted. Similarly, any message that is processed by Call Processing which causes a software exception, is also deleted. Although these enhancements made to exception handling may cause devices to time out waiting for responses, it reduces the occurrence of major resets.

NOTE: The system may still perform a major reset if multiple minor resets occur and the system cannot overcome them.

54.32 Elimination of Scheduled Delayed Major Resets: Prior to v9.0, when certain programming changes were made to the database, the system required a major reset. If the programming changes were done via DB Studio, a pop-up dialog box displayed the reset options. If the programming changes were done via an administrator's endpoint, the reset options were presented on the endpoint's display. The technician or administrator could choose to complete an immediate major reset or schedule a delayed major reset. The delayed major reset option allowed the technician or administrator to schedule a reset to occur when the system was not in use.

54.33 In v9.0 and later, certain programming changes no longer require a major reset. These programming changes include the following:

- Changing extensions
- Swapping extensions
- Creating off-node devices

- Changing ACD Agent IDs
- Adding and removing extensions from Extension Lists
- Equipping and un-equipping devices

NOTE: Because it is no longer necessary to complete a reset for the programming changes listed above, DB Studio no longer displays the reset pop-up dialog box and the administrator's endpoint no longer displays the ENTER SYS RESET OPTION after the user completes the changes.

54.34 Addition of System Software Performance Statistics: The Software Performance Statistics log contains software availability information. This information includes detailed performance and reset statistics. The Software Performance Statistics Log is maintained across versions on both the CPC and CPS, and the information in the statistics log remains intact when the software is upgraded.

54.35 By default, the system is configured to log software performance statistics. The information included in the log is used by Inter-Tel Technical Support. The Internal Statistics Logging flag under System\Flags in OLM mode enables/disables this function. The default setting for this flag is **Yes**.

NOTE: Do not use OLM mode unless you are instructed to do so by support personnel.

E. DIAGNOSTICS MONITOR SUPPORT

54.36 The latest version of Diagnostics Monitor includes support for the v9.0 and later diagnostics features.

54.37 Dump Extension: SIP trunks is included in the Dump Extension option (System Monitor\Dump Extension).

54.38 Dump Resource Managers: The following dump resource managers commands (System Monitor\Dump Resource Managers) allow you to quickly view information without having to log in by DB Studio. The v9.0 and later commands consist of the following:

- **Call Configuration**: Displays the IP call configuration values in the system.
- **Call Diagnostics**: Displays a snapshot of the active calls in the system.
- **Caller ID Resources**: Displays the number of caller ID receivers and transmitters allocated on the system.
- **IP Port Device Manager**: Displays the list of unavailable IP port devices, IPRCs that are off-line (programmed, but not plugged in), available IP Port devices, IPRCs that are on-line, and the number of networking sockets currently established with each IPRC.
- **License Manager**: Provides a list of SIP trunks that have a SIP trunk license.
- **Network Groups**: Displays all the IP networked groups in the system.
- **Network Status Module**: Displays the entire network status module. The Network Status Module dump determines which nodes in the network are running at which versions and which nodes do not have the "ENABLE 99 NODES" license loaded.
- **Off-Node Device Container**: Displays all of the off-node devices.
- **Periodic Diagnostics Monitor**: Displays information for the periodic diagnostics feature.
- **Phantoms**: Displays all the phantoms in the system.
- **SIP Gateway**: Displays all the SIP Gateways in the system.
- **System Device Information**: Displays the number of devices in the system.

54.39 Dump Logs: The following dump log commands (System Monitor\Dump Logs) support the v9.0 or later diagnostics enhancements:

- **Network Diagnostics Log** (previously named Network Print Log): Displays specific network information.
- **Software Performance Statistic Log**: Displays system software performance statistics.

Programming

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PROGRAMMING OVERVIEW

1. INTRODUCTION

1.1 This section of the manual explains how to program the system. The Axxess® system is functional when defaulted and only requires programming to make changes to meet the customer's special needs.

NOTE: Programmers must be certified on the Axxess system to receive technical support.

A. PLAN THE PROGRAMMING SESSION

1.2 Determine the features that need to be programmed to meet the customer's needs. Then refer to the specific programs and program planning sheets. For example, if the customer wants to use the hunt group feature, see the programming information on page 6-160 and the corresponding program planning sheet. Program planning sheets are included on the system software CD-ROM, Documentation folder. You can also find all documentation on the Inter-Tel *edGe* Web site. For readers located in Europe, contact the Inter-Tel Europe office for technical documentation.

B. INITIAL PROGRAMMING GUIDELINES

1.3 When you first program an Inter-Tel® system, follow these guidelines for the initial programming. (Detailed descriptions for these steps can be found throughout this chapter.)

1. With your programming PC connected to the serial port of the phone system, start the Session Manager.
2. Do the following:
   a. Select **System** and then **Software License**. Then make sure the premium features that the customer has purchased are enabled or available on the software license. (See page 6-97.)
   b. Select **System** and then **Cabinets,** and program the cards that are installed in the system. (See page 6-107.) Delete any cards you will not be using.
   c. Disable the Voice Processor port if you will not be needing it. (See page 6-107.)
3. Select **Devices** and assign your extension numbers, as determined by your numbering plan. Also change any feature codes that conflict with extension numbers or that the customer wants to change. (See page 6-156.)
4. Change the home area codes (not applicable to Europe). (See page 6-286.)
5. Program the Voice Processor:
   a. Create a Message Notification/Retrieval application and a Voice Mail Application. Make sure they have the correct extension numbers. (See page 6-369.)
   b. Create a System Administrator's mailbox for the station that will serve as the Voice Mail administrator. You can come back later to program the rest of the mailboxes or you can do it now. (See page 6-377.)
   c. Assign the newly created System Administrator's mailbox as the System Administrator. (See page 6-345.)

6. Perform the remaining programming necessary to meet the customer's needs. This may include (but is not limited to) the following:

   - **Voice Processor**: Program other Voice Processor applications (including call routing and Fax-On-Demand), extension IDs, group lists, mailboxes, Voice Processor options, timers, custom recordings, and system information. (See page 6-104.)
   - **IP Private Networking**: Program node IP connection groups and IP connections if using an IPRC for private networking. (See page 6-172.)
   - **Devices**: Program individual hunt groups, page ports, nodes, stations, off-node devices, serial ports, trunk groups, node trunk groups, and trunks. (See page 6-156.)
   - **System-Wide Flags**: Program system flags to enable the necessary features. (See page 6-258.) Note that many installations will want to have the Handsfree Announce On System Forward Transfer flag disabled (it is enabled by default).
   - **ACD Hunt Group Agent IDs**: If needed, program the Agent IDs used in ACD hunt groups. (See page 6-263.)
   - **Maintenance**: Program call cost, freeze zones, message print, and SMDR information. (See page 6-268.)
   - **Numbering Plan**: Program toll restriction area flags, emergency number information, toll strings, classes of service, and user groups (user groups are not applicable to European systems). Also set up ARS route groups, dial rules, and facility groups. (See page 6-271.)
   - **Station-Related Information**: Program account codes, attendants, station flags, button assignments, message centers, do-not-disturb and reminder messages, primary attendants, system forwarding paths, and system speed dial. (See page 6-301.)
   - **Trunk Call Routing Tables**: Program call routing tables. (See page 6-319.)
   - **System Timers**: Program system timers, as needed. (See page 6-329.)

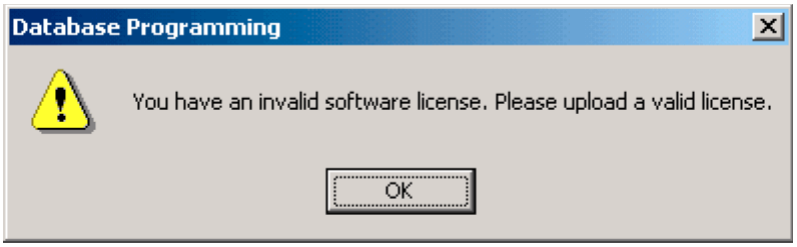
7. When you have finished programming, save your database. (See page 6-65.)

C. DETERMINE THE IP ADDRESS

NOTE: Inter-Tel recommends using the static IP address for the CPC (and CP Server). If using a BOOTP or DHCP server, the IP address may change, dropping any CT Gateway and/or Voice Processing connections.

1.4 If using a CPC, use the IP port for Database (DB) Programming. To use the port, however, you must first determine the IP address assigned to the CPC.

1.5 To determine the IP address, establish a serial or modem connection to the CPC. Then start a DB Programming session using Session Manager. Before DB Programming is initialized, the system checks for a valid software license. If you have not yet uploaded a software license (i.e., you are starting DB Programming for the first time using 11.0 software), the following message appears.



CAUTION

If the database is currently being backed up (see page 6-69), a warning message appears. If you click **Yes** to override the lock, the database backup is aborted. If you click **No**, the backup continues, and DB Programming is not initiated.

1.6 By default, the CPC is programmed to use a DHCP and BOOTP server to obtain an IP address. When powered up, the system tries to contact the BOOTP server first. If the system cannot contact the BOOTP server, it tries the DHCP server. If this fails, then the programmed static IP address (*192.168.200.201* by default) is used.

1.7 If desired, change the static IP address (see page 6-178) in DB Programming. Then reset the system. You can then use the address you entered as the IP address for the DB Programming session.

NOTE: Inter-Tel recommends that you establish a Message Print connection before you reset the system. This allows you to verify that the new IP address is assigned to the CPC.

1.8 If, however, you do not want to change the static IP address, you must determine the address that the DHCP or BOOTP server has assigned to the CPC. To do this, establish a Message Print connection, then reset the card. When the card is initialized, Message Print displays information similar to the following:

-01:030- 02:22 01-28 M2002 INF HW CP [8.6.1]: On-Line From CPC IP Port

-01:031- 02:22 01-28 *** CPC HW Address: [8.6.1] IP Address: 198.168.0.0

+Hostname: 'TESTCPC'

1.9 You can now close the current DB Programming session and establish a new session using the IP address displayed in Message Print.

NOTE: Inter-Tel strongly recommends using an IP connection to access DB Programming. IP connections are faster than serial connections.

D. PRIVATE IP NETWORKING PROGRAMMING GUIDELINES

NOTE: Prior to v9.0, the 8.x firmware IPRC could be used for private networking **or** for IP devices, as determined by the configuration chosen in DB Studio. The v9.0 and later firmware IPRC supports **both** private networking and IP devices on the same card.

ALSO: System software v9.0 and later provides a networking wizard to simplify the required programming for private networking. See page 6-87 for more information about the networking wizard.

1.10 The IPRC relies on a unique numbering plan to make connections across the network. This means that you must assign each IPRC in the network a unique extension that the other IPRCs must be able to recognize. For this reason, Inter-Tel strongly suggests that you use the P6XXY numbering convention, where *XX* is the node number and *Y* is the IPRC number. For example, if the IPRC is the second IPRC on node 1, the extension number would be P6012. It is also recommended that you complete the IP Networking Planning Sheet (available at the end of Appendix A—Networking) before you begin programming IPRC networking. If you do not follow a numbering convention and identify the extensions before you program the IPRCs, it will be difficult to successfully set up your network. (See page A-13 for a programming example.)

1.11 If using an IPRC for private IP networking, follow the guidelines below. See the reference pages for details.

NOTE: IP Networking Planning sheets are included in Appendix A — Networking for your convenience.

1. Program the equipment chassis slot where the IPRC is installed as an **IPRC** (see page 6-145).
2. Choose a "P6XXY" extension where *XX* is the node number where the IPRC resides and *Y* is the IPRC number.

   NOTE: The extension you assign to each IPRC must reflect the node where the IPRC resides. If it does not, it will be difficult to establish the IP connections between IPRCs on a multi-node system. This numbering convention also limits the number of IPRCs within a private network to 1000.

3. **Program the Voice Resources**: Program one or more voice resources as a *Networking* VoIP resource (see page 6-146).

   NOTE: While in local mode, DB Programming allows you to program up to 32 voice resources, regardless of the number of resources actually installed or the PAL configuration. The system ignores programming for voice resources rendered invalid by the PAL configuration.

4. **Program the IP Connection**: The IPRC screen contains a link to the IPRC's IP connection. Program the IP connection as described on page 6-172. Whenever the system creates an IP connection, the system broadcasts the IP information to the other nodes as a database update, and the new IP connection becomes an off-node device on the other nodes (provided the remote node does not block the database change such as when a DB Programming session is active on the remote node).

NOTE: This does not automatically configure the IPRC for use by the network. You must manually program networking parameters locally as well as on each remote node.

5. **Program the IP Connection for the remote node**: You can also create IP connections to represent IPRCs on other nodes within the network. To see these off-node IP connection fields, you must have a node created in System\Devices and Feature Codes\Nodes.
6. Create an off-node IP connection in the IP Connections field.
7. Enter the "P6XXY" extension of the off-node IP connection. Again, you must assign a unique extension to each IP connection within the network.

NOTE: When creating off-node connections and other IP-related extension numbers, Inter-Tel recommends using a numbering plan that associates the extension to the device and the node on which it resides. For example, the first IPRC on node 2 would be P6021 (P6 followed by the node number, then the IPRC number). The second IPRC would be P6022, etc.

8. Program a remote IP address, remote audio receive port, and remote listening port in the IP Connection screen (see page 6-176).
9. **Program the Node IP Connection Group**: The off-node IP connection screen also contains a link to the Node IP Connection Group field to which the IP connection belongs. The system automatically creates a node IP connection group for each remote node you program. A node IP connection group corresponds to an IP network connection between the local node and a remote node. Program the Node IP Connection Group field as described on page 6-185.

NOTE: Changes to the Node IP Connection Group parameters do not take effect until the next call. Changes do not affect existing calls.

E. IP DEVICES PROGRAMMING GUIDELINES

NOTE: Prior to v9.0, the 8.x firmware IPRC could be used for private networking **or** for IP devices, as determined by the configuration chosen in DB Studio. The v9.0 and later firmware IPRC supports **both** private networking and IP devices on the same card.

Programming IP Devices

1.12 This section describes how to program the v8.1 and later firmware IPRC for use with IP devices. Unlike the v1.x firmware device IPRC, you do not program the v8.1 or later firmware IPRC as a DKSC-16.

NOTICE

The IP SoftPhone requires v1.x firmware and is not supported by IPRCs that have v8.1 or later firmware. Other IP endpoints, such as the Model 8660 Phone, however, require the 8.1 or later IPRCs to support the new features. The Model 8602 IP softphone requires IPRC firmware v9.0.0 or later. For these reasons, Inter-Tel recommends that you do not connect IP SoftPhones and other IP endpoints to the same IPRC. If you do, you may lose feature or phone functionality.

1. Program the equipment chassis slot where the IPRC is installed as an **IPRC** (see page 6-145).
2. Select the IP extension used to access the IP port on the card and click **OK**. This extension must be unique.

   NOTE: Inter-Tel strongly recommends that you use a universal numbering plan when programming IP connections. The convention of P6nnc where *nn* is the node number and *c* is the IPRC number is recommended. For example, for the second IPRC on node 3, you would use P6032, and the first IPRC on node 2 would be P6021.

3. **Program the Voice Resources**: Program one or more voice resources as a *IP Endpoint* (see page 6-146).
4. Program the IP Connection information for the IPRC (System\Cabinets\<*IPRC*>\IP Connection) as described on page 6-172.

   NOTE: Inter-Tel recommends that you use the default port settings when possible.

5. Program the IP devices (see page 6-147).
6. Program the devices to use P2P audio, if desired. See page 6-183 for details.

Programming the Model 8602 IP Softphone

1.13 To program the Model 8602:

NOTE: A Category B software license (part number 840.0415) and a headset for audio are required to support the Model 8602. IP softphones cannot have attached devices, such as DSS/BLF units.

1. Ensure an Internet Protocol Resource Card (IPRC) is installed on the system.
2. Double-click the **Devices** under System\Cabinets\IPRC.

3. Select **IP Softphone** from the device option. Right-click on the screen and select **Create IP Softphone**. The Get IP Extension dialog appears.

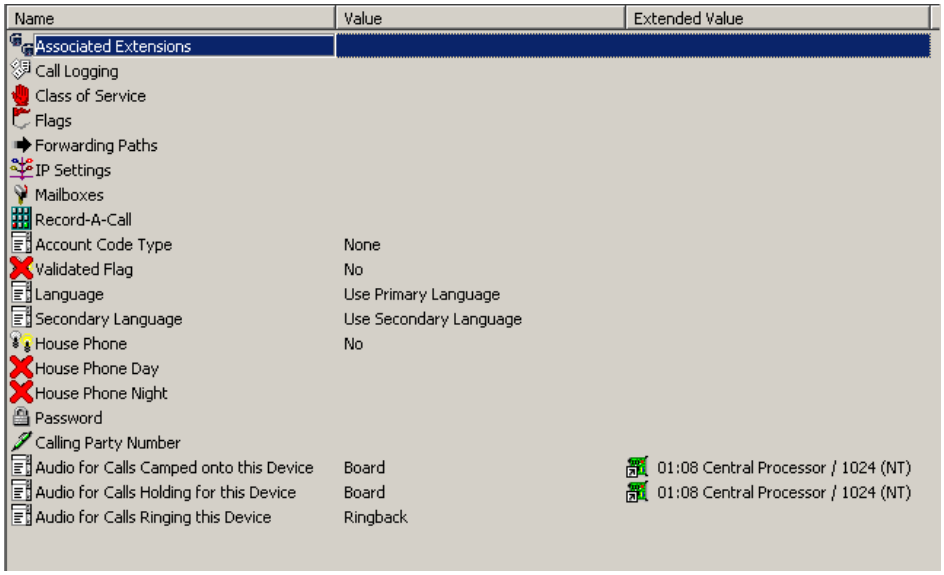
4. *EITHER,* Select a starting extension for the devices. Then click a device in the list and enter the appropriate device ID.

   OR, Click **Browse** to import the device IDs from a `.txt` file. (This opens the standard Windows Open dialog.) When you import the file, the device IDs are read from the top. This means that if there are six device IDs in the file, but there are eight devices, only the first six will be assigned an address. In addition, if there are eight device IDs in the file, but only six devices, only the first six addresses will be used.

NOTE: All device IDs must be unique. The user must choose a device ID that does not conflict with any other Inter-Tel IP endpoints configured into the system. Inter-Tel recommends that the administrator use a defined device ID scheme such as 86:02:00:xx:xx:xx. For troubleshooting purposes, Inter-Tel recommends using the extension number as the last digits of the Device ID.

5. Click **OK** to accept the extension and device IDs. The system then verifies that the IPRC has enough voice resources for the number of devices you programmed. If the number of devices exceeds the number of resources, an error message appears. Although the system allows you to program more devices than available resources, some IP devices may not come online unless the number of voice channels matches the number of IP devices. To ensure there are enough voice channels, make sure you have programmed all of the channels as VoIP. If all of the channels are programmed for VoIP, consider adding another IPRC to support additional devices you want to install.

6. Double-click the **IP Softphone** that you have just created. The device-specific fields are displayed. Program the following fields as described on page 6-202 or online help.



1.14 Keep in mind the following considerations when programming a Model 8602 IP softphone:

- The use of a password is highly recommended for all Model 8602 configurations to prevent unauthorized users from accessing the system using a valid Model 8602 device ID.
- The Model 8602 only supports American and British English. Other languages are currently not supported.
- The Model 8602 must not be configured in peer-to-peer network groups because of its mobility. If the Model 8602 connects to the system outside of the normal network, then peer-to-peer calls will fail.
- The use of RFC-2833 DTMF is recommended. The Model 8602's in-band DTMF usage (that is, G.711 and G.729 DTMF encodings) always follows the endpoint's speech encoding. Therefore, if the Model 8602 is configured to use G.729 speech encoding, then G.729 DTMF will be used if the Model 8602's DTMF encoding setting is either G.711 or G.729. The use of G.729 DTMF encodings will negatively impact the transmission of DTMF tones from the Model 8602 and will cause issues when trying to use Interactive Voice Response (IVR) type systems. The DTMF Encoding Setting and Speech Encoding Setting fields are located under System\IP-Related Information\Call Configuration.
- The Model 8602 only supports three G.-711 or six G-729 frames per packet. The Model 8602 will default to these values if the frames per packet setting is set any higher. The Audio Frame/IP Packet setting can be programmed in System\IP-Related Information\Call Configuration.

2. SYSTEM SETUP FOR PROGRAMMING

2.1 There are three methods that can be used for programming the phone system.

- On-site programming with an attached PC
- Programming using a stand-alone PC
- Remote programming using a modem connection

2.2 Each is described separately in the following pages. Only one programming session should be in operation at a time.

2.3 For on-site, remote, or stand-alone programming, the computer must be an IBM AT-compatible personal computer (PC) with a minimum of:

- Pentium II 266 MHz or higher microprocessor
- Windows 98, Windows 2000, Windows NT 4.0, Windows ME, or Windows XP
- 32 MB RAM
- 100 MB hard drive space for installation; 80-100 MB hard drive space afterwards
- VGA graphics monitor
- 1.44-megabyte, 3.5-inch double sided/high density floppy disk drive
- CD-ROM
- A mouse is optional but recommended (if a serial mouse is used, install on an unassigned COM port)

A. INSTALLING THE PROGRAMMING SOFTWARE

NOTE: If you have a previously installed version of DB Programming, you may want to uninstall the older version. The latest version of software is installed in a folder named *Inter-Tel DB Programming* and older versions were named *Inter-Tel Axxess DB Programming*. If you do not delete the older version, make sure that you use the current versions when selecting the Session Manger, Board Utility, and Converter programs supplied with the software. Also, make sure any desktop icons point to the correct version. The proper way to uninstall programs under Windows is to use Add/Remove Programs from the Control Panel. When this is used, DB Programming uninstalls completely.

2.4 The following instructions assume that Windows software is installed on the PC and the user is familiar with Windows procedures. Also, there should be enough space on the hard drive(s) where the common files are located, where the files for the specific version will be installed, and where the operating system is located so the installation program can write temporary files and install files on the system directory.

NOTE: To install DB Programming on Windows NT, Windows 2000, or Windows XP, you must have local administrative rights.

2.5 To install the programming software:

1. Ensure that Windows is running.
2. Insert the DB Programming CD into the PC's CD-ROM drive.
3. Open the CD folders and click **DATABASE_PROGRAMMING**.

4. Click on the version folder to be installed. For example, the V5352 folder represents Version 5.352 Software.

NOTE: If you desire, you may make a copy of the DB Programming folder on your DB Programming PC for emergency recovery.

5. Click either **FLOPPY_DISK_INSTALLATION** or **SINGLE_FILE_INSTALLATION**. The Floppy_Disk_Installation is used to make the floppy disks for any PC or laptop not having a CD-ROM drive. The Single_File_Installation is used to load the required programming software onto a PC or laptop with a CD-ROM unit.
6. Click on the **XXXX_PROD_DB_PROG** (where XXXX is the version number). The executable file starts running.
7. After reading the license agreement, click **I Agree** to begin the installation.
8. A Welcome screen appears; click **Next** to continue.
9. A dialog box appears that asks for the desired destination directory name. Select **Next** to accept the default directory. Or, click **Browse** and select the directory where the DB Programming files will be stored on the selected drive. Click **Next** to continue (or **Cancel** to exit without performing the installation).

NOTE: Although you can select the desired directory, use the default directory for possible impact on any earlier explorer version.

10. Follow the directions outlined by the installation wizard, clicking **Next** to advance to the next screen. The software installation will start loading files. During the installation, a window displays the percentage of the files that have been copied.

NOTE: If you have an older version of AXXESSORY Connect installed on your programming PC, you may see an error message concerning TAPI. System versions 5.1 and later DB Programming requires TAPI version 2.1 or newer to use the modem to connect to remote systems. Your version of AXXESSORY Connect may use an older version of TAPI. If so, you will be asked if you want to upgrade the TAPI components. If you answer Yes, TAPI 2.1 will be installed and your AXXESSORY Connect may be inoperable. However, if you answer No, the Session Manager will not be able to set up a modem session unless you reinstall DB Programming to upgrade the TAPI version.

11. The last installation window gives you the opportunity to view a “Read Me” file that explains the enhancements that have been made to the System. If you would like to view it now, click **Yes**. If not, click **No** (you can still view it later by selecting **Readme** in the programming group).
12. Click **Finish**. Then click **OK** to restart your PC and complete the installation. After the restart, the software is loaded on your PC and ready to use.
13. To begin programming, select **Start** - **Programs** - **Inter-Tel DB Programming** - **Session Manager** (or select the **Session Manager** icon on your desktop).

B. ON-SITE PROGRAMMING

2.6 For on-site programming, the PC is connected directly to the phone system. An RS232 cable from COM 1 or COM 2 on the PC is connected to one of the RS232 connectors on the CPC, CPU, or Options card.

NOTE: ***Do not*** connect the programming terminal to an RS232 port that has been programmed for System OAI output (SMDR or error report output is allowed). In the default state, the RS232 ports are ***not*** programmed for output.

2.7 If you attempt to establish a direct-connect programming session using a PC COM port that is connected to a modem, the connection will not be allowed and a warning will appear.

C. STAND-ALONE LOCAL PROGRAMMING

2.8 To use a PC for stand-alone programming, the system software must be loaded on the hard drive. The system database information can then be stored on double-sided/high density floppy disks for later transfer to the phone system and Voice Processor using Save/Restore.

D. REMOTE PROGRAMMING VIA MODEM

2.9 A programmer can use modem access to the system database for the purpose of customer programming or maintenance/diagnostics from a location other than the customer site. This allows the service personnel flexibility in making database changes without visiting the site and, in system trouble situations, the service personnel may perform preliminary investigations before going on site.

2.10 If using a Call Processing Card (CPC), you can connect directly to the card using the built-in modem. If the chassis is equipped with a CPU card, however, the system must be equipped with an auto-answer modem connected to a system RS232 port.

2.11 To use the remote programming feature, the user dials the telephone number that directly rings in to the modem, is transferred to the modem, or dials the modem extension number using DISA or a station. When the modem circuit rings, the CPC or auto-answer modem automatically answers the call and generates modem tone. The calling party may then connect the programming computer's modem and proceed with the programming session. When the session is completed, the calling party hangs up or disconnects the call from the modem. When this happens the system modem no longer hears modem tone and disconnects.

2.12 Digital display phones with PC Data Port Modules that are enabled for RS232 communication (see page 6-205) may dial up the modem directly and, using the attached PC, can generate programming reports or monitor SMDR (they cannot perform DB Programming). However, the MDPM circuit should be programmed with camp-on tones disabled, extended ring cadence enabled, different alerting cadence disabled, and different ringback cadence disabled (see page 6-210). See page 2-100 for details about connecting to the CPC modem.

NOTE: Use the IP port for remote programming.

2.13 To set up or check the modem parameters, use the Windows Settings/Control Panel as you would for any PC modem application.

2.14 If a remote programming session is active on a system with a Voice Processing Unit, and the communications connection fails, the remote programming session will be terminated.

NOTE: If used, Save/Restore will take longer to perform via modem than with a direct connection due to the slower bit rate. If the Restore function is used, the associated system reset, when completed, will disconnect all calls including the modem connection.

2.15 If the modem connection is lost during the programming session, allow a minute for the system modem to reset, then re-establish the call. All changes that were saved by a system update before the connection was lost will be retained in phone system memory.

2.16 **Troubleshooting Modem Connections**: If the modem is not connecting to the phone system, try the tips listed below.

1. Try running the Windows Hyperterminal program under the Accessories menu. The modem may not be configured properly if Hyperterminal is unable to dial out.
2. Lower the Maximum speed in the modem settings to 9600.
3. Select **Connection** in the modem settings dialog. Select **Advanced** and type "X1" in the Extra settings entry. Sometimes this setting is necessary to tell the modem to disable dial tone detection. Some PBXs do not transmit the dial tone signal in the format the modem is expecting.
4. Generate a modem log file and review it for clues to the problem. Select **Connection** in the modem settings dialog. Select **Advanced** and check the box labeled "Record a log file." When you check this box, Windows then creates a log file named Modemlog.txt in the directory where Windows is installed. This is useful for monitoring calls and for troubleshooting.

   NOTE: For Windows NT, the name of the log file is ModemLog_name of modem.txt where name of modem is the actual name of the modem you are using (e.g., ModemLog_U.S. Robotics 56K Voice Win.txt).

5. Check to be sure that DB Programming is set up according to the phone system modem kit instructions.
6. The modem connected to the phone system may need to be reprogrammed. Please see the Inter-Tel modem kit instructions for more details on this operation.

3. UPLOAD UTILITY

3.1 The Upload Utility allows you to upload different versions of software to board or IP devices. If necessary, you can upload software to multiple IP devices at once.

3.2 In addition, you can save your settings so that you do not have to configure information for every upload. For example, you can create an entry used strictly for IPRCs on Node 2, or, you can also create one for all hard IP endpoints, such as the Model 8660, in your network.

NOTE: Inter-Tel recommends that you upgrade the firmware on the IP endpoints BEFORE you upgrade the IPRC. If you upgrade the IPRC first, the endpoints will not be able to connect to the IPRC, but they will continuously try. During these connection attempts, the Upload Utility will not be able to connect with the endpoints.

3.3 For complete information about the Upload Utility, refer to the Inter-Tel Upload Utility online help.

4. DATABASE CONVERTER UTILITY

4.1 The conversion program does not support converting version 5.1 or earlier databases to version 9.1 or later databases. If you must convert a pre-v5.1 database, do the following steps:

1. Use an earlier version of DB Programming to convert the pre-v5.1 database to a v9.x database.
2. Use v9.1 DB Programming to convert it to a v9.1 database.

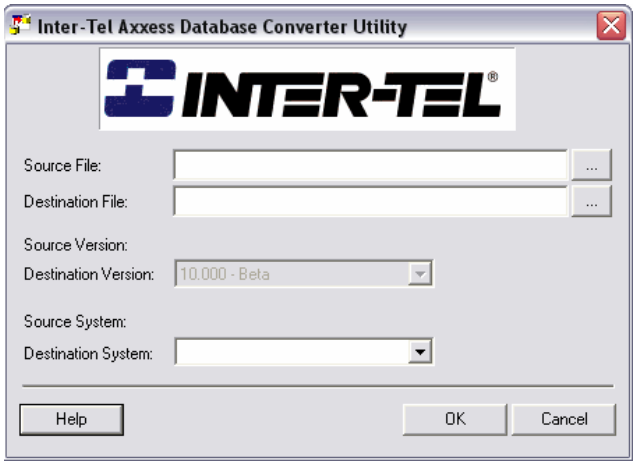
4.2 There are two different conversion programs loaded on the computer when a PC has DB Programming v9.0 or earlier and v9.1 or later loaded. The programs are stored in the following locations:

- **`C:\Inter-Tel\Update\AutoConv.exe`** (for v9.0 and earlier systems)
- **`C:\Inter-Tel\Update\AxxessAutoConversion.exe`** (for v9.1 and later systems)

4.3 Short cut icons can be created for both conversion programs to be used on the same machine. All of the fields are automatically populated.

4.4 To convert an older database to a newer version:

1. Select Start - Programs - Inter-Tel DB Programming - **DB Converter**. The main screen appears, as shown below.



2. Verify that the desired source file path is selected in the **Source File** text box. Click the Browse (…) button to change the path.
3. Verify that the desired destination file path is selected in the **Destination File** text box. Click the Browse (…) button to change the path.
4. Verify that the destination version to which you want the database converted is selected in the **Destination Version** drop-down list box. To change the version, select the desired version from the drop-down list box. The Destination Version drop-down box is grayed out initially until you have entered a source file and whenever the source file is being changed.
5. Verify that the desired destination system for the new database is selected in the **Destination System** drop-down list box. The available options are as follows:
   - **64 (CPC - Single)**: The new database uses a Call Processing Card (CPC) and have up to 64 devices.
   - **128 (CPC - Dual)**: The new database uses a CPC and have up to 128 devices.
   - **256 (CPC - Dual)**: The new database uses a CPC and have up to 256 devices.

- **512 (Call Processing Server)**: The new database uses a Call Processing Server and have up to 512 devices.
- **1024 (Call Processing Server)**: The new database uses a Call Processing Server and have up to 1024 devices.
- **ATM Call Processing Server**: The new database uses an ATM Call Processing Server (Windows 2000) and have up to 1792 devices.

NOTE: The options provided in both the Destination Version and Destination System will be reset. You must re-select the desired Destination Version and System if the defaults are not the correct ones.

6. When all settings are correct, click **OK**. Or, click **Cancel** to exit the application without performing the conversion.

NOTE: When converting to v9.1, the Database Conversion program enables the Call Logging feature for all existing digital and IP endpoints.

5. DATABASE MERGE UTILITY

5.1 The Merge Utility is designed for sites that are converting a currently networked system to an ATM System. Taking the information from existing nodes, this application merges multiple databases into one file. This not only helps to maintain the integrity of the current information, but it also eliminates the time-consuming task of having to manually retype entries.

NOTE: When the Merge Utility compares the version of the merging databases, it uses the first three digits. For example, 6.003 and 6.004 databases are both considered 6.00. Therefore, if you have a 6.003 database, you don't need to convert it to a 6.004 database.

A. MERGE UTILITY BASICS

5.2 To successfully merge the databases, the Merge Utility uses the following four main elements:

- **Master Database**: Acts as the main file to which all databases are merged (see below).
- **Merged Database**: Contains the information that is added to the master database (see page 6-24).
- **Merged Items**: Consist of the specific items (e.g., devices and cabinets, extension IDs, etc.) that can be merged (see page 6-24).
- **Merge Log**: Logs any changes made and problems that occurred during the merge process (see page 6-27).

5.3 Each of these elements is described in the following sections.

Master Database

5.4 To combine multiple databases, the Merge Utility uses an existing database and saves it as a master database. Information from subsequent databases is then added to this master file until all databases are merged. Although any number of databases can be merged, the master data-base device limits are enforced for each device type. Because this application retrieves the required information and copies it to the master database, existing databases are not deleted.

5.5 When selecting the master database, consider the following:

- Using the system that is attached to the Voice Processing Unit as the master database will speed up the merging process because all of the mailboxes and other associated voice mail data will not need to be copied.
- The programming for any devices or items NOT mentioned in this manual will remain as originally programmed in the master database.
- A v6.0 or later non-ATM database will be automatically converted to a v7.0 or later ATM database.
- The master must use one of the following number plans:
  - **Option 1**: Consists of the following defaults:
    - Trunk Groups - 9201
    - Node Trunk Groups - 9501
    - Node Route Groups - 9701
    - Trunks - 94000
    - ARS - 9200

If the application detects this numbering plan, the listed devices and the entries in the Device Baseline Extension table are automatically converted to the values associated with the Option 2 plan (see below).

— **Option 2**: Consists of the following defaults:

- Trunk Groups - 92001
- Node Trunk Groups - 97501
- Node Route Groups - 97001
- Trunks - 94000
- ARS - 92000

NOTE: This option conforms to the v6.0 and later numbering plan.

Regardless of the plan detected, all trunk extensions will be recalculated to optimize the available range. The extensions for the node and CO trunk groups in the merging databases will also be updated to the new extension plan.

NOTE: You should set the device extensions to match the values in the Device Baseline Extensions table before starting the merge process.

Merged Databases

5.6 To successfully merge multiple databases, each database must be Version 6.0 (or later) and be programmed for the same product type and country. You should also run DB Test to ensure there is no corrupted information. In addition, you should run an import/export between all databases that will be merged to ensure all off-node extensions are current.

NOTE: The merged database node information is automatically deleted from the master once the data is merged. This may require some manual adjustment of the route groups.

Merged Items

5.7 The following items are merged into the master database:

- **Nodes**: The specific node number information for the merged nodes is removed from the master database. For example, if merging node 2 with node 1 (master) on a 20-node system, node 2 information is removed from the master database (not the information on the other 18 nodes). All merged devices are then updated to reflect the master node as the devices' local node.
- **Boards**: Networked PRI cards that connect the nodes being merged are deleted from the master database and are not merged into the master. This occurs because the physical connection between the nodes no longer exists, and these cards are not needed. The list of boards and their associated devices that are merged is as follows:
  - — EVMC
  - — T1 - OPX, Ground Start, Loop Start, DID [DDI in Europe], E&M Trunks
  - — T1 PRI/E1 PRI - B-Channel Trunks
  - — LSC/LSC-D - Loop Start Trunks
  - — LGCG/LGC-D - Loop Start, Ground Start Trunks

- BRS/BRU - Station/Station, Trunk/Trunk Ports
- DKSC/16 - Keysets
- EKSC - Keysets
- SLC/16 - Single lines
- Analog Keyset Card - Keysets
- OPC - Serial Port

- **Call Routing Tables, Keymaps, Speed Dials, and Forwarding Paths**: When call routing tables, keymaps, speed dials, and forwarding paths are merged, they are assigned the next available master database ID that corresponds to the type. For instance, if the master database contains forwarding paths with IDs 1, 2, 4, 6, and 7, the merged forwarding paths are assigned IDs 3, 5, 8, 9, 10, etc. Each item is then merged until the database maximum is reached. If this limit is exceeded, all references to items not merged are set to the first ID in the corresponding list.

  Speed dials and forwarding paths that are already included in the master database are not merged. Instead, all references to these items are re-assigned to the matching item in the master database.

- **Voice Mail**: If voice processing is enabled in one of the databases that is being merged, the voice processing flag in the master database is also enabled. In addition, the following information is copied if there are no entries in the corresponding table in the master database:
  - Audiotex Recordings
  - Auto Attendant
  - AVDAP Information
  - Disk Statistics
  - DTMF Detection
  - Fax-On-Demand
  - Maintenance AVDAP
  - Reports AVDAP
  - SMDR AVDAP
  - Subscriber Statistics
  - Timers And Limits AVDAP
  - Voice Mail
- **Local Node Devices**: The local node devices that are merged include:
  - ACD Agent IDs
  - CO Trunk Groups
  - Digit Translation Nodes
  - Extension Lists
  - Extension IDs
  - Group Lists
  - Hunt Groups

- Mailboxes
- Node Trunk Groups
- Page Ports
- Page Zones
- Serial Ports
- Stations
- Voice Mail Applications
- Voice Mail Nodes

To merge local node devices, the master database is first searched for off-node devices of the same type and extension. All matching off-node devices are then deleted from the master database before the information from the merged database is added.

If a matching off-node device is not found in the master database, a new entry is added to the master devices table. Unless a conflict is detected, the existing extension is used in the master table. If a conflict is detected, new extensions are generated until the conflict is resolved. (For trunks, however, the "P#" conflict extension is immediately generated.)

Because a local page zone can contain an off-node page zone as a device, page zones require a special clean up operation. In some cases, off-node page zones become local page zones. Because local page zones are invalid device entries, when the merge is complete, each device in a page zone is validated for existence and removed if it cannot be found. Then, any unused page zones (page zones with empty device lists) are deleted.

Finally, all trunk group music-on-hold board IDs are updated to indicate the new board location. If the music-on-hold board ID was an Options card that could not be merged, the new ID for that card becomes the CPC or CPU card in slot 8.

- **Off-Node Devices**: The list of off-node devices that are merged includes the following:
  - Hunt Groups
  - Page Ports
  - Page Zones
  - Mailboxes
  - Stations
  - Voice Mail Applications

  To merge off-node devices, the master database is first searched for local and off-node devices of the same type and extension. All matching devices (off-node and local) are then deleted from the master database before the information from the merged database is added.

  If a matching device is not found in the master database, a new entry is added to the master devices table. Unless a conflict is detected, the existing extension is used in the master table. If a conflict is detected, a "P#" conflict extension is generated and assigned to the off-node device.

5.8 In addition to the details defined for various database items (see *Merged Items*), the following occurs during a database merge:

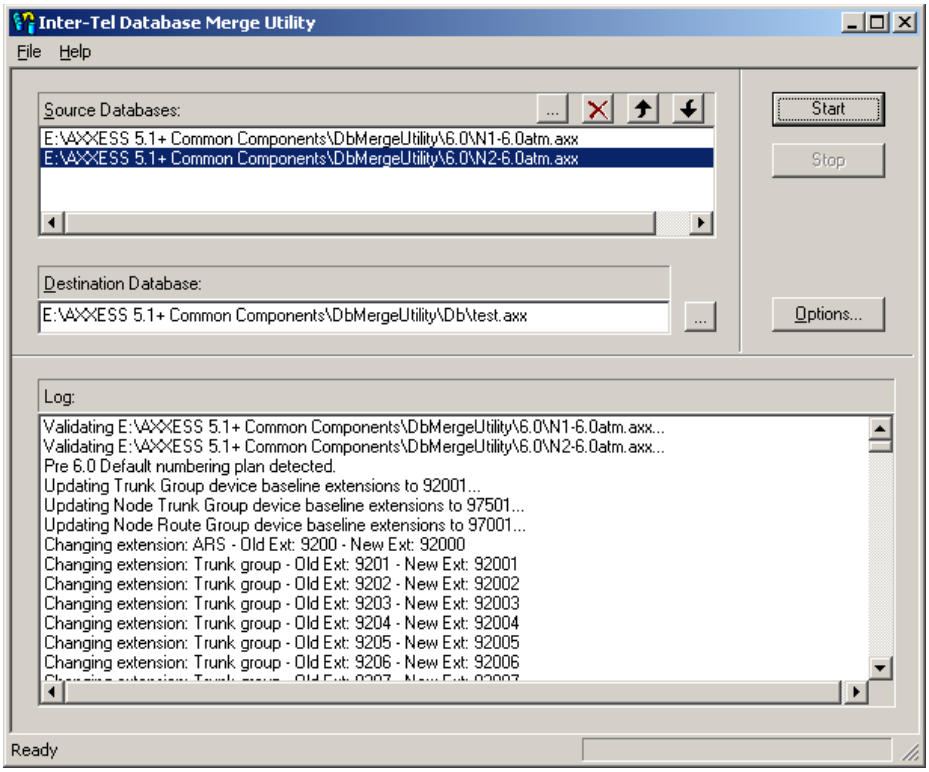
- When a device or item is merged, its associated mailbox is merged at the same time. If a device limit is exceeded, however, the associated mailbox is not merged.
- Each device is merged until the database maximum is reached. If this limit is exceeded, all references to items that cannot be merged are cleaned up in the same way that DB Test cleans up an unused device. In some cases, this involves removing the item from a list, setting the device to "None," or resetting the entry to another item of the same type.
- The Merge Utility attempts to merge all items from each database, even if some items can no longer be merged due to system capacity limits.
- All details of the merge, including every extension and hardware address change, are displayed in the merge log.

Merge Log

5.9 During the merge process, every problem and change, including hardware and extension address changes, are displayed in the merge log file. As the items are merged into the master database, any off-node devices are deleted, and any references to these devices are resolved to the new local node device.

B. MERGE UTILITY OPERATION

5.10 To start the DB Merge Utility application, select Start - Programs - Inter-Tel DB Programming - **DB Merge**. The main screen appears, as shown below.



5.11 This screen consists of the following elements:

- **Source Databases**: Contains the names and locations of the databases to be merged. The master database is always listed first. Additional databases will be merged in the order in which they are displayed.
- **Destination Database**: Displays the path and file name of the final database. This file is created when the merge operation is complete.
- **Log**: Lists the details of the database merge. The text displayed in this window is automatically saved to a **.txt** file with the same name as the destination database.
- **Command Buttons**: Perform various actions, as described in the following table.

Table 6-1: *Merge Utility Command Buttons*

Button	Keystrokes	Function
...	CTRL + B	Invokes a browse window that allows you to search for databases to be merged (or for the destination database).
	DELETE	Deletes the selected database from the Source Databases list.
	CTRL + Up Arrow	Moves the selected database up in the Source Databases list.
	CTRL + Down Arrow	Moves the selected database down in the Source Databases list.

Table 6-1: *Merge Utility Command Buttons (Continued)*

Button	Keystrokes	Function
Options	ALT + O	Displays additional options.
Start	CTRL + ENTER	Begins the merge process.
Stop	ESC	Cancels the merge process.

- **Status Bar**: Displays the current status of the merge process.

5.12 The following instructions detail how to merge multiple databases using the Merge Utility.

Merging a Database

5.13 To merge a database:

1. Click the browse button (...) in the Source Databases section to browse for the database. When you click **Open**, the database will be added to the list.

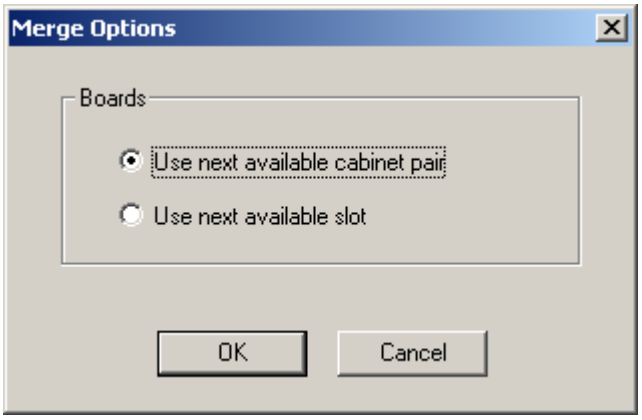
   NOTE: To delete a database from the Source Databases list, select that database and click **Delete** (**DELETE**).

2. Use the arrow buttons to place the databases in the desired order. Databases are merged from the top down.

   NOTE: The first database is always considered the master database.

3. Enter the path and name of the destination database in the space provided. Or, click the browse button to browse for a path location.

4. Click **Options** to define how the boards are assigned. The following screen appears.



You have the following options:

- **Use next available chassis pair**: Merges boards until there are no more empty cabinet pairs. When selected, vacant slots in the merged databases are mirrored in the master database. This allows you to retain the existing cabinet configuration without having to move boards. This is the default selection.
- **Use next available slot**: Assigns boards to the next available slot until there are no more empty slots.

5. Click **OK** to accept the merge option. You are returned to the main screen.
6. Click **Start**. The application will begin merging the databases.

NOTE: Click **Stop** (**ESC**) to terminate the merge process.

5.14 The status bar at the bottom of the application will track the merge process. Any errors and/or changes will be displayed in the Log area, as described below.

Understanding the Log Display

5.15 While the application is merging the information, the Log area will display the changes and/or problems the application encounters. This includes the following:

- **Validating <*Database*>**: Displays the database path and name that the application is currently validating. The application will ensure the database is the correct version and that the country and product type matches the other databases.
- **Merging**: Identifies the database and the corresponding items that are currently being added to the master database.
- **Adding**: Details any changes the application is making. Typically, the old value is listed first, followed by the new value.
- **Merge Completed**: Indicates that the merge is complete.

5.16 In addition, the log displays any embedded error messages, which are preceded by three asterisks (***). If desired, you can search for an error message by searching for the asterisk.

NOTE: If a database access error occurs during the merge, ensure that you are using the latest version of the Merge Utility and run the merge one more time. If the error still occurs, send the affected database files to Inter-Tel engineering for review.

5.17 After the merge is complete, the text shown in the Log area is saved to a **`.txt`** file with the same name as the final database. This file is also stored in the same location as the destination database. If a file already exists with this name, a dialog box will prompt you to overwrite the file. If you select **Yes**, the existing file is overwritten. If you select **No**, the merge process is stopped, and you are prompted to provide a new name for the file.

Completing the Merge

5.18 When the application has finished the merge, perform the following steps:

1. Review the Merge Utility log (see the previous section).
2. Run DB Test and use the repair option if any inconsistencies are found.
3. Load the database in a local session and update the node route groups.
4. Export the merged database to all nodes in the system.
5. Import from all nodes in the system to the merged database.
6. Review all information and settings in the merged database for accuracy. You may need to manually change devices and settings before the database will work.

6. INTER-TEL DATABASE TEST AND REPAIR UTILITY

6.1 The Inter-Tel Database Test and Repair Utility tests databases for corruption. In most cases, it is able to repair the corruption.

6.2 You can choose among eight tests to run. Each test can be enabled for testing only, or it can be enabled for test and repair.

CAUTION

If repairs are enabled, the given database will be modified. Therefore, you must make a backup copy of your database before running this program. This will give you a backup copy in case the repairs were undesirable.

A. RUN A TEST

1. Enter the path and name of the database (.axx) file in the Database To Test field. (Or, you can use the Browse button to locate the file.)
2. Do one of the following:
   - — Select the Test to run from the list of tests in the left pane of the window. Then select **Enable** or **Enable/Repair** from the Test drop-down menu.
   - — To run all tests, select **Enable All** or **Enable/Repair All** from the Test drop-down menu.
3. If desired, set the options for the test by selecting **Options** from the File drop-down menu.
4. Select **Start Testing** from the File drop-down menu. The results of the test are shown in the right pane of the window.
5. If you want to save the results to a file, select **Save Results** from the File drop-down menu.

NOTE: These tests do not find all of the corruptions that are possible. However, they cover a large number of them. Frequently found corruptions are described in Common Errors. If the Repair function is enabled and the repair status is Not Repaired, the program was unable to fix the problem. In that case, contact Inter-Tel for assistance. Sometimes, a repair can cause another corruption. If this happens and the required test has not been run, DB Test will notify you that another test needs to be run to repair the new corruption. One example would be a station with a 0.0.0 hardware address. The Hardware Address test will fix the problem by deleting the station. However, the Referential Integrity test must run in order to delete the references to this station. There are a few errors that DB Test detects but is unable to repair. If you come across such an error, contact Technical Support for assistance.

When the Database Test and Repair utility makes a repair by deleting records, it is possible for further corruption to be generated. All of the corruption can be removed by running the proper test. For example, if extension 1000 was deleted because of a major corruption, all references to extension 1000 would still be in the database. The Referential Integrity test would be able to clean up these references. Therefore, once the repairs are made, the utility may warn you that other tests/repairs need to be run in order for clean up to occur. If the proper test automatically runs after the deletion occurs, then no warning will be necessary since the test has already run. However, you will notice additional error messages as the new corruption is repaired.

B. MENUS

6.3 The menu options include the following:

File Menu

Table 6-2: *File Menu Options*

Menu Option	Action
Start Testing	Runs the enabled tests. Repairs errors if repairs are enabled.
Save Results	Saves the errors/warnings to a file.
Options	Enables user to program utility options.
Exit	Closes the program.

Test Menu

6.4 You can also view these options by right-clicking in the Test list.

Table 6-3: *Test Menu Options*

Menu Option	Action
Enable/Repair	Enables the selected test for test and repair.
Enable	Enables the selected test for testing only.
Disable	Disables the selected test.
Enable/Repair All	Enables all tests for test and repair.
Enable All	Enables all tests for testing only.
Disable All	Disables all tests.
Properties	Displays the test properties—a description of what the test is for.

Toolbar

6.5 The toolbar contains the following icons that provide shortcuts to the drop-down menu options.

Table 6-4: *Toolbar Icons*

Icon	Menu Option	Action
Stoplight	Start Testing	Runs the enabled tests. Repairs errors if repairs are enabled.
Floppy disk	Save Results	Saves the errors/warnings to a file.
Red dot	Options	Enables user to program utility options.
Lit light bulb with wrench	Enable/Repair	Enables the selected test for test and repair.
Lit light bulb	Enable	Enables the selected test for testing only.
Unlit light bulb	Disable	Disables the selected test
Paper	Properties	Displays a description of the test.

Options

6.6 When you select **Options** from the File menu, you can set the following options for the test:

- **Log File**: Enter the path where the log file is stored.
- **Browse**: Browse to locate the log file.
- **Repair System Tables**: Check this box if system tables should be repaired.
- **Compact Database**: Check this box if compaction should occur after testing/repairing the database. (This reduces the size of the resulting database.)
- **Display Warnings**: Check this box to display warnings as well as errors in the output window.
- **Max. Errors**: This is the maximum number of errors you will allow the test to find to find before quitting the test. The default is 500.

C. TESTS

6.7 Tests include the following:

- Associated Mailboxes (see page 6-33)
- Boards (see page 6-34)
- Cleanup (see page 6-34)
- Devices (see page 6-35)
- Dynamic Enumerations (see page 6-35)
- Enumerations (see page 6-36)
- Extension Conflicts (see page 6-36)
- Hardware Addresses (see page 6-37)
- Referential Integrity (see page 6-37)
- Static Records (see page 6-38)
- Miscellaneous (see page 6-38)

Associated Mailboxes

6.8 Each mailbox in the database has a flag to indicate whether or not it is associated with another device. This test searches for two cases. The first case is when a mailbox's associated flag enabled, but there is no valid corresponding device with the same extension to be associated. The second case is when a mailbox's associated flag is disabled, but there is a valid corresponding device with the same extension to be associated.

6.9 Possible Error Messages:

- **Mailbox x<*extension*> is not associated, but has multiple corresponding devices**: In this case, mailbox x1000 has an associated flag disabled. It has more than one valid corresponding device with the same extension. This case will not be repaired. It is not valid for a mailbox to be associated to more than one device. The Repair Status will be Not Repaired. To repair this case, run the Extension Conflicts test. This test will repair the multiple devices sharing one extension. Then, rerun the Associated Mailboxes test.
- **Mailbox x<*extension*> is not associated, but has a corresponding device**: In this case, mailbox x1000 has an associated flag disabled. However, it has a valid corresponding device with the same extension. This corruption is repaired by enabling the associated flag. The Repair Status will be Set Associated Flag to true.

- **Mailbox x<*extension*> is associated, but has no corresponding device**: In this case, mailbox x1000 has an associated flag enabled. However, it does not have a valid corresponding device with the same extension. This corruption is repaired by disabling the associated flag. The Repair Status will be Set Associated Flag to false.
- **You have exceeded the maximum number of mailbox licenses**: The database is currently programmed for the maximum number of licenses allowed. You cannot add any more mailboxes. For additional licenses, contact your Inter-Tel dealer.

Boards

6.10 This test checks to make sure that each board has the proper entries in all corresponding tables.

6.11 Possible Error Messages:

- **The board in slot <*slot number*> has no records in <*table name*>**: In this case, the board is required to have records in its table. However, none exist. This corruption is repaired by adding default records to the table. The Repair Status will either be Record Added if one record was added or Records Added if more than one record was added.
- **The board in slot <*slot number*> has an incorrect number of records in <*table name*>**: In this case, the board does not have the proper number of records in its table. This corruption is repaired by adding default records to the table. The Repair Status will be Records Added.
- **The board in slot <*slot number*> has an incorrect record sequence in <*table name*>**: In this case, the board has the proper number of records in the table. However, the sequence numbers are incorrect. For example, the records should be numbered 1 through 10. However, they are numbered 0 through 9. This corruption is repaired by deleting the record that has an out-of-bounds sequence number and adding a default record for the each missing sequence number. In the given example, record 0 would be deleted and record 10 would be added. The Repair Status will be Added/Deleted Records, Deleted Records, or Records Added depending on the action that was taken.

Cleanup

6.12 The Cleanup test detects and removes objects that are not in use. The following objects are removed if not used:

- Hunt Groups
- Facility Groups
- Route Groups
- CO Trunk Groups
- Page Zones
- Standard Account Codes
- Forced Account Codes
- System Speed Dials

Devices

6.13 This test checks to make sure that each device has the proper entries in all corresponding tables.

6.14 Possible Error Messages:

- **Invalid table <*table name*>**: This is only a warning. It indicates that it was looking for a table, but could not find it in the database. This can happen if a new table is added in a later version of a major release. If the test is being run on the older version, this warning will appear. For example, if Table A was added in version 5.111, running this test on databases previous to version 5.111 will produce a warning that Table A could not be found.
- **Invalid column <*table name*>:<*column name*>**: This is only a warning. It indicates that it was looking for a column, but could not find it in the database. This can happen if a new column is added in a later version of a major release. If the test is being run on the older version, this warning will appear. For example, if Column A was added in version 5.111, running this test on databases previous to version 5.111 will produce a warning that Column A could not be found.
- **Ext <*extension number*> has no records in <*table name*>**: This corruption indicates that the given device has no records in the table indicated. This corruption can be repaired in more than one way. If the table that is missing the records is critical, then the device is non-recoverable and will be deleted completely. In this case, the Repair Status will be Deleted Device. If the table that is missing the records is not critical, the corruption will be repaired by adding default records to the table. The Repair Status will either be Record Added if one record was added or Records Added if more than record was added.
- **Ext %s has an incorrect number of records in %s**: In this case, the device does not have the proper number of records in a table. This corruption is repaired by adding default records to the table. The Repair Status will be Records Added.
- **Ext %s has an incorrect record sequence in %s**: In this case, the device has the proper number of records in the table. However, the sequence numbers are incorrect. For example, the records should be numbered 1 through 10. However, they are numbered 0 through 9. This corruption is repaired by deleting the record that has an out-of-bounds sequence number and adding a default record for each missing sequence number. In the given example, record 0 would be deleted and record 10 would be added. The Repair Status will be Added/Deleted Records, Deleted Records, or Records Added depending on the action that was taken.

Dynamic Enumerations

6.15 This test checks to make sure that all dynamic enumeration fields contain valid values. A dynamic enumeration field is an enumeration field that depends on another field. If the master field is set to A, then the field can contain one set of values. If the master field is set to B, then the field can contain a different set of values.

6.16 Possible Error Messages:

- **Invalid table <*table name*>**: This is only a warning. It indicates that it was looking for a table, but could not find it in the database. This can happen if a new table is added in a later version of a major release. If the test is being run on the older version, this warning will appear. For example, if Table A was added in version 5.111, running this test on databases previous to version 5.111 will produce a warning that Table A could not be found.

- **Invalid column <*table name*>:<*column name*>**: This is only a warning. It indicates that it was looking for a column, but could not find it in the database. This can happen if a new column is added in a later version of a major release. If the test is being run on the older version, this warning will appear. For example, if Column A was added in version 5.111, running this test on databases previous to version 5.111 will produce a warning that Column A could not be found.
- **<*table name*>/<*column name*> contains an invalid enumeration value [<*invalid value*>]**: The table has an incorrect value in the indicated column. This corruption is repaired by changing the field to a correct value. The Repair Status will be Value Set to <*value*> where <*value*> is the field's new contents.

Enumerations

6.17 Each enumeration field has a set of values that are valid. An error is detected if the field has a value that is out of this range.

6.18 Possible Error Messages:

- **Invalid table <*table name*>**: This is only a warning. It indicates that it was looking for a table named, but could not find it in the database. This can happen if a new table is added in a later version of a major release. If the test is being run on the older version, this warning will appear. For example, if Table A was added in version 5.111, running this test on databases previous to version 5.111 will produce a warning that Table A could not be found.
- **Invalid column <*table name*>:<*column name*>**: This is only a warning. It indicates that it was looking for a column named, but could not find it in the database. This can happen if a new column is added in a later version of a major release. If the test is being run on the older version, this warning will appear. For example, if Column A was added in version 5.111, running this test on databases previous to version 5.111 will produce a warning that Column A could not be found.
- **<*table name*>/<*column name*> contains an invalid enumeration value [<*invalid value*>]**: The table has an incorrect value in the indicated column. This corruption is repaired by changing the field to a correct value. The Repair Status will be Value Set to <*value*> where <*value*> is the field's new contents.

Extension Conflicts

6.19 This test checks for extension conflicts.

6.20 Possible Error Messages:

- **Found blank extension, device type <*device type*>**: In this case, a device was found with a blank extension. The extension is changed to P#XXX where XXX can be from 000 to 999. The Repair Status will be Set to Extension <*new extension*>.
- **Found conflict with extension <extension number>, device types <*device type 1*>, <*device type 2*>**: Two devices have the same extension number, and the two device types are not compatible. This corruption is repaired by changing the extension number of the second device <d*evice type 2*> to P#XXX where XXX can be from 000 to 999. The Repair Status will be Set to Extension <*new extension*>.

NOTE: Only 1000 extension conflicts can be cleaned up at a time because there are only 1000 possible extensions in P#XXX to change them to. Therefore, starting at the 1001 extension, the Repair Status will be Not Repaired. After an extension is set to P#XXX, you can go into DB Programming and change the extension to a valid number.

Hardware Addresses

6.21 This test checks to make sure that each hardware address is valid.

6.22 Possible Error Messages:

- **Ext <*extension number*> refers to an invalid board slot[<*slot number*>.<*port*>.<*circuit*> Type**:**<*board type*>]**: In this case, the device is on a board that is in a slot that it should not be. Repairs are made by deleting the device. The Repair Status will be Deleted Device.
- **Ext <*extension number*> refers to an invalid port [<*slot number*>.<*port*>.<*circuit*> Type**:**<*board type*>]**: In this case, the device has an invalid port number. Repairs are made by deleting the device. The Repair Status will be Deleted Device.
- **Ext <extension number> refers to an invalid circuit <*slot number*>.<*port*>.<*circuit*> Type**:**<*board type*>]**: In this case, the device has an invalid circuit number. Repairs are made by deleting the device. The Repair Status will be Deleted Device.
- **Ext <*extension number*> refers to an unrecognized board [<*slot number*>.<*port*>.<*circuit*> Type**:**<*board type*>]**: In this case, the device is on a board with an invalid type. Repairs are made by deleting the device. The Repair Status will be Deleted Device.
- **Ext <*extension number*> refers to an invalid board type [<*slot number*>.<*port*>.<*circuit*> Type**:**<*board type*>]**: In this case, the device is on a board that it should not be. For example, a loop start device cannot be on a single line card. Repairs are made by deleting the device. The Repair Status will be Deleted Device.

Referential Integrity

6.23 This test checks to make sure that the relationships between tables in the database are valid.

6.24 Possible Error Messages:

- **Invalid table <*table name*> or Invalid source table <*table name*>**: This is only a warning. It indicates that it was looking for a table, but could not find it in the database. This can happen if a new table is added in a later version of a major release. If the test is being run on the older version, this warning will appear. For example, if Table A was added in version 5.111, running this test on databases previous to version 5.111 will produce a warning that Table A could not be found.
- **Invalid column <*table name*>:<*column name*> or Invalid source column <*table name*>:<*column name*>**: This is only a warning. It indicates that it was looking for a column, but could not find it in the database. This can happen if a new column is added in a later version of a major release. If the test is being run on the older version, this warning will appear. For example, if Column A was added in version 5.111, running this test on databases previous to version 5.111 will produce a warning that Column A could not be found.
- **<*table name*>/<*column_name*> contains an invalid reference to <*source table name*>/<*source column name*> [<*invalid reference*>]**: The indicated value contains an invalid reference to another table or column.
- **<*table name*>/<*column name*> contains a reference to a device of the wrong type [DevID**:**<*device ID*>, Type**:**<*device type*>]**: The indicated value contains a reference to a device that is not the proper type. For example, if the System Administrator Mailbox field contains a reference to a phone it is invalid, because the System Administrator Mailbox can only be a mailbox.

- ***<table name>/<column name>*** **contains a reference to a board of the wrong type [BoardID:*<board ID>*, Type:*<board type>*]**: The indicated value contains a reference to a board that is not the proper type. For example, the Primary Rate card records are connected to a Board record for a keyset card.

6.25 **Repairs**:

6.26 Repairs for Referential Integrity are different depending on the field that has the corruption. The possibilities are as follows:

- If a lack of a valid reference makes the record no longer useful, a repair is made by deleting the record. The Repair Status will be Record Deleted.
- If a reference is referring back to a device or object that is not valid or not present, several repairs can be made. They are as follows:
  - — **Set to Extension *<extension number>* [*<device ID>*]**: In this case, the repair was made by replacing the bad reference with a device of the same extension, but of the proper type.
  - — **Set to None Device**: In this case, the repair was made by replacing the bad reference with None.
  - — **Set to Current Device**: In this case, the repair was made by replacing the bad reference with the current device.
  - — **Set to First Available**: In this case, the repair was made by replacing the bad reference with the first device/object found.
- It is possible for the repair to be complicated, and therefore not attempted. In this case, the Repair Status will be Not Repaired.

Static Records

6.27 This test ensures that all static records are present in the database.

6.28 Possible Error Messages:

- ***<Table name>*** **contains an invalid number of records [*<current number>* instead of *<desired number>*]**: The table should have the "desired number" of records. However, it has the "current number" records. This corruption is repaired by deleting unneeded records and adding records that are missing. The Repair Status will be: Added/Deleted Records, Added Records, or Deleted Records.
- ***<Table name>*** **is missing *<specific record>***: This corruption occurs when a specific record is missing. Only some of these corruptions will be repaired. The Repair Status can be: Rebuilt Table, Created None Device, Created Voice Processing Local Node, Created Unused Trunk Group, or Not Repaired.

Miscellaneous

6.29 This test checks for miscellaneous business rules.

6.30 **Possible Error Message**:

6.31 **Station *<extension>* is its own attendant**: This error message indicates that the station is programmed to serve as its own attendant. The DB Test will set the attendant to none and the Repair Status will show [Set to None Device]. This test ensures that all static records are present in the database.

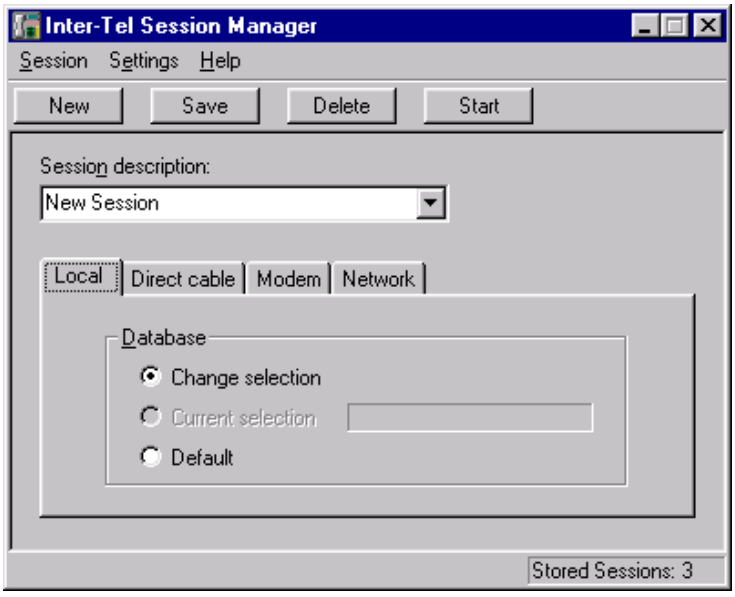
D. COMMON ERRORS

6.32 The following table lists common error messages.

Table 6-5: *Common Errors*

Test	Error
Hardware Address	Hardware Address Ext _____ refers to an unrecognized board [0.0.0 Type 262] A device has a hardware address of 0.0.0. DB Test will remove the device and cleanup references to it.
Referential Integrity	[Applications] / [transfer_recall_destination_device_id] contains a reference to a device of the wrong type [DevID:1, Type 0] [Stations] / [transfer_recall_destination_device_id] contains a reference to a device of the wrong type [DevID:1, Type 0] The Transfer Recall Destination field is not pointing to a valid device. DB Test changes this field to point to the current device. This corruption is a known problem and is fixed in 5.2.
Boards	The board in slot ____ has an incorrect number of records in [DSPs]. DB Test creates the necessary records.
Static Records	[BoardsInformation] contains an invalid number of records [0 instead of 16] DB Test creates the necessary records. This corruption occurred in early versions of 5.1.
Referential Integrity	[Report...] / [report_id] contains an invalid reference to [Reports] / [report_id][0] DB Test removes these unneeded records. This was a corruption in the default 5.1 database.
Referential Integrity	[PrimaryRateBoards] / [board_id] contains a reference to a board of the wrong type [BoardID:___, Type:253] DB Test removes this unnecessary record.
Static Records	[MessagePrintFreezeOnStrings] contains an invalid number of records [0 instead of 6] DB Test creates the necessary records. This corruption occurred in early versions of 5.2.
Devices	Ext ____ has no records in [CallRoutingAnnouncements] DB Test creates the necessary records.
Boards	The board in slot ____ has no records in T1CallType DB Test creates the necessary records. This corruption occurred in early versions of 5.2.
Dynamic Enumerations	[Keysets] / [attached_device_baud_rate_id] contains an invalid enumeration value [4] This corruption will be fixed starting in production versions of 5.2.

7. SESSION MANAGER



7.1 The Session Manager, shown above, is the application you will use to start all of your programming "sessions." Each session can have unique connection settings. This allows you to enter information once for each database you will be programming and save their settings. For example, you may want to have a session stored for each node in a networked system. An unlimited number of sessions can be stored. In addition, you can export/import session settings to use saved information on a different PC. The session manager also allows multiple sessions to be running at the same time as long as each one uses a different database. This provides more flexibility for programming or viewing databases at numerous sites.

7.2 The following pages describe the features and operations contained in the session manager.

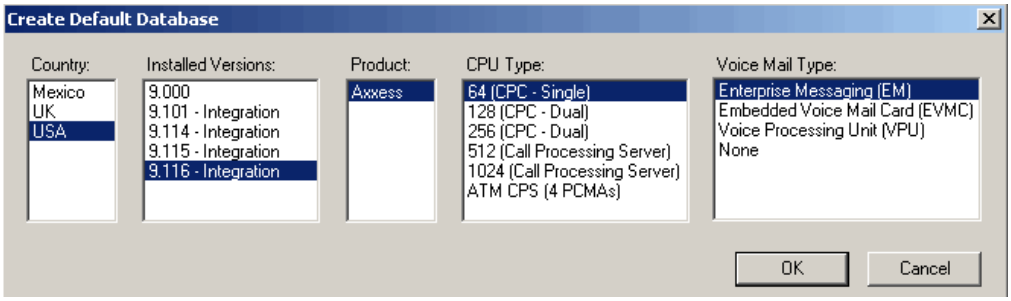
A. SESSION MENU AND BUTTON BAR

- **New**: To create a new session, select **New** on the button bar or from the Session drop-down menu. A default set of values will be displayed in the program fields. (The system version will be defaulted to the latest version installed on the PC.)
- **Save**: After you have modified the session description and connection type information, choose **Save** on the button bar or from the Session drop-down menu.
- **Delete**: Select **Delete** or menu option to remove the currently displayed session. If a local database exists for this session, it will also be deleted.
- **Start**: To start a session, scroll to the desired session in the Session Description box (or create a new session). Then select **Start** from the Session drop-down menu or the button bar at the top of the screen.
- **Export Session Settings File**: If you have sessions that you want to save for use on another computer, select **Export Session Settings File** from the Session menu. Then select the desired name and location of the file. When you click **Save**, the information for all session settings is saved as a **`.txt`** file.
- **Import Session Settings File**: To import existing session information, select **Import Session Settings File** from the Session menu. Then select the name of the **`.txt`** file that you want to import. When you click **Open**, the sessions are automatically added to the registry (i.e., existing sessions are **not** overwritten) and available for use in the Session Manager.

NOTE: You cannot export/import individual session settings. Information for all sessions is saved to and/or imported from the same file.

7.3 When you first start a session, a dialog will appear displaying the startup progress. (A Cancel button is provided on the startup dialog to quickly terminate a session that may have inadvertently been started.) As part of the startup process, a DB Programming Password window may appear, as shown below. If so, enter your password and select **OK** to begin. (By default, there is no password. See page 6-95 for password programming instructions.) The next screen you will see is the DB Programming screen (see page 6-49). This is where all of the Voice Processor and System programming will take place. The screen may take several seconds to a couple of minutes to display.

7.4 When you choose the Default option on the Local tab and click **OK**, the following screen appears.



7.5 The Session Manager will prompt you to enter the Voice Mail type you will be using in addition to the Country, Version, Product, and CPU type. When you choose the Default option on the Local tab and click **OK**, the following dialog box appears. The Voice Mail List provides the following choices:

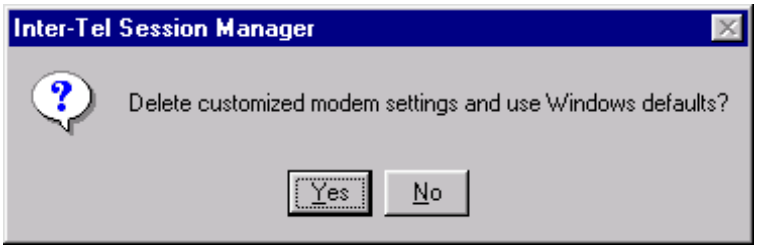
- Enterprise Messaging (EM)—this is the default selection
- Embedded Voice Mail Card (EVMC)
- Voice Processing Unit (VPU)
- None

7.6 Highlight the appropriate choices and then click **OK**.

B. SETTINGS MENU

7.7 Language: The Settings menu has a Language option that sets the language that will be used for programming the System database. (The default is American English for the U.S. and British English for Europe.)

7.8 Default Modem Settings: When you select this option, the following screen appears. If you select **Yes**, any custom Modem settings you have made will be replaced by the Windows default settings. If you select **No**, the Modem settings for your PC will remain unchanged.



7.9 Network Connections: Allows you to change various parameters when using an IP connection. These network settings should only be changed if the link to Call Processing goes down while DB Programming is receiving large amounts of data (e.g., when saving a database or when accessing mailbox or station folders). The Network Connection parameters include:

- **Receive Buffer Size**: Select the number of bytes that are used as a buffer when DB Programming is receiving data from Call Processing. If DB Programming is shutting down when receiving large amounts of data, ping the network. If the network connection is faster than the current Read Timeout value, increase this setting. By default, this is 64000. Click **OK** to accept the new setting.
- **Read Timeout**: Select the number of milliseconds that DB Programming will wait for data packets from Call Processing. If DB Programming does not receive data before this timer expires, DB Programming posts a warning message (see below) and may shut down. If this occurs, check the link with Call Processing. If the link is valid, ping the network. If the network connection appears to be slower than the current Read Timeout value, increase this setting. By default, this is 1000 milliseconds (1 second). Click **OK** to accept the new setting.

NOTE: Unless necessary, use the default setting. If you set this value too high, it may slow down other operations.

7.10 Use Data Compression: When this Settings menu option is checked, DB Programming information going through the CPC or CPU to the Voice Processing PC will be compressed to speed up data transfers.

CAUTION

You should not disable this flag unless instructed to do so by Inter-Tel personnel.

C. STATUS BAR

7.11 The status bar at the bottom of the screen shows the descriptions of the menu options and the total number of sessions saved.

D. SESSION MANAGER FIELDS

7.12 **Session Description**: For a new session, enter the desired description in the Session Description field, up to 40 characters. To display a previously saved session, open the session description drop down list box and select the desired entry.

7.13 **Session Type Tabs**: Four types of sessions are supported, as shown on the tabs at the bottom of the screen: local database, direct cable, modem, and network. The Session Manager requires you to enter the following information before a session can begin.

- **Local Database**: A local session allows you to change your database selection, use the current selection, or use a default database.
  - — *Change Selection*: When a local session is started, you will be prompted to select the database for this session. If a Current Selection exists for this session, it will be deleted.
  - — *Current Selection*: This option uses the same database from the previous local session. If a previous selection was made, the database version will be displayed next to the current selection option.
  - — *Default*: When you start a local session, a screen will be displayed asking you to select the version of the System database and whether it is a dual or quad system.
- **Direct Cable**: For a direct connection, you must select the communications port and set the bit rate to match that of the system associated with the session. Use the list boxes to scroll to the proper settings.
  - — *Port*: Select the PC COM port that is used for the direct connection.
  - — *Speed*: Set the bit per second (bps) rate for the specified COM port.
  - — *System version*: This is the version of the system that will be connected to the Session Manager. If the session manager connects to the system and the version is incorrect, the session manager will attempt to reconnect with the correct version if it is installed on a PC. Updating the system version on this tab will also update it on every other tab it appears on.
- **Modem**: To set up a modem connection, you must enter the phone number and type of the modem device before beginning the session.
  - — *Phone Number*: Enter the phone number you will dial to reach the System's modem.
  - — *Modem*: Select the name of the modem to use. A drop down list box appears with the names of all the modems installed in your PC.
  - — *System Version*: This is the version of the system that will be connected to the Session Manager. If the session manager connects to the system and the version is incorrect, the session manager will attempt to reconnect with the correct version if it is installed on a PC. Updating the system version on this tab will also update it on every other tab it appears on.
  - — *Settings*: If you select this button, the settings for the selected modem will be displayed.

- **Network**: For a network connection, you must enter the IP address of the system associated with the session.
  - — *IP Address*: Enter the IP address of the system.
  - — *System Version*: This is the version of the system that will be connected to the Session Manager. If the session manager connects to the system and the version is incorrect, the session manager will attempt to reconnect with the correct version if it is installed on the PC. Updating the system version on this tab will also update it on every other tab it appears on.

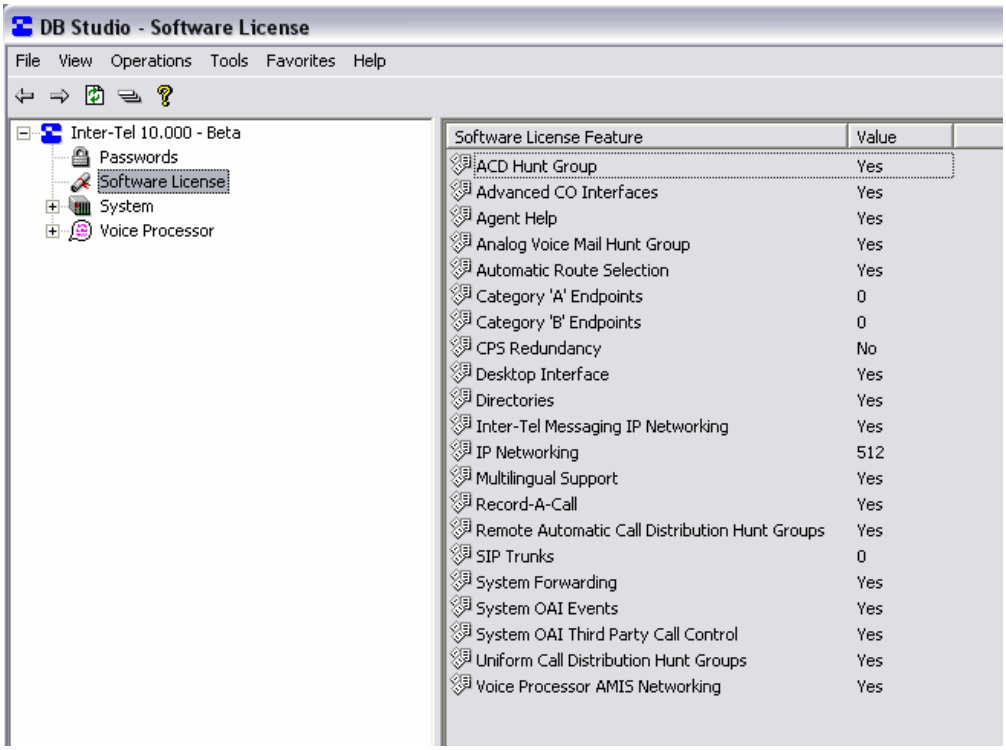
8. SOFTWARE LICENSE UPLOAD

8.1 After you are in DB Programming, you must upload software license. If you do not upload the software license, the system considers the license as "invalid" and issues Alarm 125 (see page 7-20).

NOTE: Licenses are tracked by and specific to the serial number on the CPC or CP Server security key. If you attempt to upload a license to a CPC or CP Server with a different serial number, an error message appears, and the upload fails.

8.2 To upload a license:

1. Select Software License Operations - **Upload Software License** from the Operations menu.
2. Enter the path and name of the license file or click **Browse** to search for the file.
3. Click **Start**. The system compares the new software license to the existing license, if any, as shown below. For details about changing software licenses, see page 7-105.



NOTE: If you click **Start** before selecting a license (`.isl`), a message appears, informing you that you have not identified a file name.

4. Click **Finish**. If you click **Cancel**, the license is not uploaded.

8.3 If the license is invalid, an error message appears, indicating why the upload failed. A license may be considered invalid for a variety of reasons, as described in the following table.

Table 6-6: *Software License Upload Error Messages*

Error Message	Possible Cause	Corrective Action
Corrupted Software License. Unable to load Software License.	The software license is corrupted.	Download the file from the *edGe* Web site again. If this error occurs again, contact the Inter-Tel Software Maintenance Center. For readers located in Europe, download the software from the Inter-Tel Europe FTP server.
The specified Software License is not valid for this specific CPU. Unable to load Software License.	The serial number on the CPC does not match the number in the license.	Either install the CPC identified on the software license or load a license that has the correct serial number.
The specified Software License has expired. Unable to load Software License.	The software license is no longer valid or has expired.	Load (or request) a new Software License.
Too many devices equipped. Unable to load software license.	The current database is programmed for more devices than the software license allows.	Either use local mode to unequip some devices or upgrade the capacity limit on the license.
Software License version is not supported. Unable to load Software License.	The license you have attempted to load is for a different version of system software.	Either install the correct version of system software or upload a software license that specifies the correct system version.
Software License is already loaded. Unable to load Software License.	A valid software license is already loaded on the system.	If the software license you are attempting to load is the correct license, start a new DB Programming session with a default database. Then upload the software license before restoring the database.
Software License is not valid for the current CPU type.	The CPU type identified in the software license does not match the CPU type detected in the system.	Either install the correct CPC or upload a license that identifies the installed CPU type.
Security key is either invalid or missing.	The software license serial number does not match the serial number on CP Server security key, or the security key is not present on the CP Server.	Make sure the CP Server security key is present and properly seated on the parallel port. If the key is present, make sure the license reflects the correct serial number. If necessary, transfer the license to the correct serial number (see page 7-103).
The specified Software License is not valid for this software version. Unable to load Software License.	The software license product version does not match the version installed on the system.	Either install the correct system version or upgrade the license to reflect the correct system version.

Table 6-6: *Software License Upload Error Messages (Continued)*

Error Message	Possible Cause	Corrective Action
Cannot upload to a redundant backup Call Processing Server. You must make this Call Processing Server the master to upload a Software License.	The Call Processing Server is the backup server in a redundant system, and software licenses are valid only for master Call Processing Servers.	Make sure the master Call Processing Server has the correct software license key (part no. 827.9775) installed on the parallel port. If the correct software license key is installed, you may have to transfer the license (see page 7-103) to a different security key.
The specified Software License encountered an unknown upload error. Unable to load Software License.	An unidentified error occurred.	Contact the Inter-Tel Software Maintenance Center.

8.4 After you have successfully loaded the software license, you can begin programming the system.

9. 99-NODE NETWORK CONFIGURATION

9.1 This section describes how to configure the 99 nodes feature. For complete information about 99 nodes support, see page 1-12.

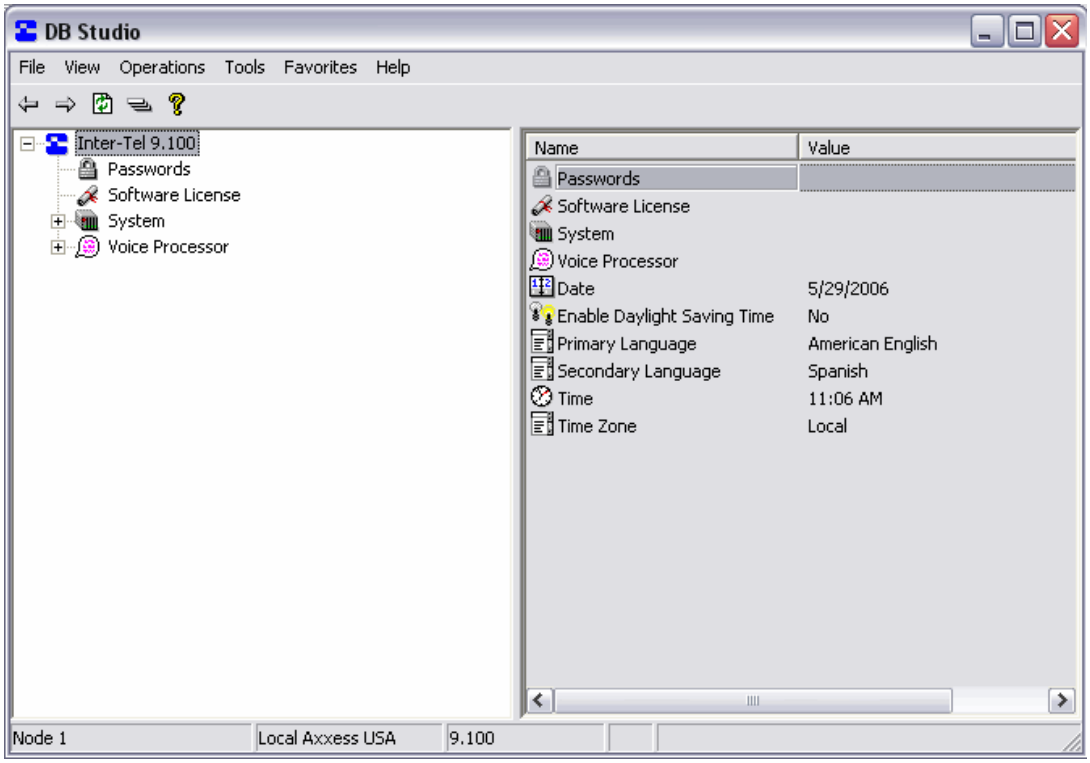
9.2 To configure a 99-node network:

1. In DB Programming remote mode, go to Operations - Database Operations - **Database Save**, and save the existing database for each node.
2. Perform a Database Conversion and upgrade existing nodes to Axxess v10.0 or later software. Ensure all nodes in the network are upgraded.
3. Verify that the "AXXESS LICENSE ENABLE 99 NODES" (part no. 840.0632) license is enabled for each node.
4. Go to Operations - Database Operations - **Database Restore**, and restore the converted database for each node. This allows the system to run properly in an extended node network.

NOTE: You can create more than 64 nodes in local mode. However, if you attempt to load a license without 99-node support, DB Programming alerts you that the license you are attempting to upload is not valid with the current Database. It also suggests that you delete all nodes with numbers greater than 63 if you want to upload this license.

9.3 System Backward Compatibility: When the network has been upgraded to v10.0 or later, the network can still support a downgrade of systems. However, when you create a remote node 64 through 99, this introduces a "no-turning back" point in the network. The creation of a remote node greater than 63 changes the internal protocol for private networking. In general, the Inter-Tel 5000 can be used as a node within a network with Axxess nodes. However, after the Inter-Tel 5000 system is set up for 99-node capacity, it can no longer be used in a network with Axxess nodes of version 7.0 (for PRI private networking) or version 8.1 (for IP networking) because they do not support 99-node networks. At this time, it is imperative that you do not downgrade any nodes to below v10.0. Older versions of software cannot interpret the protocol change correctly.

10. DB PROGRAMMING WINDOW



CAUTION

If the DB Programming session is slow to respond, shut down any other programs, including anti-virus programs, that may be running. If this does not help, make sure the PC meets the minimum requirements (see page 2-99).

10.1 All programming is done through the programming window shown above or one of its dialog boxes. The DB Programming interface, called DB Programming Studio, makes it easier to move between programming areas when creating the customer's database. It functions very much like Windows Explorer and follows many of the same conventions.

10.2 The status bar at the bottom of the window shows the description and number of the Node being programmed, the type of programming session, and the name assigned to the programming session in Session Manager. It also displays status and error messages when applicable.

10.3 The window provides access to all of the programming areas. In the window, you can see both the hierarchy of the database and all the items in each programming area. When you click on an element in the left side of the window, the contents of that programming area will appear on the right side of the window.

10.4 Some tips for using the DB Programming Studio interface include:

- While programming, it is best to display the information in Details view so that the input fields are visible.

- In Details mode, you can sort a list of items by clicking the column headings. To sort files in reverse order, click the column heading once more.
- In many programming areas, you can click your right mouse button to see a menu of available commands.

10.5 For a complete list that shows the location of each DB Programming field, see page 6-95.

10.6 The toolbar has special icons for the following:

Back and Forward: These buttons (and the Back and Forward options in the View Menu) allow you to move Back and Forward between views.

Refresh: This button is the same as the Refresh option in the View menu. It allows you to refresh the display. You can also use the F5 button.

Collapse All: This button collapses all expanded folders in the left pane and returns the focus to the root folder.

Help: Select this button to display the Help About DB Studio screen.

10.7 There are several drop-down menus along the top of the window that are used as described below. At this level, there are also several system-wide parameters that you can program. See page 6-104 for Date, Time, and Time Zone programming instructions.

A. FILE MENU

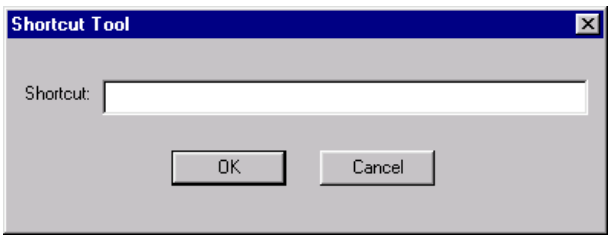
10.8 This drop-down menu includes only the Exit option. You can use it to exit your programming session.

B. VIEW MENU

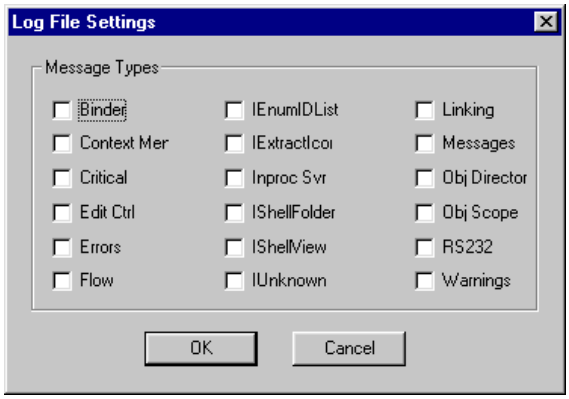
10.9 The View drop-down menu includes the following options:

- **Tool Bar**: Enabling or disabling this options determines whether the tool bar at the top of the window is displayed.
- **Status Bar**: Enabling or disabling this option determines whether the status bar at the bottom of the screen appears. The status bar shows the name of the session.
- **Back and Forward**: These options (and the Back and Forward buttons shown above) allow you to move Back and Forward between views.
- **Collapse All**: This option (and the tool bar button shown above), collapses all expanded folders in the left pane.
- **Errors/Warning Window**: This option allows you to enable or disable the errors and warnings window that appears at the bottom of the screen.
- **On-Line Monitor**: This option enables or disable On-Line Monitor mode. On-Line Monitor is used for troubleshooting system problems. ***It should only be used when requested by Inter-Tel Technical Support or Engineering personnel.***

- **Shortcut Tool (CTRL+T)**: When you select **Shortcut Tool** from the View menu or press **CTRL+T**, the following screen appears. To quickly jump to a specific programming area, enter its abbreviation. (See page 6-95 for a list of keyboard shortcuts.)



- **System Device Info**: Selecting this option displays all of the system devices. The TDM and IP devices are displayed at the top of the list, followed by the system's non-physical devices at the bottom.
- **Log File Settings**: Selecting this option displays the following window. It allows you to select the types of log files that will be saved.



- **Refresh**: Select this to refresh the screen.

C. OPERATIONS MENU

10.10 The Operations drop-down menu has several options. Each of these options is described in detail in the following pages.

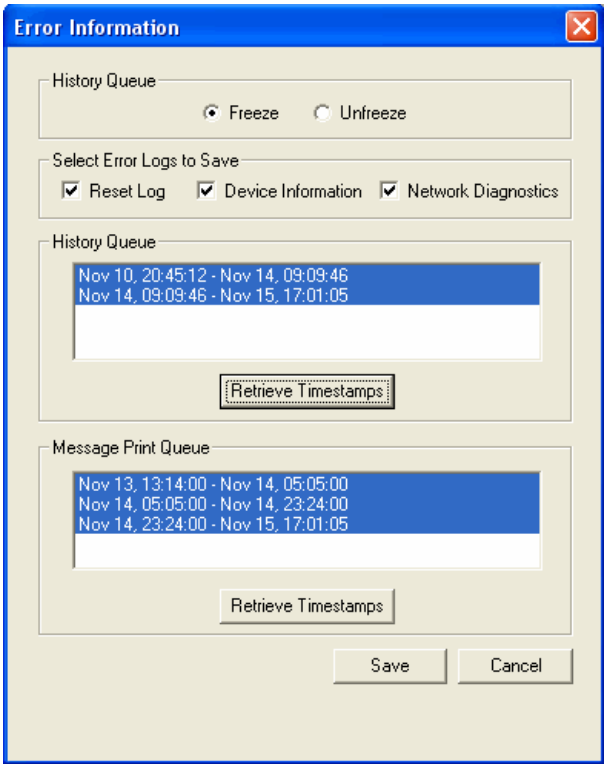
- Database Operations
  - — Database Save and Restore (see page 6-65)
  - — Voice Processor Save and Restore (see page 6-71)
  - — Backup Database Save (see page 6-69)
  - — Default Database (see page 6-70)
- Error Information (see page 6-52)
- Export/Import Devices (see page 6-83)
- IP Device Status (see page 7-78)
- Print Reports (see page 6-82)
- Software License Operations
  - — Upload Software License (see page 6-45)
- System Manager CA Certificate Upload (see page 6-86)
- Reset Now (used for immediate system resets, see page 6-271 for more information)

10.11 Error Information: If you need to do system troubleshooting, you can freeze and unfreeze the system history queue and/or save error information using this option.

10.12 To retrieve history and Message Print queue blocks:

NOTE: If Automatic Diagnostics Delivery (ADD) is running, you cannot retrieve Message Print timestamps or save Message Print logs. If you attempt to retrieve Message Print timestamps or save Message Print logs while ADD is running, an error message appears.

1. Select **Error Information** from the Operations menu. The following screen appears.



NOTE: When you select the Network Diagnostics option before you complete a system freeze, the freeze includes a Network Diagnostics Log file with an extension of **`.ndl`**. This log file captures settings for CPS, CPC, IPRC, and IP settings, as well as insufficient bandwidth alarms. This information can be obtained using Diagnostics Monitor (see page 5-324).

2. If the system is not already frozen, select **Freeze**. Otherwise, continue to the next step.
3. Click **Retrieve Timestamps** to view the timestamps associated with the history and Message Print queues. The blocks are listed based on the time intervals.
4. Highlight the timestamps that you want to save.

NOTE: To select multiple timestamps that are listed consecutively, hold down **SHIFT** while selecting the first and last desired timestamp. To select multiple individual timestamps, hold down **CTRL** while making your selection.

5. Click **Save**. The standard Windows browse screen appears.
6. Select the destination for the files (maximum 65 characters, including the freeze file names) and click **OK**.

10.13 When DB Programming saves the files, it uses the naming convention specified earlier.

NOTE: The history and Message Print queues are stored in RAM, not battery-backed RAM. This means that if the system loses power, the queue information is lost. Inter-Tel recommends, therefore, that you retrieve history and Message Print information before powering down when troubleshooting the system. Queue files, however, are not affected by minor or major resets.

D. FAVORITES MENU

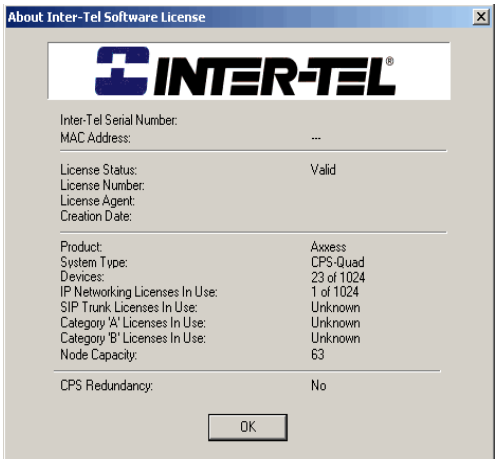
10.14 With the Favorites menu, you can add shortcuts to your most frequently visited folders to the Favorites list. Then you can then simply open the Favorites menu and select the desired entry from the list. The menu also has options for sorting the list and deleting entries.

E. HELP MENU

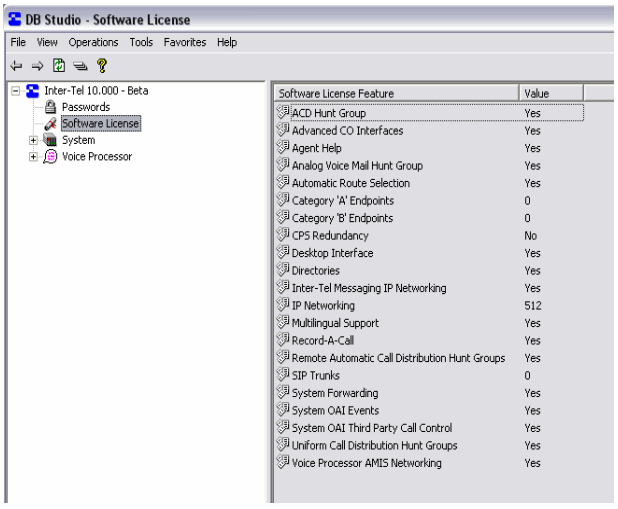
10.15 The Help drop-down menu allows access to the on-line Help files. This drop-down menu also includes entries titled "About DB Studio," "About Inter-Tel DB Programming," and "About Inter-Tel Software License" that, when selected, displays the software part number and version and software license information.

10.16 Because there is now a Software License folder, much of the information previously located on the About Inter-Tel Software License dialog box no longer displays in the About Inter-Tel Software License menu option. The dialog box contains main license and system information along with the current status of dynamic usage information. Also, the current node capacity appears. See the following examples. See page 6-97 for details about the Software License directory.

Help - About Inter-Tel Software License



Software License Folder

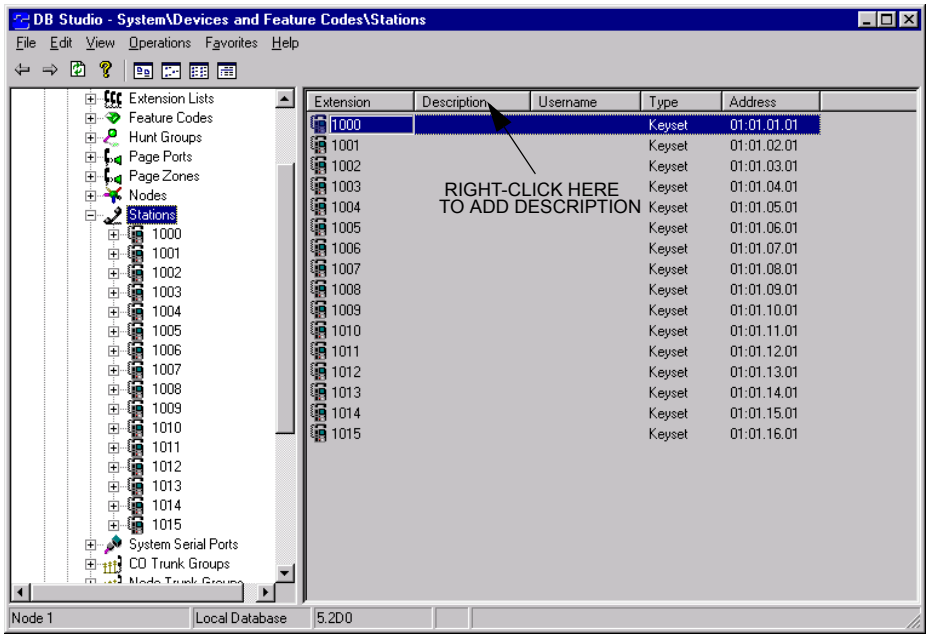


NOTE: To check the call processing software version at an Administrator's phone, you can enter 9900 [9100 in Europe] to enter Diagnostics mode, then enter 9928 [9128] to view the version information for the call processing software. When finished, enter 9900 [9100] again to exit Diagnostics mode.

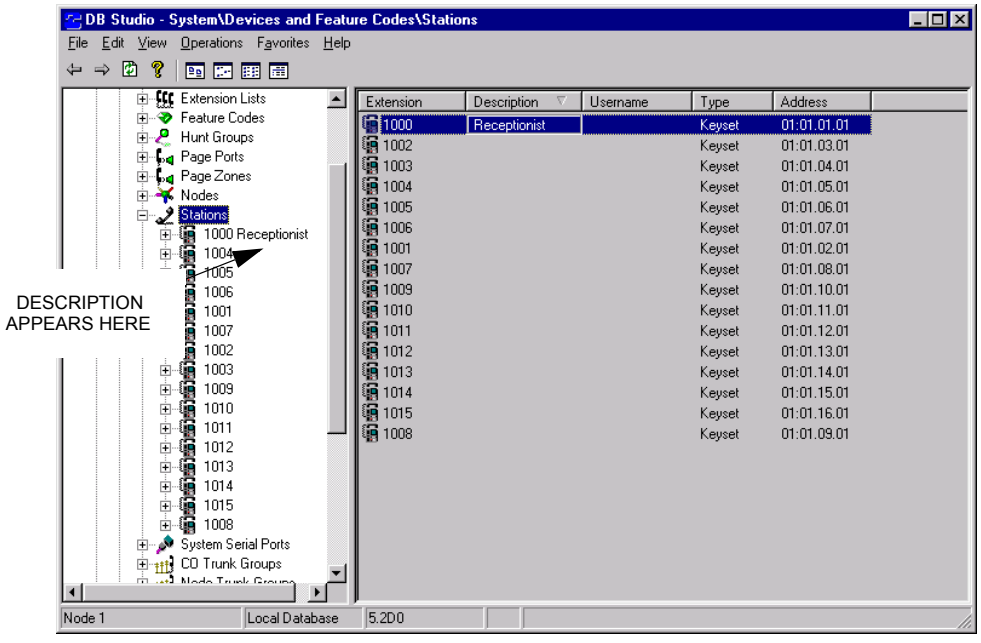
F. CHANGING INFORMATION DISPLAYS IN THE PROGRAMMING WINDOW

10.17 You can add information to the left side of the programming window to make items easier to identify. You do this by opening the folder to which you want to add information. For example, look at System - Devices - Stations, as shown below. Each Node has a Description, Username, and Node Number that are not displayed on the left side of the screen.

10.18 To add the information, right-click on the field you want displayed. For example, right-click on the Description field, if you want it to appear with nodes on the left.



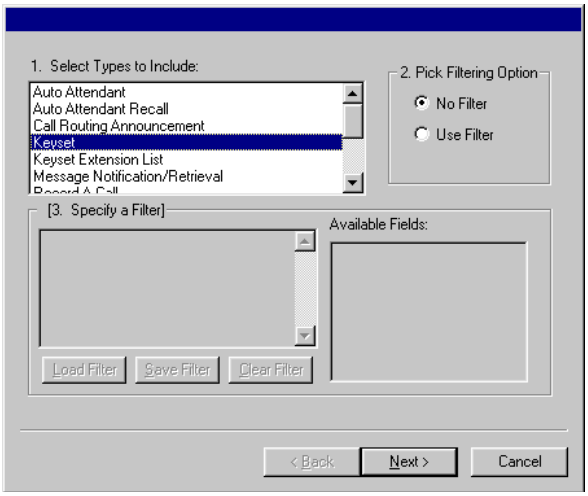
10.19 You will see an option to add the field to the folder name. Select that option and the field will appear in the folder name at left, as shown in the example below. To remove the field, repeat the procedure and select the option to remove the field. (When adding fields to the folder name, the recommended limit is three fields.)



G. FILTERING LISTS OF DEVICES

10.20 DB Programming contains a filtering feature which increases the speed with which you can sort the items in a list of devices. This allows much faster importation of data for the selection process and can be used to quickly find devices that meet certain criteria when you are searching for them.

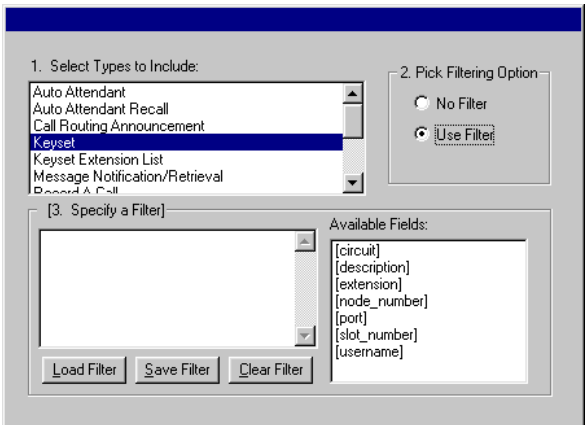
10.21 Filtering is available each time you choose to add to a list (for example, adding to a list of members of hunt groups, adding to a list of stations with a particular class of service, adding to a list of Audiotex recordings, etc.). The first step in adding to the list brings up a dialog box as shown below:



10.22 First, you must select a device type to include. The sample box shown above was copied from the interface for adding members to a hunt group. If **Next** is selected with "No Filter" selected, the next screen would show a list of all devices of the type(s) selected.

10.23 With filtering, it is possible to narrow your search even further. To make use of the filtering option, select one device type (filtering cannot be used if more than one device type is selected). Then enable **Use Filter**. This will activate the "Specify a Filter" and "Available Fields" areas of the screen. The Specify a Filter box is used to provide a statement specifying the exact search parameters. The Available Fields box provides a list of the fields that can be used in building the search statement.

10.24 Electing to use filtering will also make available the "Load Filter," "Save Filter," and "Clear Filter" boxes which are used to save and restore filter statements to and from your disk drive. When the "Use filter" option is selected, the screen will change to look like this:



10.25 If no filter statement is specified, clicking **Next** will cause an error because there is no filter statement to check.

SQL Commands

10.26 To use the filtering option, a filter statement must appear in the "Specify A Filter" box. The statement must be presented in SQL (Structured Query Language) format. SQL recognizes the following operators:

Table 6-7: *SQL Operators*

<	Less than
<=	Less than or equal to (order of symbols must be correct)
=	Equal to
>=	Greater than or equal to (order of symbols must be correct)
>	Greater than
BETWEEN	A value that is greater than or equal to a specified lesser value and less than or equal to a specified greater value. For example 3, 4, 5 and 6 are BETWEEN 3 and 6.
NOT	Negates a statement.
IN	Same as OR
AND	Boolean operation indicating that two statements must both be true
OR	Boolean operation indicating that either of two statements must be true
?	Wildcard character for a single character
*	Wildcard character for any number of characters
LIKE	Used for equalities that are not exact matches. See text for examples.

10.27 The fields that are used to construct the SQL statement are provided in the Available Fields list box. This box serves two purposes. First, it shows what fields can be used to construct the SQL search statement and, second, it provides quick access to them. Double-click on any of these fields and it will appear where the cursor exists in the Specify a Filter box.

SQL Statements

10.28 SQL commands and Available Fields are combined to construct a SQL expression. For example, assume that you want to specify extension number 1234. The SQL expression to do this is:

[extension] = "1234"

10.29 Note that the "extension" field to be searched is in square brackets. The value to be matched in that field (the number 1234), is placed in quotes. The spaces on either side of the equal sign are not required but help make the statement easier to read.

10.30 To place this expression into the Specify a Filter box, you can place the cursor in the box and type the expression. Or, you can place the cursor into the Specify a Filter box, then point at and double-click the desired field in the Available Fields box. This will cause the field to appear at the cursor location in the filter specification box. At that point, you type the remainder of the expression. If **Next** is pressed, the system will present a new box showing any phone with an extension of 1234. Remember that the device selected (keyset) was specified in the Select Types To Include box.

10.31 You can use the equal (=) and less than (<) or greater than (>) commands to widen the field of the search. For example, assume a database consisting of eight phones, all phones, with the extension numbers 1001 through 1008. You could use the following commands to narrow the list.

Table 6-8: *SQL Command Examples (1)*

SQL Statement	Resulting List
[extension] < “1005”	1001, 1002, 1003, 1004
[extension] <= “1005”	1001, 1002, 1003, 1004, 1005
[extension] = “1005”	1005
[extension] > “1005”	1006, 1007, 1008
[extension] >= “1005”	1005, 1006, 1007, 1008

10.32 You can also use Boolean operators. For the next chart, assume the same database of eight stations.

Table 6-9: *SQL Command Examples (2)*

SQL Statement	Resulting List
[extension] = “1005” OR [extension] = “1006”	1005, 1006
[extension] = “1005” AND [extension] = “1006”	Nothing found
[extension] NOT BETWEEN “1005” AND “1007”	1001, 1002, 1003, 1004, 1008
[extension] BETWEEN “1005” AND “1008”	1005, 1006, 1007, 1008
[extension] >= “1005” AND [extension] <= “1008”	1005, 1006, 1007, 1008

10.33 Notice how the last two expressions return the same result. You can use either the < and > symbols or you can use the Boolean expression BETWEEN for the same things.

10.34 Also note that the third expression returned nothing even though it seems like it should have returned 1005 and 1006. The reason it didn’t do this is because that expression requests a list of phones that have the extension number 1005 and also have the extension number 1006. To simplify, the AND expression applies to an individual device. To satisfy this, both numbers would have to be assigned to the same station. This is not allowed by DB Programming. The correct expression would be to use the operator “OR” which would request a list of phones having the extension 1005 or the extension 1006.

10.35 Some of the fields do not require quotes around the values. These values are considered to be numbers, not string expressions. String expressions, such as extension numbers and user names must always be in quotes. Numeric expressions, such as node and port numbers, should not be placed in quotes. This chart specifies the type of each field:

Table 6-10: *SQL Field Quotes*

Field	Type	Requires Quotes?
[circuit]	Number	No
[description]	String	Yes
[extension]	String	Yes
[node_number]	Number	No
[port]	Number	No
[slot_number]	Number	No
[username]	String	Yes

10.36 In the event that you forget which type it is, you will see an error message pointing out that you have a "Data type mismatch" and you must return to the statement and enter it correctly.

Wildcards

10.37 Wildcard characters that can be used for filtering are:

- Asterisk (*) means any character or characters are accepted.
- Question mark (?) is used to indicate a match of a single character.

10.38 Because filtering uses statements composed of letters as well as numbers, the equal sign means that an exact match must be present. Therefore, the expression *[username] = "FRED*"* will only return a username that has the letters F, R, E, D and an asterisk; that is, an actual asterisk as the fifth character of the username. It would not return "FREDDY" or "FRED SMITH." To use the wildcard, the SQL statement must tell the search engine to locate something that does not exactly equal the search string specified in the quotes. To do this, we use the SQL command "LIKE."

10.39 The expression *[username] LIKE "FRED*"* returns usernames of "FRED," "FREDDY," "FRED SMITH" or any other username that starts with the letters F, R, E, D.

10.40 In a like manner, the question mark is used to indicate only a single character space is a wildcard.

10.41 Here are some additional examples:

Table 6-11: *SQL Command Examples with Wildcards*

SQL Statement	Search Result
[username] = "FRED?"	FRED?
[username] LIKE "FRED?"	FRED plus any single character
[username] LIKE "FRED*"	FRED followed by any or no additional characters
[username] LIKE "FRED??"	FRED followed by any two characters
[username] LIKE "FRED?*"	FRED plus at least one more character
[username] > "b"	Any username except those beginning with the letter A
[username] < "a"	Any username that would occur before those beginning with the letter A, (e.g.: blanks)
[extension] LIKE "12??"	Extensions in the range 1200 through 1299 inclusive

Combining Commands

10.42 It is possible to combine search patterns using the Boolean operators we have already discussed, as shown in the following examples:

Table 6-12: *SQL Command Combination Examples*

SQL Statement	Search Result
[username] = "FRED" AND [extension] = "1234"	Only 1234 with a username of FRED
[username] = "FRED" OR [username] = "WILMA"	Any phone with a username FRED or WILMA
[username] = "FRED" OR [extension] = "1234"	Extension 1234 as well as any with FRED as the username

Table 6-12: *SQL Command Combination Examples (Continued)*

SQL Statement	Search Result
[extension] LIKE “12*” AND [username] LIKE “F*”	Extensions beginning with the digits 1 and 2 that have usernames beginning with the letter F
[extension] < “1234” AND [username] = “FRED”	Extensions with a number lower than 1234 that also have a username of FRED
[slot_number] = 1 AND [extension] >= “1000”	Stations on board number 1 having extension numbers 1000 and higher

Clear Filter, Save Filter, and Load Filter Command Buttons

10.43 There are three command buttons below the Specify A Filter box. They are used as follows:

- **Clear Filter**: You can remove the entire filter specification by selecting **Clear Filter**. This will remove the entire entry in the Specify A Filter box.
- **Save Filter**: There will probably be instances where you have a complex, or a commonly-used search that you will want to use over again. When you have a search statement that you want to keep for future use, you can save it to floppy or hard disk for later retrieval. The way to do this is to select **Save Filter** when the search statement is in the Specify A Filter box. A dialog box will appear asking you to specify a location and file name. The PC will then save the file in that location with an extension of “FLT.”
- **Load Filter**: When you want to recall a saved search, select **Load Filter**. A dialog box will allow you to find your previously saved search string and restore it to the Specify A Filter box.

H. NAVIGATING WITHOUT A MOUSE

10.44 For those times when a mouse is not available, or if you simply prefer to use keyboard commands, the information presented here can be used to navigate the programming window and perform those commands normally associated with the mouse.

Use Details View

10.45 If you haven't already done so, make sure that you are in Details View. This insures that all the necessary information is shown in the programming windows. To make sure that your machine is set up in this way, press **ALT+V** to drop down the View menu. Then press **D** to specify Details View.

Window Panes

10.46 The programming window is divided into two panes. The left pane is a list of folders, while the right pane is a list of folders and files. In relation to System programming, the left side of the screen (left pane) may be thought of as the "programming areas" (like Station Programming, Hunt Group Programming, etc.). The right pane contains the individual components (Specific Stations and Specific Hunt Groups).

Mouse Click Substitutions

10.47 The movement of the programming focus—that part of the screen where you want to make changes—is primarily controlled by the arrow keys and by the **TAB** and **SHIFT+TAB** keys. These keys move the programming focus right and left, respectively. In addition, you need to know that **ENTER** takes the place of a single click, and **CTRL+ENTER** takes the place of a double-click.

- **Arrow Keys**: With the highlight in the left pane, you can navigate using the arrow keys. Note that placing the highlight on any item on the left pane automatically shows the contents of that item in the right pane, whenever applicable. For example, place the highlight on "System" in the left pane and the computer shows the contents of the system in the right pane: Cabinets, Devices and Feature Codes, Flags, etc. At this point, the programming focus is on the words "System" in the left pane. It is highlighted and there is a dashed line around the words. This is the programming focus. It is possible to have more than one highlight on the screen at one time, but there can only be a single focus.
- **Expand (+) and Collapse (-)**: If the item in the left pane is expandable (which is indicated by a small "+" sign to the left of the item), it is possible to expand it to show all of its contents in the left pane, by pressing the plus (**+**) key on the numeric keypad. Note that the numeric keypad must be used for this function. Likewise, to collapse an expanded left pane list (indicated by a small "-" sign to the left of the item), press the minus (**-**) button on the numeric keypad.
- **Tab/Shift to Change Focus Between Panes**: To shift the programming focus between the left pane and the right pane, use **TAB** and/or the **SHIFT+TAB** key. To select items, use **CTRL+ENTER**. To practice using these keys, do the following:
  a. Place the highlight on System in the left pane. The contents of System appear on the right.
  b. To select Devices and Feature Codes for programming, press **TAB**. Then press the down arrow key. The highlight in the right pane will move down to Devices and Feature Codes. To move from the right pane to the left pane, use **SHIFT+TAB**.

NOTE: You can use **TAB** and **SHIFT+TAB** to move to the drop down directory path window as well. This window gives a shortcut definition to your current programming selection and can be used to navigate to any point above your current location using the arrow keys.

c. At this point, your programming focus and highlight is in the right pane on Devices and Feature Codes. If you want to select the **Devices** and **Feature Codes** area in the right pane, press **CTRL+ENTER**. (This is the equivalent of a double-click.) On the right side of the screen, Devices and Feature Codes will expand showing Extension Lists, Feature Codes, Hunt Groups, etc. Notice that the programming highlight has moved back to the left pane. This is because you have selected Devices and Feature Codes, but nothing below it. There is no highlight in the right pane.

d. Now move the highlight to the right pane using **TAB** and expand Extension Lists by pressing **CTRL+ENTER** while the highlight is on it. Again, the highlight will automatically move back to the left pane.

e. Put the highlight back in the right pane and use the arrow key to move down to Keyset and press **CTRL+ENTER** again. You now see a list of Keyset Extension Lists in the right pane. Notice that the highlight has again moved to the left pane. Move it back to the right pane using **TAB**.

- **CTRL+N to Add Items In The Right Pane**: If you want to add an item to a list (such as the Keyset Extension List in the example above), you can right-click with the mouse. However, if you're not using a mouse, you can use **CTRL+N**, ("N" as in "New") to bring up the prompt that allows you to add to the list. Pressing the down arrow key will highlight this prompt. Pressing **ENTER** (equivalent of a single click) will create a new item.
- **CTRL+M to Edit An Existing Item**: If you wanted to edit an existing item, use the arrow keys to highlight the entry and press **CTRL+M**, ("M" as in "Modify"). This brings up the dialog allowing you to explore (open) or delete the item.
- **ESC Closes Without Change**: Pressing **ESC** tells the system that you do not want to make a change. If a pop-up dialog box is present on the screen, it will be closed.
- **Changing A Flag**: Flags have two states (on/off, enabled/disabled, yes/no, etc.). To change a flags, you must highlight it, select it and change the state of the flag.
  - — When viewing a list of flags, press **TAB** to move the programming highlight to the right pane. Then use the down arrow key to move to the desired flag. The highlight is currently on the name of the flag, not the flag itself. You must move the highlight to the flag to change it by pressing the right arrow key. You can see the focus of the highlight move from the name of the flag to the state of the flag.
  - — Press **ENTER** (single click) and the check box for the flag appears. You can then change the status by using the **SPACE BAR** to toggle between the choices. When the check box shows the desired status, press **ENTER**.

- **Selecting From A Drop-Down List**: Move the programming focus to the list box, as described above for changing a flag. Then, with the programming focus on the box, press **ENTER** to select this area to program. The box will "open up" and highlight its current Value. Then, you can use the up and down arrow keys to increment or decrement the value. You can also type in a value directly. When the desired value is shown, press **ENTER**. An example of drop down lists is the Cabinets programming folder. Follow these steps to practice using drop-down lists:

  a. Press **CTRL+ENTER** (double-click) to open the Cabinets programming. The highlight will move to the left side of the screen. Use Tab to move it back to the right and use the down arrow to select the first card slot (01:01).

  b. Press **ENTER** (single click) to open the drop-down list for this card slot. You can now program the card for the slot using one of the three following methods:

  - Use the arrow keys to scroll through the entries in the list. Only one entry will display. When you have the desired value in the box, press **ENTER**.
  - Press **ALT+Down Arrow** to show the entire list of choices. The drop-down box will show all possibilities with the current choice highlighted. You can use the up and down arrow keys to select the value desired. When you have done so, press **ENTER**. This will collapse the box with your value selected. If you would like to collapse the box without changing the selection you can press **ESC** (highlight removed) or press **ALT+Up Arrow** (highlight remains).
  - Press the first letter of the device type you want to specify. In other words, if you want to specify a "Loop Start Card," press **L**. Each subsequent press of the L key will rotate through the choices beginning with the letter L.

Keystroke Summary

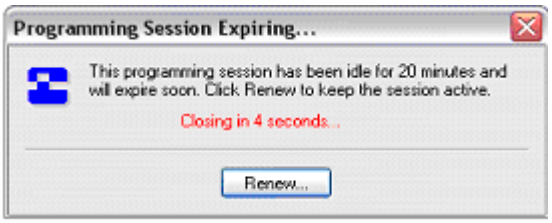
TABLE 6-13: *Keystroke Summary*

Key	Action
TAB **SHIFT+TAB**	Moves the highlight between left and right pane or between drop-down path box (immediately below Menu Bar) and the left pane.
ALT+TAB	Switches between open programs in Windows.
Left or Right Arrow	Moves programming focus to left or right.
Up or Down Arrow	Moves programming focus up or down. Scrolls up or down in drop down list boxes.
SPACE BAR	Changes status of a flag when the flag is selected for programming.
ENTER	Equivalent to a single click. Also used to accept a highlighted item in a drop-down list box.
CTRL+ENTER	Equivalent to a double-click.
ESC	When programming an item, it restores the original value (no change made) and removes focus from that item. If a drop-down list box is open, it closes the box with no change.

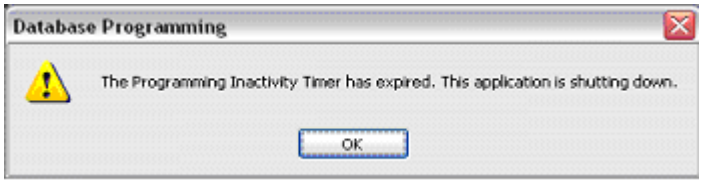
I. SESSION INACTIVITY TIMER

10.48 In systems running v9.0 and later, DB Studio has an inactivity timer that automatically shuts down the application when it has been idle for a specified amount of time. An idle session is one where no data has been sent or received between DB Studio and Call Processing, and the user has not browsed any pre-cached folders or menu items during the specified period.

10.49 When the session has been idle for the specified amount of time, the inactivity timer expires and the dialog box displayed on the right appears.



10.50 If you click **Renew** before the 10-second countdown expires, the DB Studio session is renewed. If you do not click **Renew** before the 10-second countdown expires, the connection to CP is dropped and the following dialog box appears, warning that DB Studio is shutting down.



10.51 The programmable idle period is a field in the main folder and can be modified only in the On-Line Monitor (OLM) mode. The valid range is 20 - 120 minutes, and the default is 20 minutes.

NOTE: Do not use OLM mode unless you are instructed to do so by support personnel.

11. DATABASE OPERATIONS

CAUTION

The Voice Processing Unit must remain attached when saving and restoring the database. If the unit is not attached, converted voice mail information, such as non-associated mailboxes and cascade levels, will not be restored.

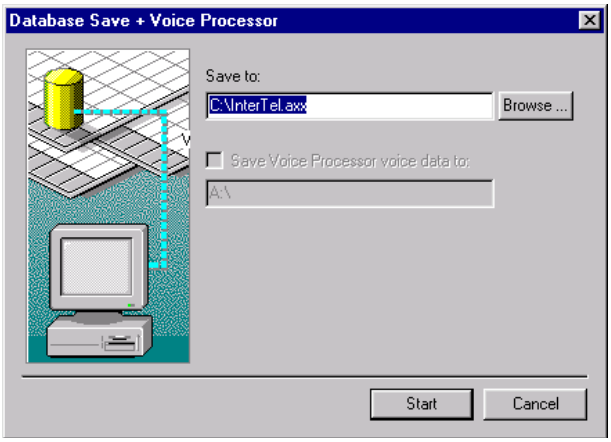
11.1 The Database Operations submenu, which is available under the Operations menu, allows you to save, restore, default, and backup the database. Each of these options is described in the following sections.

NOTE: Performing a database Save or Restore operation will cause the phone system to slow down. If at all possible, these operations should not be performed during normal business hours when the system is being used to place and receive calls.

A. DATABASE SAVE

11.2 After the System and Voice Processor databases have been programmed for the first time, you should perform a database save. Store this database in a safe place to be used in the event that the CPC or CPU database ever becomes corrupted and must be re-entered. This saved database can be used as a starting point for the re-entry. In addition, you should keep a current copy of the database that you can update every time you make changes. When this option is selected, the window shown below appears. It allows you to save the database to disk during a programming session.

- **Save To**: Enter the name of the file where the database will be saved in this box. Or, select **Browse** to locate the file. This file is saved to the local database PC.
- **Save Voice Processor voice data to**: *(Available in remote mode only when a Voice Processing PC is attached.)* Select this option to save the Voice Processor database to the Voice Processing PC attached to the system. Then enter the destination path for the database.
- **Start**: To begin the operation, select **Start**. If you chose to save to a file that already existed, it will ask if you want to overwrite it. When the operation is finished, a message will show that the operation was successful. (During a save operation you may need to insert a second disk. If so, replace the first disk with a blank, formatted floppy disk.)
- **Cancel**: To stop an operation before it is completed, select **Cancel**. A message will show that the operation was aborted by the user.



B. DATABASE RESTORE

CAUTION

Poor line quality may cause data transmission problems when the modem connection exceeds 19200 baud. For this reason, Inter-Tel does not recommend using the CPC modem to restore the database. If you attempt a restore using the modem, the database may become corrupt.

11.3 When this option is selected, the window shown below appears. It allows you to restore the database to disk during a programming session.

NOTICE

If the Database Restore feature is used during a remote programming session, all calls are disconnected except the modem connection. Before restoring the database, ensure that the circuit card being used for the modem connection will not be reconfigured during the restoration (changed to a different type of card).

- **Restore From**: Enter the name of the file where the database was saved in this box. Or, select **Browse** to locate the file.
- **Start**: To begin the operation, select **Start**. When the operation is finished, it will automatically close the programming session.
- **Cancel**: To stop an operation before it is completed, select **Cancel**. A message will show that the operation was aborted by the user.

NOTICE

Canceling a restore operation will cause the system to default the database and terminate the programming session.

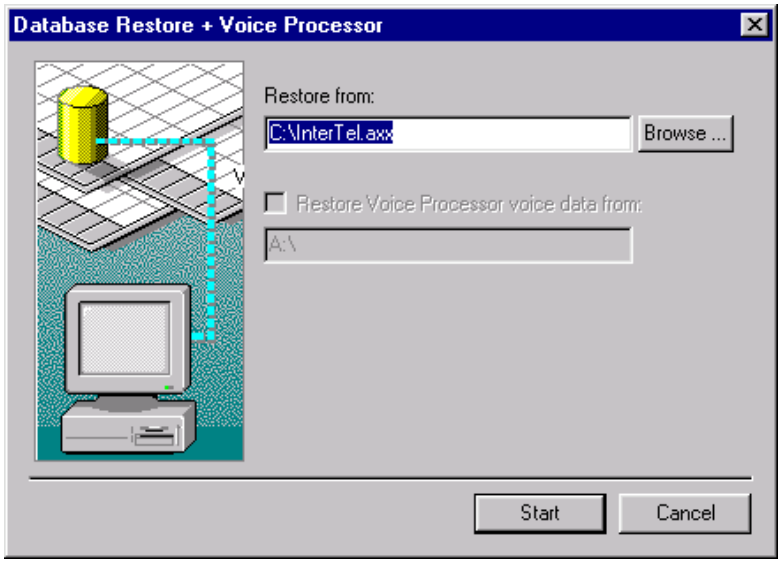
— If the number of multi-protocol endpoints in the database being restored exceeds the number indicated in the license, an error message appears, and the database is not restored. (This applies to software versions 8.1 and later.)

— If the premium features enabled in the database being restored do not match the premium features available in the software license, an error message appears, and the system is inoperative. (This applies to software versions 8.0 and later.)

— If the database being restored requires more Premium Feature units than the system can support, you will see a warning message and the system will be inoperative. (This applies to software versions 7.0 and earlier only.)

— If the database being restored contains Voice Processor information and the system does not have an operating Voice Processor, you will see a warning message and no Voice Processor information will be loaded during the restore operation.

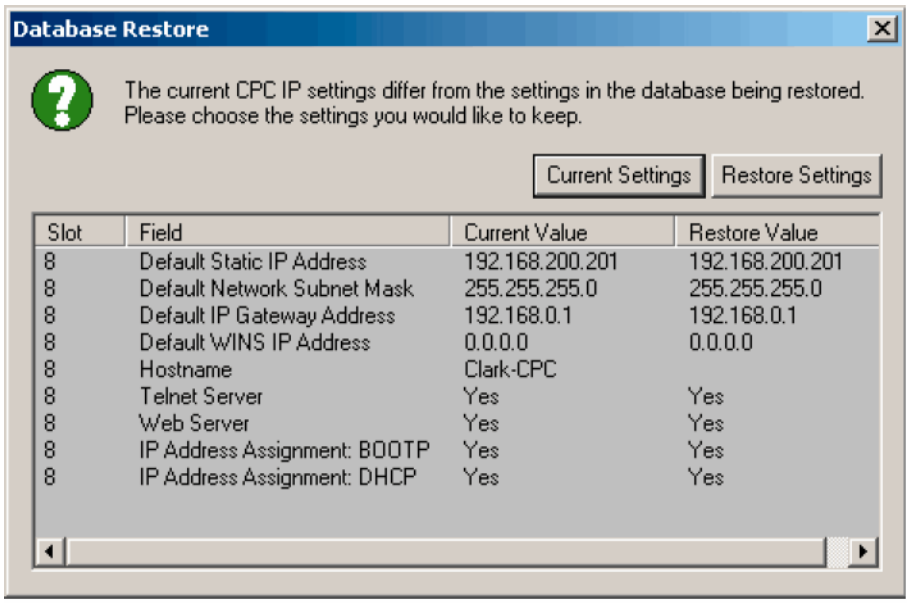
— If restoring a database in Local Mode, you may receive an error message that says, "Database is incompatible," due to the differences in CPU types. It is highly recommended that you use the **Change Selection** method in Session Manager when doing database restores in a local session. This will load any database over an existing local session database that was loaded previously without any problems.



11.4 When a database is restored, the system checks the following against the currently loaded license:

- CPU type
- Number of devices
- Software licenses

11.5 If one or more of these items in the database differ from what is programmed in the license, an error message appears, and the database is not restored. If the license is valid for the database to be restored, the system checks the current IP settings for the CPC. If any differences are detected, a screen similar to the following appears.



11.6 This screen displays the current IP settings (Current Value column) and the settings detected in the database to be restored (Restore Value). If you click **Current Settings**, the current settings, as shown in the Current Value column, are retained. If you click **Restore Settings**, the IP settings are automatically changed to those shown in the Restore Value column, overwriting the current settings.

11.7 If, however, you defaulted the database or uploaded new software, the system considers the static database as "corrupt." When the system is powered up, a backup database is not detected, and the IP settings programmed in the restored database are automatically used.

C. DATABASE BACKUP

NOTE: Version 8.0 or later software is required to use the backup database feature.

11.8 As of version 8.0, a backup copy of the database is automatically stored on the CP Server's hard drive or in flash memory in the CPC. This backup remains available, even if the system experiences a power failure or if the battery backup loses power. If the system is powered up and a corrupt database file is detected, the backup file in flash memory or on the hard drive is restored.

NOTE: The Voice Processor's database is not stored in flash memory.

11.9 This backup database is designed to recover in the event of a total power failure and is not designed to replace the static database. Changes made to the static database are, therefore, not reflected in the backup database until the database is automatically or manually backed up. For example, if you default the database, the static database is defaulted, not the backup database.

11.10 In addition, the backup database allows the system to automatically convert databases when the software is upgraded. For example, if the system is upgraded from v8.01 to v8.02, the v8.01 static database is detected and considered corrupt. When this occurs, the system restores the backup database and converts the information to a v8.02 static database. The backup database version, however, does not change until a backup is performed (i.e., it remains a v8.01 database).

11.11 The system repeatedly attempted to restore a backup database but was interrupted by power-ups. If the static database is corrupted, the system attempts to restore the backup database. If the system detects problems with the backup database, it may default the static database instead of restoring the backup database. The database will be defaulted if one of the following occurs:

- There is no backup database available.
- The system was saving the backup database when the power down occurred.
- The system repeatedly attempted to restore a backup database but was interrupted by power ups. The system will default the database once it has exceeded the number of restore attempts, as specified in DB Programming (see page 6-107). If this occurs, the backup database remains in flash memory for troubleshooting purposes.

NOTE: You can use the * button to prevent the backup database from loading onto the system. See page 7-88 for details.

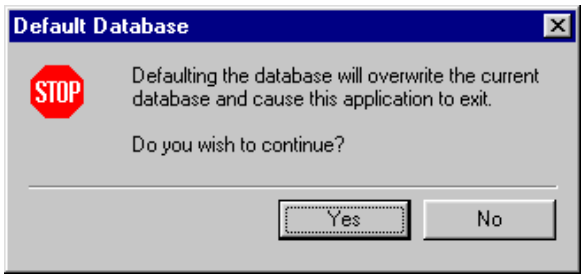
11.12 If desired, you can force DB Programming to save the backup file immediately, as described below.

11.13 To force a backup database save:

1. Select Database Operations then **Backup Database Save** from the Operations menu.
2. Click **Start** to create the backup. The backup file is saved and stamped with the new date/time.

D. DEFAULT DATABASE

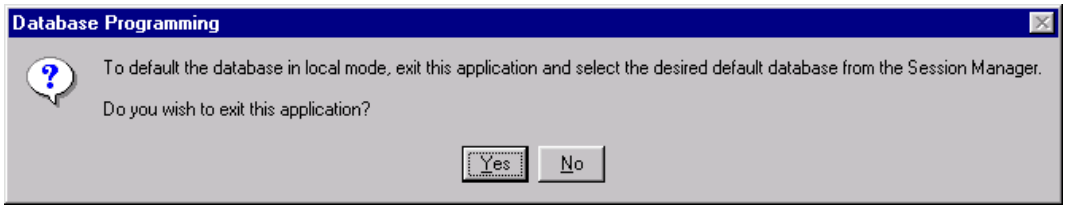
11.14 On an Windows-based Voice Processing Unit, you can default the database to return the call processing and Voice Processor databases to default values. When you select **Default Database**, a window appears, as shown below, that warns that defaulting the database will overwrite the current database and asks if you want to continue.



11.15 Select **Yes** to default the database or select **No** to leave it unchanged.

NOTE: Defaulting the database will end the programming session and drop all calls. It will also cause Voice Processing to stop. Voice Processing will restart after the database default operation is complete. However, the Voice Processing Unit will not reset.

11.16 If you are programming in Local Mode when you select **Default Database**, you will see the following warning:



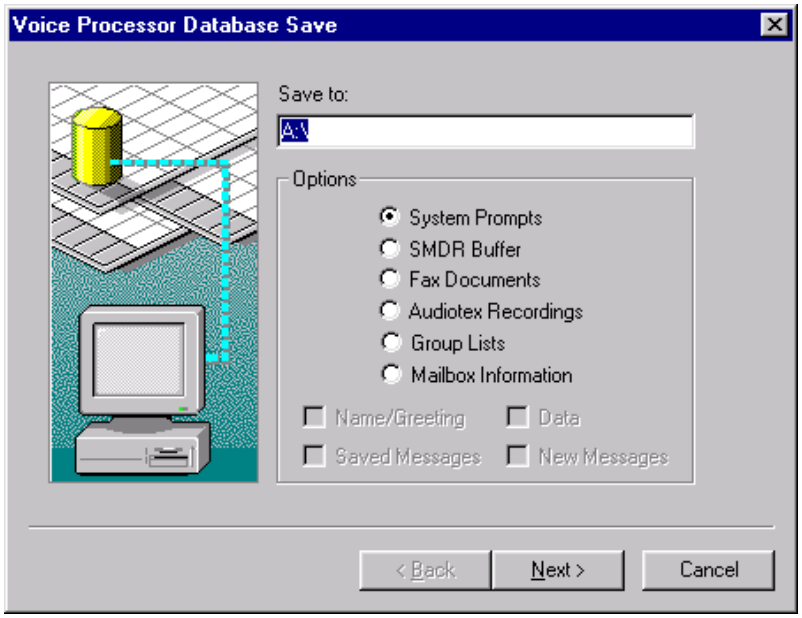
12. VOICE PROCESSOR SAVE AND RESTORE

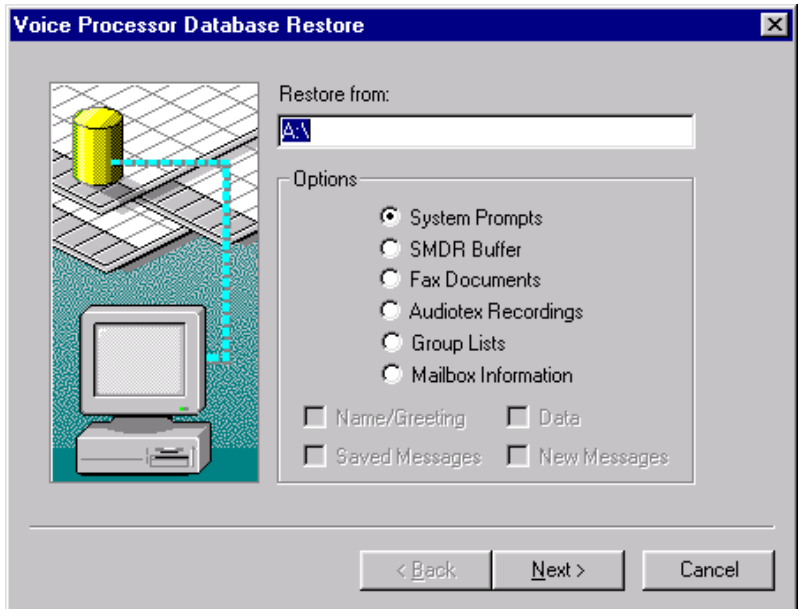
12.1 The selective Voice Processor Save and Restore options are available under the Database Operations submenu in the Operations drop-down menu.

NOTE: Ensure the VPU is attached during a restore. If the VPU is not attached, the system will not restore the non-associated mailbox information.

12.2 To ensure that you have enough diskettes for saving the data, check the Disk Usage Statistics report (as shown on page 6-345) to determine the number of minutes used for the voice mail data. Each diskette will hold 5-6 minutes of data. To calculate the number of diskettes needed, divide the number of minutes used (shown in the statistics) by 5.

12.3 When you select either Voice Processor Save or Voice Processor Restore from Operations - Database Operations, one of the following windows appears:





A. SAVE TO OR RESTORE TO

12.4 Determine which drive (on the Voice Processing PC) will be the save destination or restore source. Selecting the desired drive places a dot in the option button.

NOTE: Be sure to place a diskette in the correct drive on the Voice Processing PC before selecting **Next.**

ALSO: When the Voice Processing Unit performs a save operation, it erases files and/or directories from the UNC_path. Therefore, be careful when specifying the save/restore path. For example, do not specify C:\ as the save/restore path because the Voice Processing Unit would delete the entire C drive.

B. OPTIONS

12.5 Select information you want to save or restore by placing a dot in the corresponding button. If you select Mailbox Information, you will also need to select the types of information by placing checks in the appropriate check boxes.

NOTE: Note that restoring Voice Processor data **will not** allow you to create new mailboxes or group lists. You can only restore data to mailboxes and groups lists that already exist in the System database.

12.6 The options are as follows:

- **System Prompts**: The system prompts (default and customized) will be saved or restored. Restored information will overwrite the current prompts, if any.
- **SMDR Buffer**: If you select Save, the current call record information in the SMDR buffer will be saved to the selected drive. The buffer will not be cleared. If you select Restore, the SMDR call records contained on the disk in the selected drive will overwrite the existing SMDR buffer contents.
- **Fax Documents**: All fax documents in the database will be saved or restored. Restored documents will overwrite existing documents of the same number, if any.
- **Audiotex Recordings**: All Audiotex recordings will be saved or restored. Restored recordings will overwrite the current recordings of the same number, if any.
- **Group Lists**: All group lists and group list directory information in the database will be saved or restored. Restored information will overwrite existing information for group lists of the same number, if any.
- **Mailbox Information**: You can select the mailboxes information to be included. Depending on the items checked, the following will be saved or restored:
  - — **Name/Greeting**: Name and greetings recorded by the Subscribers, including primary and alternate greetings and directory names for extension IDs.
  - — **Saved Messages**: Messages that have been saved by the Subscribers.
  - — **Data**: User statistics for each mailbox and extension ID.
  - — **New Messages**: New and undelivered messages that have not been heard by the Subscribers.

NOTE: Restored mailbox information will overwrite the existing information in the database. If any certified messages are restored, the "certified" flag will be cleared so that the sender does not receive duplicate receipt messages. If a message that is being restored is already in the mailbox, the message will not be copied from the disk. It will not create a duplicate copy of the message.

C. COMPLETING THE SAVE/RESTORE

12.7 Selecting Recordings, Group Lists, or Mailboxes: If you are saving recordings, group lists, or mailbox information, you must select the items you want to save. When you select **Next**, a window appears that allows you to select the items that you want to save. You can select several consecutive items by selecting one, holding down **SHIFT** and then selecting the last. Or you can select individual items by holding down **CTRL** while clicking on the desired items. When you have finished, select **Next** again.

12.8 Next/Back: When you select **Next**, the selected information appears for reference. If it is not correct, select **Back** and correct the information. If it is correct, select **Finish** to continue.

12.9 Finish: When you select **Finish**, the information will be saved or restored to the selected drive. The following windows may appear:

- A restore operation begins with a warning that the database is about to be overwritten with the new data. You are given the choice of allowing the overwrite (select **Yes**) or canceling the restore process (select **No**).

  NOTE: Use a separate disk for each type of save (System Prompts, SMDR Buffer, and Fax Documents). Each time you perform a save operation, ***ALL FILES*** on the disk are erased and replaced with the new saved information.

- During a save or restore process, a window appears that shows the percentage of the database that has been copied. To cancel the save or restore operation, select **Cancel**.

  NOTE: Canceling a restore operation will cause the system to default the Voice Processor database.

- If restoring mailbox information, group lists, or Audiotex messages, place the ***last*** disk of the saved information in the drive. This disk has the file that tells the system which information was saved and has the requested information included in the saved data. The system checks to see if the requested information is contained on the disks. If not, you will see an error message. If the information is available, you will be instructed to insert the appropriate disk(s) to complete the restore operation. Only the information you checked in the Save/Restore windows will be restored to the system, even if other information is present on the disks.

12.10 The Voice Processor applications and/or mailboxes will not be usable while they are being saved or restored and callers will receive reorder tones. This is necessary to prevent users from making new recordings and causing database errors. *If a mailbox is busy when you attempt to restore it,* you will receive a warning message with the following three options:

- *Abort the Save/Restore operation*: This terminates the entire save or restore operation, even if you were performing the operation for more than one mailbox.
- *Cancel the operation for that mailbox*: This terminates the save or restore operation for the busy mailbox.

- *Try again*: If you are saving or restoring multiple mailboxes, it places the busy mailbox at the end of the list and retries it later. If only one mailbox is being restored, it immediately tries again. If the mailbox is still busy when retried, you receive the three options again.

D. CREATING CUSTOM SAVE/RESTORE DISKS

12.11 By copying selected files from disk to disk, you can create a set of disks containing selected information from a Save operation. For example, you can have one set of disks with saved mailbox messages and another with saved Audiotex messages and group lists.

12.12 When restoring custom data files, any fields not present on the disks will not be modified during a restore. For example, if a mailbox's programmable data file contains only the username field, restoring the file will affect only the mailbox's username.

12.13 Custom-made disks must conform to the following rules:

- The disks must correctly use the naming conventions, as described in paragraph 12.14 on the next page, for each of the individual save categories.
- Each disk must contain a DISK.* file in the root directory that specifies the contents of the save. The file names must correspond to the disk number of the save (DISK.001 for the first disk, DISK.002 for the second disk, etc.). Additionally, all of the DISK.* files must have the same time and date stamp.
- The last disk must contain a DISKINFO file in the root directory that specifies objects saved on the disks as well as the starting disk numbers of each of the objects. Each line of the file contains the object name and a disk number. The object name is MBxxxxx, GLxxxxx, or Rxxxxx where "xxxxx" is the mailbox number, group list number, or Audiotex recording number. The disk number specifies the first disk containing the data. This information allows the system to determine the contents of the save and to determine which disks contain what data. For example, a DISKINFO file with the following contents describes a save in which the information for mailbox 1000 starts on disk 1 and the information for mailbox 1001 starts on disk 2:

  MB1000 1
  MB1001 2

- If an object (such as a mailbox, group list, or recording) continues from one disk to another, no other objects can continue from that disk to another disk.
- Only one file may continue from one disk to the next. For example, if the alternate greeting for mailbox 1221 continues from disk 2 to disk 3 (creating a split file), all of the other mailboxes saved on disk 2 must complete on disk 2. Additionally, none of the other information for mailbox 1221 can be split between disks 2 and 3.
- If mailbox information spans more than one disk, the information must appear in the following order on the disks. For example, if a mailbox's primary greeting appears on disk 3, its recorded name cannot appear in any subsequent disks. It must appear on disks 1, 2, or 3.
  - — Programmable data
  - — Recorded name
  - — Primary greeting
  - — Alternate greeting
  - — New messages (envelope first, then voice data)
  - — Saved messages (envelope first, then voice data)

- If a group list spans more than one disk, the programmable data must appear before the recorded name.
- If an Audiotex recording spans more than one disk, the programmable data must appear before the recording.
- The Voice Processor stores database-programmable data and message envelopes in text files. These files all consist of lines of information that specify one field per line. Each line must appear in the format *field_name=value* where "field_name" is one of the fields listed below and "value" is a valid value for that field. Valid field names and values are as follows:

Table 6-14: *Voice Processor Valid Field Names and Values*

Field	Value
Numerical	An integer
Flag	Either TRUE or FALSE
Time	In the format: HH:MM
Date	In the format: MM/DD/YYYY HH:MM
String	A case-sensitive string of characters
Enumeration	Strings specifying one of several possible values (e.g., the transfer method can be unannounced, announced, or screened)

- File names for new and saved messages indicate message numbers that start with 1 and are in sequence on the disks. For example, N0001 must be the first new message. If there are additional new messages, they must be N0002, N0003, N0004,... (in that order).

12.14 During a save operation, one directory is created on the floppy disk for each mailbox, group list, or Audiotex message that is saved. The directories are named as follows:

- Mailbox directories are identified as A:\MAILBOX. The mailbox directories are named "MBxxxxx" where xxxxx is the 1- to 5-digit mailbox number. For example, the information for mailbox 1221 would be in the directory A:\MAILBOX\MB1221. The data and voice messages for one mailbox may not fit on a single floppy disk. In fact, it is very possible that the information for one mailbox may span several disks. If this is the case, the mailbox information will be saved on each of the subsequent disks under the same directory name.
- Group list directories are identified as A:\GROUPLST. The group list directories are named "GLxxxxx" where xxxxx is the 1- to 5-digit group list number. For example, the information for group list 5000 would be in the directory A:\GROUPLST\GL5000. The data and recording files for a group list may not fit on a single floppy disk. In this case, the information spans several consecutive disks.
- Audiotex directories are identified as A:\AUDIOTEX. The recording directories are named "Rxxx" where xxx is the 3-digit recording number. For example, the information for recording 1 would be in the directory A:\AUDIOTEX\R001. The data and voice data files for a recording may not fit on a single floppy disk. In this case, the information spans several consecutive disks.

12.15 When information spans several disks, the directory on the first disk ends with a "001" suffix. The suffix increases with each disk required to save the information. The directory on the last disk ends with an "END" suffix. For example, if mailbox information spans three disks, the directory on the first disk would be A:\MAILBOX\MB1221.001, the directory on the second disk would be A:\MAILBOX\MB1221.002, and the directory on the last disk would be A:\MAILBOX\MB1221.END. (If all of the mailbox information fits on one disk, there is no suffix on the directory name.)

12.16 Recordings, such as messages and directory names, may actually consist of multiple files (called segments). When these multi-file recordings are saved, they will have the filenames that use the following naming convention. In the filenames below, "y" is a letter (A, B, C, etc.) that specifies the voice-data segment. The xxxx is a sequence number that indicates the message index (0001, 0002, 0003, etc.). R24 and R32 specify the voice-data recording format.

Mailbox voice data:

DATA: Programmable data
NAME_y.R24: Company directory name
GR1_y.R24: Primary Greeting
GR2_y.R24: Alternate Greeting
Nxxxx.ENV: New message envelope
Nxxxx_y.R24 and Nxxxx_y.R32: New message
Sxxxx.ENV: Saved message envelope
Sxxxx_y.R24 and Sxxxx_y.R32: Saved message

Group list voice data:

DATA: Programmable data
NAME_y.R24: Company directory name

Audiotex voice data:

DATA: Programmable data
REC_y.R24 or REC_y.R32: Recording

12.17 All user-recorded voice data (such as greetings and mailbox messages) are in R24 format. System-recorded voice data (such as a voice mail message from the system informing a subscriber that there were 3 bad login attempts) are in R32 format. Most of the saved voice data will be in R24 format.

12.18 Files containing the programmable data and message-envelope are text files. These files always appear on a single floppy disk. They will never split and be saved across multiple disks. However, voice data (such as directory name, greetings, or messages) can split across multiple disks. When this happens, an incomplete file's filename ends in a hyphen (-). For example, if a new message 1 (consisting of only one segment) spans three disks, it would be split into the following files:

First disk:

A:\MAILBOX\MB1221.001\N0001.ENV
A:\MAILBOX\MB1221.001\N0001_A-.R24

Second disk:

A:\MAILBOX\MB1221.002\N0001_A-.R24

Last disk:

A:\MAILBOX\MB1221.END\N0001_A.R24

12.19 Note that if the message consisted of multiple segments, each segment would have a separate base filename. For example, suppose new message 1 consists of two segments. The first segment spans 3 disks and the second segment spans two. The message would be split into the following files:

First disk:

A:\MAILBOX\MB1221.001\N0001.ENV
A:\MAILBOX\MB1221.001\N0001_A-.R24

Second disk:

A:\MAILBOX\MB1221.002\N0001_A-.R24

Third disk:

A:\MAILBOX\MB1221.003\N0001_A.R24

Fourth disk:

A:\MAILBOX\MB1221.004\N0001_B-.R24

Last disk:

A:\MAILBOX\MB1221.END\N0001_B.R24

12.20 Similar conventions apply to continued segments for greetings, recorded names, and saved messages, all of which can also span several diskettes.

12.21 You can combine split voice-data files into a single file by using the DOS copy command. For example, a mailbox message is split across two disks in these files:

A:\MAILBOX\MB1000.001\N0001_A-.R24
A:\MAILBOX\MB1000.END\N0001_A-.R24

12.22 You can copy the two files to the hard drive and use the + syntax to combine them, as follows:

```
copy a:\mailboxbox\mb1000.00\n10001_a-.r24 c:\tmp
copy a:\mailboxbox1\mb000.end\n0001_a.r24 c:\tmp
copy c:\tmp\n0001_a-.r24+c:\tmp\n0001_a.r24
c:\tmp\n0001_a.r24
```

NOTE: The order of the files in the + expression is important. The copy command concatenates the files from left to right. Also, note that the copy command overwrites the original n0001_a.r24 in the example above. When combining files, you can join R24-format files only with other R24-format files and R32 format files only with other R32-format files.

E. VOICE MAIL SAVE/RESTORE TO A PC NETWORK DRIVE (B DRIVE MAPPING)

NOTE: In order for this feature to work, the Voice Processing PC must not have a B drive (either real or mapped). Otherwise, the Voice Processing will not be able to perform its own B-drive mapping.

12.23 The Windows-based Voice Processing allows the user to save or restore the database, voice data, and selected information using a network directory in addition to the Voice Processing hard drive or floppy drive. The Voice Processing Unit provides this support by mapping the B drive to the desired target directory as follows:

12.24 The AVDAP Service must log on as a specific user, not a System Account, because drive mappings in Windows-based PC are Profile-specific, and a user must log on to the local machine. Permanent drive mappings for that user, based on the stored profile, will be applied. By default, AVDAP logs on as a System Account and thus will not be allowed a drive map.

- It may be necessary to create this user.
- The user can be a Local User or a Domain User.
- You must know the user's password to use the account. It is suggested that you do not allow the user to change the password and that the password never expires.
- The user must be a member of the Local Machine's Administrator Group.
- The user must have proper rights and permissions on the target share. Full Control or its equivalent is suggested. Read/write minimally.

12.25 To map the B drive to a directory on the network, you must add a new command option to the avdap.opt file in the root directory where the Voice Processing software is installed. For example, if the Voice Processing software is installed in *C*:*AVDAP*, the avdap.opt will have an absolute filename *C*:*AVDAP**avdap.opt*. Note that capitalization is unimportant. The format of the command option to map the B drive to a network directory is:

-map_b=UNC_path

12.26 The file name is specific and has a required specific location of: "\AVDAP\avdap.opt" It may be necessary to create this file. It is a ASCII text file and can be created or edited with Notepad or any text editing program. *UNC_path* is the name of the target directory in the Universal Naming Convention as follows:

- *If the target directory is a Netware path,* the UNC_path can specify a directory path (for example, **\\nwserver\sy2\ud\avdap\save**).
- *If the target directory is on another Windows-based machine,* the UNC_path must specify a share name (for example, **\\ntserver\save_avdap_share**).
- The destination must be in UNC format: \\machine name\share name.

12.27 An example of the complete line in avdap.opt is: **-map_b=\\ntserver\vm_save**

12.28 It is not possible to map any other drive than B. *-map_b=* is a command specific to AVDAP, it is not a Windows-based command. AVDAP only allows mapping of the B drive.

12.29 The AVDAP Service must be started anew after changing its log on field and editing the avdap.opt file.

12.30 Success or failure of the B drive mapping is evident from either the AVDAP Monitor window or viewing the appropriate log file in \AVDAP\DB\MSGS. Drive mapping is attempted during AVDAP's startup.

Example of a failed mapping

[00105 1-28-1999 13:46:42 LOGON]Unable to map B: drive [67]

[00106 1-28-1999 13:46:42 LOGON]Blank username may be causing the failure

Example of a successful mapping

[00108 1-28-1999 13:48:25 LOGON]B: drive mapped to "\\machine\bdrive"

12.31 When you are ready to perform Voice Processor save or restore, specify the B drive as the save/restore drive from DB Programming. You will see a warning that says "Before AVDAP performs a save operation, it will first delete everything in the target directory including all subdirectories inside it. Therefore, it would be a bad idea to specify C:\ as the save/restore path as AVDAP will end up deleting the entire C: drive." Take this warning into consideration when backing up more than one item. Backing up Audiotex recordings to the B drive will create files and folders in the target share. If a mailbox save is then done, prior to moving the existing Audiotex data, the MB save will delete the Audiotex data.

12.32 For example, suppose a site wants to perform saves to the C:\AVDAPSV directory on the Voice Processing PC itself (machine name VM_PC). The system administrator needs to create a share for the C:\AVDAPSV directory. For example, the new shared directory is called VM_SAVE. The voice mail administrator would then need to edit the file avdap.opt and add the following line:

-map_b=\\VM_PC\VM_SAVE

12.33 To save to a directory on a different PC, the system administrator needs to create the shared directory on the target PC, give the Voice Processing's account read/write access to the directory, and specify the -map_b line that refers to the target PC's share.

CAUTION

When the Voice Processor performs a save operation, it erases files and/or directories from the UNC_path. Therefore, be careful when specifying the save/restore path. For example, do not specify C:\ as the save/restore path because the Voice Processor would delete the entire C drive.

F. VOICE MAIL SAVE/RESTORE TO A TAPE DRIVE

12.34 The Windows-based Voice Processor allows the user to save or restore the database, voice data, and selected information using a tape drive in addition to the Voice Processor hard drive or floppy drive.

12.35 To back up data to a tape drive, you will need a DOS boot disk. To perform the backup, do the following:

1. Delete any unnecessary voice mail messages.
2. Save the Voice Processor database, as described on page 6-71, without the voice data.
3. Turn off the Voice Processing PC and connect the tape drive.

> **NOTE:** If you are using a tape drive that connects to the parallel port, you may not have to turn OFF the Voice Processing PC.

4. Insert the DOS boot disk in the Voice Processing floppy drive and turn on the Voice Processing PC.
5. View the config.sys file. It should contain the following:
   Device = himem.sys
   Device = EMM386.EXE
   DOS = HIGH
   buffers = 30
   files = 30
6. Install the tape drive's backup software on the C drive, following the manufacturer's instructions.
7. Prepare the tape drive to receive files, according to the manufacturer's instructions.
8. Save the database and index files, located in the \avdap\db directory, to the tape drive.
9. Save the message files, located in the \avdap\db \rcrdings directory, to the tape drive.
10. Save the files located in the \avdap\db\sys directory to the tape drive.

12.36 To restore data from the tape drive, you will need to use the DOS boot disk again. To perform the backup, do the following:

1. Turn off the Voice Processing PC.
2. Insert the DOS boot disk in the Voice Processing floppy drive and turn on the Voice Processing PC.
3. Install the tape drive's backup software on the C drive, following the manufacturer's instructions.
4. Restore the database and index files from the tape drive to the \avdap\db directory on the Voice Processing PC. Overwrite existing files.
5. Restore the message files to the \avdap\db\rcrdings directory on the Voice Processing PC. Overwrite existing files.
6. Restore the files to the \avdap\db\sys directory on the Voice Processing PC. Overwrite existing files.
7. Reboot the Voice Processing PC and restore the database following the normal procedure, described on page 6-71.

G. VOICE MAIL SAVE/RESTORE TO A ZIP DRIVE

12.37 The Voice Processing Unit allows the user to save or restore the database, voice data, and selected information using a ZIP drive in addition to the Voice Processing Unit hard drive or floppy drive.

12.38 You can order a Zip Drive Backup Kit (part number 828.1460) that contains instructions and the following items:

- Iomega Zip Drive (part no. 520.5200) - External parallel port version, which includes the AC wall transformer, parallel port cable, and User's Manual.
- Two Zip disks (part no. 839.0002)
- 3.5 inch (1.44 MB) Save Boot Floppy disk (part no. 827.8813)
- 3.5 inch (1.44 MB) Restore Boot Floppy disk (part no. 827.8814)

12.39 In addition to the above, you will need the following to perform the backup or restore of the voice data.

- A PC (desktop or laptop) running the System DB Programming software.
- The appropriate serial cable to connect between your programming PC's COM port and the serial port on the system.
- Additional Zip disks as required (depends on the "Total Storage Used" on the Voice Processing Unit).

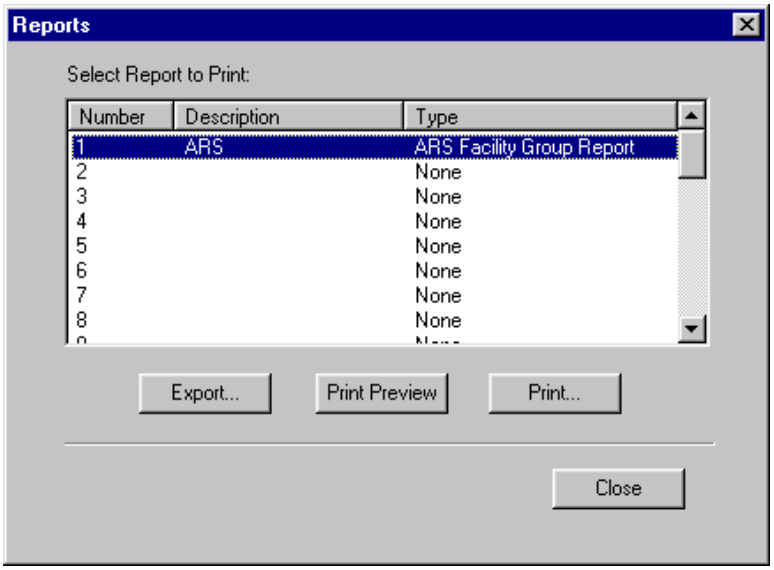
NOTE: The Zip Disks must be formatted and blank.

- A monitor, keyboard and mouse with the appropriate sized connectors (serial, PS/2, etc.) connected to the Voice Processing Unit.
- A suitable AC power source for the large AC wall transformer needed for the Zip Drive.
- A suitable amount of time when the customer's voice mail system can be down for the save/restore procedure to complete. The Zip Drive Backup/Restore utility requires the voice mail to be shut down for the duration of the save/restore operation.

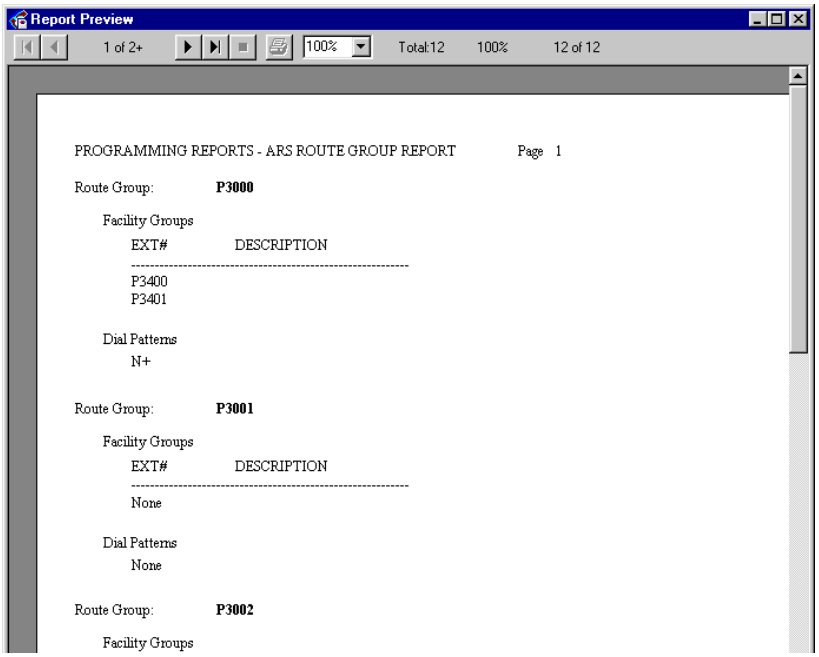
13. PRINTING REPORTS

NOTE: The Print Report menu option is disabled if a report is currently being generated.

13.1 Customized report programming is described in detail on page 6-339. To print the reports or export them to files, select **Print Reports** from the Operations drop-down menu. The following screen appears. You can see all available reports by scrolling through the Select Report To Print box. When you find the report you want to print or export, highlight it and select **Print** or **Export**.



13.2 If you want to view the report before printing it, select **Print Preview**. A screen appears, as shown below, that shows you the report exactly as it will print. You can page through the report using the arrow buttons at the top of the screen and change the view size (shown 100% below). To print from this screen, select the printer icon.



14. EXPORT/IMPORT DEVICES

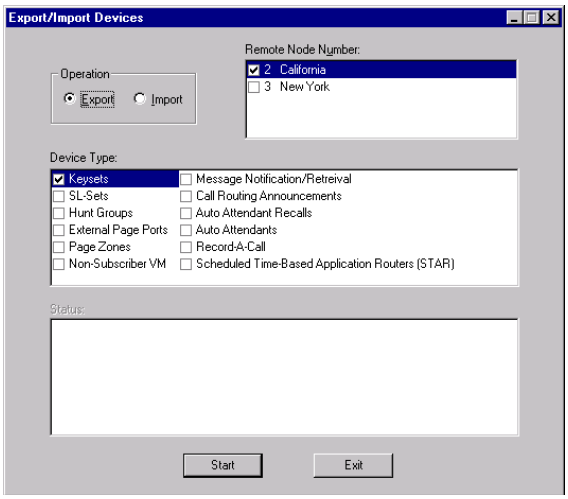
NOTE: The Voice Processor Enable flag is enabled by default. If you do not have a Voice Processing Unit connected to the System, make sure you disable this flag before you attempt to import or export information over the network.

14.1 When another node adds a new extension or changes an existing extension, username, or description, the network broadcasts the new extension information to the other nodes in the network. Or, programmers can choose to export local device information to other nodes, or import devices from the other nodes to the local node, using the Export/Import option in this menu. If there is a conflicting (duplicate) extension in a broadcast from another node, the receiving node ignores the new extension in favor of its existing extension number.

14.2 If the network is unable to export or import an extension to a node (automatically or using the Export/Import feature) because there is an active programming session on that node, the node is unable to communicate with its Voice Processor port, the node is down, or the links to the node are down, the new extension will not be added to or changed on that node. (If Message Print is enabled, error messages will indicate any unsuccessful broadcasts.) You must manually add or change the new extension in the node's database or try to export or import it later. You may want to check each node to verify that their off-node device lists are programmed properly to allow access between the nodes.

NOTE: Extension numbers that exist before the network is established will not be automatically broadcast to other nodes, until they are modified. They must be exported or imported using the Export/Import option or programmed manually at each of the other nodes. Also, each node can have only 8000 off-node device entries in its database. After that limit has been reached, new devices received through network broadcasts cannot be added to the node's database.

14.3 To export or import devices, select **Export/Import Devices** from the Operations menu. The following window appears.



NOTE: You must have a programmed Node to export information to or import information from it.

14.4 Select the following information, then select **Start**. (Or select **Exit** to cancel the operation without exporting any information.)

- **Operation**: Select the desired option to determine whether you will be importing devices into the local node or exporting local devices to a remote node.
- **Remote Node Number**: Select the node(s) you want to export the information to or import the information from, by placing checks in the appropriate boxes.
- **Device Type**: Select the types of devices you want to export or import by checking the appropriate boxes. You can select any combination of devices.

> **NOTE:** If running system versions 8.1 and later, IP device information is automatically included if you import or export all keysets. In addition, IP SLAs are automatically included if you import or export all single-line sets.

14.5 When you select **Start** to begin the export or import operation, the lower box in the screen shows the status of the export or import.

14.6 If you are exporting information, the node you are exporting from is listed as "Node X: Export Source." The destination node or nodes will show the node number and the current status of the export. After the import from a node is completed, either successfully or unsuccessfully, the import source node displays the final status of the import. The screen will show messages as explained in the chart below.

14.7 After completing an import operation, you must terminate DB Programming to update the database.

14.8 If the local node cannot communicate with the remote node, the export or import will fail. Before you attempt to export or import information, verify that you can reach the remote node by dialing the extension assigned to the remote node. If you hear dial tone, you should be able to successfully export or import information. If you receive a "NOT REACHABLE" message, you must determine why calls are not properly being routed to the specified remote node before you can export or import information. If you camp on to a node while trying to reach the remote node, wait for the node trunk group to become available so that you can guarantee the remote node can be reached.

Table 6-15: *Remote Node Connection Status*

Status	Meaning
Starting Up	Indicates that the node is attempting to initiate the export.
Creation XX%	This node is XX percent done with the creation portion of the export.
Deletion XX%	This node is XX percent done with the deletion portion of the export.
COMPLETED	This node successfully completed the export.
Waiting...	When importing, this status indicates that this node is waiting for another node to finish before it can become an import source
Error - Not Reachable	The export to the node failed because the export source node could not communicate with this node.
Error - KTS Programming	The export to the node failed because someone on the node was performing Keyset Programming.
Error - Voice Processor	The export to the node failed because the link to the Voice Processor is down.
Error - DB Programming	The export to the node failed because someone on the node was performing DB Programming.

14.9 After you attempt to export or import information, check Message Print output for error messages:

- If there is an active programming session on the remote node, you cannot export information to that node until the programming session is terminated. Also, if the remote node has a Voice Processor and the link is down, the export will fail on the remote node.
- If an existing extension on the remote node conflicts with an exported extension number, you will see one of the error messages listed in the following table. The error message indicates the resulting action on the remote node. In the example used in the table below, the device information was exported from local node 1 to remote node 2.

Table 6-16: *Remote Node Connection Error Messages*

Extension # Exported from Local Node 1	Existing Extension # on Remote Node 2	Results on Remote Node 2
Phone 1000	Phone 10	WRN DB CP Device Info Import For ID(1:'1000') Failed: Conflict with existing extension
Phone 1000	Phone 100	WRN DB CP Device Info Import For ID(1:'1000') Failed: Conflict with existing extension
Phone 1000	Phone 1000	WRN DB CP Device Info Import For ID(1:'1000') Failed: Conflict with existing extension
Phone 1000	Off-Node Device ID(3:1000)	WRN DB CP Device Info Import For ID(1:'1000') Failed: Conflict with existing extension
Phone 1000	Unassociated Mailbox 1000	Off-Node Device ID(1:1000) created and Mailbox 1000 is associated with the off-node device
Hunt Group 2000	Unassociated Mailbox 2000	Off-Node Device ID(1:2000) created and Mailbox 2000 is associated with the off-node device
Phone 1000	Unassociated Mailbox 100	Off-Node Device ID(1:1000) created and Unassociated Mailbox is deleted
Phone 1000	Off-Node SL-Set ID (1:1000)	Off-Node Device ID(1:1000) deleted and Off-Node Device ID(1:1000) created
Page Port 91000 [99000 in Europe]	Unassociated Mailbox 91000 [or 99000]	Off-Node Device ID(1:91000 or 1:99000) created and Mailbox 91000 [or 99000] remains unassociated
Phone 1000	Off-Node Device ID (1:10XX)	WRN DB CP Device Info Import For ID(1:'1000') Failed: New extension matches existing wildcard

15. SYSTEM MANAGER CA CERTIFICATE UPLOAD

15.1 If the node will connect to System Manager, you must upload the Certification Authority (CA) certificate. This allows the node to use Secure Socket Layer (SSL) for a secure connection.

NOTE: You cannot upload the CA certificate while in local mode.

15.2 To upload the CA certificate:

1. Select **System Manager CA Certificate Upload** from the Operations menu.
2. When prompted, click **Browse** and locate the CA certificate (usually C:\Inter-Tel\System Manager\CA). Certificates have a **`.cer`** extension.
3. Click **Start**. The CA certificate is uploaded.

15.3 For details about using System Manager, see the *System Manager Installation Manual*.

16. NETWORKING WIZARD

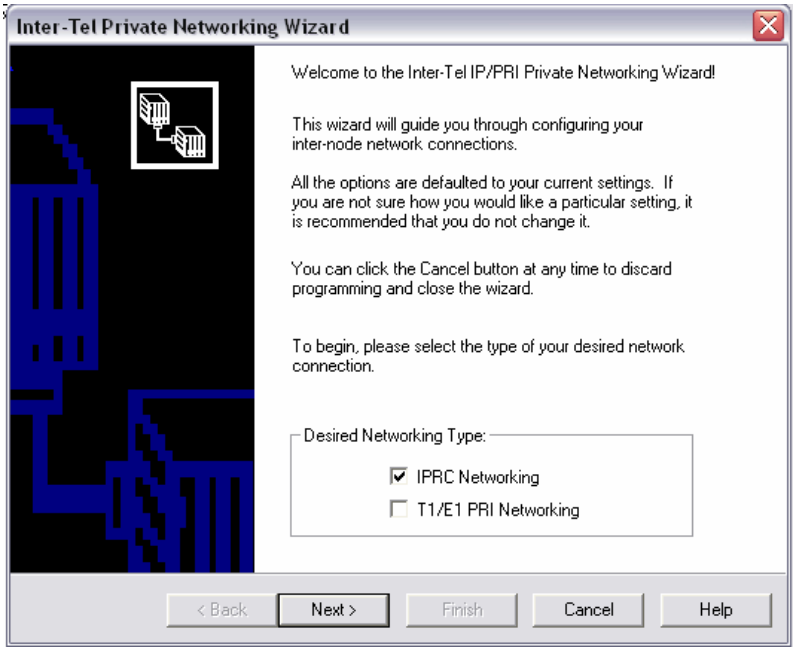
16.1 To access the networking wizard, select **Networking Wizard** from the Operations menu, or under System\Devices and Feature Codes\Nodes\<Node>, right-click in a blank portion of the right window and select **Networking Wizard**. For troubleshooting information, see page 7-65.

NOTE: The **Back** button on the wizard screens allows you to return to a previous screen to modify programming. To finalize programming done through the networking wizard, you must click the **Finish** button.

16.2 The Networking Wizard allows you to program the following information:

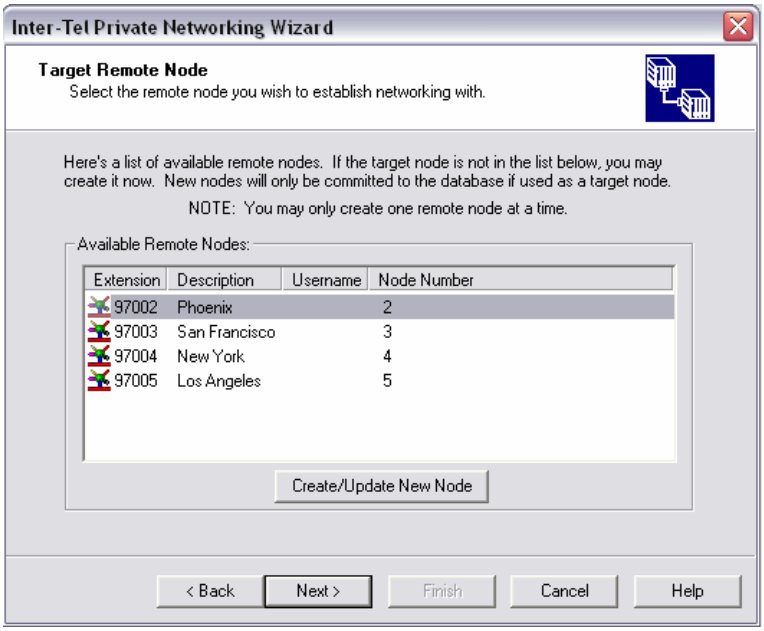
- Selecting the Networking Type—IPRC or T1/E1 (see page 6-87)
- Selecting/Creating the Target Remote Node (see page 6-87)
- Selecting/Creating Networking Boards (see page 6-89)
- Updating the IP/PRI Configuration (see page 6-91)
- Configuring IPRC Private Networking (see page 6-91)
- Configuring T1/E1Private Networking (see page 6-92)
- Updating Node Trunk/IP Connection Group Settings (see page 6-93)
- Completing the Private Networking Setup (see page 6-94)

16.3 Selecting the Networking Type (IPRC or T1/E1): The first screen of the networking wizard allows you to choose the type of private networking you would like to configure. By default, IPRC Networking is selected.



16.4 Selecting/Creating the Target Remote Node: The second screen of the networking wizard allows you to choose which node you would like to network with. If the node already exists, select it from the list and then click **Next**. The **Create/Update New Node** button allows you to create and configure a new node as part of the Networking wizard.

NOTE: You can create only one node at a time while you are using the networking wizard.



16.5 To create and configure a new node:

1. Click **Create/Update New Node**. The Create Remote Node dialog box opens, as shown below.

NOTE: If the node limit has been reached, an error appears.

2. Select an Extension for the node using the drop-down list box.
3. Enter a Description for the node.
4. Enter the Username for the node.
5. Select the Node Number using the drop-down list box. Make sure the node number you select for the new node complies with your numbering plan.
6. Click **OK**.

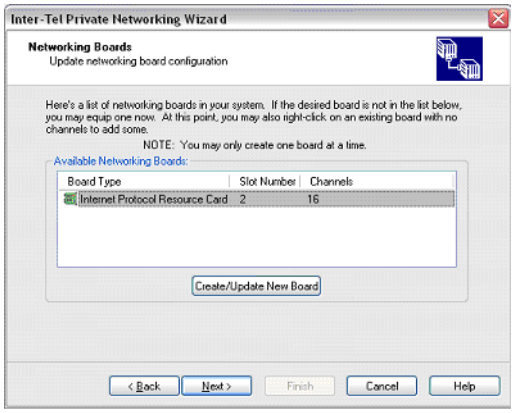
16.6 The new node is added to the Target Remote Node screen of the wizard. If you want to network to this new node, select it from the list and click **Next**.

NOTE: Although the new node is available on the wizard's screen, it will not be added to the database unless you select it as the target node and complete the networking wizard. If you create the first remote node on the system using the networking wizard, DB Studio automatically shuts down when you click **Finish**.

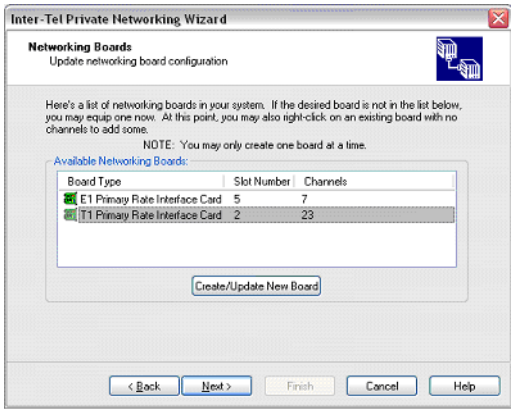
16.7 **Selecting/Creating Networking Boards**: The next screen of the networking wizard allows you to select a networking board from the list of boards in the system. If you are configuring IPRC networking, a list of available IPRCs appears. If you are configuring T1/E1 networking, a list of available T1/E1 PRI Cards appears.

16.8 Select a board from the list and then click **Next**. If the system does not have an available networking board, you can create and configure one by clicking **Create/Update New Board**.

IPRC Networking

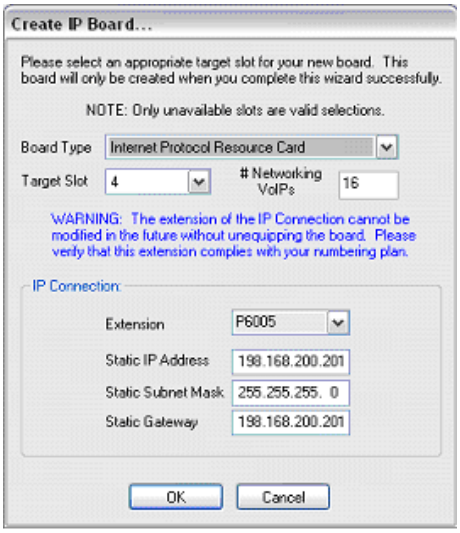


T1/E1 Networking



16.9 **To create and configure a new IPRC:**

1. Click **Create/Update New Board**. The Create IP Board dialog box appears as shown below. The Board Type is pre-populated as Internet Protocol Resource Card, and cannot be changed.



2. Select the Target Slot for the card using the drop-down list box.
3. Select the number of Networking VoIP resources for the IPRC.

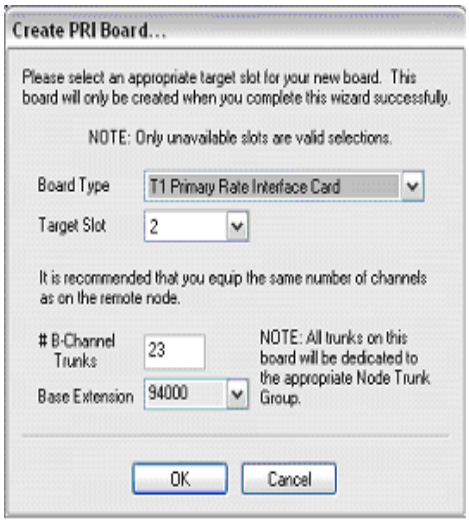
NOTE: The number of Networking VoIP resources is either 16 channels or the maximum number of Networking VoIP resources that may be created without exceeding the device limit (not exceeding 32).

4. Select the Extension for the IP connection using the drop-down list box. This extension is not easily modified. Make sure the extension you select for the new IP connection complies with your numbering plan.
5. Update the Static IP Address, if necessary.
6. Update the Static Subnet Mask, if necessary.
7. Update the Static Gateway, if necessary.
8. Click **OK**.

16.10 The new board is added to the Networking Boards screen of the wizard. If you want to use this new board, select it from the list and click **Next**.

16.11 To create and configure a new T1/E1 PRI Card:

1. Click **Create/Update New Board**. The Create PRI Board dialog box appears as shown below.



2. Select **T1 Primary Rate Interface Card** or **E1 Primary Rate Interface Card** using the drop-down list box.
3. Select the Target Slot for the card using the drop-down list box.
4. Select the number of B-Channel trunks to equip for the card.

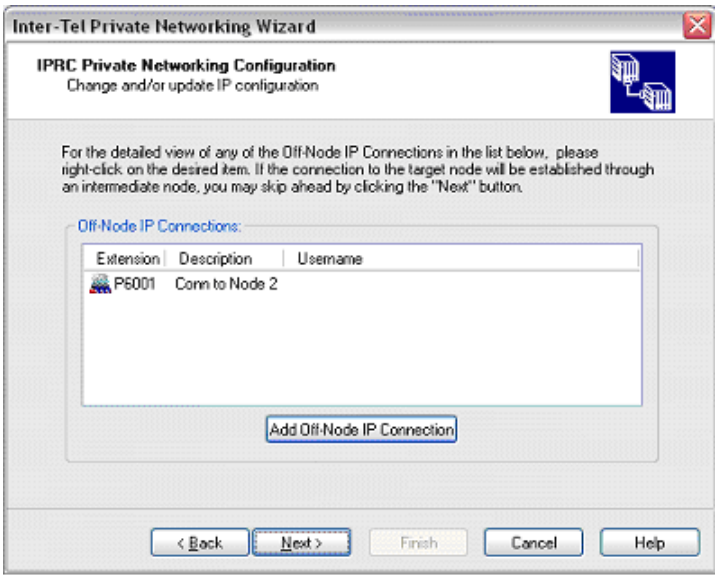
NOTE: For T1 PRI boards, the default number of channels is 23. For E1 PRI boards, the default number of channels is 30.

5. Select the Base Extension for the trunk using the drop-down list box.
6. Click **OK**.

16.12 The new board is added to the Networking Boards screen of the wizard. If you want to use this new board, select it from the list and click **Next**.

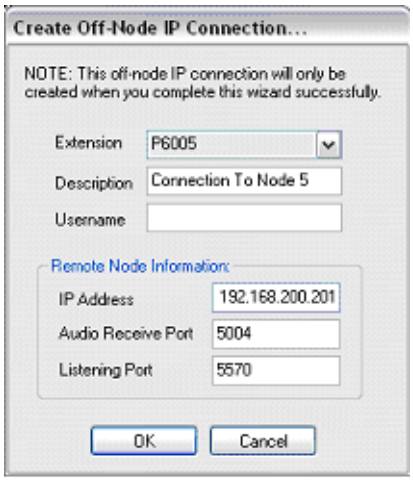
16.13 Updating the IP/PRI Configuration: The fourth screen of the wizard allows you to configure private networking for the IPRC or the T1/E1 PRI card. This screen differs according to the type of networking that you are configuring.

16.14 Configuring IPRC Private Networking: Select an existing Off-Node IP connection and then click **Next**. If there are off-node IP connections, you can create one by clicking **Add Off-Node IP Connection.**



16.15 To create an off-node IP connection:

1. Click **Add Off-Node IP Connection.** The Create Off-Node IP Connection dialog box opens as shown below.



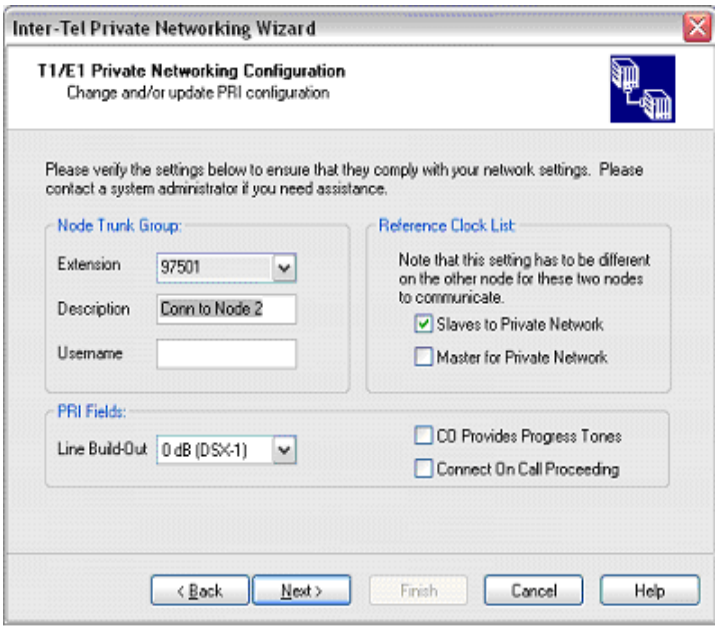
2. Select an Extension for the connection using the drop-down list box.
3. Enter a Description for the connection.
4. Enter the Username for the connection.

5. Update the IP Address for the remote node.
6. Update the Audio Receive Port for the remote node, if necessary.
7. Update the Listening Port for the remote node, if necessary.
8. Click **OK**.

16.16 The new IP connection is added to the IPRC Private Networking Configuration screen of the wizard.

16.17 Configuring T1/E1 Private Networking: If a Node Trunk Group exists for the target node, its information appears on the fourth screen of the networking wizard as shown below. You can change or update the PRI configuration by modifying the fields on this screen.

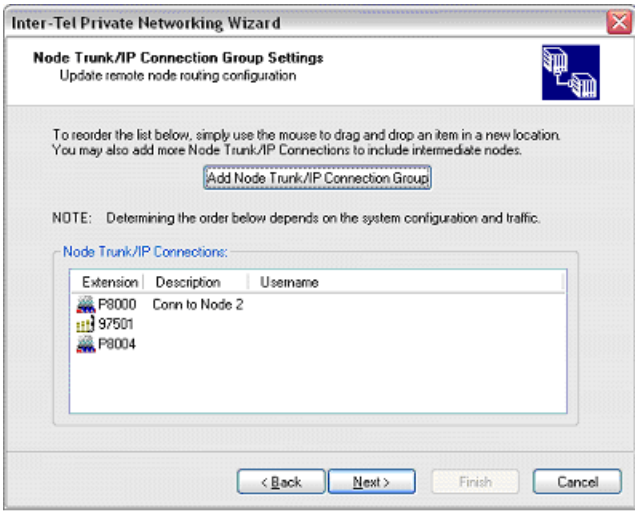
NOTE: A warning appears on this screen if the board you selected is being used in a connection to another node.



16.18 To update PRI configuration:

1. Select an Extension for the Node Trunk Group using the drop-down list box.
2. Enter a Description for the Node Trunk Group.
3. Enter a Username for the Node Trunk Group.
4. Select a Line Build-Out using the drop-down list box, if necessary.
5. Modify the Reference Clock List, if necessary.
6. Select CO Provides Progress Tones, if desired.
7. Select Connect On Call Proceeding, if desired.
8. Click **Next**.

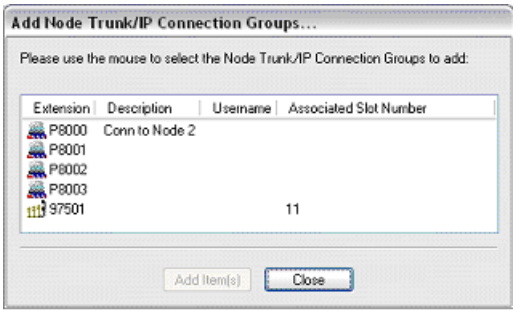
16.19 Updating Node Trunk/IP Connection Group Settings: The fifth screen of the networking wizard displays the Node Trunk/IP Connections for the target node.



16.20 The Add Node Trunk/IP Connection Group button allows you to add Node Trunk/IP Connections. After you have added Node Trunk/IP Connections, you can reorder the list by dropping and dragging with the mouse.

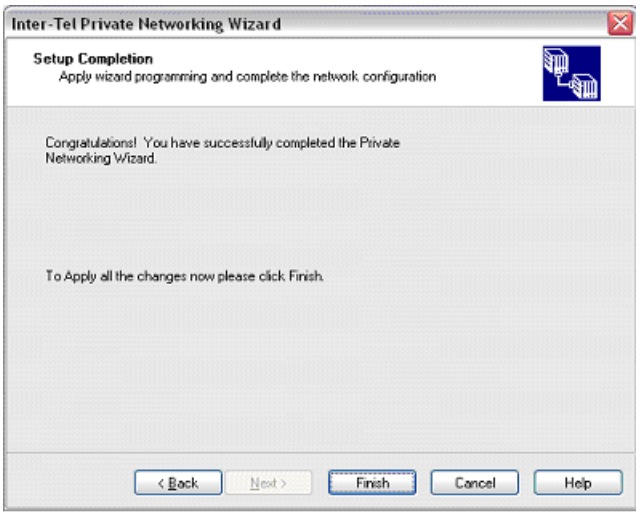
16.21 To add Node Trunk/IP Connections:

1. Click **Add Node Trunk/IP Connection Group**. The Add Node Trunk/IP connection Groups dialog box appears as shown below. The dialog box is pre-populated with the Node Trunk/IP Connections Groups in the system.



2. Select one or more Node Trunks or IP Connections and click **Add Item(s).**
3. Click **Close**. The Node Trunks/IP Connections you selected are added to the Node Trunk/IP Connection Group Settings page of the wizard.
4. Based on your system's configuration and traffic, if necessary, reorder the list of Node Trunk/IP Connections by dropping and dragging them with the mouse.
5. Click **Next**.

16.22 Completing the Private Networking Setup: The sixth and final page of the networking wizard informs you that the setup is complete. At this point all of the information has been gathered and the **Finish** button is enabled.



16.23 Click **Finish** to complete the wizard and save the private networking programming to the database.

NOTE: A warning message may appear stating that DB Studio is shutting down if you created the first remote node using the networking wizard.

17. PROGRAMMING FOLDERS

17.1 When you view the main directory tree, you will see the following folders:

- Passwords folder (see below)
- Software License folder (see page 6-97)
- System folder (see page 6-99)
- Voice Processor folder (see page 6-101)

A. PASSWORD FOLDER

Passwords

17.2 Passwords are very important to system security. Without sufficient password protection, the database is vulnerable to unauthorized access.

17.3 The system passwords are located in the Passwords folder (the shortcut is "PASS"). The Password folder contains ten password entries that allow you to set the description (up to 20 characters) and define the password. The password can be up to 8 digits in length. You can also program the access rights for passwords 2-10. The first password is special because it is the administrator's password. This selection cannot have its access rights customized and thus the administrator will always be able to use everything in DB Programming.

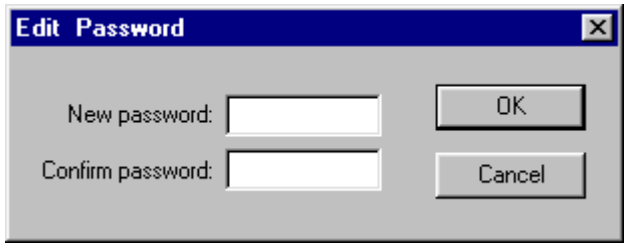
17.4 As of v8.0, administrators can use their password (password #1) as the old password when changing other passwords (e.g., network, socket, etc.). For example, if changing a socket connection password, the administrator password is accepted as a valid entry in the Old password text box. This allows administrators to change passwords that may have been forgotten.

17.5 In v9.006 or later system, there is a new Windows administrator password. Instead of a blank admin password, "itpassw" is now the default admin password. To prevent unauthorized access, field technicians should change the default system password to a higher security password as soon as possible (for example, at least eight characters, combination of upper and lower case and numbers, etc.).

17.6 When you right-click on a password, you will see an option for editing the Access Rights and for Editing the password.

17.7 To program a password:

1. Right-click on the desired password number and select **Edit Password**. The following screen displays.



2. Enter the new password (up to 8 characters) in the New Password box. The characters will not appear on the screen when typed; they will appear as asterisks (***).
3. Retype the password exactly as before in the Confirm Password box.
4. Click **OK** to exit and save the password. If the entered passwords match, you will return to the Password field. If not, you must re-enter the new password and verify it again. If you make a mistake while entering the password or want to leave it unchanged, click **Cancel**.

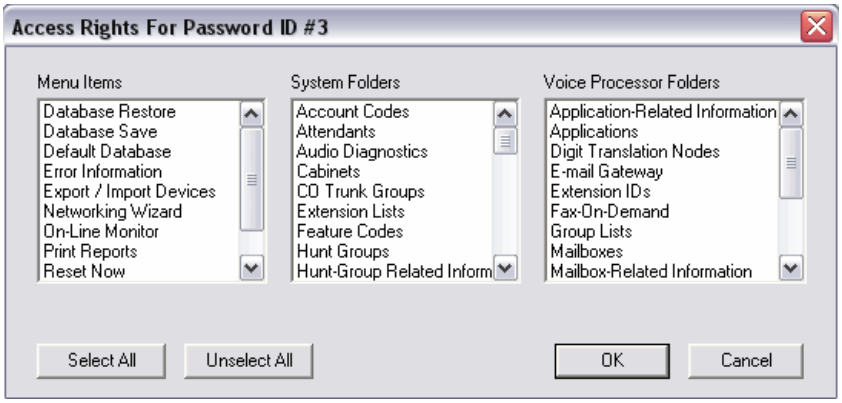
17.8 A system password will be requested at the start of each programming session, unless there are no passwords in the database.

17.9 Passwords can be programmed in any order or left blank. If one or more passwords are left blank, press **ENTER** when the Request System Password dialog appears. This allows access to the highest level containing a blank password.

Access Rights

17.10 To program access rights:

1. Right-click on the desired password number and select **Edit Access Rights**. The following screen displays.



NOTE: You cannot program access rights for the Administrator (password #1).

2. Select the menu items, system folders, and voice processor folders to which the password user will have access. You can select these options in any combination for each password by scrolling through the list and highlighting the desired options (hold down the **CTRL** key to highlight several topics). Selecting one of these folders gives you access to all subfolders and the items they contain. Or, click **Select All/Unselect All** to select/clear every item in every list. To select/deselect multiple items that are listed consecutively, press **SHIFT** while selecting the first and last item within a list. To select/deselect multiple items that are not listed consecutively, press **CTRL** while making your selection.

   There are two special cases to note. In order to gain access to the items in the System root folder, you must select the **Cabinet** folder. Similarly, to gain access to the items in the Voice Processor root folder, you must select the **Voice Processor Networking** folder.

3. Click **OK**.

NOTE: Although a user can be given access to the Password folder, the user will be allowed to edit only his own description and password. The user is not allowed to edit other passwords or descriptions.

B. SOFTWARE LICENSE FOLDER

17.11 The Software License folder displays the licenses currently uploaded to the system. The features common to all licenses are displayed in this folder along with the value for each feature according to the current software license. This folder is read-only.

17.12 The detailed information about software license features is shown in the table below. The values shown are for local mode. In remote mode, the values match the software license that is loaded. If there is no license loaded or the current license on the system is invalid, or if connected to DB Programming in remote mode, the fields will display with a red "X." The available software license features are listed in the table below.

Table 6-17: *Software Licenses*

Field	Description
ACD Hunt Group	Indicates whether the ACD Hunt Groups software license is uploaded to the system.
Advanced CO Interfaces	Indicates whether the Advanced CO Interfaces software license is uploaded to the system.
Agent Help	Indicates whether the Agent Help software license is uploaded to the system.
Analog Voice Mail Hunt Group	Indicates whether the Analog Voice Mail Hunt Group software license is uploaded to the system.
Automatic Route Selection	Indicates whether the Automatic Route Selection software license is uploaded to the system.
Category A Endpoints*	Indicates how many Category A endpoint licenses are uploaded to the system. Category A license includes the following endpoints: **Multi-protocol endpoints in ITP mode**: • Model 8600 • Model 8622 (and discontinued Model 8620) • Model 8662 • Model 8690 **Non-Inter-Tel endpoint**: OAISYS Net Phone, which emulates an Inter-Tel Model 8662 in ITP mode
Category B Endpoints*	Indicates how many Category B endpoint licenses are uploaded to the system. Category B license includes the Model 8602 IP softphone.
CPS Redundancy	Indicates whether the CPS Redundancy software license is uploaded to the system.
Desktop Interface	Indicates whether the Desktop Interface software license is uploaded to the system.
Directories	Indicates whether the Directories software license is uploaded to the system.
Extended Node Capacity	Indicated whether the Axxess License Enable 99 Nodes software license is uploaded to the system.
Inter-Tel Messaging IP Networking	Indicates whether the Inter-Tel Messaging IP Networking software license is uploaded to the system.
IP Networking	Indicates how many IP networking licenses are uploaded to the system.

Table 6-17: *Software Licenses (Continued)*

Field	Description
Multilingual Support	Indicates whether the Multilingual Support software license is uploaded to the system.
Remote Automatic Call Distribution Hunt Groups	Indicates whether the Remote Automatic Call Distribution Hunt Groups software license is uploaded to the system.
SIP Trunks	Indicates how many SIP trunk licenses are uploaded to the system.
System Forwarding	Indicates whether the System Forwarding software license is uploaded to the system.
System OAI Events	Indicates whether the System OAI Events software license is uploaded to the system.
System OAI Third Party Call Control	Indicates whether the System OAI Third Party Call Control software license is uploaded to the system.
Uniform Call Distribution Hunt Groups	Indicates whether the Uniform Call Distribution Hunt Groups software license is uploaded to the system.
Voice Processor AMIS Networking	Indicates whether the Voice Processor AMIS Networking software license is uploaded to the system.

* DB Programming supports licensing for two endpoint categories—A and B. DB Programming displays which license type each endpoint is using in the IP Device Status dialog (see page 7-78). The About license dialog displays the total number of licenses in each category currently in use.

C. SYSTEM FOLDER

17.13 If you open the System folder (the shortcut is "AXX"), you will see the following directories. Opening each of those directories reveals their programming fields. The shortcuts for each of the programming areas are shown in parentheses in the chart below. See page 6-51 for information about using the Shortcut option.

Table 6-18: *System Directory*

Folder	Ref.	Programming Field
Cabinets (CNFG, CONF, SETU)	6-110	Error Thresholds (T1ERR) Reference Clock List (T1REF) Resources Sockets Circuit Cards
Devices and Feature Codes (DEV)	6-156	Extension Lists (EXTL or LIST) Feature Codes (FEAT) Hunt Groups (HUNT or UCD) IP Ports (IPP) IP Connections (IPC) Modems (MDM) Network Groups (NETG) Node IP Connection Groups (NIPCG) Nodes (NRG or NODE) Page Ports (PPORT) Page Zones (PAGE) Stations (STN) System Serial Ports (PORT, SPA, or SPP) CO Trunk Groups (TRNK) Node Trunk Groups (NTG) Trunks (ILIN or INDT)
Flags (MISC or SYS)	6-258	System-Wide Flags
Hunt Group-Related Information (HRI)	6-263	ACD Agent IDs (AGID)
IP-Related Information	6-265	Call Configurations
Maintenance (MAIN)	6-268	Call Cost (COST) Freeze Zones (FRZ) Message Print (MPP or MPRNT) SMDR (SMDR) Major Reset Scheduling (MRS)

Table 6-18: *System Directory (Continued)*

Folder	Ref.	Programming Field
Numbering Plan (NUM)	6-280	Area Flags (AREA) Classes Of Service (CLASS) Device Baseline Extensions (DEVBE) Dial Rules (DIAL) Emergency (EMER) Facility Groups (FAC) Home Area Codes (HOME) – U.S. only Route Groups (ARS) Toll Strings (TOLL) User Groups (USER) – U.S. only
Station-Related Information (SRI)	6-301	Account Codes (ACCT) Attendants (ATT) Flags (SFEA) Key Assignments - DSS (DSS or KAD) Key Assignments - Keyset (KAS or KEY) Message Centers (CENT) DND and Reminder Messages (DND or MESG) Primary Attendants (PRI) System Forwarding Paths (SFWD) System Speed Dial (SSPD) Keyset DB Programming Password
Trunk-Related Information (TRI)	6-319	Call Routing Tables (CALL)
Timers and Limits (TIMR)	6-329	System Timers
Report Programming (CREP or REP)	6-339	System Reports 1-30
System Manager (SYSM)	6-341	IP Address Port Username Password Enable Connection
System-Wide Fields (AXX)	6-107	Periodic Backup Database Save Time Periodic Backup Database Save Retry Attempts Enable Periodic Backup Database Save Voice Processor Enable

D. VOICE PROCESSOR FOLDER

17.14 If you open the Voice Processor folder (the shortcut is "ASYS" or "TALK"), you will see the following directories. Opening each of those directories reveals their programming fields. The shortcuts for each of the programming areas are shown in parentheses in the chart below. See page 6-51 for information about using the Shortcut option.

Table 6-19: *Voice Processor Directory*

Folder	Ref.	Programming Field
Application-Related Information (AMISC or AMSC)	6-353	Audiotex Recordings (AUDIO or REC) Time Slot Groups (SLOT) Voice Mail Information (VMI) Auto Attendant Information (AAI)
Devices (ADEV)	6-356	Applications (AAPP or APPL) Digit Translation Nodes (DIGIT or TRANS) Extension IDs (AEX, EXID, or EXTID) Group Lists (AGL or GROUP) Mailboxes (AMAIL, AMB, or MAIL) Nodes (ANODE)
E-Mail Gateway (EMAIL)	6-398	E-Mail System E-Mail Address E-Mail Real Name E-Mail Username E-Mail SMTP Server E-Mail POP Server Administrator E-Mail Address Gateway Password
Fax-On-Demand (FAX or AFAX)	6-401	Timers and Limits Fax Documents Allow International Calls Outgoing Access, Prefix, and Termination Start and Stop Time Days of the Week Format: Local Fax ID, Company Name, Logo Doc.

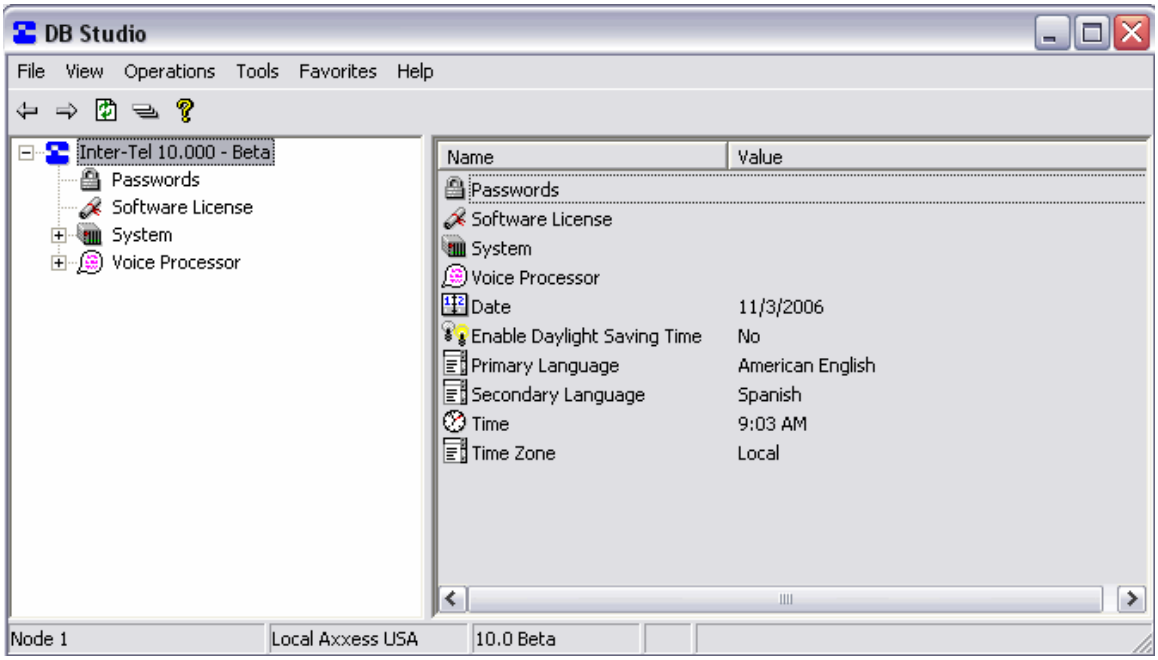
Table 6-19: *Voice Processor Directory (Continued)*

Folder	Ref.	Programming Field
Mailbox-Related Information (MRI)	6-406	Number of Mailboxes Number of Messages Sent Number of Messages Received Number of Messages Received From Remote Nodes Number of New Messages Number of Saved Messages Total Length of New and Saved Messages Average Mailbox Percent Full Number of Times Mailbox was Full Number of Times Mailbox was 80% Full Number of Time 3 Bad Passwords Were Entered Number of Mailboxes Currently Full Number of Mailboxes Currently More Than 80% Full
Maintenance (AMAIN)	6-407	Auto Reboot Auto Maintenance Day Enable Diagnostics Enable Auto Schedule Auto Schedule Start Time
Networking (ANET)	6-411	AMIS Timers and Limits Validate Off-Node Mailboxes Undeliverable Messages Destination Type
Report Parameters (AREP)	6-418	Automatic Report Generation Manual Report Generation Output Device Output Destination
SMDR Buffer (ASMDR)	6-422	SMDR Buffer Size SMDR Buffer-Box Bit Rate Enable SMDR Buffering
Timers and Limits (TIMR)	6-423	Timers and Limits DTMF Detection Information Number of Voice Channels

Table 6-19: *Voice Processor Directory (Continued)*

Folder	Ref.	Programming Field
System-Wide Parameters (ASYS or TALK)	6-344	Dial-0 Destination Type
		Total Storage on the System
		Total Storage Used
		Total Storage Remaining
		System Administrator Mailbox
		Alternate Tone Detection
		Volume
		Save Message On Return Call
		Identification Prompt
		Management Command Port
		Management Event Port
		Speech Recognition (ASR) Setting
		Speech Recognition (ASR) Enabled
		Monitor Password
		Windows Networking Username
		Windows Networking Domain
		Windows Networking Password

18. SYSTEM-WIDE PROGRAMMING FIELD



18.1 In the screen shown above, there are system-wide fields for setting the system date, time, time zone, and primary and secondary languages. They are programmed as follows.

A. DATE

18.2 When you select **Date**, you can edit the month, day, and/or year. If you select the scroll arrow, a small calendar appears. Scroll to the correct month and then select the correct date. Press **ENTER** or select another field to save your change.

B. TIME

18.3 To change the time, select the current Value. Then highlight the hour or minute digits and scroll to the desired setting. When finished, press **ENTER** or select another field to save your change.

C. TIME ZONE

18.4 To change the time zone, select the current Value. Then scroll to the desired setting. When finished, press **ENTER** or select another field to save your change.

NOTE: Chassis within an ATM-based node cannot span time zones.

D. PRIMARY AND SECONDARY LANGUAGES

18.5 The System provides a choice between American English, British English, Spanish, and Japanese prompts and displays. Note that Spanish voice prompts use Mexico City Spanish Dialect for VPU and EVMC, and American Spanish Dialect for EM. The system selects the language to use for each call, as determined by the trunk, station, and Voice Processor programming. Japanese prompts can be viewed only on the digital display phones; Analog phones do not support Japanese displays. To select the primary and secondary languages, select the drop-down list boxes and scroll to the desired settings. Make sure that the appropriate prompts have been loaded into the Voice Processor as described on page 3-226. Otherwise, if a user tries to access those prompts, the user hears the default American or British English instead.

E. DAYLIGHT SAVING TIME

NOTE: Inter-Tel recommends that you do **NOT** schedule automatic functions, such as backups and resets, to occur at 2:00 AM. If you do, the system may not perform the function when the time changes.

18.6 **Enable Daylight Saving Time**: This flag determines whether or not the phone system automatically adjusts the time whenever Daylight Saving Time changes. If enabled, the time changes on the days specified on page 5-164. If disabled, the system does not recognize Daylight Saving Time. By default, this flag is *disabled*. To enable this flag, select the check box to place a check mark in it. Then press **ENTER** or click on another field to save the change. To disable the option, select the check box again to remove the check mark.

NOTICE

Station Message Detail Recording (SMDR) generates call costs based on the difference between the start and stop times of a call. System time changes **WILL** affect this calculation. If Daylight Saving Time is enabled for the system, and the time changes while SMDR is tracking a call, the call cost will be inaccurate. For example, if a call starts at 1:30 AM and ends at 2:30 AM on the night that Daylight Saving Time goes into effect, the call cost will be for 2 hours (1:30 to 3:30) instead of 1 hour (1:30 to 2:30).

SYSTEM PROGRAMMING

1. INTRODUCTION

1.1 When you click on **System** in the left side of the programming window, the following programming areas are displayed on the window on the right:

- Cabinets (see page 6-110)
- Devices and Feature Codes (see page 6-156)
- Flags (see page 6-258)
- Hunt Group-Related Information (see page 6-263)
- IP-Related Information (see page 6-265)
- Maintenance (see page 6-268)
- Numbering Plan (see page 6-280)
- Station-Related Information (see page 6-301)
- Trunk-Related Information (see page 6-319)
- Timers and Limits (see page 6-329)
- Report Programming (see page 6-339)
- System Manager (see page 6-341)

1.2 Also shown are the following system-wide parameters (see page 6-107):

- Periodic Backup Database Save Time
- Periodic Backup Database Save Retry Attempts
- Enable Periodic Backup Database Save
- Voice Processor Enable

NOTE: As of version 8.2, the major reset parameters are located under System\Maintenance\Major Reset Scheduling.

1.3 Each of these programming areas are described individually in the following pages.

2. SYSTEM-WIDE PARAMETERS

2.1 This section describes system-wide parameters.

A. BACKUP DATABASE PARAMETERS

2.2 As of version 8.0, a backup copy of the database is automatically stored on the CP Server's hard drive or in flash memory in the CPC. This backup remains available, even if the system experiences a power failure or if the battery backup loses power. If the system is powered up and a corrupt database file is detected, the backup file in flash memory or on the hard drive is restored. You can program an automatic backup as described below.

2.3 **Periodic Backup Database Save Time**: Select the time of day when a scheduled database backup will occur. This must be at least 15 minutes after the Scheduled Reset Time; otherwise, the backup may not occur. If you enter a time that is less than 15 minutes after the Scheduled Reset Time, a message appears, requesting a new value. By default, this is *2*:*00AM*. To program the time, select the current hours, minutes, or AM/PM value and then scroll to or enter new setting in the text box. Press **ENTER** or select another field to save the change.

2.4 **Periodic Backup Database Save Retry Attempts**: Select the number of times that DB Programming will attempt to save the database. If the database is not saved before the system retries the number of times specified in this field, the database is not saved until the next day. The range is *0-10*, and the default is *3*.

2.5 **Enable Periodic Backup Database Save**: Enable this flag to save the database at the time indicated in Periodic Backup Database Save Time. This flag must be enabled to backup the database at the scheduled time. By default, this is *enabled*. To enable this flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

NOTE: If the system detects that the database has not changed since the last backup, the save is not performed.

B. VOICE PROCESSOR ENABLE

2.6 The Voice Processor Enable database field is auto-enabled by Call Processing. For example, if the Voice Processor Enable flag is set to "No" (disabled) and the user successfully connects up a Voice Processor unit, Call Processing will auto-enable the flag. However, if DB Programming is currently programming the switch, the auto-enable flag *will not* get auto-enabled.

2.7 **To disable this flag,** select **No**. Then press **ENTER** or select another field to save the change.

NOTE: This flag is enabled by default. If you do not have a Voice Processing Unit connected to the System, make sure you disable this flag before you attempt to import or export information over the network. Also if DB Programming is currently programming the switch, the Voice Processor Enable flag will not be automatically enabled.

Selecting a Voice Processor Type in Local Mode

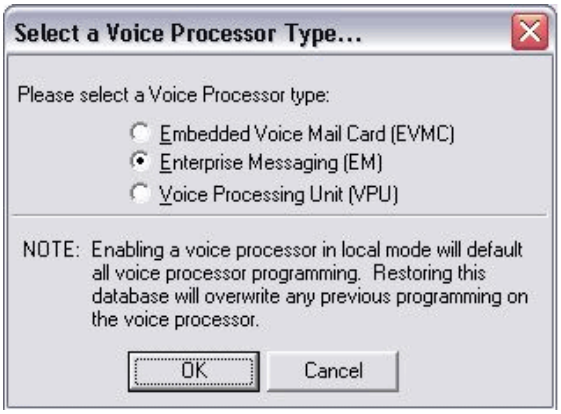
2.8 With the maintenance release of v9.1, when you enable a voice processor in local mode, you may be prompted to choose a voice processor type. The Select a Voice Processor Type dialog appears *only* if you have saved the database from a remote session that did *not* have the voice processor physically connected. For the Inter-Tel-recommended process, see page 6-109.

2.9 The Select a Voice Processor Type dialog allows you to identify a known type of voice processor that has not been enabled in the system configuration. It does *not*, however, allow you to change a voice processor type after the database has been created.

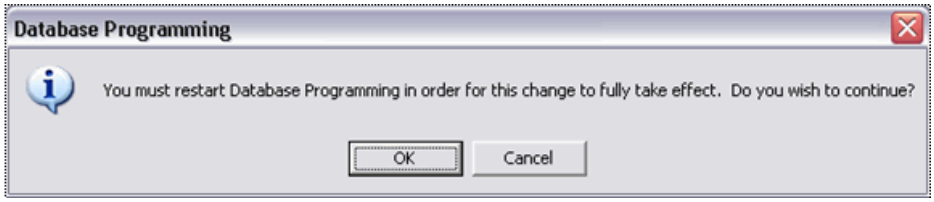
2.10 To enable a voice processor in local mode:

1. Launch Session Manager and start a session of DB Programming.
2. From the left pane of the DB Studio main screen, select the System folder. Contents of the System folder appear in the right pane.
3. Toggle the value of the "Voice Processor Enable" feature to **Yes**.

> **NOTE:** The Select a Voice Processor Type dialog appears only if you have saved the database from a remote session that does *not* have a voice processor physically connected. After the Voice Processor function is enabled and a database is saved in a remote session, the Select a Voice Processor Type dialog does *not* display again.



4. Select the option button for the voice processor type you know is connected to the system.
5. Click **OK** to continue programming the selected voice processor. If you select **Cancel**, the Voice Processor Enable field in the System folder toggles to **No.** If you select a voice processor type, or if one already exists in the database, you must restart the system in order for the changes to take effect. The following message window appears.



6. Click **OK.** The voice processor type is updated in DB Programming and the session closes. Clicking **Cancel** toggles the Voice Processor Enable value in the System folder to "No" cancels any Select a Voice Processor Type selection made, and displays the DB Studio main window.

7. Restart a local DB Programming session.
8. After the voice processor is enabled, it can be configured and programmed in accordance with customer requirements. See the following graphic showing a sample of the Voice Processor folder contents.
9. From the DB Studio menu bar, select Operations – Database Operations – **Database Save**. Identify the location where you want to store the system database, then check the **Save Voice Data to:** box and identify the location where you want to save the voice processor database.
10. Click **Start**. System and voice processor databases are saved to the locations indicated.

Review and Update for Enabling a Voice Processor

2.11 Inter-Tel Axxess platforms support any of the following voice processors: Enterprise Messaging (EM), an Embedded Voice Mail Card (EVMC), or a Voice Processing Unit (VPU). And, for proper configuration, DB Programming requires that the voice processor type is identified. After the VP is physically connected, the system senses which type it is.

2.12 As with previous versions, a voice processor connected to an Inter-Tel Axxess platform auto-enables when the system boots up. To avoid the risk of overwriting an established database with default values, Inter-Tel recommends connecting the voice processor first, then connecting with a remote mode session of DB Programming, and then saving.

2.13 To save a voice processor database in remote mode:

1. Connect the voice processing unit equipment to the system in accordance with Inter-Tel installation instructions. See the applicable installation manual.
2. In remote mode, launch Session Manager and start a DB Programming session.

3. CABINETS

3.1 The Cabinets programming area allows you to designate the type of interface card that is installed in each of the card slots, and designate the circuits making up the card. In addition, there is information which tells you the number of timeslots required by the database configuration. This is done so that you are aware of the number of timeslots in use by the system. Although the basic system configuration is non-blocking, certain configurations of Options card ports can lead to circumstances wherein more timeslots are assigned than are available, which may result in blocking if an attempt is made to use all resources at once. (See page 2-17 in SPECIFICATIONS for further discussion of time slot allotment.) Because the cardfile has universal slots, any type of card (except the CPC or CPU) can be installed in any slot. You program these fields to "tell" the database which type of card is in each slot.

3.2 The number of available slots depends on the number of cabinets (chassis) installed:

- In a single-chassis system there are 8 slots. The only type of card that can be programmed into card slot 8 is a CPC (or CPU).
- In a dual-chassis system, there are 16 slots. The only type of card that can be programmed into card slot 8 is a CPC (or CPU). Slot 16 is not used.
- In a triple-chassis system, there are 24 slots. The only type of card that can be programmed into card slots 8 and 24 is a CPU. Slot 16 is not used.
- In a quad-chassis system, there are 32 slots. The only type of card that can be programmed into card slots 8 and 24 is a CPU. Slots 16 and 32 are not used.
- In an eight-chassis ATM system, there are 64 slots. The only type of card that can be programmed into card slots 8, 24, 40, and 56 is a CPU. Slots 16, 32, 48, and 64 are not used.

3.3 The system will not allow an empty slot (or any card type other than a CPC or CPU) in slots 8, 24, 40, and 56. For an ATM-based system, these slots default to a PCMA Card and cannot be changed. Nothing can be programmed in slots 16, 32, 48, and 64.

3.4 The CPC or CPU card used in each slot also depends on the number of chassis. The available CPU card types are:

- **CPC64 or CPU64**: This is used in single-chassis systems with 64 or fewer devices. It is ideal for the CPU64 configuration (see Appendix D — CPC64/CPU64 Configuration).
- **CPC-Dual or CPU128**: This is used in a single- or dual-chassis system with 128 or fewer devices.
- **CPC-Dual or CPU020-D/PCM-D (CPU 256)**: This is used in a single- or dual-chassis system. It provides support for up to 256 devices.

NOTE: A CPC is required to run version 8.0 or later on a 64, 128, or 256 system.

- **CPU020-Q/PCM-D**: This CPU/PCM combination is not recommended, but can be used temporarily in a single- or -dual chassis system if a CPU020-D is not available. It must be programmed as a CPU020-D/PCM-D combination.
- **CPU020-D/PCM-F**: This CPU/PCM combination is not recommended, but can be used temporarily in a single- or -dual chassis system if a PCM-D is not available. It must be programmed as a CPU020-D/PCM-D combination. Note that the PCM card will remain in a constant state of alarm, indicating that it cannot achieve local or remote sync; this can be ignored.

- **CPU020-Q/PCM-F (CPU 512 Master)**: This combination can be used as the "master" CPU in a triple- or quad-chassis system. The PCM-F provides the connectors for the fiber-optic cable that connects the "master" CPU to the "slave" CPU. It can also be used in place of any of the combinations above.
- **CPU020-X/PCM-F (CPU 512 Slave)**: This combination is used only in the "slave" chassis in a triple- or quad-chassis system. The PCM-F provides the connectors for the fiber-optic cable that connects the "master" CPU to the "slave" CPU.
- **CPU020/PCM/EAC and NT- or 2000-based CPU**: This CPU is designed to support heavily loaded systems with high call traffic. The call processing software has been moved from the CPU to a Windows NT or 2000 Call Processing (CP) Server that communicates with the System via a 10Base-T Ethernet connection. For more information about this special CPU, see Appendix C — CP Server.

  NOTE: A 512 or higher system requires a CP Server to run version 8.0 or later.

- **CPU020/PCMA and CP Server**: This CPU is designed to work with an ATM Switch which supports large capacity systems consisting of up to eight chassis and 1792 devices. The call processing software has been moved to Windows 2000 CP Server that communicates with the System via the ATM Switch. For more information about this special CPU, see Appendix C — CP Server.

NOTE: When choosing the CPU types for use in a network, consider the number of devices the CPU type will need to communicate with, both on the local node as well as on remote nodes. The more devices it must handle, the more powerful the CPU type must be. For example, although the CPU020-Q/PCM-F can support a fully loaded quad-chassis system, it may slow down if it has to support four chassis ***and*** off-node devices. For optimum performance in a network, the nodes should not be filled to maximum capacity and they should use the most powerful CPU types.

3.5 See SPECIFICATIONS and INSTALLATION for further information regarding the CPU type selection.

3.6 When changing the card type for equipped slots, existing circuits must first be reconfigured. Failure to redefine the associated circuits before inserting another type of interface card of a different device type will produce an error message.

3.7 If multiple OPCs, T1Cs, T1/PRI, E1/PRI, T1/E1/PRIs, and/or DKSC-16s are installed in expanded chassis, resource blocking can be prevented by splitting the cards between the cardfiles and/or installing them in the same cardfile as the LGCs or LSCs. In tri-/quad-chassis installations, the master T1C or E1C (if one exists) should be located in the master dual-chassis.

3.8 The fields used in Cabinet programming include the following. They are described in detail in the following pages.

- Error Thresholds (see page 6-112)
- Reference Clock List (see page 6-112)
- Resources (see page 6-113)
- Sockets (see page 6-114)
- Circuit Cards (see page 6-114)

A. ERROR THRESHOLDS

3.9 Specifications include several error thresholds. When an error exceeds one of these thresholds, an error message is sent to the error message output port (see page 6-126).

3.10 You can set the thresholds (number per hour and number per day) for T1/E1/PRI or BRI errors, if necessary. (These are set to default values that, in most cases, will not need to be changed.) Possible errors include the following:

- **BPV (Bipolar Violations)**: This is a non-zero signal element that has the same polarity as the previously received non-zero element.
- **CRC-6 (Cyclic Redundancy Check)**: *(T1/E1/PRI Only)* The calculation carried out on a set of transmitted bits by the transmitter does not match the calculation performed by the System.
- **OOF/COFA (Out of Frame/Change of Frame Adjustment)**: *(T1/E1/PRI Only)* OOF is a state in which the frame alignment that is received is not consistent with that which is transmitted. COFA occurs when the System realigns its receiver to the proper frame alignment signal. This could be caused by an incorrect line build-out (LBO) value.
- **CS (Controlled Slips)**: *(T1/E1/PRI Only)* The System replicated or deleted one 192-bit digital signal (DS1) frame due to a lack of frequency synchronization. This could be caused by an incorrect line build-out (LBO) value.
- **ES (Errored Seconds)**: These are seconds in which at least one error occurred.
- **SES (Severely Errored Seconds)**: This is a second during which transmission performance is degraded below an acceptable level.
- **UAS (Unavailable Seconds)**: *(T1/E1/PRI Only)* This is the time interval during which the T1 or E1/PRI span is unavailable for service. This time begins with 10 or more consecutive Severely Errored Seconds and ends with 10 or more Non-Severely Errored Seconds.

3.11 To program a threshold, double-click on **Error Thresholds** to view the list. Then, to change a specific threshold, click on its Value or Daily field and enter the new number. Then press **ENTER** or click on another field to save the change.

B. REFERENCE CLOCK LIST

3.12 One T1 or E1 card is designated as the "Slave from Public Network" and is the system reference card. All other cards are the backup cards.

3.13 The reference clock list has three categories that determine whether the card is a master or a slave and if it is in a public network or a private network. They are:

- Slaves to Public Network
- Slaves to Private Network
- Masters for Private Network

3.14 All cards configured in the system appear in this window. Place the cards in the proper categories by clicking on their Value and scrolling the list box to the proper setting.

NOTE: In tri-/quad-chassis installations, the master card (if one exists) should be located in the master dual-chassis.

3.15 If there is more than one "Slaves from Public Network," the first card is the system reference card. The other cards that appear below it in the list are the backup cards (in the order they appear in the window).

C. RESOURCES

3.16 The Resources folder contains amperage and time slot information, as described in the following sections.

Amps Required

3.17 This information includes the amperage for 5 and 24 volts and is provided as a reference to aid in selection of a power supply. Choose a power supply that is rated to provide ***more than*** the indicated requirement. The amperage requirement is based on the programmed system configuration. However, in a quad-chassis configuration, the amperage requirement for chassis 1 and 3 will always be "9.00."

Available Time Slots

3.18 This line is a non-programmable reference that tells you how many timeslots are available for use by the system resources. The available number decreases as the following resources are programmed:

- OPC DTMF receivers
- OPC Caller ID [CLIP] receivers
- OPC Caller ID [CLIP] transmitters
- OPC conference circuits
- OPC standard speakerphone resources (Phones on DKSC-16 cards will not use OPC speakerphone timeslots unless all resources on their cards are unavailable at the time of the speakerphone request.)
- OPC enhanced speakerphone resources for phones on DKSC and DKSC-16 circuits
- DKSC, AKSC, EKSC, LSC, and SLC circuits
- Single-Line Adapters (SLAs)
- Digital Display Phone Modem Data Port Modules (MDPMs) and Analog and Executone® Phone Data Port Modules (DPMs)

NOTE: Programing T1 or E1 and DKSC-16 resources does not reduce the number of available timeslots. These cards allocate their voice channel timeslots dynamically, on an as-needed basis, from the pool of available timeslots rather than requiring dedicated voice channels for each card.

Time Slots Required for Non-Blocking

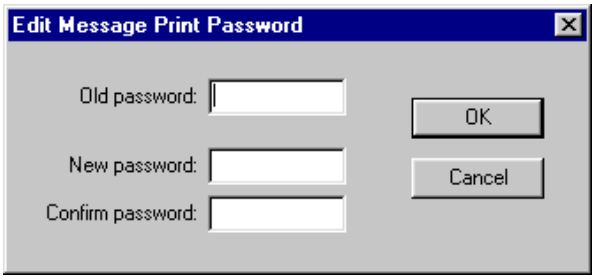
3.19 This line shows how many of the available timeslots are required to support the DSP functions as programmed on all of the OPC cards combined. If this number is greater than the "Available Time Slots" number, voice channel blocking may occur. This information is provided for reference and is not programmable.

D. SOCKETS

3.20 If using a CPC or a CP Server (CPU/EAC and Windows NT- or 2000-based CPU or ATM and Windows 2000-based CPU), you can use the Sockets folder to set the sockets for TCP/IP connections to Message Print, SMDR, System OAI, or the User System Monitor. If running system version 7.0 or earlier without a CP Server, the Sockets folder is still visible, but a red "X" appears to indicate the option is not available. (For more information, see Appendix C — CP Server.)

3.21 To program the sockets, open the Cabinets folder and double-click on **Sockets**. You will see the list of available sockets. To program an individual socket, do the following:

- **Password**: Right-click on the current Value for the network password and select Edit Password to change the password for the socket. The following window appears. Enter the old password, if one exists. Then enter the new password in the New Password and Confirm Password text boxes and select **OK**.



- **Enabled**: To enable the socket, select **No** and then place a check mark in the check box. To disable the socket, select **Yes** and remove the check from the check box. Then press **ENTER** or select another field to save your change.

E. CIRCUIT CARDS

3.22 To select the type of card that will be assigned to a card slot, click on the desired card slot and then scroll to the desired card type in the list box. The following cards are configurable:

- Call Processing Card or Central Processing Unit Card (see page 6-116)
- Options Card (see page 6-117)
- Digital Keyset Card (see page 6-118)
- Analog Keyset Card (see page 6-122)
- Executone Keyset Card (see page 6-123)
- Single-Line Card (see page 6-124)
- Loop Start Card, LSC Daughter card, Loops/Ground Start Card, or LGC-D (see page 6-125)
- T1 Card, T1C with optional Primary Rate Interface daughter card, or E1/PRI card (see page 6-126)
- Basic Rate Interface card (see page 6-136)
- Internet Protocol Resource Card (see page 6-145)

3.23 When a card is changed, one of the following may occur:

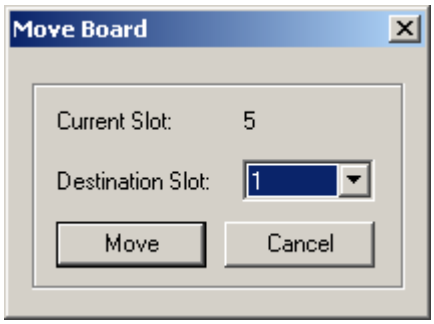
- When a card slot is modified to contain a DKSC, DKSC-16, SLC or SLC-16 the new stations will be included in all station lists in the database, except Keyset Map assignments. Phones will be included in Keyset Map number 1.
- If a DKSC-16 is changed to a DKSC, the device programming for circuits 1-8 will be preserved, but circuits 9-16 will be unequipped. If a DKSC is changed to a DKSC-16, circuits 1-8 will remain unchanged, and any previous programming for circuits 9-16 will be restored (they will be at default if they were never programmed).
- An SLC-16 cannot be installed in a card slot that is programmed for an SLC. The ports must be specifically programmed for an SLC-16. However, if port programming is changed from SLC to SLC-16, the existing eight circuits will retain their individual programming and the new circuits will be in the default state. If the port is changed from an SLC-16 to an SLC, all circuits will retain their programming, but only the first eight will be available.
- When a card slot is modified to contain a Loop Start Card (with or without a Daughter Card) or T1, E1/PRI, or T1/E1/PRI Card the new ports will be included in the "Unused CO" trunk group.
- LSC and LSC/D loop start devices are compatible. If the card type is changed to a compatible type, the loop start trunk programming will be preserved.
- When a card slot is modified to Nothing Installed, or where the existing card type is changed to another card type, ***all*** reference to the stations or trunks that were on the card in that slot will be removed from ***all*** areas of the Database Manager (unless the card was first moved to another slot).
- The card slot that contains the keyset card with the primary attendant's station cannot be changed to another type of card or deleted until the attendant's station is changed to another location on another keyset card. If you attempt to do so, you will receive a warning message.
- If you change a CPU128 slot to a CPU020/PCM, you cannot change back.
- The Embedded Voice Mail Card (EVMC) should be placed in slot 7 of the equipment chassis. No further programming is required for the EVMC ports.
- The PCMA Card contains a MAC Address field. This information is not programmable. For details about the MAC Address, see the Call Processing Configuration Utility information on page C-20 in Appendix C — CP Server.

3.24 See page 1-5 for equipment chassis board capacities. To program an individual card, double-click on it (or right-click and select **Explore Card**). Then follow the programming instructions for that type of card, located in the following pages.

3.25 If you change the card type, you will see a window that asks you to confirm the change. Select **Yes** to continue or **No** to cancel the change.

Moving Cards

3.26 A card can be moved to any unoccupied slot (except slots 8 and 16, which are reserved for CPCs or CPU Cards). To do this, highlight the existing card and drag and drop it to the new location. Or, highlight the card to move and press **CTRL** + the up/down arrow to move the card up or down in the list. You can also move a card using the Move Board shortcut menu option. To do this, right-click on the desired card or press **CTRL + M**. Then select **Move Board**. The Move Board dialog appears, as shown below. Select the destination slot and click **Move** or click **Cancel** to terminate the move process. (This can only be done while in local programming mode.)

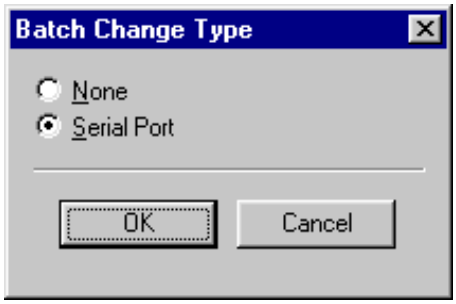


CPC or CPU Card Configuration

3.27 You can configure the serial ports and page port on the CPC and CPU. If a CPC is installed, you can also configure the modem and IP port.

3.28 **Ports**: *(You cannot change the first port.)* If you equip a Serial Port, its number is assigned automatically. If you want to see the programming for this port (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the port number.

- **To equip a single port:** To select the port type, select the current Port description and use the drop-down list box to scroll to the new port type. Then press **ENTER** or click on another field to save the change.
- **To equip several circuits at once:** Highlight the ports you want to equip. (To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items.) Then right-click and select **Batch Change Type**. The following window appears. Place a dot in the port type you want to assign and select **OK**.



3.29 **Page Port**: You cannot disable the page port. Its number is shown for your reference. If you want to see the programming for the page port (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the page port number.

3.30 **Modem** *(CPC only)*: Each CPC has a built-in modem. To access programming options for the modem, either double-click the modem extension or open System\Devices and Feature Codes\Modems. See page 6-180 for details.

3.31 **IP Port** *(CPC only)*: The CPC also has a 10/100-Base T Ethernet port that allows you to upload Call Processing software, upload software licenses, and access DB Programming. In addition, the port can be used to establish the following connections:

- Voice mail
- System OAI (Level 2)
- Desktop OAI
- Diagnostics Monitor (for Message Print, SMDR, or Monitor)

NOTE: The IP Port does not support Simple Network Management Protocol (SNMP).

3.32 Before you program the IP port, Inter-Tel recommends that you consult the on-site network administrator for information on any standard practices that are followed, including the use of special characters in hostnames. Providing a static IP address and a hostname is generally considered the "best" practice. To program the port, either double-click the port extension or open System\Devices and Feature Codes\IP Ports. See page 6-178 for details.

Options Card Configuration

NOTE: DB Programming restricts the programming of DSP chips that are not physically present on the system during a remote programming session. There are no restrictions to DSP programming in local mode. After equipping a new Options card during a remote session, you must exit from DB Programming to apply your changes and allow the system to check for the presence of DSPs. Then, when you re-enter DB Programming, you will be allowed to program the DSPs for the new Options card.

3.33 An Options Card provides onboard Digital Signal Processors (DSPs). Each of the DSPs installed on the OPC (up to four) may be configured as one of several available resources. (Caller information features require OPC programming to ensure that resources are available to support the features. Caller ID [CLIP] uses Caller ID [CLIP] receivers and DID/ANI uses DTMF receivers.) The Options Card can also be specified as having a Voice Processor (OPC to AIC) connection to support additional ports.

3.34 **Serial Port**: To change the serial port, select its description. Then use the drop-down list box to scroll to the desired setting (Serial Port or None). Press **ENTER** or click on another field to save your change. If you create a Serial Port, its number is assigned automatically. If you want to see the programming for this port (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the port number.

3.35 **DSPs**: The DSPs on the card and their current configurations are shown in the list. To program a DSP, click on its current Value and then scroll to the desired option. (See SPECIFICATIONS and INSTALLATION for information regarding the equipment supported by these options.) The options are as follows:

- **None**: If the DSP will not be used or is not equipped, select this option.
- **8 Caller ID [CLIP] Receivers**: To enable eight receivers for Caller ID [CLIP] information, select this option.
- **6 Caller ID [CLIP] Transmitters**: *(U.S. only)* To enable six transmitters for Caller ID [CLIP] to single-line phones, select this option.

- **6 DTMF Receivers**: To enable six additional DTMF receivers select this option.
- **16 Conference Circuits**: To enable 16 conference circuits, select this option.
- **12 Conference Circuits and 4 DTMF**: This enables 12 conference circuits and 4 DTMF receivers.
- **4 Enhanced Speakerphones**: Select this option to support four shared enhanced speakerphone resource.
- **8 Standard Speakerphones**: Select this option to support eight standard shared speakerphone resources.

NOTE: Each DKSC-16 and DKSC-16+ has DSP resources to support speakerphones. Their phones only use OPC speakerphone resources when the DKSC-16/DKSC-16+ resources are unavailable. DKSCs do not have on-board DSPs; their phones require the OPC resources for speakerphones.

3.36 **Voice Processor Connection**: If more than eight Voice Processor ports are equipped, enable this option by selecting the current Value and then selecting the check box to place a check in it. (Or, remove the check mark if the Voice Processor has eight or fewer ports.) The Voice Processor connection should be enabled on only the first OPC installed when there are multiple OPCs. When this option is enabled, you must connect the cable from the OPC to the Voice Processor's AIC. See page 3-213 for details.

3.37 **Time Slots Required**: *This is used for reference only.* This tells you how many time slots are needed to support the resources, as configured, to prevent voice channel blocking (including 24 timeslots for a Voice Processor connection, if enabled). For further discussion of time slots, see page 2-17 in SPECIFICATIONS.

Digital Keyset Card Configuration

3.38 When you double-click on a Digital Keyset Card (DKSC or DKSC-16), a list of the circuits appears on the right of the screen. This allows you to configure the DKSC, DKSC-16, or DKSC-16+ card ports to support the various devices. DKSC and DKSC-16/DKSC-16+ capacities include the following:

- Phone ports can be assigned for Single-Line Adapter (SLA) use. (Two single-line sets, two DID trunks, or one of each can be connected to an SLA on a DKSC or DKSC-16 port.) See INSTALLATION, page 3-211, for special installation instructions.
- To support the Off-Hook Voice Announce feature, a digital display phone must be equipped with a PCDPM. (See page 3-191 in INSTALLATION for more information.) These phone ports do not require special configuration programming.
- Up to 20 DSS/BLF units can be connected to digital display phones with PCDPMs. Each of those phones can have up to four DSS/BLF Units in addition to using the Off-Hook Voice Announce feature that is available to all phones with PCDPMs.
- A digital display phone with a PCDPM (that does not have a DSS/BLF unit attached) can use a Desktop Interface device.
- Digital display phone ports with PCDPMs can be designated as system serial ports. (There can be up to 32 system serial ports per system, including phones, OPC, CPC, and CPU ports.) Phone system serial ports can be used for system programming, SMDR report generation, and programming report generation. For proper serial port operation, the phone circuit should also be programmed with camp-on tones disabled, extended ring cadence enabled, different alerting cadence disabled, and different ringback cadence disabled (see page 6-210).

- Modem Data Port Modules (MDPMs) can be programmed for use on digital display phones. A digital display phone with an MDPM and a PCDPM can use the off-hook voice announce and have a single-line set or data device connected.

3.39 When multiple purposes are assigned to one DKSC port, the circuit numbers are assigned as follows:

- **Keyset**: XX.XX.01
- **Keyset with MDPM**: The phone is XX.XX.01 and the MDPM is XX.XX.02
- **Dual Single-Line Sets on an SLA**: The single-line set circuit numbers are XX.XX.01 and XX.XX.02
- **Single-Line Set and DID trunk on an SLA**: The single-line set circuit number is XX.XX.01 and the DID trunk is XX.XX.02
- **Dual DID Trunks on an SLA**: The DID trunk circuit numbers are XX.XX.01 and XX.XX.02
- **Keyset with System Serial Port**: The phone is XX.XX.01 and the serial port is XX.XX.02.
- **Keyset with MDPM and System Serial Port**: The phone is XX.XX.01, the MDPM is XX.XX.02, and the serial port is XX.XX.03.
- **Single-Port LSA**: XX.XX.01

3.40 When a DKSC or DKSC-16 port is changed from single purpose (e.g., Keyset) to multiple purpose (e.g., Keyset with RS232 port), the database will retain all programming for the original single port and it will create new additional station(s) in all other applicable areas of programming. The original single port will retain its extension number; the new ports will have extension numbers assigned as you exit from this programming area.

3.41 If a DKSC or DKSC-16 port is converted from Keyset to an SLA function, the keyset programming is erased and the two SLA circuits are at default values.

3.42 If a DKSC or DKSC-16 port is changed from an SLA function to Keyset with MDPM, the port programming for the SLA circuits is erased and the MDPM is set at default values.

3.43 If a DKSC-16 is changed to a DKSC, the device programming for circuits 1-8 will be preserved, but circuits 9-16 will be unequipped. If a DKSC is changed to a DKSC-16, circuits 1-8 will remain unchanged, and any previous programming for circuits 9-16 will be restored (they will be at default if they were never programmed).

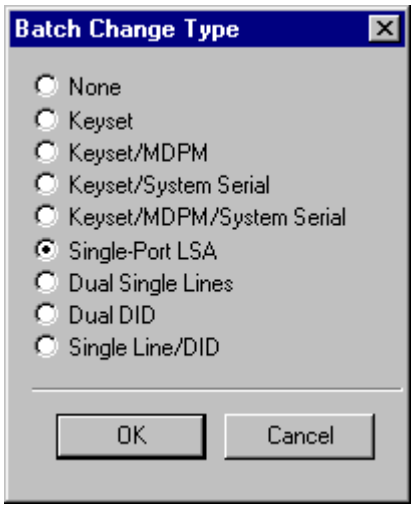
3.44 The circuit type options are as follows:

- **None**: To disable the port, select **None**.
- **Keyset**: This option is selected if the port will be used for installing a phone. The phone can have an attached DSS/BLF Unit, but cannot be used as a serial port or have a data device connection. (The phone may have a PCDPM that is used only for off-hook voice announce.) If you want to see the programming for this phone (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the port number.
- **Keyset/MDPM**: If the port will support a phone that is equipped with a PCDPM and an MDPM, but is used only for off-hook voice announce and a data device connection, select this option. The MDPM circuit number appears in the Circuit 2 column. If you want to see the programming for this port (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the port number.

- **Dual Single-Lines**: If the port will be used for installing two single-line sets on an SLA, select this option. The circuit numbers for the Single-Line ports appear in the Circuit 1 and Circuit 2 column. If you want to see the programming for these ports (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the port number.
- **Single-Line/DID**: If the port will be used for installing one single-line set and one DID trunk on an SLA, select this option. The circuit numbers for the Single-Line port appears in the Circuit 1 column and the DID trunk appears in the Circuit 2 column. If you want to see the programming for these ports (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the port number.
- **Dual DID**: If the port will be used for installing two DID trunks on an SLA, select this option. The circuit numbers for the DID trunks appear in the Circuit 1 and Circuit 2 column. If you want to see the programming for these ports (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the port number.
- **Keyset/System Serial**: To configure the port for a phone with a system serial port, select **Keyset/System Serial**. (The phone may also use its PCDPM for off-hook voice announce.) The System Serial Port number automatically appears in the Circuit 2 column. If you want to see the programming for this port (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the port number.
- **Keyset/MDPM/System Serial**: To configure a port for a phone with a PCDPM and a MDPM that will be used as a system serial port **and** for off-hook voice announce and a data device connection, select this option. The MDPM circuit number appears in the Circuit 1 column and the System Serial Port number appears in the Circuit 3 column. If you want to see the programming for this port (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the port number.
- **Single Port LSA**: To configure a port for use with an IP Device, such as the IP Phone Plus or Model 8660, select this option. The Loop Start Adapter (LSA) has the same functionality as a standard loop trunk. To access the programming for this port (as performed in Devices and Feature Codes\Trunks, described on page 6-244), double-click on the port number.

3.45 To equip a single port: To select the port type for a specific circuit, select the current Port description and use the drop-down list box to scroll to the new port type. Then press **ENTER** or click on another field to save the change.

3.46 To equip several circuits at once: Highlight the circuits you want to equip. (To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items.) Then right-click and select **Batch Change Type**. The window on the right appears. Place a dot in the circuit type you want to assign and select **OK**.



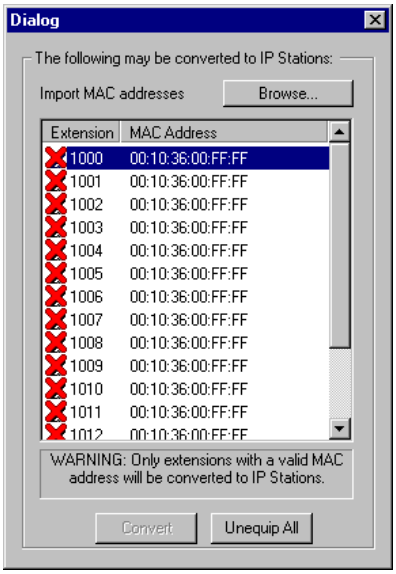
3.47 If the new devices require extension numbers, you will be prompted to select the first number in the batch. The available extension numbers will be assigned sequentially to the newly equipped circuits.

3.48 If you attempt to exceed the device limit of the System, you will see an error message telling you the operation cannot be performed.

Converting a DKSC-16 to a Version 8.2 or Later Firmware IPRC

NOTE: This feature is available only in local mode.

3.49 If a slot is programmed as a DKSC-16, and there are keysets and dual single-line ports programmed on that card, the following dialog box appears if you change that card to a device v8.2 or later firmware IPRC. (This occurs in local mode only.)



3.50 This dialog lists all of the keyset and dual single-line ports that were programmed for the DKSC-16. It allows you to convert these extensions to IP phones and/or IP SLAs. If you convert a device, the circuit is programmed based on the following:

- *If the circuit was programmed for a keyset,* it is converted to an IP device.
- *If the circuit was programmed for a dual single-line port,* it is converted to an IP SLA. Only one circuit, however, is used.
- *If the circuit was programmed as any other device (e.g., Keyset/MDPM),* it is not listed in the dialog and is automatically unequipped.

3.51 While in this dialog, you can enter the MAC address of any IP device or IP SLA that terminates at this card. Or, you can import MAC addresses from a **`.txt`** file (see page 6-148 for more information).

3.52 After you have entered or imported the MAC addresses for the appropriate devices, click **Convert**. If the MAC address is valid, the extension is converted. If the MAC address is invalid or if no MAC address is specified (i.e., a red X appears next to the extension), that circuit is unequipped. Circuits are equipped beginning with port 2.

3.53 If you want to manually assign the circuits, click **Unequip All**. Then program the circuits as described on page 6-147.

NOTE: The first card slot does not require a keyset card (e.g., DKSC16). You can change this slot to any card, other than a CPC or CPU card. If, however, you change the first slot to something other than a keyset card, you must reprogram the Auto Attendant in a keyset card.

Analog Keyset Card Configuration

- AKSC card ports support the following devices:
- 8-, 12-, and 24-Line AIM Analog Phones
- 8-, 12-, and 24-Line DVK Analog Phones
- 12-, and 24-Line GMX Analog Phones
- 8-, 12-, and 24-Line IMX Analog Phones (including Dual-Circuit Phones)
- Data Port Modules (DPMs) connected to Analog Phones. An analog DPM can be used for connecting a data device or a loud-ringing adapter to the phone.

3.54 When multiple purposes are assigned to one AKSC port, the circuit numbers are assigned as follows:

- **Keyset with DPM**: The phone is XX.XX.01 and the DPM is XX.XX.02
- **Dual-Circuit Keysets**: The first phone (on the "A" side) is XX.XX.01 and the second phone (on the "B" side) is XX.XX.02

3.55 When an AKSC port is changed from single purpose (Keyset) to multiple purpose (Keyset with DPM or Dual-Circuit Keyset), the database will retain all programming for the original single port and it will create the new additional station, with default values, in all other applicable areas of programming. The original single port will retain its extension number; the new ports will have extension numbers assigned as you exit from this programming area.

3.56 If the port is changed from a Dual-Circuit Keyset function to a single Keyset, the XX.XX.01 circuit programming is preserved and the XX.XX.02 circuit programming is erased from the database.

3.57 The circuit type options are as follows:

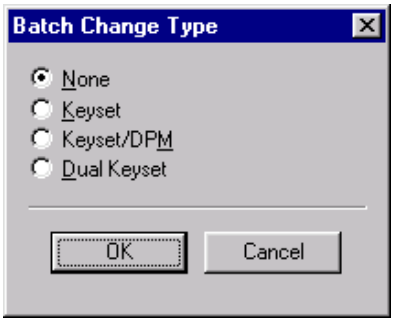
- **None**: To disable the port, select **None**.
- **Keyset**: This option is selected if the port will be used for installing a single analog phone that will not have an DPM connection. The phone connected to this circuit will have primary and secondary voice paths, allowing use of Off-Hook Voice Announce (OHVA) feature. If you want to see the programming for this phone (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the port number.
- **Keyset/DPM**: If the port will support a single analog phone that is equipped with an DPM, select this option. The device connected to the DPM shares the secondary voice path of the attached phone. The DPM circuit number appears in the Circuit 2 column. If you want to see the programming for this port (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the port number.

  NOTE: The system does not send ring voltage to the device connected to the DPM; it sends a signal to the DPM which, in turn, causes the device to ring.

- **Dual-Circuit Keysets**: Dual-circuit analog phones can be installed on the AKSC port. Each phone has one voice path. They cannot use the OHVA feature or have attached DPMs. The circuit numbers for the phones appear in the Circuit 1 and Circuit 2 columns. If you want to see the programming for this port (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the port number.

3.58 **To equip a single port:** To select the port type for a specific circuit, select the current Port description and use the drop-down list box to scroll to the new port type. Then press **ENTER** or click on another field to save the change.

3.59 **To equip several ports at once:** Highlight the ports you want to equip. (To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items.) Then right-click and select **Batch Change Type**. The following window appears. Place a dot in the circuit type you want to assign and select **OK**.



3.60 If the new devices require extension numbers, you will be prompted to select the first number in the batch. The available extension numbers will be assigned sequentially to the newly equipped ports.

3.61 If you attempt to exceed the device limit of the phone system, you will see an error message telling you the operation cannot be performed.

Executone Keyset Card Configuration

3.62 **To program an individual card,** select System\Cabinets then double-click on the card (or right-click and select **Explore Card**).

3.63 EKSC cards support Executone IDS Phones, Wave Phones, and DSS Consoles.

3.64 **To equip a single circuit:** To select the port type for a specific circuit, select the current Port description and use the drop-down list box to scroll to the new port type. Then press **ENTER** or click on another field to save the change.

3.65 **To equip several circuits at once:** Highlight the ports you want to equip. (To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items.) Then right-click and select **Batch Change Type**.

3.66 The circuit types are as follows:

- **None**: To disable the port, select None.
- **Keyset**: This option is selected if the port will be used for installing an Executone Phone.
- **DSS**: This option is selected if the port will be used for installing an Executone 48-key DSS Console (not the 60-key DSS/BLF Unit).

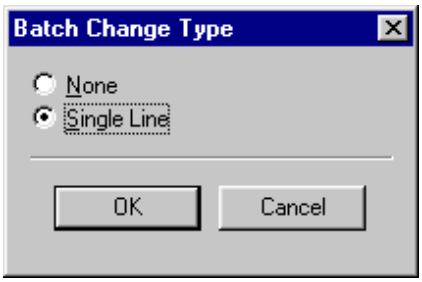
3.67 If you want to see the programming for a circuit (as performed in Devices and Feature Code programming), double-click on the port number.

Single-Line Card Configuration

3.68 The ports on the card and their current configuration are shown in the list.

3.69 **To equip a single port:** To select the port type for a specific circuit, select the current Port description and use the drop-down list box to scroll to the new port type. Then press **ENTER** or click on another field to save the change.

3.70 **To equip several ports at once:** Highlight the ports you want to equip. (To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items.) Then right-click and select **Batch Change Type**. The following window appears. Place a dot in the port type you want to assign and select **OK**.



3.71 If the new devices require extension numbers, you will be prompted to select the first number in the batch. The available extension numbers will be assigned sequentially to the newly equipped ports.

3.72 If you attempt to exceed the device limit of the System, you will see an error message telling you the operation cannot be performed.

3.73 The circuit number for the single-line set appears in the Circuit 1 column. If you want to see the programming for this station (as performed in Devices and Feature Code programming, described on page 6-156), double-click on the port number.

3.74 When the Singe Line 16 Cards (SLC-16s) are programmed in DB Programming, and there are SLCs installed in the associated slots and vice versa, the cards are not allowed to come online. They are, instead, logged as “Ignored” when plugged in. This change prevents a potential one-way audio problem where the system fails to allocate the necessary voice channels for a hardware board. The problem only exists for hardware boards that do not match the DB Programming board type.

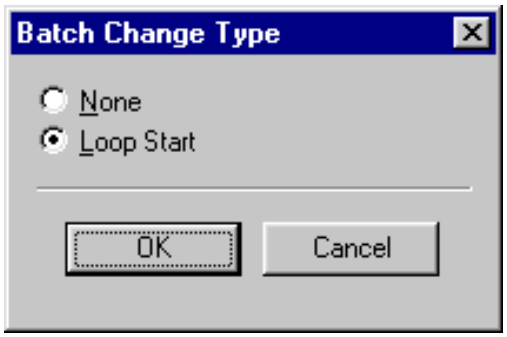
Loop Start Card Or Loop Start/Ground Start Card Configuration

3.75 The loop and ground start trunk cards (LSC, LSC-D, LGC, and LGC-D) are shown in the list. To program a port on one of the trunk cards, double-click on that card to display its ports.

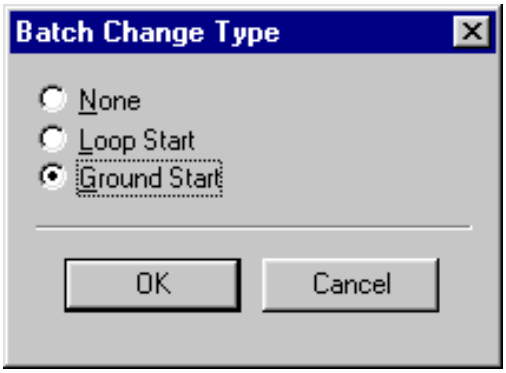
3.76 **To equip a single port:** To select the port type for a specific circuit, select the current Port description and use the drop-down list box to scroll to the new port type. Then press **ENTER** or click on another field to save the change.

3.77 **To equip several ports at once:** Highlight the ports you want to equip. (To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items.) Then right-click and select **Batch Change Type**. One of the following windows appears. Place a dot in the port type you want to assign and select **OK**.

Loop Start Options:



Loop/Ground Start Options:



3.78 If the new devices require extension numbers, you will be prompted to select the first number in the batch. The available extension numbers will be assigned sequentially to the newly equipped ports.

3.79 If you attempt to exceed the device limit of the System, you will see an error message telling you the operation cannot be performed.

3.80 When the Loop Start Card/Daughter cards (LSC-Ds) are programmed in DB Programming, and there are LSCs installed in the associated slots and vice versa, the cards are not allowed to come online. They are, instead, logged as “Ignored” when plugged in. This change prevents a potential one-way audio problem where the system fails to allocate the necessary voice channels for a hardware board. The problem only exists for hardware boards that do not match the DB Programming board type.

T1, T1/PRI, or E1/PRI Card Configuration and Programming

NOTE: DB Programming will not allow the system to equip PRI unless the premium feature for "Advanced CO Interfaces" is licensed.

3.81 The T1 cards are shown in the list. T1 and T1/E1 Cards are represented as "T1 Card" and T1/PRI and T1/E1/PRI Cards are represented as "T1 Primary Rate Card." E1/PRI and T1/E1/PRI Cards are represented as "E1 Primary Rate Card." To program a port on one of the T1 cards, double-click on that card to display its programming fields.

3.82 Card Programming includes the following fields:

- T1 or E1 Circuits (see page 6-126)
- T1/E1 Diagnostics (see page 6-127)
- Timers—PRI-equipped Cards Only (see page 6-129)
- Call Type—T1/PRI Only (see page 6-130)
- Overlap Sending—E1/PRI Only (see page 6-131)
- Description (see page 6-130)
- Framing Type—T1C and E1/PRI only (see page 6-132)
- Zero Code Suppression Scheme—T1C and E1/PRI only (see page 6-132)
- CRC Processing—E1/PRI Only (see page 6-133)
- Line Build-Out—T1C and E1/PRI only (see page 6-133)
- ISDN Switch Type—T1/PRI Only (see page 6-133)
- CO Provides Progress Tones—T1/PRI Only (see page 6-134)
- Connect on Call Proceeding—PRI-equipped Cards Only (see page 6-134)
- Operator System Access—T1/PRI Only (see page 6-134)
- Send International Toll Digits (see page 6-135)
- Retry ARS Call If Call Rejected (see page 6-135)
- Use Delayed Facility for CID Propagation Method (reserved for future use)

NOTE: System software v9.0 and later provides a networking wizard to simplify the required programming for T1/E1 PRI networking. See page page 6-87 for more information about the networking wizard.

T1 or E1 Circuits

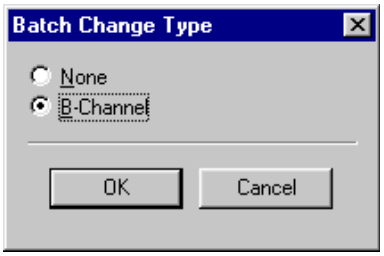
3.83 The following information describes how to equip circuits.

- **To equip a single port:** To select the port type for a specific circuit, select the current Port description and use the drop-down list box to scroll to the new port type. Then press **ENTER** or click on another field to save the change.

- **To equip several circuits at once:** Highlight the circuits you want to equip. (To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items.) Then right-click and select **Batch Change Type**. One of the following windows appears. Place a dot in the circuit type you want to assign and select **OK**.

T1 Options:

T1/PRI or E1/PRI Options:



3.84 If the new devices require extension numbers, you will be prompted to select the first number in the batch. The available extension numbers will be assigned sequentially to the newly equipped circuits. If you attempt to exceed the device limit of the System, you will see an error message telling you the operation cannot be performed.

T1/E1 Diagnostics

3.85 T1 Diagnostics is used for troubleshooting T1 or E1. It shows the status of various conditions for the selected card.

- **Hourly Display**: Double-click on **Hourly** to see the number of error that have occurred in each hour for the past 24 hours.
- **Daily Display**: Double-click on **Daily** to see the number of errors that have occurred during several 24-hour periods.
- **Hourly Summary**: The hourly summary shows the number of errors that have occurred in the past hour.
- **Daily Summary**: The daily summary shows the number of errors that have occurred in the past 24 hours.
- **Last Error Update Hour and Day**: These fields show the date and time that the information was last updated. (Information is automatically updated whenever a new card is selected.)
- **T1/E1 Status**: Double-click on **T1/E1 Status** to see the following fields. They show the status of the T1 or E1 card and allow you to perform diagnostics. To update the information, right-click anywhere and select **Update All Statuses**.
  - — **Make Busy**: This option can be used to busy-out all of the circuits. When it is enabled, the card is busied out, just as it would be if the busy-out switch on the card was used. To change this option, right-click on the current Value and then select **Enable Make Busy**. To turn off the Make busy, right-click again and select **Disable Make Busy**.

NOTE: The T1 or E1 circuits should be busied out before any diagnostics procedures are performed. If not, the card will go into Red Alarm and drop all active calls.

If the Make Busy LED is flashing rapidly on the card, someone pressed the Make Busy switch and the Busyout Switch Timeout timer expired before the card was removed. If this occurs, you must disable this Make Busy field.

— **Loopback Tests**: These options can be used for enabling and disabling the following T1 loopback diagnostics and alarm. During the tests, LED indications and error messages can be used to provide troubleshooting information. (The testing options cannot be used in stand-alone programming mode.) To enable or disable one of the tests, right-click on its current Value and then select **Enable**. To end the test, right-click again and select **Disable**. The loopback tests are as follows:

- *Digital Local Loopback:* Allows you to check the transmission and reception of alarm conditions between systems in a private T1 or E1 network. This tests the framing chip on the card.
- *Digital Remote Loopback:* Allows you to check the transmission and reception of test patterns conditions generated by the T1C or E1C. This tests the framing chip on the card.
- *Transceiver Local Loopback:* Allows you to check the transmission and reception of alarm conditions generated by the T1 or E1 Card. This tests the transceiver chip on the card.
- *Transceiver Remote Loopback:* Allows you to check the transmission and reception of test patterns between the T1C or E1C and the public network. This tests the transceiver chip on the card.
- *Transmit Blue Alarm:* This sends a Blue alarm over the public or private network interface for testing purposes.

NOTE: If the T1 or E1 Loopback timeout timer expires during a loopback test, the loopback will be terminated automatically.

- **Line Build-Out**: This information is shown here for reference purposes only. To program Line Build-out for the T1 or E1 Card, see page 6-133.

Timers

3.86 *(Timers are programmed only for PRI-equipped Cards.)* The ISDN timer default values, listed below, have been carefully selected to ensure proper system operation under most circumstances. Occasionally, one or more of the timers may need to be adjusted. To program a timer, select its current Value and enter the new value in the text box. Then press **ENTER** or click on another field to save the change.

Table 6-20: *ISDN Timers*

Timer	Default Value	Range	Purpose
Call Proceeding (T310)	25	1-255 seconds	*This is used for AT&T 5ESS custom switch type only.* If the appropriate Call Proceeding response (Alerting, Connect, or Disconnect) is not received before this timer expires, the requesting system sends a Disconnect.
Connect Ack. (T313)	9	1-255 seconds	If the appropriate Call Proceeding response (Alerting, Connect, or Disconnect) is not received before this timer expires, the requesting system sends a Disconnect.
Data Link Disconnect (T309)	90	1-255 seconds	*(T1/PRI Only)* This timer is initiated when the data link layer is disconnected. The timer is terminated when the data link is re-established.
Disconnect (T305)	4	1-255 seconds	If the appropriate Disconnect response (Disconnect or Release) is not received before this timer expires, the requesting system sends a Release.
Idle Link (T203)	100	1-65535 tenths	This timer controls how long the ISDN Data Link layer will remain idle before transmitting a Receive Ready (RR) Supervisory frame or an information frame, once multi-frame operation has been established.
Overlap Receiving (T302)*	15	1-255 seconds	This timer determines the number of seconds that the system will wait between messages during the overlap receiving process. This timer is started whenever the system receives a setup acknowledge message. It is then restarted every time the system receives an information message during the overlap receiving process. After the last information message is received, the system has until this timer expires to send a call progress message (for example, alerting, connected, etc.). If the system does not send a call progress message or if the system does not receive an information message before this timer expires, the call is cleared.
Overlap Sending (T304)*	20	1-255 seconds	This timer determines the number of seconds that the system will wait between messages during the overlap sending process. This timer is started when the system receives the setup acknowledge message from the network provider. It is then restarted every time the system sends an information message during the overlap sending process. After the last information message is sent, the system waits for a call progress message (for example, alerting, connected, etc.) from the network provider. If the system does not receive a call progress message or if the system does not send an information message before this timer expires, the call is cleared.

Table 6-20: *ISDN Timers (Continued)*

Timer	Default Value	Range	Purpose
Release (T308)	4	1-255 seconds	If the appropriate Release response (Release Complete) is not received before this timer expires, the requesting system sends another Release or releases the B-channel and Call Reference value.
Restart Acknowledge (T316)	120	1-255 seconds	The system initiates this timer when a Channel Restart message is sent. The timer is terminated when the Restart Acknowledge is received from the public network. If the Restart Acknowledge is not received before this timer expires, the system sends a second Restart message to the public network. If the timer expires a second time and a Restart Acknowledge has not been received from the public network, the system idles the respective channel.
Resume Acknowledge (T318)	9 seconds	1-255 seconds	*(E1/PRI Only)* If the appropriate Resume response (Resume Acknowledge or Resume Reject) is not received before this timer expires, the requesting system sends a Release Request.
Re-Transmission (T200)	10	1-65535 tenths	This timer controls the amount of time the ISDN Data Link layer will wait for an acknowledgment from the network before it transmits a frame again.
Setup Request Ack. (T303)	4 (9 for 4ESS)	1-255 seconds	If the appropriate Setup Request response (Alerting, Connect, Call Proceeding, Setup Acknowledge, or Release) is not received before this timer expires, the requesting system sends another Setup Request or clears the call.
Suspend Acknowledge (T319)	9 seconds	1-255 seconds	*(E1/PRI Only)* If the appropriate Suspend response (Disconnect, Release, Suspend Acknowledge, or Suspend Reject) is not received before this timer expires, the requesting system sends a Suspend Reject.
XID Request (T204)	20 tenths	1-65535 tenths	This timer controls how long the ISDN Link Data layer will wait for an XID response from the network before transmitting the XID Request again. (XID parameter negotiation is not supported.)

* Currently, the overlap sending/receiving protocols are supported with PRI Net 5 and BRI Net 3 switch types, which are prevalent in the Europe and Mexico markets.

Call Type

3.87 *(T1/PRI Only)* All Primary rate ISDN calls have a specific number type (International, National, Network, or Local/Subscriber) and numbering plan field (ISDN, Telephony, or Private) indicated in the call setup message to the public network peer. In certain cases, front end equipment interfacing with the ISDN user (CPE) peer requires a specific number type and/or numbering plan. The number type and numbering plan are programmable per call type for each equipped primary rate card. The operation of this feature is transparent to the user. After the number type and numbering plan are programmed, the Primary rate call setup message will include this information for each call. To set the Primary Rate Call Type for a T1/PRI card, select **System** and **Cabinets**, then double-click on the card (or right-click and select Explore Card). Then select **Call Type**.

Description

3.88 You can enter a description for the card. Select its current Value and enter the description in the text box. Then press **ENTER** or click on another field to save the change.

Overlap Sending

3.89 *(E1/PRI or BRI Only)* If using a PRI-equipped card with PRI Net 5 and BRI Net 3 switch types, you can program the system to use the overlap sending and receiving protocols. These protocols allow you to program the switch to send called number digits either in the setup message or the information message. To access the programming options, double-click Overlap Sending. You then have the following options:

- **Immediate Overlap Sending**: Enable this flag to have the system immediately send the setup message to the network provider. If disabled, all called number digits are collected before sending the setup message to the network provider. You should enable this flag if you want the system to seize a B-channel immediately or if you want the network provider to provide immediate feedback for invalid numbers. You should disable this flag if the network provider does not support overlap receiving. By default, this flag is *disabled*.

  NOTE: Because Automatic Route Selection (ARS) calls require the phone system to collect digits before determining the most cost-effective route, this flag is ignored for ARS calls.

- **Maximum Digits in Called Number IE**: Select the maximum number of called number digits that the system will include in the setup and/or information messages. This field may be required if the network provider only allows calls with a specified number of maximum digits. The number of digits programmed here and the Overlap Sending Conditions value determine the number of called number digits sent in the called number information elements (IEs) for setup and information messages. The valid range is 1 - 48, and the default is *20*.

  NOTE: If the Immediate Overlap Sending flag is enabled, this field is ignored.

- **Overlap Receiving**: Enable this flag if the network provider does not send digits "en bloc" to the phone system. If enabled, the system will not route the call until it has received a sending complete indication from the network provider or until the Overlap Receiving Timeout Timer has expired (see below). You should enable this flag if the network provider sends any called number digits in subsequent information messages. You should disable this flag if the network provider sends all of the called number digits in the setup message ("en bloc"). By default, this is *disabled*.
- **Overlap Receiving Timeout Timer**: Select the number of seconds that the system will wait to receive any called number digits from the network provider. You may need to adjust this value if the network provider does not send a sending complete indication to signal the end of the called number digits. The valid range is 1 - 255, and the default is *30* seconds.
- **Overlap Sending Conditions**: Select how the system will send digits if the Immediate Overlap Sending flag is disabled. Whether or not the overlap sending protocol is used depends on the value selected here. You have the following options:
  - — *Overlap Sending Disabled:* Sends all digits "en bloc" as part of the called number IE in the setup message. This option does not allow overlap sending.
  - — *Overflow Is Equal Access Only:* Sends any overflow digits as equal access digits in the setup message. This option does not allow overlap sending. This is the default setting.
  - — *Overflow Digits Sent In Overlap:* Sends any overflow digits in subsequent information messages. This option allows overlap sending.

— *Overlap And Equal Access Facility:* Sends any equal access digits in the setup message, but sends overflow digits that are not considered equal access in subsequent information messages. This option allows overlap sending.

— *Overlap Sending Enabled For All Calls:* Sends all digits in subsequent information messages (that is, none are included in the setup message). The number of digits sent in each information message is limited by the Maximum Digits in Called Number IE setting. This option allows overlap sending.

- **Sending Complete Indication**: Select how the phone system will indicate that it has finished sending digits to the network provider. If Sending Complete IE is selected, the phone system includes a sending complete information element in the setup or information message. If '#' Digit in Called Number IE is selected, the phone system inserts a pound [hash in Europe] digit (#) at the end of the digits in the called number information element. By default, this is *Sending Complete IE*.

NOTE: This field is ignored if the Transmit Sending Complete Indication flag is disabled.

- **Transmit Sending Complete Indication**: Enable this flag to have the phone system send the complete indication type selected in the Sending Complete Indication field. If this flag is disabled, the phone system will not send any indication to the network provider that no more digits follow. By default, this is *disabled*.

NOTE: The Transmit Sending Complete Indication and Sending Complete Indication fields also affect ISDN trunks that use the "en bloc" method for delivering digits. For example, if the Transmit Sending Complete Indication flag is enabled and the Sending Complete Indication value is set to '#' Digit in Called Number IE, the system will insert a pound [hash] digit (#) after the collected digits.

Framing Type

3.90 The type of framing scheme used by the T1/E1 trunks connected to the card can be as follows:

- T1 Trunks—D4 Superframe or Extended Superframe (ESF)
- E1 Trunks—Common Channel Signaling or Channel Associated Signaling

3.91 Select the current Value and use the drop-down list box to scroll to the desired setting. Then press **ENTER** or click on another field to save the change.

3.92 **Zero Code Suppression Scheme**: The trunk zero-suppression scheme supports the followings:

- Trunks on the card—AMI (Bit 7), Bipolar Eight Zero Substitution (B8ZS), or None
- T1/PRI and T1/E1/PRI cards—B8ZS or None
- E1/PRI cards—HDB3 or None

3.93 To change the setting, select the current Value and use the drop-down list box to scroll to the desired setting. Then press **ENTER** or click on another field to save the change.

CRC Processing

3.94 *(E1/PRI Only)* The CRC Processing flag is a configuration flag that can be enabled/disabled for each E1 card. The setting for this flag must match the setting for the remote-end E1 signal. When CRC Processing is enabled only at one end of an E1 connection, the E1 signal with the CRC enabled will never reach the CRC Multiframe synchronization state and will transmit the RAI signal to the remote end until either the CRC Processing is disabled for that E1 signal or the CRC Processing is enabled for the remote-end E1 signal.

3.95 This flag enables the transmission of a CRC Multiframe pattern and a CRC-4 value computed/transmitted for every 16 frames of data. At the same time, the CRC Processing flag enables the E1 Card to look for this CRC Multiframe pattern and CRC-4 value on the received E1 signal. When CRC Processing is enabled on both ends of an E1 connection, both ends will reach the CRC Multiframe synchronization. Then, in this multiframe synchronized state, both ends will compute a CRC-4 value and transmit it to the other end along with the E1 data. On the receive ends of the E1 connection, both receivers will then calculate their own CRC-4 value for the received data and if the received and the computed CRC-4 values match, the corresponding 16 frames of E1 data are assumed to be have been received with no bit errors.

Line Build-Out

3.96 The LBO attenuation of the T1 trunk connected to the card value is determined by the distance to the nearest public network T1 repeater. The selections are shown below. (The Auto option should be used during initial installation. The other settings can be used if the build-out needs to be adjusted later.) To change the setting, select the current Value and use the drop-down list box to scroll to the desired setting. Then press **ENTER** or click on another field to save the change.

- 0 dB (DSX-1)
- -7.5 dB (this is the best setting for T1/E1/PRI Cards)
- -15 dB
- -22.5 dB
- Auto

ISDN Switch Type

3.97 *(Used for PRI-equipped Cards only)* The System supports primary-rate B-channels.

- T1/E1/PRI Cards support AT&T 4ESS Custom, AT&T 5ESS Custom, DMS-100 or National ISDN2, Private Networking and IP Private Networking.
- E1/PRI Card supports Private Networking, PRI NET-5, and IP Private Networking.

3.98 If the card being programmed will be used to connect two nodes in a network, select **Private Networking**. To change the setting, select the current Value and use the drop-down list box to scroll to the desired setting. Then press **ENTER** or click on another field to save the change. If you selected Private Networking as the ISDN Switch Type, you will be prompted to select the node that will contain the trunks on this card. If you want to change the node later, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the existing Node Number. An option box appears.
2. Select **Change Node Number**. A window appears that asks you to select the type of device to include.
3. Select **Local Node** or **Remote Node** and then select **Next**. The list of nodes appears. (You can view a list by selecting List or view details by selecting Details.)
4. Highlight the node and then select **Finish**. Your selection appears as the Node Number.

NOTE: If you change the ISDN Switch Type to Private Networking, the system automatically places the trunks in the first available node trunk group. (See page 6-243 for more information on node trunk groups.) If you change to any other ISDN Switch Type option, all of the trunks are moved to the "Unused" trunk group.

CO Provides Progress Tones

3.99 *(Used for PRI-equipped Cards only)* When the System provides local ringback, some central offices/local exchanges (particularly in Europe) take too long to send the System the connect message when a call is received. When the CO connects a call too slowly, the user can answer before the caller is on the line, causing the caller to miss the greeting (such as "Welcome to Inter-Tel. How may I help you?"). If this flag is enabled, the System does not provide local ringback. Instead, the System connects the call to the line and the caller hears CO ringback. That way, when the called party answers, the calling party is already on the line. The flag is *disabled* by default. To change the setting, select the current Value and then place a check mark in the box by selecting it (to remove the check, select the box again). Then press **ENTER** or select another field to save the change.

NOTE: This flag should be enabled for PRI spans that are used for placing outgoing calls from another node (e.g., Node 2 uses ARS to place outgoing calls using the PRI span on Node 1). If it is not enabled, some of the calls from Node 2 may not go out to the public network properly. Enable this flag for **all** nodes in a system.

Connect On Call Proceeding

3.100 *(Used for PRI-equipped Cards only)* This flag affects outgoing Primary Rate ISDN calls. If it is enabled, the system will connect the B-channel as soon as the Call Proceeding message is received from the CO interface. Some CO interfaces, especially SS7 interfaces, do not send progress indicators when they are playing tones or announcements on the B-channel. They play the tone or announcement prior to sending the progress message, after the call proceeding. In this case, if the flag is *disabled,* the user experiences audio clipping. If the flag is enabled, the channel is opened when the call proceeding message is received and the system does not have to wait for a progress indicator (indicating alerting, busy, reorder tones or announcements).

Operator System Access

3.101 *(Used for PRI-equipped Cards set for National ISDN2 only)* If National ISDN 2 is selected, there is an option to enable Operator System Access. Operator System Access (OSA) is required by some central offices. If this flag is enabled, the user will be allowed to request access to an operator services system.

NOTE: If using ARS with National ISDN 2, and OSA is enabled, you must remove Dial Rule #2 Echo Toll Field from the Operator and International Operator Route groups. (See page 6-293.)

Send International Toll Digits

3.102 *(Used for T1/PRI or E1/PRI trunks only)* If enabled (default), the system sends international toll digits (for example, 011) in the dial string in addition to the number type. If disabled, the toll digits are stripped from the dial string for international calls using ISDN trunks. You should enable this flag if the CO ignores the number type. By default, this is *enabled*.

Retry ARS Call If Call Rejected

3.103 This flag is for ISDN messaging for outgoing calls using ARS. If set to **Yes**, the system will retry the additional channels and trunk groups programmed in ARS if the call is rejected by the telco when receiving a specific cause value.

3.104 The symptom that will appear because of this flag is when an outgoing call is rejected by a telco, where it causes the system (via ARS) to cycle through all the ISDN channels before releasing/disconnecting the call.

3.105 This flag must remain enabled or set to **Yes** on T1/PRI or E1/PRI circuits connected to Interprise 3200s. This flag is enabled by default. The flag should be disabled on Pubic ISDN circuits connected to a telco. If not using ARS with ISDN services, this flag does not apply.

3.106 When this flag is set to **No**, the system will not retry the ARS call, regardless of the reason it was rejected. If this flag is set to **Yes**, the system will retry the ARS call as long as the cause of the rejection is NOT one of the following causes:

- Unallocated or unassigned number
- The user is busy
- No user is responding
- Call rejected
- Number has changed
- Destination is out of order
- Number format is invalid
- There was a temporary failure
- Congestion in the switching equipment
- Requested circuit/channel is not available
- Outgoing calls are barred
- Incoming calls are barred
- Service or option not available, unspecified
- Service or option not implemented, unspecified
- Mandatory information element is missing
- Interworking, unspecified

BRI Card Configuration

3.107 To program an individual card, select System\Cabinets then double-click on the card (or right-click and select Explore Card).

NOTE: DB Programming will not allow the system to equip BRI cards unless the Premium Feature for "Advanced CO Interfaces" is licensed.

3.108 The BRI cards are shown in the list. BRS Cards are represented as "Basic Rate Interface - S/T" and BRU Cards are represented as "Basic Rate Interface - U."

3.109 **To equip a single circuit:** To select the port type for a specific circuit, select the current Port description and use the drop-down list box to scroll to the new port type. Then press **ENTER** or click on another field to save the change.

3.110 **To equip several circuits at once:** Highlight the ports you want to equip. (To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items.) Then right-click and select the Batch Change Type option.

3.111 To program a port on one of the BRI cards, double-click on that card and then on Ports to display its programming fields. Card programming includes the following fields:

- **Port**: Most fields may be programmed for both station and trunk ports. In DB Programming, choose System\Cabinets\the card's folder\Ports to access the programmable fields. For fields that apply specifically to either station or trunk ports only, that information is noted. See for more details.
- **Make Busy**: This option can be used to busy-out all of the circuits. When it is enabled, the card is busied out, just as it would be if the busy-out switch on the card was used. To change this option, right-click on the current Value and then select Enable Make Busy. To turn off the Make busy, right-click again and select Disable Make Busy.

3.112 Card Programming includes the following fields:

- Timers (see page 6-137)
- Diagnostics (see page 6-138)
- *SPID/DN/TEI [or TEI in Europe] (see page 6-139)
- *Call Type (see page 6-140)
- Overlap Sending (see page 6-141)
- Description (see page 6-140)
- ISDN [local exchange] Switch Type (see page 6-142)
- *CO Provides Progress Tones (see page 6-142)
- *Connect on Call Proceeding (see page 6-143)
- *Retry ARS Call If Call Rejected (see page 6-143)
- **Forward if Busy to Other Device (see page 6-144)
- *Process ISDN Data Link Down (see page 6-144)
- Use Delayed Facility for CID Propagation Method (reserved for future used)

* Trunk ports only.

** Station ports only.

Timers

3.113 The BRI timers are programmed in System\Cabinets\Ports in the each port's folder. The timers listed in the following table can be programmed for each port.

Table 6-21: *BRI Timers*

Timer	Default Value	Range	Purpose
Busy No Answer Timer (T301)	0	0-255 seconds	*Applicable to the System for Japan only.* If the Call Delivered response (connect message) is not received before this timer expires, the System disconnects the call.
Call Proceeding (T310)	25	1-255 seconds	*Used for AT&T 5ESS custom switch type only.* If the appropriate Call Proceeding response (Alerting, Connect, or Disconnect) is not received before this timer expires, the requesting system sends a Disconnect.
Connect Acknowledge (T313)	4	1-255 seconds	If the appropriate Connect Acknowledge is not received before this timer expires, the requesting system sends a Disconnect.
Data Link Disconnect (T309)	90	1-255 seconds	This timer is initiated when the data link layer is disconnected and is terminated when the data link is re-established. If the link is re-established before this timer expires, the call is reconnected. If the timer expires before the link is established, the call is released, and the B-channel is cleared.
Data Link Down (T321)	30	1-255 seconds	This timer is initiated when the data link layer fails and is terminated when the data link is re-established. If this timer expires before the link is re-established, the system attempts to establish the link by sending a Data Link Establish Request.
Disconnect (T305)	4	1-255 seconds	If the Disconnect or Release response is not received before this timer expires, the requesting system sends a Release.
Idle Link (T203)	100	1-65535 tenths	This timer controls how long the ISDN Data Link layer will remain idle before transmitting a Receive Ready (RR) Supervisory frame or an information frame, once multi-frame operation has been established.
Overlap Receiving (T302)*	15	1-255 seconds	This timer determines the number of seconds that the system will wait between messages during the overlap receiving process. This timer is started whenever the system receives a setup acknowledge message. It is then restarted every time the system receives an information message during the overlap receiving process. After the last information message is received, the system has until this timer expires to send a call progress message (for example, alerting, connected, etc.). If the system does not send a call progress message or if the system does not receive an information message before this timer expires, the call is cleared
Overlap Sending (T304)*	20	1-255 seconds	This timer determines the number of seconds that the system will wait between messages during the overlap sending process. This timer is started when the system receives the setup acknowledge message from the network provider. It is then restarted every time the system sends an information message during the overlap sending process. After the last information message is sent, the system waits for a call progress message (for example, alerting, connected, etc.) from the network provider. If the system does not receive a call progress message or if the system does not send an information message before this timer expires, the call is cleared.
Release (T308)	4	1-255 seconds	If the appropriate Release response (Release Complete) is not received before this timer expires, the requesting system sends another Release or releases the B-channel and Call Reference value.

Table 6-21: *BRI Timers (Continued)*

Timer	Default Value	Range	Purpose
Restart Acknowledge (T316)	120	1-255 seconds	The system initiates this timer when a Channel Restart message is sent. The timer is terminated when the Restart Acknowledge is received from the public network. If the Restart Acknowledge is not received before this timer expires, the system sends a second Restart message to the public network. If the timer expires a second time and a Restart Acknowledge has not been received from the public network, the system idles the respective channel.
Retransmit TE1 Request UI Messages (T202)	20	1-65535 tenths	Controls the amount of time the ISDN Data Link layer will wait for a response from the network before it re-transmits a UI frame during the TEI assignment procedure.
Resume Acknowledge (T318)	4	1-255 seconds	If the appropriate Resume response (Resume Acknowledge or Resume Reject) is not received before this timer expires, the requesting system sends a Release Request.
Re-Transmission (T200)	10	1-65535 tenths	This timer controls the amount of time the ISDN Data Link layer will wait for an acknowledgment from the network before it transmits a frame again.
Setup Request Acknowledge (T303)	4	1-255 seconds	If the appropriate Setup Request response (Alerting, Connect, Call Proceeding, Setup Acknowledge, or Release) is not received before this timer expires, the requesting system sends another Setup Request or clears the call.
Status Enquiry (T322)	4	1-255 second	The system is expected to respond to the Network Status Enquiry within this time interval. If the system does not respond before the timer expires, the network will send a second Status Enquiry and will again expect the system's status response within this interval. Although the system does not actively use this timer, it is provided as a reference to gauge the system's response.
Suspend Acknowledge (T319)	4	1-255 seconds	If the appropriate Suspend response (Disconnect, Release, Suspend Acknowledge, or Suspend Reject) is not received before this timer expires, the requesting system sends a Suspend Reject.
XID Request (T204)	20	1-65535 tenths	This timer controls how long the ISDN Link Data layer will wait for an XID response from the network before transmitting the XID Request again. (XID parameter negotiation is not supported.)

* Currently, the overlap sending/receiving protocols are supported with PRI Net 5 and BRI Net 3 switch types, which are prevalent in the Europe and Mexico markets.

Diagnostics

3.114 The Diagnostics folders, located within each trunk port folder in DB Programming, are used for troubleshooting. The following information appears:

- **Hourly Display**: Double-click on Hourly to see the number of errors that have occurred during each hour for the past 24 hours.
- **Daily Display**: Double-click on Daily to see the number of errors that have occurred during several 24-hour periods.
- **Hourly Summary**: The hourly summary shows the number of errors that have occurred in the past hour.
- **Daily Summary**: The daily summary shows the number of errors that have occurred in the past 24 hours.

- **Last Error Update Hour and Day**: These fields show the date and time that the information was last updated. (Information is automatically updated whenever a new card is selected.)
- **BRI Port Status**: This folder, located in the Diagnostics folder, shows the following status indicators:
  - **Out of Sync Alarm**: When this field displays a NO value, it indicates that the port is synchronized with the terminal connection. If it is a trunk port, this indicates that the connection with the CO is operational. If it is a station port, it indicates that the signal is being transmitted to any devices connected to the port. If the value is YES, then the port is not synchronized with its terminal connection, but may still have signal.
  - **Loss of Signal (LOS) Alarm**: This field indicates that the port either does (a value of NO) or does not (a value of YES) recognize the device or CO trunk it is connected to. When the Loss of Signal field has a value of YES, the Out of Sync value should also be YES. When the Loss of Signal field has a value of NO, the Out of Sync Alarm field can assume either value and provides additional information about BRI status.
  - **2B Loopback**: When the 2B Loopback Test is active, data entering the device on the B1/B2 receive channels is demodulated and remodulated back out on the B1/B2 transmit channels. Data is received and transmitted on the same B-channel, either B1 or B2, and does not cross channels. This test checks the transmission and reception between the BRS Card and the public network, verifying the integrity of the BRS chip transceiver and the physical interface from the card to the public network. BRS station ports cannot perform loopback tests on the D-channel.
  - **2B+D Loopback**: When the 2B+D Loopback Test is active, data is received and transmitted as described for the 2B Loopback Test with the addition of the D-channel. This test checks the transmission and reception patterns between the BRU Card and the public network, confirms the integrity of the physical interface between the card and the public network, and tests the BRU chip transceiver.

NOTE: When the 2B+D Loopback Test is active, data is received and transmitted as described for the 2B Loopback Test with the addition of the D-channel. This test checks the transmission and reception patterns between the BRU Card and the public network, confirms the integrity of the physical interface between the card and the public network, and tests the BRU chip transceiver.

SPID/DN/TEI

3.115 *(For trunk ports only.)* The SPID/DN/TEI or TEI option applies only for BRI trunk ports only. Each BRI trunk can have up to 16 SPID/DN/TEI or TEI pairs.

- **Directory Number (DN)**: *(U.S. Only)* Is the phone number. No DN numbers are assigned by default.
- **Service Profile Identification (SPID)**: *(U.S. Only)* Contains the bearer information so that the CO can associate the number with a set of bearer types, The CO will reject calls to the directory number based on the bearer types assigned to that number. This also allows the CO to charge a higher rate for numbers that have more bearer capabilities (for example, numbers which support data or video as opposed to just speech). No SPIDs are assigned by default.

- **Terminal Endpoint Identifier (TEI)**: Contains value that Network uses to identify the terminal. This value can be fixed or automatically assigned by the Central Office [local exchange]. In a point-to-point setup, there is only one TEI. In a point-to-multipoint setup, there is at least two. For the U.S., each SPID/DN pair has an associated TEI. The default is set to Automatic.

NOTE: In the U.S., some Central Offices do not allow 16 SPIDs, and some allow as few as six. Also, SPID/DN/TEI pairs are only defined for North American CO switch types, so they are defined only for the U.S. Systems. Other countries do not support SPID/DN.

3.116 To program this field, select System\Cabinets and double-click on the **Basic Rate Interface - U or S/T** Card (or right-click and select Explore Card). Select Trunk\Trunk from the Ports and choose the desired extension number, then double-click on the **Ports**.

- When a BRI port comes on line, its SPIDs and Directory Numbers are registered. Each time a SPID/DN pair is registered, the system displays "BRI Trunk Port [B.p.0] Registered SPID <*SPID*>."
- If a BRI trunk port does not have any SPIDs programmed, the message "Trunk Port [b.p.0] does not have any SPIDs programmed" appears to explain why the port is not coming on line:
- If the Central Office rejects one of the programmed SPIDs, the system displays the message "BRI Trunk Port [b.p.0] has an invalid SPID <*SPID*>" to explain why the number may not be working.

Call Type

3.117 *(For trunk ports only.)* Select Call Type in each trunk port's folder. All Basic Rate ISDN calls have a specific number type and numbering plan field indicated in the call setup message to the public network peer. In certain cases, front end equipment interfacing with the ISDN user (CPE—Customer Provided Equipment) peer requires a specific number type and/or numbering plan. In the past the System was unable to support this requirement because the number type and numbering plan options were not programmable. Now the number type and numbering plan are programmable per call type for each equipped basic rate card. The operation of this feature is transparent to the user. After the number type and numbering plan are programmed, the basic rate call setup message will include this information for each call.

Description

3.118 The user can enter a description for the card. Select the field and enter the description in the text box. Then press **ENTER** or click on another field to save the change.

3.119 To program this field, select System\Cabinets then double-click on the card (or right-click and select **Explore Card**).

Overlap Sending

3.120 If using a PRI-equipped card with PRI Net 5 and BRI Net 3 switch types, you can program the system to use the overlap sending and receiving protocols. These protocols allow you to program the switch to send called number digits either in the setup message or the information message. To access the programming options, double-click Overlap Sending. You then have the following options:

- **Immediate Overlap Sending**: With this flag enabled, the system does not collect the Called Number digits on non-Automatic Route Selection (ARS) calls before sending a setup message to the network provider. Instead, the system will send each digits as it is dialed in subsequent information messages (one message per digit) until the network has received enough digits to route the call. Disabling this flag allows the system to collect the Caller Number digits for ALL calls (ARS and non-ARS type calls). After all digits are collected, the system sends the Called Number digits based on the programmable "Overlap Sending Condition." This flag should be enabled in order for the system to seize a B-channel immediately or if the network provider is to provide immediate feedback for invalid numbers. This flag should be disabled if the network provider does not support overlap receiving. By default, this flag is *disabled*.

  NOTE: Because ARS calls require the phone system to collect digits before determining the most cost-effective route, this flag is ignored for ARS calls.

- **Maximum Digits in Called Number IE**: Select the maximum number of Called Number digits that the system will include in the setup and/or information messages. This field may be required if the network provider only allows calls with a specified number of maximum digits. The number of digits programmed here and the Overlap Sending Conditions value determine the number of Called Number digits sent in the Called Number information elements (IEs) for setup and information messages. The valid range is 1 - 48, and the default is *20*.

  NOTE: If the Immediate Overlap Sending flag is enabled, this field is ignored.

- **Overlap Receiving**: Enable this flag if the network provider does not send digits "en bloc" to the phone system. If enabled, the system will not route the call until it has received a sending complete indication from the network provider or until the Overlap Receiving Timeout Timer has expired (see below). You should enable this flag if the network provider sends any Called Number digits in subsequent information messages. You should disable this flag if the network provider sends all of the Called Number digits in the setup message ("en bloc"). By default, this is *disabled*.

- **Overlap Receiving Timeout Timer**: Select the number of seconds that the system will wait to receive any Called Number digits from the network provider. You may need to adjust this value if the network provider does not send a sending complete indication to signal the end of the Called Number digits. The valid range is 1 - 255, and the default is *30* seconds.

- **Overlap Sending Conditions**: Select how the system will send digits if the Immediate Overlap Sending flag is disabled. Whether or not the overlap sending protocol is used depends on the value selected here. You have the following options:

  — *Overlap Sending Disabled:* Sends all digits "en bloc" as part of the Called Number IE in the setup message. This option does not allow overlap sending.

  — *Overflow Is Equal Access Only:* Sends any overflow digits as equal access digits in the setup message. This option does not allow overlap sending. This is the default setting.

— *Overflow Digits Sent In Overlap:* Sends any overflow digits in subsequent information messages. This option allows overlap sending.

— *Overlap And Equal Access Facility:* Sends any equal access digits in the setup message, but sends overflow digits that are not considered equal access in subsequent information messages. This option allows overlap sending.

— *Overlap Sending Enabled For All Calls:* Sends all digits in subsequent information messages (that is, none are included in the setup message). The number of digits sent in each information message is limited by the Maximum Digits in Called Number IE setting. This option allows overlap sending.

- **Sending Complete Indication**: Select how the phone system will indicate that it has finished sending digits to the network provider. If Sending Complete IE is selected, the phone system includes a sending complete information element in the setup or information message. If '#' Digit in Called Number IE is selected, the phone system inserts a pound [hash] digit (#) at the end of the digits in the Called Number information element. By default, this is *Sending Complete IE*.

NOTE: This field is ignored if the Transmit Sending Complete Indication flag is disabled.

- **Transmit Sending Complete Indication**: Enable this flag to have the phone system send the complete indication type selected in the Sending Complete Indication field. If this flag is disabled, the phone system will not send any indication to the network provider that no more digits follow. By default, this is *disabled*.

NOTE: The Transmit Sending Complete Indication and Sending Complete Indication fields also affect ISDN trunks that use the "en bloc" method for delivering digits. For example, if the Transmit Sending Complete Indication flag is enabled and the Sending Complete Indication value is set to '#' Digit in Called Number IE, the system will insert a pound [hash] digit (#) after the collected digits.

ISDN [Local Exchange] Switch Type

3.121 The BRI Cards support AT&T 5ESS Custom, National ISDN1, National ISDN2, and DMS-100 switch types in the United States and NET-3 in Europe. No other switch types are currently supported by the System. The default type in European systems is National ISDN1. To change the setting, select the current value and use the drop-down list box to scroll to the desired setting. Then press **ENTER** or select another field to save the change.

CO Provides Progress Tones

3.122 *(For trunk ports only.)* When the System provides local ringback, some central offices/ local exchanges (particularly in Europe) take too long to send the System the connect message when a call is received. When the CO [local exchange] connects a call too slowly, the user can answer before the caller is on the line, causing the caller to miss the greeting (such as "Welcome to Inter-Tel. How may I help you?"). If this flag is enabled, the System does not provide local ringback. Instead, the System connects the call to the line and the caller hears CO [local exchange] ringback. That way, when the called party answers, the calling party is already on the line. The flag is *disabled* by default. Most sites will not require this flag and it is best to leave it disabled unless the delayed-connect problem occurs. To change the setting, select the current Value and then place a check mark in the box by selecting it (to remove the check, select the box again). then press **ENTER** or select another field to save the change.

Connect On Call Proceeding

3.123 *(For trunk ports only.)* This flag affects outgoing Basic or Primary Rate ISDN calls. If it is enabled, the system will connect the B-channel as soon as the Call Proceeding message is received from the Central Office [local exchange] interface. Some Central Office [local exchange] interfaces, especially SS7 interfaces, do not send progress indicators when they are playing tones or announcements on the B-channel. They play the tone or announcement prior to sending the progress message, after the call proceeding. In this case, if the flag is *disabled,* the user experiences audio clipping. If the flag is enabled, the channel is opened when the call proceeding message is received and the system does not have to wait for a progress indicator (indicating alerting, busy, reorder tones or announcements).

Retry ARS Call If Call Rejected

3.124 *(For trunk ports only.)* This flag is for ISDN messaging for outgoing calls using ARS. If set to **Yes**, the system will retry the additional channels and trunk groups programmed in ARS if the call is rejected by the telco when receiving a specific cause value.

3.125 The symptom that will appear because of this flag is when an outgoing call is rejected by a telco, where it causes the system (via ARS) to cycle through all the ISDN channels before releasing/disconnecting the call.

3.126 This flag is enabled by default. The flag should be disabled on Pubic ISDN circuits connected to a telco. If not using ARS with ISDN services, this flag does not apply.

3.127 When this flag is set to **No**, the system will not retry the ARS call, regardless of the reason it was rejected. If this flag is set to **Yes**, the system will retry the ARS call as long as the cause of the rejection is NOT one of the following causes:

- Unallocated or unassigned number
- User is busy
- No user is responding
- Call rejected
- Number has changed
- Destination is out of order
- Number format is invalid
- There was a temporary failure
- Congestion in the switching equipment
- Requested circuit/channel is not available
- Outgoing calls are barred
- Incoming calls are barred
- Service or option not available, unspecified
- Service or option not implemented, unspecified
- Mandatory information element is missing
- Interworking, unspecified

Forward if Busy to Other Device

3.128 *(For station ports only.)* This port level flag enables calls coming in to one device on a BRI port to be automatically forwarded to the other device on that port when the first device is busy. This is mainly used for video conference units.

Process ISDN Data Link Down

3.129 *(For trunk ports only.)* This field allows a system to determine whether or not the BRI trunk port processes an ISDN Data Link Down message. When this flag is set to Yes and a BRI trunk port receives an ISDN Data Link Down message, then a BRI trunk port drops all calls and attempts to bring the ISDN data link up. When this flag is set to No and a BRI trunk port receives an ISDN data link down message, then the BRI trunk ignores the ISDN data link down. Some Central Offices [local exchanges] will release its TEI after a B-channel trunk has been idle for some time. This causes an ISDN data link down message to be sent from the CO [local exchange] to the switch. In this case, if the switch processes the ISDN data link down message, it may drop the call on the other B-channel trunk on that port. The default value of this field is to ignore the ISDN data link down message.

3.130 To program this field, select **System\Cabinets** then double-click on the card (or right-click and select **Explore Card**).

Internet Protocol Resources Card

NOTE: Prior to v9.0, the 8.x firmware IPRC could be used for private networking **or** for IP devices, as determined by the configuration chosen in DB Studio. The v9.0 and later firmware IPRC supports **both** private networking and IP devices on the same card.

3.131 Version 9.0 or later system software introduces an improved IPRC that supports both IP devices and private networking. When version 8.1.x or later IPRC firmware is used with a v9.0 or later Axxess system, the IPRC allows an Axxess system to share IP endpoint and IP networking capabilities on the same IPRC. The physical components of the IPRC did not change for v9.0.0, however, because the v9.0.0 IPRC supports **both** IP devices and networking, the associated programming fields for the 32-device IPRC and the networking IPRC are combined for the v9.0.0 IPRC.

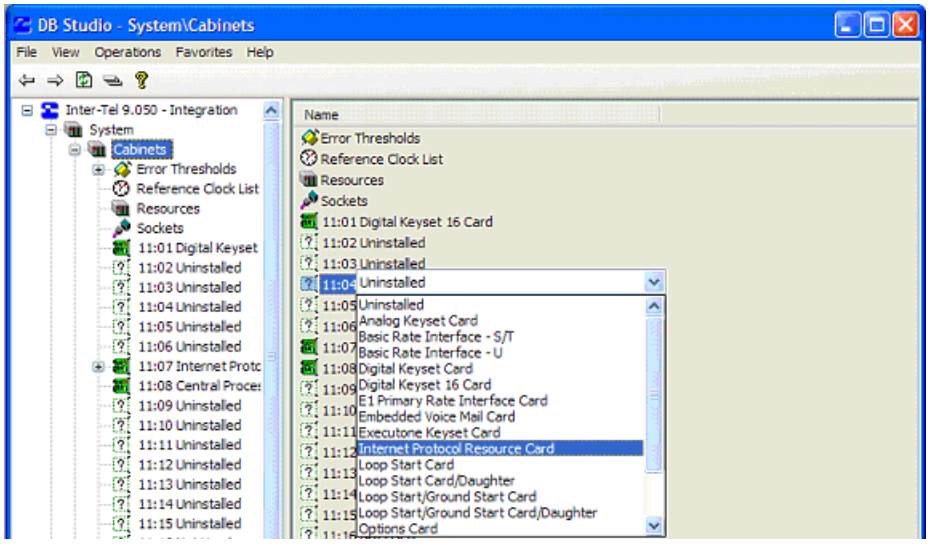
3.132 For more information about the v9.0.0 IPRC, see page 3-147. For troubleshooting information, see page 7-59.

NOTE: When using the networking feature in systems running software version 7.0 and later, to avoid one-way audio, do not use two IPRCs on separate networks.

3.133 Converting a pre-9.0.0 IPRC to a v9.0.0 IPRC: Part of the upgrade process from pre-9.0 system software to v9.0 or later is converting the database with DB Converter (see page 6-29). During the conversion process all existing IPRCs (Networking and 32-Device) are converted to the v9.0.0 IPRC configuration. When these cards have been converted, VoIP resources programmed on an IPRC Networking are converted to networking VoIP resources and VoIP resources programmed on an IPRC 32-Device are converted to endpoint VoIP resources.

NOTE: DB Converter provides a Help option with instructions for converting a database.

3.134 Equipping the v9.0.0 IPRC: To equip an IPRC, select **Internet Protocol Resource Card** as the card type for the corresponding card slot in the Systems/Cabinets folder as shown below.



3.135 When you equip a v9.0.0 IPRC, DB Studio automatically equips an IP connection corresponding to the IP port on the IPRC. You are prompted to enter an extension for the IP connection, and you must enter a non-dialable extension beginning with "P6."

NOTE: Inter-Tel strongly advises installers to create a universal numbering plan that includes IP connections for forward compatibility. One such plan is to use the convention of "PSnnc" where P is the pause digit, S is the number 6 or 7, nn is the node number, and c is a number between 0-9 identifying the specific IPRC to which the IP connection corresponds.

3.136 Programming the IPRC Fields: The v9.0.0 IPRC programming folders and fields are a combination of the networking IPRC and the 32-device IPRC fields used prior to v9.0.0. The following sections describe the folders and fields for the v9.0.0 IPRC.

Voice Resources

3.137 The Voice Resources field provides a link to the voice channels available for use by the IPRC. Double-click on Voice Resources to display the list of channels. Then enable the channels as described below. By default, all channels are set to *Out of Service* (disabled). Options for this field include:

- Out of Service
- Networking
- IP Endpoint

3.138 Any channels that are enabled are considered "devices" and affect the maximum number of devices the system supports. For example, if the system currently supports 512 devices and you program 32 voice resource channels, the remaining number of devices available is 480 (512 - 32). By default, this field is set to Out of Service.

NOTE: The maximum number of channels is 32. In remote mode, DB Programming displays all 32 channels (a red "X" indicates that the channel is offline).

ALSO: To use a v9.0.0 IPRC for private networking, at least one of the VoIP resources must be programmed for networking.

3.139 To enable a single channel: Select the current **Value** for the channel and use the drop-down list box to select Networking or IP Endpoint. Then press **ENTER** or click on another field to save the change.

3.140 To enable several channels at once: Highlight the channels you want to enable. (To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items.) Then right-click and select the Batch Change Voice Resources option. Select the value you want assigned to the channels and click **OK**.

Devices

3.141 The Devices field allows you to configure IP devices that terminate at the IPRC. For each device, this folder hides the hardware address, device type and the extension. IP devices that support v8.x firmware include the IP phones, Media Gateway Control Protocol (MGCP) gateways, SIP gateways, IP single-line adapters (IP SLAs), and multi-protocol endpoints (i.e., those that support SIP and ITP). This section describes how to program all of these devices. See the following pages for information about programming devices:

- IP endpoints and IP SLAs (see page 6-148)
- Model 8602 IP softphone (see page 6-149)
- MGCP gateways and endpoints (see page 6-151)
- MGCP endpoints (see page 6-152)
- SIP trunks (see page 6-152)

NOTE: In system software v9.0 and later, IP Devices display only the node and the board number (WW:XX) in the Address field. Earlier software versions displayed an address for IP endpoints that was similar to other station devices using the WW:XX:YY:ZZ convention (see page 2-37).

3.142 To program the circuits for devices, the MAC address of the IP phone or IP SLA is required. If desired, you can store the MAC addresses in a text (**`.txt`**) file and import the information. This file should have only one MAC address per line, and each address must be in the format *XX:XX:XX:XX:XX:XX*, where *XX* represents one or two characters. If desired, you can enter comments after the MAC address entry, but the address must be listed first and there must be at least one space between the address and your comments. Any other text that is not a valid MAC address is ignored. In other words, you can enter comments or other information in the file, but DB Programming only reads those lines that begin with a valid MAC address. See below for an example.

```
This file has the MAC addresses for node 59.
12:22:33:44:55:66 IP 2000
00:23:43:32:21:44 IPSLA
55:FF:33:11:22:33
34:43:54:42:2:2
00:22:33:34:44:43
J. Doe's Extension 00:11:22:33:44:55
```

This is ignored.

This MAC address is ignored.

3.143 To program the circuits for IP endpoints and IP SLAs:

1. Make sure the software license supports the required number of multi-protocol endpoints. Although DB Programming allows you to program more devices than the software license allows, only those devices that are licensed will come online.

   NOTE: The license regulates the number of multi-protocol endpoints only, such as the Model 8600, Model 8662, Model 8620, and Model 8690. It does not regulate the number of endpoints that use IP only, such as the Model 8660 and IP PhonePlus, or the number of MGCP gateways/endpoints.

2. Under System\Cabinets\Internet Protocol Resource Card, expand the Devices folder.
3. Highlight the circuits to program, right-click, and select **Batch Change Type**.
4. Select **IP Phone** or **IP SLA** and click **OK**.
5. Select the starting extension for the devices.
6. *EITHER,* Click a device in the list and enter the appropriate Media Access Control (MAC) address. To program a phantom extension, change the MAC address to FF:FF:FF:FF:FF:FF (the default is 00:10:36:00:FF:FF). The system then considers this device offline.

   OR, Click **Browse** to import the MAC addresses from a `.txt` file. (This invokes the standard Windows Open dialog box.) When you import the file, the MAC addresses are read from the top. This means that if there are six MAC addresses in the file, but there are eight devices, only the first six will be assigned an address. In addition, if there are eight MAC addresses in the file, but only six devices, only the first six addresses will be used.

   NOTE: All MAC addresses must be unique.

7. Click **OK** to accept the MAC address(es).
8. Program the MAC addresses for all of the listed devices.
9. Click **OK** to exit the screen.

3.144 When you click **OK**, the system verifies that the IPRC has enough voice resources for the number of devices you programmed. If the number of devices exceeds the number of resources, an error message appears. Although the system allows you to program more devices than available resources, some IP devices may not come online unless the number of voice channels is greater than or equal to the number of IP devices.

3.145 To program the circuits for the Model 8602 IP Softphone:

NOTE: A Category B software license (part number 840.0415) and a headset for audio are required to support the Model 8602. IP softphones cannot have attached devices, such as DSS/BLF units.

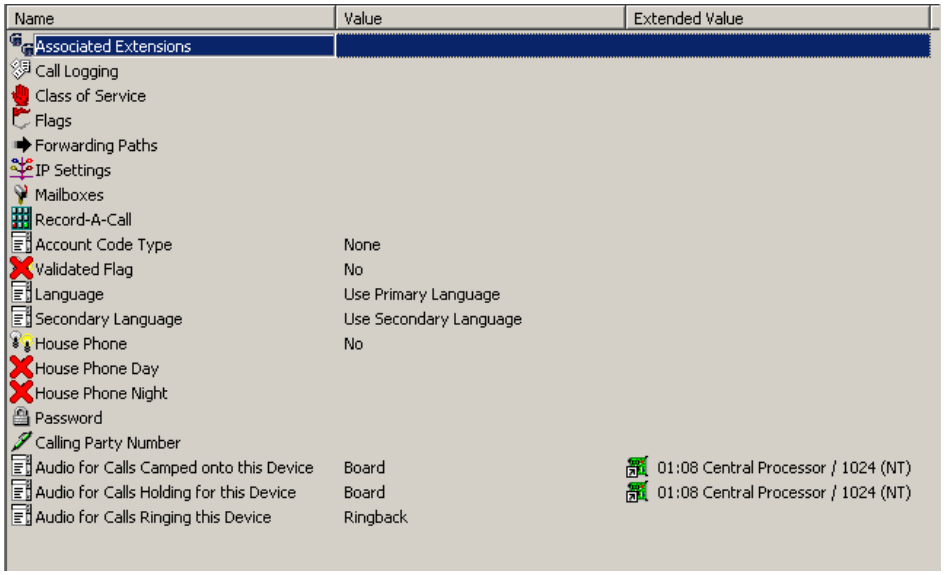
1. Ensure an Internet Protocol Resource Card (IPRC) is installed on the system.
2. Double-click the **Devices** under System\Cabinets\IPRC.
3. Select **IP Softphone** from the device option. Right-click on the screen and select **Create IP Softphone**. The Get IP Extension dialog box appears.
4. *EITHER,* Select a starting extension for the devices. Then click a device in the list and enter the appropriate device ID.

   OR, Click **Browse** to import the device IDs from a `.txt` file. (This opens the standard Windows Open dialog box.) When you import the file, the device IDs are read from the top. This means that if there are six device IDs in the file, but there are eight devices, only the first six will be assigned an address. In addition, if there are eight device IDs in the file, but only six devices, only the first six addresses will be used.

   NOTE: All device IDs must be unique. The user must choose a device ID that does not conflict with any other Inter-Tel IP endpoints configured into the system. Inter-Tel recommends that the administrator use a defined device ID scheme such as 86:02:00:xx:xx:xx. For troubleshooting purposes, Inter-Tel recommends using the extension number as the last digits of the Device ID.

5. Click **OK** to accept the extension and device IDs. The system then verifies that the IPRC has enough voice resources for the number of devices you programmed. If the number of devices exceeds the number of resources, an error message appears. Although the system allows you to program more devices than available resources, some IP devices may not come online unless the number of voice channels matches the number of IP devices. To ensure there are enough voice channels, make sure you have programmed all of the channels as VoIP. If all of the channels are programmed for VoIP, consider adding another IPRC to support additional devices you want to install.

6. Double-click the **IP Softphone** that you have just created. The device-specific fields are displayed. Program the following fields as described in this manual or use online help.



3.146 Consider the following considerations when programming a Model 8602 IP softphone:

- The use of a password is highly recommended for all Model 8602 configurations to prevent unauthorized users from accessing the system using a valid Model 8602 device ID.
- The Model 8602 only supports American and British English. Other languages are currently not supported.
- The Model 8602 must not be configured in peer-to-peer network groups because of its mobility. If the Model 8602 connects to the system outside of the normal network, then peer-to-peer calls will fail.
- The use of RFC-2833 DTMF is recommended. The DTMF encoding will follow the endpoint's speech encoding unless the DTMF encoding is set to RFC-2833. When the Model 8602 is configured to use G.729 speech encoding then G.729 DTMF will be used, and when the Model 8602 is configured to use G.711 speech encoding then G.711 DTMF will be used. However, if RFC-2833 DTMF encoding is used, then the endpoint will no longer "automatically" follow the speech encoding setting. The use of G.729 DTMF encodings will negatively impact the transmission of DTMF tones from the Model 8602 and will cause issues when trying to use Interactive Voice Response (IVR) type systems. As such, the use of RFC-2833 DTMF is recommended.
- The Model 8602 only supports three G.-711 or six G-729 frames per packet. The Model 8602 will default to these values if the frames per packet setting is set any higher. The Audio Frame/IP Packet setting can be programmed in System\IP-Related Information\Call Configuration.

3.147 To program the circuits for MGCP gateways and endpoints:

CAUTION

Inter-Tel recommends that you install an MGCP gateway and its associated endpoints on one node only. Installing an MGCP gateway and its endpoints on more than one node may lead to undesired results.

NOTE: You must program an MGCP gateway before you can program an MGCP endpoint.

1. Under System\Cabinets\Internet Protocol Resource Card, expand the Devices folder.
2. Highlight the circuits to program, right-click, and select **Batch Change Type**.
3. Select **MGCP Gateway and Endpoint** and click **OK**.
4. Select the starting extension for the gateway and endpoints.
5. Enter a name, up to 18 characters, that identifies the gateway on the network. If you are batch creating more than one gateway, each gateway is automatically assigned the base name followed by the port number. For example, if you are batch creating three gateways with a base name of MGCPGW, the gateway in the first port is assigned the name MGCPGW01; the second is assigned MGCPGW02; etc. Each gateway on the IPRC must have a unique name, and each name is case-sensitive.
6. Click the endpoint associated with the first port on the gateway and enter a unique endpoint name, up to 20 characters. Inter-Tel recommends that you use a naming convention to avoid resolution problems. By default, the endpoints are named ACgw*X*, where *X* is a number (0-3) that identifies the port. For example, the endpoint in the first port is ACgw0; the endpoint in the second port is ACgw1; etc. Each endpoint must have a unique name, and each name is case-sensitive.

   NOTE: The endpoint and gateway names entered in DB Programming must match those programmed in the gateway configuration. For details, refer to the *Inter-Tel Axxess/5000 Quintum SIP Gateway Installation Manual* (document part no. 835.3123) or the *Inter-Tel Axxess/5000 AudioCodes MP-114 SIP Gateway Installation Manual* (document part no. 835.3202).

7. Click **OK** to accept the endpoint name.
8. Assign up to three additional endpoints to the gateway and program the endpoint names.
9. Click **OK** to exit the screen.

   NOTE: If you unequip an MGCP gateway (i.e., change the circuit value to None), any MGCP endpoints assigned to that gateway are automatically unequipped.

3.148 To program the circuits for MGCP endpoints:

1. Under System\Cabinets\Internet Protocol Resource Card, expand the Devices folder.
2. Highlight the circuits to program, right-click, and select **Batch Change Type**.
3. Select **MGCP Endpoint** and click **OK**.

NOTE: If there is not an MGCP gateway currently programmed on the card, an error message appears. Program at least one circuit to be an MGCP Gateway and Endpoint before attempting to program additional endpoints.

4. Select the starting extension for the endpoints.
5. Click an endpoint in the list and enter an endpoint name, up to 20 characters. Inter-Tel recommends that you use a naming convention to avoid resolution problems. By default, the endpoints are named ACgw*X*, where *X* is a number (0-3) that identifies the port. For example, the endpoint in the first port is ACgw0; the endpoint in the second port is ACgw1; etc. (Hostnames are case-sensitive.)

NOTE: The endpoint names entered in DB Programming must match those programmed in the gateway configuration. For details, refer to the *Inter-Tel Axxess/5000 Quintum SIP Gateway Installation Manual* (document part no. 835.3123) or the *Inter-Tel Axxess/5000 AudioCodes MP-114 SIP Gateway Installation Manual* (document part no. 835.3202).

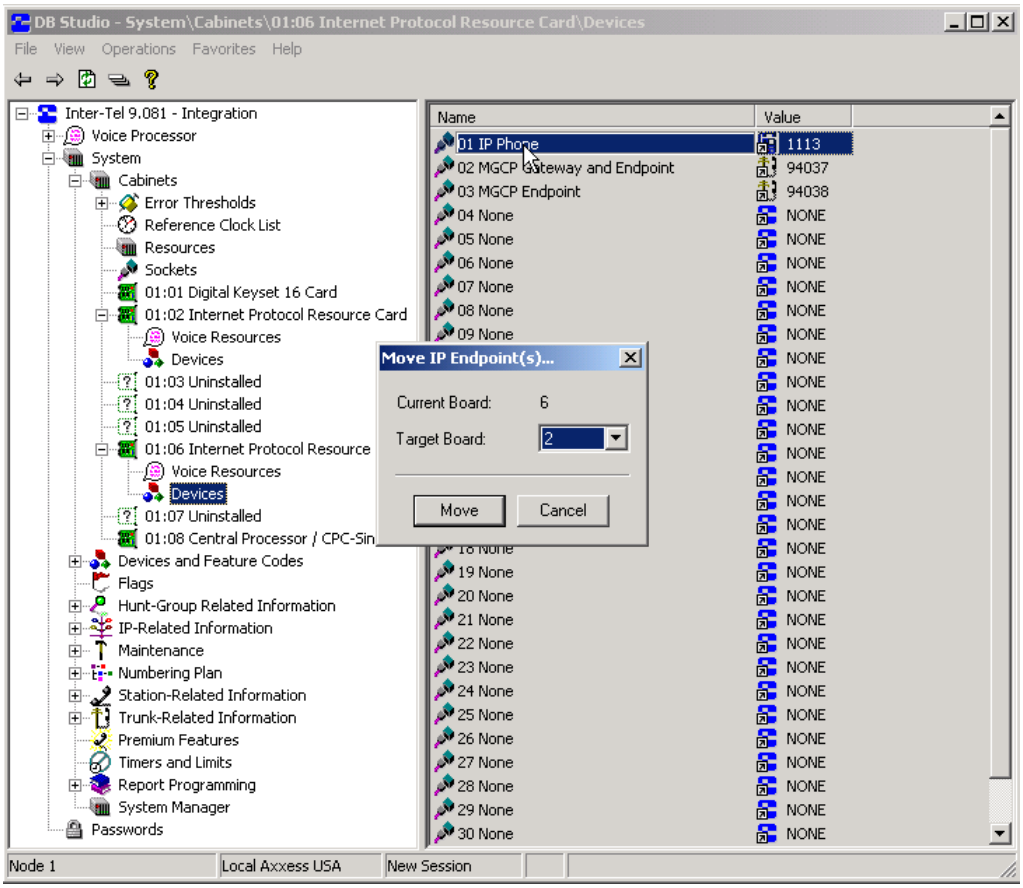
6. Click **OK** to accept the name.
7. Select the extension for the MGCP gateway that the endpoints will use and click **OK**.

3.149 To program the circuits for SIP trunks:

1. Under System\Cabinets\Internet Protocol Resource Card, expand the Devices folder.
2. Highlight the circuits to program, right-click, and select **Batch Change Type**.
3. Select **SIP Trunk** and click **OK**.
4. Select the starting extension for the SIP trunk and click **OK**.

NOTE: For complete information about installing and configuring a SIP gateway, refer to the *Inter-Tel Axxess/5000 Quintum SIP Gateway Installation Manual* (document part no. 835.3123) or the *Inter-Tel Axxess/5000 AudioCodes MP-114 SIP Gateway Installation Manual* (document part no. 835.3202).

3.150 Moving IP Endpoints: You can select IP Endpoints from one Devices folder and move them to another, across IPRC cards in local and remote programming modes. Moving an MGCP gateway and endpoints is allowed only when all of the MGCP Endpoints are selected along with the associated MGCP Gateway.



IP Connection

3.151 The IP Connection field provides a link to the IPRC's IP connection (see page 6-172 for details). You must assign a unique extension (P6000-P6999) to each IP connection within the private network. The available range is P6000 to P6999. Whenever the system creates an IP connection, the system broadcasts the new IP connection's extension to the other nodes as a database update, and the new IP connection's extension becomes an off-node device on the other nodes (provided the remote node does not block the database change such as when a DB Programming session is active on the remote node).

NOTE: Programming the IP connection does not automatically configure the IPRC for use by the network. You must manually program networking parameters locally as well as on each remote node (see page 6-176).

Description

3.152 This field provides a description for the IPRC. Enter the description, up to 20 characters long, in the text box. Then press **ENTER** or click on another field to accept the value.

Web and Telnet Password

3.153 The password is used to access the web server or telnet configuration tools. To change the password, right-click on the field and select **Edit Password**. Then enter the password, up to nine characters long, in the New Password and Confirm Password text boxes. Click **OK** to accept the new password.

NOTE: Because this password can be used to access the IPRC via the Web or a Telnet session, Inter-Tel recommends that you change the password.

Web and Telnet Server

3.154 If set to *yes*, allows you to use the Web/Telnet server configuration tool. You should use the Web/Telnet configuration tool only for troubleshooting purposes because the IPRC does not propagate changes made through the Web/Telnet configuration tool to call processing. In addition, call processing may override any changes made through the Web/Telnet configuration tool. By default, these fields are set to *Yes*.

NOTE: To upload boot code to the IPRC using the Board Utility, you must enable the Web Server option.

DTMF Configuration

3.155 You can program each IPRC to use a specific set of values that affect DTMF detection. This configuration allows you to make the system more or less sensitive to DTMF tones, which ultimately affects the amount of talkoff. (In general, the more sensitive the system is to DTMF tones, the more likely talkoff is to occur.)

NOTE: To configure DTMF detection, IPRC firmware version 8.2.0 or later and Call Processing v8.224 or later is required.

3.156 To change the DTMF detection settings, select the desired configuration in the DTMF Configuration field. Then press **ENTER** of click another field to save your changes. You have the following options:

- **Industry Standard**: Uses the industry standard settings for detecting DTMF tones. If selected, the system is the most sensitive to DTMF tones, resulting in the most talkoff.
- **Inter-Tel 1**: Provides less talkoff without drastically affecting DTMF detection. If selected, the system is less sensitive to DTMF tones than the Industry Standard option, resulting in less talkoff.
- **Inter-Tel 2**: Provides significantly less talkoff than Inter-Tel 1, but DTMF detection is moderately affected. This is the default setting.
- **Inter-Tel 3**: Provides the least amount of talkoff, but DTMF detection is significantly compromised. If selected, the system may not detect some dialed DTMF digits.
- **Custom**: Uses the DTMF settings that are programmed in on-line monitor (OLM) mode.

NOTICE

Inter-Tel recommends that you do not change the individual DTMF detection settings unless instructed to do so by Inter-Tel Technical Support or Engineering personnel. Inter-Tel is not responsible for significant talkoff or loss of DTMF detection if these settings are configured without approval from a Technical Support representative.

3.157 Exporting and Importing IPRCs: During an export or import, programming information is maintained across nodes for IPRCs that are used for networking only. For example, if Node A imports an IPRC from Node B and Node B's IPRC is not used for networking, Node A will remove the programming for Node B's IPRC. Similarly, if Node B exports the IPRC to Node A, Node A will remove the programming for the IPRC.

4. DEVICES AND FEATURE CODES

4.1 The extension numbers and feature codes are preset to carefully selected default values. Changing the codes or numbers can cause existing assignments to be erased. For example, if 300, 305, and 306 are assigned as feature codes and you attempt to assign 30 as another feature code, you would receive a warning message, because 30 makes up part of existing codes. The warning message allows you to clear all of the existing numbers (300, 305, and 306) and use the new one (30) by selecting **OK**, or to leave the existing numbers unchanged by selecting Cancel.

4.2 Each node in the network has its own extension and feature code programming. However, the network should have a universal numbering plan so that extension numbers on the various nodes do not overlap and do not conflict with feature codes. That is, when planning the extension numbers for each of the nodes in the network, set aside a block of extension numbers (for stations, hunt groups, Voice Processor applications, etc.) for each node.

4.3 Devices and Feature Codes programming include the following fields:

- DSS Consoles—U.S. Only (see page 6-157)
- Extension Lists (see page 6-158)
- Feature Codes (see page 6-159)
- Hunt Groups (see page 6-160)
- IP Connections (see page 6-172)
- IP Ports (see page 6-178)
- Modems (see page 6-180)
- Network Groups (see page 6-182)
- Node IP Connection Groups (see page 6-185)
- Nodes (see page 6-190)
- Page Ports (see page 6-193)
- Page Zones (see page 6-194)
- Phantom Devices (see page 6-196)
- Stations (see page 6-198)
- System Serial Ports (see page 6-227)
- CO Trunk Groups (see page 6-228)
- Node Trunk Groups (see page 6-243)
- Trunks (see page 6-244):

A. DSS CONSOLES—U.S. ONLY

4.4 This information is programmed for Executone DSS Consoles that are installed on the EKSCs. To view the available DSS Consoles, select System\Devices and Feature Codes and double-click on **DSS Consoles**.

4.5 **Description and Username**: At this level you can program the description and username for the DSS Consoles. You can enter a new description or username for the off-node device by selecting the appropriate field and then entering the new information in the text box. Then press **ENTER** or select another field to save the change. Descriptions can contain up to 20 characters and usernames can contain up to 10 characters. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

4.6 Double-click on a DSS Console to program the following:

- **Associated Keyset**: To determine the phone that will use this DSS Console, select the current value for Associated Keyset and enter the station's extension number. Or you can do the following:
  a. Right-click on the Associated Keyset value and select **Change Associated Keyset**. A window appears that asks you to select the type of device to include.
  b. Select **Next**. The list of phones appears. (You can view them in a list by selecting **List** or view details by selecting **Details**.)
  c. Highlight the phone you want to use and then select **Finish**. Your selection appears in the Associated Keyset field.
- **DSS Map Group**: To determine the DSS keymap that will be used by this DSS Console, select the current value and enter the new map number. Or you can do the following:
  a. Right-click on the DSS Map Group value and select **Change DSS Map Group**. A window appears that asks you to select the type of device to include.
  b. Select **Next**. The list of maps appears. (You can view them in a list by selecting **List** or view details by selecting **Details**.)
  c. Highlight the map you want to use and then select **Finish**. Your selection appears in the DSS Map Group field.

B. EXTENSION LISTS

4.7 An extension list is a group of intercom extensions or trunk group extensions. These lists are used when programming features that use common lists. For example, a group of stations could be assigned to the same paging zone and have ring-in for the same trunk groups. With an extension list, you would have to enter only one list number instead of entering all of the extensions. Extension lists of stations can also be included in hunt groups.

4.8 There are several types of possible extension lists:

- **ACD Agent ID**: Includes only ACD hunt group Agent IDs. If no Agent IDs have been created (as described on page 6-263), this option cannot be used.
- **Keyset**: Includes only phone extension numbers. The *Auto: All Keysets (PP052)* extension list is provided by default and cannot be deleted. This list contains all of the phones programmed and is automatically updated when phones are added/deleted.
- **Single-Line**: Includes only single-line set and MDPM extension numbers.
- **Station**: Can include phones, MDPMs, and/or single-line extension numbers. The *Auto: All Stations (PP051)* extension list is provided by default and cannot be deleted. This list contains all of the stations programmed and is automatically updated when stations are added/deleted.
- **CO Trunk Group**: Includes only trunk group access numbers.
- **Hunt Group Members**: Includes only stations that are members of hunt groups.

4.9 To view the available lists, double-click on **Extension Lists**. To program or change a name for an extension list, double-click on the desired list. Then do the following:

NOTE: You cannot change or delete the *Auto: All Keysets* or *Auto: All Stations* extension lists.

- **To create an extension list:** Right-click on the type of list you want to create and select **Create Extension List**. The new list automatically appears in the list for that extension list type.
- **To program a description for the extension list:** Select **Description** and enter the name, up to 20 characters, in the text box. When finished, press **ENTER** or select another field to save the change.
- **To program a list:** Double-click on the list. Then right-click anywhere in the list's window and select **Add To List** to add devices to the list. A window appears that asks you to select the type(s) of device to include. Select the device type(s) and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.) Highlight the devices you want to add to the list and then select **Finish**. Your selection appears in the extension list window.
- **To delete an extension list:** Highlight the extension list, and right-click. Then select **Delete**. The list is automatically removed.

NOTE: When using an extension list for ring-in or hunt groups, do not exceed 30 stations per list. The system can send ring signal to up to 30 stations.

C. FEATURE CODES

4.10 The feature codes are preset to carefully selected default values. Changing the codes can cause existing assignments to be erased. For example, if 300, 305, and 306 are assigned as feature codes and you attempt to assign 30 as another feature code, you would receive a warning message, because 30 makes up part of existing codes. The warning message allows you to change the existing numbers (300, 305, and 306) individually or to leave the existing numbers unchanged by selecting Cancel.

NOTE: If you change feature codes, they will not be automatically updated in the feature code directory used by Desktop Interface devices that are installed on the system. Those directories will have to be updated manually. However, any devices installed after the change will have the new feature code information. Unlike feature codes, extension numbers are updated in the intercom directories when the devices are activated and when updates are requested.

4.11 **To program individual feature codes:** Select its current Feature Code and enter the new number in the text box (or scroll to an available number). Feature code descriptions cannot be changed.

4.12 **To program several feature codes at once:** Highlight the codes you want to change. Then right-click and select **Batch Extension Change**. When the Get Extension screen appears, select the number you want to assign to the first highlighted feature code (the other highlighted codes will be numbered consecutively after this number). When you select **OK**, the feature codes are automatically renumbered and re-sorted in the list.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items. (You may have to continue to hold **SHIFT** or **CTRL** while right-clicking to display the option without changing the highlighted selections.)

4.13 For a complete list of the feature code and their descriptions, see page 5-14.

D. HUNT GROUPS

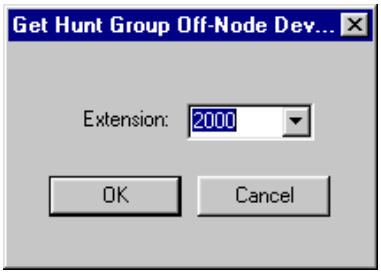
4.14 See page 5-32 in FEATURES for an explanation of hunt groups.

Node-Spanning Hunt Groups

4.15 There is a software license called Remote Automatic Call Distribution Hunt Groups. It allows ACD hunt groups to span nodes. Node-spanning ACD hunt groups can have either members or ACD Agent IDs. Agent IDs are considered global throughout all nodes in which the Agent exists. This means that if you create Agent ID 100 on one node, you must also create Agent ID 100 on all other nodes that have a station that may want to login to the hunt group using that ID.

Remote (Off-Node) Hunt Groups

4.16 When you double-click on a remote node, a list of the existing off-node hunt groups for that node, if any, appears. You can create or delete off-node hunt groups and/or enter descriptions and usernames. To create an off-node hunt group right-click anywhere in the right half of the window and select **Create Off-Node Hunt Group**. The following window appears. Enter the desired extension number in the text box or scroll to the desired number. Then select **OK** to continue. (Or select **Cancel** to exit without creating the device.) The new hunt group appears in the list without a description or username.



Local Hunt Groups

4.17 When you double-click on the Local node, a list of the existing local hunt groups, if any, appears. You can create or delete hunt groups and/or enter descriptions and usernames. You can then program the hunt group, as described below. To create a new hunt group, right-click anywhere in the right half of the window and select **Create Hunt Group**. The following window appears. Enter the desired extension number in the test box or scroll to the desired number. Then select **OK** to continue. (Or select **Cancel** to exit without creating the hunt group.) The new hunt group appears in the list without a description or username.

Deleting Hunt Groups

4.18 To delete a hunt group, highlight it and right-click. Then select **Delete**. The hunt group is automatically removed from the list.

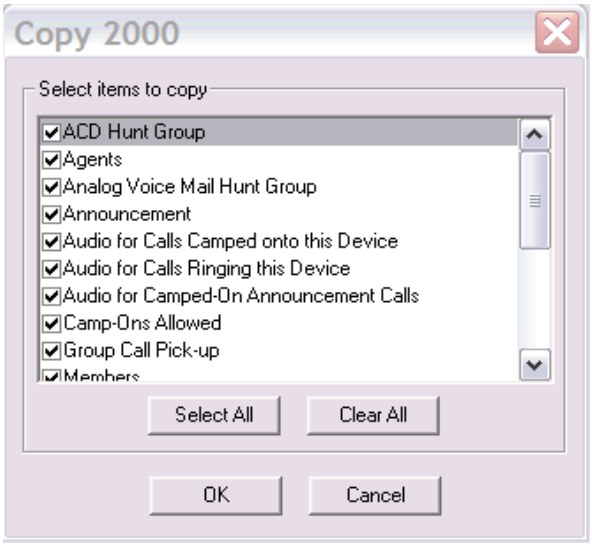
Programming Hunt Group Extensions, Descriptions, and Usernames

4.19 Extension: To change the hunt group's extension, select its current Extension and enter the new number in the text box (or scroll to an available number).

4.20 Batch Extension Change: To change several hunt group extension numbers at once, highlight the hunt groups you want to change. Then right-click and select **Batch Extension Change**. When the Get Extension screen appears, select the number you want to assign to the first highlighted hunt group (the other highlighted hunt groups will be numbered consecutively after this number). When you select **OK**, the hunt groups are automatically renumbered and re-sorted in the list.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items. (You may have to continue to hold **SHIFT** or **CTRL** while right-clicking to display the option without changing the highlighted selections.)

4.21 Copy Hunt Group: To copy a hunt group, right-click on the hunt group extension and then select **Copy**. To paste the programming information into another hunt group, right-click the hunt group where you want the information pasted. Then select **Paste**. The Copy dialog box (shown below) allows you to select the attributes you want to copy. Select the desired attributes and click **OK**.



4.22 Descriptions and Usernames: You can enter a new description or username for the hunt groups by selecting the appropriate field and then entering the new information in the text box. Then press **ENTER** or select another field to save the change. Descriptions can contain up to 20 characters and hunt group usernames can contain up to 10 characters. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

Programming Local Hunt Groups

4.23 When you select an individual Local hunt group, the following options appear. Use the instructions in the following pages to program the hunt group information.

- Agents (see below)
- Members (see page 6-163)
- Supervisors (see page 6-164)
- Timers (see page 6-165)
- ACD Agent No Answer - DND Message Additional Text (see page 6-166)
- ACD Agent No Answer - DND Message Number (see page 6-166)
- ACD Hunt Group flag (see page 6-167)
- Analog Voice Mail Hunt Group (see page 6-167)
- Announcement and Overflow Stations (see page 6-167)
- Audio for Calls Camped onto this Device (see page 6-168)
- Audio for Calls Ringing this Device (see page 6-168)
- Audio for Camped-On Announcement Calls (see page 6-168)
- Camp-Ons Allowed (see page 6-168)
- Group Call Pick-Up (see page 6-169)
- Priority Level (see page 6-169)
- Recall Destination Station (see page 6-169)
- Restart ACD Idle Time Upon Login (see page 6-169)
- Return ACD Calls to Hunt Group (see page 6-170)
- Search Type (see page 6-170)
- Send Camp-On Notifications to Members in DND (see page 6-170)
- Use ACD Agent IDs (see page 6-171)

Agents

4.24 If the hunt group is programmed to use ACD Agent IDs, each agent is assigned an Agent ID number which the agent enters during the login procedure. The hunt group calls are routed to logged in agents, according to their Agent ID number instead of their station extension. Because the Agent ID is not associated with any station extension, the agent can use any station in the system to log in and does not have to use the same station every time. There can be up to 1000 Agent IDs in the system. ACD Agent IDs can be included in Extension Lists which allows several ACD Agents to receive a call at once. To add or delete Agent IDs, double-click on **Agent IDs** to see the current list of IDs, if any. Then do the following:

- **To add Agent IDs to the hunt group:** Note that this is an *ordered list*. Place the agents in the list in the order you want them to be accessed when the hunt group receives a call.

  a. *To add to the bottom of the list:* Do not highlight any existing agents and right-click anywhere in the right side of the window. An option box appears.

     To add to the list above an existing agent: Highlight the agent and then right-click. An option box appears.

b. Select **Add To Agents List**. A window appears that asks you to select the type of device to include.

c. Select **ACD Agent ID** and/or **ACD Agent ID Extension List** and then select **Next**. The list of agents/extension lists appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

NOTE: To highlight a series of IDs, hold down **SHIFT** while selecting the first and last IDs you want. To highlight two or more IDs that are not consecutive, hold down **CTRL** while selecting the desired IDs.

d. Highlight the appropriate items and then select **Add Items**. When you have added all the Agent IDs, select **Finish**. Your selections appear in the list.

NOTE: ACD Agent IDs are programmed in Hunt Group-Related Information. If you want to see the programming for a specific ACD Agent ID, double-click on its number to jump to the Hunt Group-Related programming area.

- **To move an agent to another location in the list:** Drag and drop the agent to the new position. Or, highlight the agent to move and press **CTRL** + the up/down arrow to move the agent up or down in the list.
- **To delete Agent IDs:** Highlight the agent ID(s) and press **DELETE** on your keyboard, or right-click and select **Remove Selected Items**.

Members

4.25 Prepare a list of the stations/extension lists to be included in each of the hunt groups. If desired, a station or extension list can appear more than once in a hunt group list or can be in more than one hunt group. If an extension list is included in an ACD hunt group set for Longest Idle or Balanced Call Count distribution, it will treat each station in the extension list as a separate agent; it will not ring all of the stations on the list at once. If the hunt group is set for linear or distributed order, a call will ring at all stations on an extension list at once when the call reaches that point in the hunt group list. Therefore, to create an "all ring" type of hunt group, you can program the hunt group as either linear or distributed and then assign an extension list as the only hunt group member. To add or delete hunt group members, double-click on **Members** to see the current list. Then do the following.

NOTE: When programming hunt group **Members** in the Local\<*hunt group no.*> folder, be aware that adding a large extension list can result in slowing down system performance. Adding an Extension List that contains more than 60 members may cause a system slowdown because when the list is called, ALL members of the list are called at the same time.

- **To add members to the hunt group:** Note that this is an *ordered list*. Place the devices in the list in the order you want them to be accessed when the hunt group receives a call.

  a. *To add to the bottom of the list:* Do not highlight any existing members and right-click anywhere in the right side of the window. An option box appears.

  To add to the list above an existing member: Highlight the member and then right-click. An option box appears.

  b. Select **Add To Members List**. A window appears that asks you to select the type of device to include.

c. Select the desired device type(s) and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

d. Highlight the appropriate items and then select **Add Items**. When you have added all the members, select **Finish**. Your selections appear in the list.

NOTE: If you want to see the programming for a specific member, double-click on its number to jump to its programming area.

- **To move a member to another location in the list:** Drag and drop the member to the new position. Or, highlight the member to move and press **CTRL** + the up/down arrow to move the member up or down in the list.
- **To delete members:** To delete one or more member from the list, highlight them and press **DELETE** on your keyboard, or right-click and select **Remove Selected Items**.

NOTE: A system reset is no longer required for UCD or ACD hunt groups after the hunt group member list changes.

Supervisors

4.26 (*Hunt group supervisors cannot be off-node devices.)* If the UCD Hunt Groups premium feature is licensed (see page 6-97), hunt groups can have one or more stations assigned as the hunt group supervisor(s). (An extension list can be included in the list of supervisors.) If desired, a station can be assigned as the supervisor for more than one hunt group. If not using Agent IDs, ACD hunt group supervisors with display endpoints will receive visual camp-on displays if they are programmed as members of the hunt group and they have the ACD Agent Logout feature enabled. If a Hunt Group is using ACD Agent IDs, the supervisor must be logged on to the group to receive camp-on indications. If not using ACD Agent IDs, you can add the supervisor as a hunt group member, set the Hunt Group flag for the supervisor's station to "Remove," and the supervisor will still receive hunt group camp-on indications. To add or delete supervisors, double-click on **Supervisors** to see the current list of supervisors, if any. Then do the following:

- **To add supervisors to the hunt group:**

  a. Right-click anywhere in the right side of the window. An option box appears.

  b. Select **Add To Supervisors List**. A window appears that asks you to select the type of device to include.

  c. Select the station and/or extension list type(s) and then select **Next**. The list of stations/extension lists appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

d. Highlight the appropriate items and then select **Add Items**. When you have added all the supervisors, select **Finish**. Your selections appear in the list.

NOTE: If you want to see the programming for a specific station assigned as a supervisor, double-click on its extension number to jump to the Stations programming area.

- **To delete supervisors:** To delete one or more supervisor from the list, highlight them and press **DELETE** on your keyboard, or right-click and select **Remove Selected Items**.

Timers

4.27 Several timers are programmed on a hunt group-by-hunt group basis. To program a timer, select its current Value and enter the new value in the text box. Then press **ENTER** or select another field to save the change. The timers are as follows:

- **No Answer Advance**: Determines the amount of time a call will ring at a hunt group station (unanswered) before advancing to the next station on the list. It is started when the call is received at the hunt group. The default value is 18 seconds. It can be programmed for 1-255 seconds.
- **Announcement**: *(UCD Hunt Groups Only)* This determines the amount of time a call will remain unanswered before it is picked up by the hunt group's announcement station. It is started when the call is received at the hunt group. The default value is 18 seconds. It can be programmed for 1-255 seconds.
- **Overflow**: *(UCD Hunt Groups Only)* This determines the amount of time a call will circulate through the hunt group (unanswered) before being picked up by the hunt group's overflow station. This timer is started when the Announcement timer expires (or, if there is no announcement station, when the call is received by the hunt group) and it is restarted each time the call leaves the overflow station. The default value is 72 seconds. It can be programmed for 1-255 seconds.
- **Recall**: This determines the amount of time a call will circulate through the hunt group (unanswered) before being sent to the hunt group's recall destination station. The timer is started when the call is received by the hunt group. The default value is 180 seconds (3 minutes). It can be programmed for 1-65535 seconds.
- **Wrap-Up**: *(ACD Hunt Groups Only)* Each time an agent ends an ACD hunt group call, an ACD Wrap-Up Duration timer is started. Until that timer expires, the agent will not receive another call through any ACD hunt group. (However, the agent can receive other non-ACD hunt group calls, direct ring-in calls, and transfers.) The default value is 15 seconds. The range is 1 to 65535 seconds.
- **Average Connect Time Per Call**: *(UCD Hunt Groups Only)* An application announcement or overflow station message can be programmed to include the caller's queue position and/or estimated wait time. The estimated wait time is based on the Average Connect Time Per Call multiplied by the number of calls ahead of the caller in the queue, divided by the number of available hunt group members (*avg. connect time per call* x *no. of waiting calls* ÷ *available members*). The default value for Average Connect Time Per Call is 60 seconds. The range is 1-10,000 seconds.

ACD Agent No Answer - DND Message Additional Text

4.28 This flag allows OAI applications to function better with the Hunt Groups. The DND Message Additional Text flag is disabled and displays a red "X" if the "ACD Hunt Group" flag is set to "No." Enter additional text to be added to the No Answer DND message chosen for the field below. The limit for the size of entry will mimic that of the DND messages. This field will have a red "X" to indicate it is disabled, if the "DND Message for Agent No Answer" field has a value of 0.

ACD Agent No Answer - DND Message Number

4.29 This flag, added in software v9.1, allows OAI applications to function better with the Hunt Groups. Previously, when a call reached the end of the agent list in an ACD Hunt Group, the call rang at that agent until it was answered or another agent became available. In v9.1 or later, if the last agent in a Hunt Group does not answer an incoming call, the agent is automatically placed in DND and the call returns to the Hunt Group where it will be answered or camp on.

4.30 This flag and the "ACD Agent No Answer - DND Additional Text" flag allow you to determine which DND message will be used (selected by number) and what the bottom line of the DND message text will say. These flags are disabled and display a red "X" if the "ACD Hunt Group" flag is set to "No." The DND Message Number flag is a link to Endpoint-Related Information\Messages where you can select the language that you want to use and see which DND message you want to use.

4.31 Users are allowed to choose 0 – 20, which corresponds with the DND messages. A value of zero implies not allowing an agent to go into DND. It is set to zero by default. To determine whether trunk calls will be allowed to go to DND for the selected hunt group, program the "Return ACD Calls to Hunt Group" flag (see page 6-170).

4.32 To program the Hunt Group DND Trunk support options:

1. Click the Value for the ACD Agent No Answer – DND Message Number field. You are prompted to see the Messages folder.
2. Do either of the following:
   - Click **Yes** to see the Messages folder and select the appropriate language. Click **DND** to see a list of DND messages from which you can determine the number to select. Then click the Back button on the toolbar several times to return to the Hunt Group folder to enter the value.
   - Click **No** to ignore the prompt and use the value you selected.
3. Click the Value column for ACD Agent No Answer - DND Additional Text to enter any additional text you want to accompany the DND message you chose in step 2.

ACD Hunt Group

4.33 *(This option is not recommended for Analog Voice Mail hunt groups.)* If you want the hunt group to use the ACD features, enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check, select the check box again.) The ACD Hunt Group software license is required (see page 6-97). Automatic Call Distribution (ACD) hunt groups can use all of the standard and UCD hunt group features (if enabled), plus these features:

- ACD hunt groups can be programmed to distribute calls to equalize call time or call count among the available members.
- If the System OAI Events premium feature is licensed, ACD can also provide call information records that can be processed by an external device connected to a system serial port.
- ACD hunt groups can use of Agent ID numbers in place of station extensions in the hunt group list. (See Agent ID information on page 6-263.)

Analog Voice Mail Hunt Group

4.34 If the Analog Voice Processor Hunt Group premium feature is licensed (see page 6-97), the hunt group can be composed of station ports which are connected to an analog voice mail device. Enabling this option has the effect of passing along station identification when a call reaches the hunt group as a result of forwards and transfers. The purpose of this flag is to provide compatibility between the system and analog voice mail units. (The Voice Processing Unit is a digital system and does not require this flag.) If the hunt group is composed of station ports which are connected to an analog (non-Inter-Tel) voice mail device, enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check, select the check box again.)

NOTE: When an analog voice mail unit is connected to the System, the voice mail unit should use the Silent Message feature code (367) instead of the Message feature code (365). (This option is not recommended for ACD hunt groups.)

Announcement and Overflow Stations

4.35 (*Announcement and overflow stations can be assigned to off-node stations or Voice Processor applications, if they are programmed as off-node devices. See page 6-198).* If the UCD Hunt Groups premium feature is licensed (see page 6-97), hunt groups can have an announcement station and/or an overflow station. If a call to the hunt group is not answered before the Announcement timer expires, the call is picked up by the announcement station. If the call remains unanswered when the hunt group Overflow timer expires, the call is picked up by the overflow station. Announcement and overflow stations should be playback device stations (these can be Auto Attendant, Voice Mail, or Call Routing Announcement applications). Do not include these stations in the hunt group distribution list. To program an Announcement or Overflow station for the hunt group, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the existing value for the Announcement or Overflow station. An option box appears.
2. Select **Change**. A window appears that asks you to select the type of device to include.
3. Select desired device type and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the device you want as the Announcement or Overflow station and then select **Finish**. Your selection appears in the appropriate field.

Audio for Calls Camped onto this Device

4.36 The Audio for Calls Camped onto this Device field defines the audio that a caller hears when camped-on to this hunt group. To program this field, use the drop-down list to select the type of audio and press **ENTER**. Your options are Silence, Ticktone, Ringback, or Board. If you select **Board**, specify the music source in the Extended Value field. The default is *Board*, and the default extended value is *01:08 Central Processor.*

Audio for Calls Ringing this Device

4.37 The Audio for Calls Ringing this Device field defines the audio that a caller hears when ringing this hunt group. To program this field, use the drop-down list to select the type of audio and press **ENTER**. Your options are Silence, Ticktone, Ringback, or Board. If you select **Board**, specify the music source in the Extended Value field. The default is *Ringback.*

Audio for Camped-On Announcement Calls

4.38 This flag determines the functionality for camped-on hunt group calls that are listening to an Announcement Station's message when they transition from camped-on to ringing. If this flag is set for ringback, the user will switch from hearing the Announcement Station to hearing ringback when the call moves from camped-on to ringing. This is the current functionality. If this flag is set for Announcement Station, the user will continue to hear the Announcement Station when the call moves from camped-on to ringing. When the Announcement Station hangs up (i.e., the entire message is played), the user will start to hear ringback. If, however, an agent answers the call before the Announcement Station hangs up, the call is connected, and the rest of the message is not played.

NOTE: The user may hear Music-On-Hold instead of ringback depending on how the system is programmed.

Camp-Ons Allowed

4.39 If this flag is enabled, callers will be allowed to camp on to the hunt group when all members are busy. If the flag is disabled, the callers will hear busy tones when no members are available. In the default state, camp-ons are enabled. To allow callers to camp on to the hunt group, enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check, select the check box again.)

Group Call Pick-Up

4.40 The Group Call Pick-up feature allows users to reverse transfer a call that is ringing in to a hunt group or one of its stations using the hunt group's pilot number. If Group Call Pick-up is enabled for a hunt group, users can enter the Reverse Transfer feature code (4) and dial the hunt group's pilot number to pick up a call that is ringing in to any station's extension within that hunt group or to the hunt group's pilot number. (If the feature is disabled, reverse transfers using the hunt group's pilot number will reverse transfer only calls ringing to the hunt group's pilot number, not calls ringing at the individual stations within the hunt group.)

Priority Level (0-75)

4.41 If the UCD Hunt Groups premium feature is licensed (see page 6-97), hunt groups can have assigned priority levels 0-75. Some stations may be members of more than one UCD hunt group. For this reason, hunt groups are assigned a "priority level." The priority level determines which hunt group's calls should be received first when calls ring in and/or camp on to several hunt groups at once. If a station is a member of multiple hunt groups that have the same priority level, calls received by those hunt groups will be queued in the order they were received by the phone system. Select current Value for the Priority Level and enter the correct level or scroll to the desired priority level for this hunt group. Higher numbers have higher priority. The range is 0-75.

Recall Destination Station

4.42 The recall destination can be assigned to an off-node station or application, if it is programmed as an off-node device (see page 6-198). Any hunt group can have a recall destination station, regardless of whether UCD is enabled. If a call remains unanswered when the Recall timer expires, it will be sent to the recall destination station. The recall destination station should be a phone or single-line station. Do not include this station in the hunt group distribution list. Use one of the following methods to program the Recall Destination Station:

Method A

- Select the current Value and enter the new value in the text box.
- Press **ENTER**. A screen appears showing you what is associated with the number you entered.
- Select **OK**. The new number appears in the field.

Method B

- Right-click on the existing value for the Recall station. An option box appears.
- Select **Change Recall**. A window appears that asks you to select the type of device to include.
- Select desired device type and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
- Highlight the device you want as the Recall station and then select **Finish**. Your selection appears in the Recall field.

Restart ACD Idle Time Upon Login

4.43 This flag determines where an agent is placed in a longest idle queue when the agent logs back in to a hunt group. When *enabled*, the agent's idle time is reset to zero whenever the agent logs in (i.e., that agent will be least likely to receive the next distributed call). If *disabled*, the agent's idle time includes the time the agent was logged out of the hunt group (i.e., that agent will be most likely to receive the next distributed call). By default, this flag is *disabled*.

Return ACD Calls to Hunt Group

4.44 This flag was added in software v9.1 to each hunt group to allow OAI applications to function better with the Hunt Groups. Prior to v9.1, Hunt Group members could not reject calls by entering DND. However, they are now allowed to do so in a v9.1 or later system. This flag was added so that Hunt Group members who are in DND can still receive Camp-On notifications. This flag determines whether trunk calls will be allowed to go to DND for the selected hunt group. It is set to *No* by default. This flag is disabled and display a red "X" if the ACD Hunt Group flag is set to "No."

Search Type

4.45 Select the current Value for the Search Type and scroll to the search type for this hunt group. Available Search Types depend on the type of hunt group being programmed:

- **For Standard and UCD Hunt Groups**: Determine whether the calls are sent to the stations in linear or distributed order. Linear order means that the call is sent to the first station or extension list on the list and moves down the list until it reaches an available station. With distributed order, the call is sent to the station that appears on the list after the last station or extension list to receive a call (even if the call was not answered).
- **For ACD Hunt Groups**: ACD Hunt Group calls can circulate in linear or distributed order (as described above) or using one of the ACD distribution methods: Longest Idle or Balance Call Count order. Longest idle means that an incoming call is sent to the station that has not been involved in a call to this hunt group for the longest period of time. Balance Call Count means that, to balance the call load, each incoming call is sent to the station that has received the fewest calls through this hunt group. (It does not count calls that were received through other hunt groups, direct ring-in, or transfer.)

Send Camp-On Notifications to Members in DND

4.46 This flag indicates whether camp-on burst tones are sent to hunt group members that are in DND or that are logged out. When sent to display phones, the display also indicates *N* CALLS WAITING FOR <*HUNT GROUP*>, where *N* is the number of calls that are currently camped on to the hunt group. This allows hunt group members to see the hunt group queue in real-time. By default, this flag is disabled.

4.47 When this flag is enabled:

- If the hunt group is a UCD hunt group, the members receive one of the following camp-on notifications:
  - — Logged into hunt group and in DND: Tone -YES
  - — Logged into hunt group and not in DND: Tone - YES
  - — Logged out of hunt group and in DND: Tone - YES
  - — Logged out of hunt group and not in DND: Tone - YES
- If the hunt group in an ACD hunt group but does *not* use Agent IDs, the members receive one of the following camp-on notifications:
  - — Logged into hunt group and in DND: Tone -YES
  - — Logged into hunt group and not in DND: Tone - YES
  - — Logged out of hunt group and in DND: Tone - YES
  - — Logged out of hunt group and not in DND: Tone - YES

- If the hunt group is an ACD hunt group and uses ACD Agent IDs:
  - The user will get camp-on notifications if the Agent ID is **logged in** to the Hunt Group *and* in DND.
  - The user will *not* get camp-ons if the Agent ID is **logged out** of the hunt group.

4.48 When this flag is disabled:

- If the hunt group is a UCD hunt group, the members receive one of the following camp-on notifications:
  - Logged out of hunt group and not in DND: Tone - YES
  - Logged out of hunt group and in DND: Tone - NO
  - Logged into hunt group and not in DND: Tone - YES
  - Logged into hunt group and in DND: Tone - NO
- If the hunt group in an ACD hunt group but does *not* use Agent IDs, the members receive one of the following camp-on notifications:
  - Logged out of hunt group and not in DND: Tone - YES
  - Logged out of hunt group and in DND: Tone - NO
  - Logged into hunt group and not in DND: Tone - YES
  - Logged into hunt group and in DND: Tone - NO
- If the hunt group is an ACD hunt group and uses ACD Agent IDs:
  - The user will *not* get camp-on notifications if the Agent ID is **logged in** to the Hunt Group AND in DND.
  - The user will *not* get camp-ons if the Agent ID is **logged out** of the hunt group.

Use ACD Agent IDs

4.49 If the ACD Hunt Group Option is checked, you can choose to route calls according to ACD Agent ID numbers instead of station extensions. Enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check, select the check box again.)

Hunt Groups Associated Devices

4.50 The Associated Devices and References feature (see page 5-31) allows you to see the associated devices for a station, mailbox, and hunt group. This feature also allows you to query various groups in the database to locate the associated references to the extension.

4.51 To search for devices associated with the hunt group, right-click the hunt group and select **Associated Devices and References**. A dialog box, containing associated device(s) and references for the mailbox appears.

E. IP CONNECTIONS

NOTICE

Certain ports that the phone system uses for IP call control and audio may be reserved for other protocols, such as SIP (which uses 5060 by default) and may be blocked by NATs, firewalls, and/or routers. If these ports are blocked, you may experience audio or connection problems. If this occurs, change the port numbers in DB Programming so that there are no conflicts.

4.52 You can program information about each IP connection in the private network. When you access the IP Connections folder, the right side of the screen displays the local and remote nodes.

4.53 When you double-click on a node, the extension, description, and username information appears. For local nodes, the circuit address is also displayed.

4.54 **Description and Username**: All IP connections should have a description and a username. The description that appears in all IP connection lists in the database can be up to 20 characters long. The username, that will appear on display phones, can have up to 10 characters. To program the names, select the desired text box and type the entry. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

4.55 To display the parameters for a local or off-node IP connection, double-click on the connection's extension. You can then program the appropriate fields as described in the following sections.

Local Node

4.56 For the local node, the following fields are displayed:

- Backplane Call Configuration (see page 6-173)
- Static IP Address (see page 6-173)
- Static Subnet Mask (see page 6-173)
- Static Gateway (see page 6-174)
- WINS Server (see page 6-174)
- Hostname (see page 6-174)
- Audio RTP Type of Service and Data Type of Service (see page 6-174)
- Audio Stream Receive Port (see page 6-174)
- IP Terminal TCP Call Control Port (see page 6-174)
- IP Terminal General Purpose UDP Port (see page 6-175)
- MGCP Receive Port (see page 6-175)
- TCP Call Control Port (see page 6-175)
- Web Listening Port (see page 6-175)

Backplane Call Configuration

4.57 The Backplane Call Configuration defines the backplane call configuration for the IP connection. To configure this, double-click or expand this field and program the following:

- **Backplane Transmit Signal Attenuation Level**: Defines the amount of audio volume that the system removes from the audio stream. This attenuation is applied to the audio that the IPRC receives from the IP network as the IPRC injects the audio onto the PCM highway backplane of the system. Use the scroll box to make your selection. Then press **ENTER** or click on another field to save your change. The default is *0 db*, and the range is *0* to *15 db*.

  NOTE: The larger this value, the lower the volume of the signal that you hear.

- **Echo Suppression**: If set to *Yes*, the system plays the conversation in half-duplex mode, for a short period of time at the beginning of every conversation, while it adapts the echo canceller based on audio characteristics. If set to *No*, you may hear strange echo audio while the canceler adapts. By default, this is set to *Yes*.
- **Echo Suppression Sensitivity Level**: Defines the threshold at which the system suppresses the echo from the far end at the beginning of a conversation. A value closer to zero results in very little echo suppression (full duplex), and a value closer to 100% results in a large amount of echo suppression (half duplex). Use the scroll box to make your selection. Then press **ENTER** or click on another field to save your change. The default is *60%,* and the range is *0%* to *100%*.
- **Echo Saturation Blocker**: If set to *Yes*, the system plays the conversation in half-duplex mode when the volume level becomes very loud. If set to *No*, the system always plays the conversation in full-duplex mode. By default, this is set to *Yes*.

  NOTE: Full-duplex mode can lead to echo audio because the echo canceller cannot cancel echo caused by very loud or saturated audio.

Static IP Address

4.58 The Static IP Address field defines the IP connection's static IP address provided by the IP network administrator. Changing the static IP address resets the IPRC. The system broadcasts changes to an IP connection's IP address to the other nodes in the private network as a database update. The IP connection's IP address must be kept in sync throughout the network. Conflicting IP addresses can cause the system to behave erratically. The system generates a default value for this field upon creation of the IP connection. This default value is *192.168.200.201* and should be changed as appropriate when configuring the network. To change the IP address, click on the field. A dialog box appears. Enter the new IP address (do not include the periods) and click **OK**.

NOTE: Failure to properly change the default value of the Static IP Address could cause the connection to fail.

Static Subnet Mask

4.59 The Static Subnet Mask field defines the IP connection's default network subnet mask provided by the IP network administrator. Changing the default network subnet mask resets the IPRC. The default is *255.255.255.0*. To change the subnet mask, click on the field. A dialog box appears. Enter the new subnet mask (do not include the periods) and click **OK**.

Static Gateway

4.60 The Static Gateway field defines the IP connection's default IP gateway address provided by the IP network administrator. Changing the default IP gateway address resets the IPRC. The default is *192.168.200.201*. To change the IP gateway address, click on the field. A dialog box appears. Enter the new IP address (do not include the periods) and click **OK**.

WINS Server

4.61 The WINS Server field defines the IP connection's WINS IP address provided by the IP network administrator. If there is no WINS IP address, set this field to *0.0.0.0*. Changing the WINS IP address resets the IPRC. The default is *0.0.0.0*. To change the WINS IP address, click on the field. A dialog box appears. Enter the new IP address (do not include the periods) and click **OK**.

Hostname

4.62 The Hostname field defines the hostname provided by the IP network administrator. This is the name the IPRC registers with the WINS server. To change the hostname, enter the new name, up to 15 characters, in the text box. Then press **ENTER** or click on another field. (Changing the hostname resets the IPRC.) By default, this field is blank.

Audio RTP Type of Service and Data Type of Service

4.63 The Audio RTP and Data Types of Service specify the precedence for audio and data packets, respectively. The system inserts the values of these two fields into the Type of Service (ToS) field of each audio and data packet. Network devices can then use the value of the ToS field to establish precedence for IP routing. The default value is *0*, meaning the packets have no precedence. A value of *184* indicates "IP Precedence." All other values may be used by a network device using Differentiated Services (Diffserv) to establish precedence. Use the scroll box to make your selection. Then press **ENTER** or click on another field to save your change.

Audio Stream Receive Port

4.64 The Audio Stream Receive Port defines the first of 33 even port numbers that off-node IPRCs use to transmit audio data to this IPRC. You must choose an even-numbered value. The base audio port number, plus 64 additional port numbers, cannot conflict with other port numbers on the IP connection. For example, if the base audio port number is 5000, no other port numbers on the IP connection can fall in the range from 5000 to 5064. Changing the Audio Stream Receive Port resets the IPRC. The system broadcasts changes to an IP connection's Audio Stream Receive Port to the other nodes in the private network as a database update. This field corresponds to the off-node IP connection's Remote Audio Receive Port (see page 6-177). To change the port number, enter the new number in the text box. Then press **ENTER** or click on another field. The default is *5004,* and the range of values is *1026* to *65468*.

NOTE: Because SIP uses port 5060 (by default) for call control, firewalls, NATs, and routers often treat this port as a "special case" and may block its use by non-SIP packets. If your network currently uses SIP, Inter-Tel strongly recommends that you use a different port range (other than 5004-5068) for audio stream ports; otherwise, you may experience audio problems over the IP network.

IP Terminal TCP Call Control Port

4.65 The IP Terminal TCP Call Control Port defines the port number the IPRC uses for call control. To change the port number, enter the new number in the text box. Then press **ENTER** or click on another field. The range is *1024-65535*, and the default is *5566*.

IP Terminal General Purpose UDP Port

4.66 The IP Terminal General Purpose UDP Port defines the port number the IPRC uses for general purpose and broadcast messages. To change the port number, enter the new number in the text box. Then press **ENTER** or click on another field. The range is *1024-65535*, and the default is *5567*.

MGCP Receive Port

4.67 The MGCP Receive Port defines the port number the MGCP gateway and endpoints use for communication. To change the port number, enter the new number in the text box. Then press **ENTER** or click on another field. The range is *1024-65535*, and the default is *2427*.

TCP Call Control Port

4.68 The TCP Call Control Port defines the port number that off-node IPRCs use to connect call control with this IPRC. The TCP Call Control Port cannot conflict with other port numbers on the IP connection. Changing the TCP Call Control Port resets the IPRC. The system broadcasts changes to an IP connection's TCP Call Control Port to the other nodes in the private network as a database update. The IP connection's TCP Call Control Port must be kept in sync throughout the network. This field corresponds to the off-node IP connection's Remote Listening Port (see page 6-177). To change the port number, enter the new number in the text box. Then press **ENTER** or click on another field. The default is *5570,* and the range of values is *1025* to *65535*.

Web Listening Port

4.69 The Web Listening Port identifies the port number the IPRC uses for Web access (i.e., for diagnostics). To change the port number, enter the new number in the text box. Then press **ENTER** or click on another field. The range is *1-65535*, and the default is *80*.

NOTE: If using the IP port for the Board Utility, the Web Listening Port number must match the port number assigned in the Board Utility (see page 3-174).

Remote Node

4.70 You can also create off-node IP connections to represent IPRCs on other nodes within the network. To display a list of IP connection(s) for a remote node, double-click on the remote node.

NOTE: Each node in an IP network must contain an off-node IP connection for all other IP nodes in the network. For example, in a three-node network, node 1 must have an off-node IP connection for nodes 2 and 3; node 2 must have off-node IP connections for nodes 1 and 3; and node 3 must have off-node IP connections for nodes 1 and 2. Nodes that do not have the correct off-node IP connections programmed may not be able to communicate with each other via IP.

4.71 To create an off-node IP connection, right-click anywhere in the right side of the screen and select **Create Off-Node IP Connection**. You are prompted to select an extension that begins with "P6." Select a unique extension to the IP connection and click **OK**.

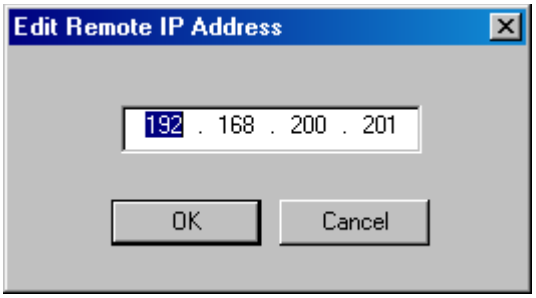
NOTE: When creating off-node connections and other IP-related extension numbers, Inter-Tel recommends using a numbering plan that associates the extension to the device and the node on which it resides. For example, the first IPRC on node 2 would be P6021 (P6 followed by the node number, then the IPRC number). The second IPRC would be P6022, etc.

4.72 For each off-node IP connection, the following fields are displayed:

- Node IP Connection Group
- Remote IP Address
- Remote Audio Receive Port
- Remote Listening Port

4.73 **Node IP Connection Group**: The Node IP Connection Group parameter provides a link to the node IP connection group to which the IP connection belongs. See page 6-185 for details.

4.74 Remote IP Address: The Remote IP Address defines the remote IP connection's IP address provided by the IP network administrator. This corresponds to the static IP address programmed for the connection on the remote node (see page 6-173). To change the address, click the current value. The Edit Remote IP Address dialog box appears, as shown below. Enter the new IP address (do not include the periods) and click **OK**. The following IP addresses are not allowed: 0.0.0.0 and 255.255.255.255.



4.75 Remote Audio Receive Port: The Remote Audio Receive Port defines the remote IP connection's audio receive port. This should match the corresponding IP connection's Audio Stream Receive Port (see page 6-174) programmed on the remote node.

NOTE: Because SIP uses port 5060 (by default) for call control, firewalls, NATs, and routers often treat this port as a "special case" and may block its use by non-SIP packets. If your network currently uses SIP, Inter-Tel strongly recommends that you use a different port range (other than 5004-5068) for audio stream ports; otherwise, you may experience audio problems over the IP network.

4.76 Remote Listening Port: The Remote Listening Port defines the remote IP connection's listening port. This should match the corresponding IP connection's TPC Call Control Port (see page 6-175) programmed on the remote node.

F. IP PORTS

4.77 Each CPC has a 10/100-Base T Ethernet port that allows you to upload Call Processing software, upload software licenses, and access DB Programming.

NOTE: The IP Port does not support Simple Network Management Protocol (SNMP).

4.78 Before you program the IP port, Inter-Tel recommends that you consult the on-site network administrator for information on any standard practices that are followed, including the use of special characters in hostnames. Providing a static IP address and a hostname is generally considered the "best" practice.

4.79 To access the IP Port programming options, either double-click the IP Port extension (PPXXX) at the CPC folder level or select the desired IP port extension from the IP Ports folder (System\Devices and Feature Codes\IP Ports). You can then program the appropriate fields as described in this section.

NOTE: When you access the IP port programming fields, a warning message appears, informing you that changes will not take effect until the system is reset.

4.80 **Description and Username**: All IP ports should have a description and a username. The description that appears can be up to 20 characters long. The username, that will appear on display phones, can have up to 10 characters. To program the names, select the desired text box and type the entry. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

4.81 **Current IP Address**: *(This field is not programmable.)* Identifies the current IP address assigned to the IP port. This value may change if using a DHCP or BOOTP server to obtain an IP address.

4.82 **Current Subnet Mask**: *(This field is not programmable.)* Displays the current subnet mask assigned to the IP port. This value may change if using a DHCP or BOOTP server to obtain an IP address.

4.83 **Current Gateway**: *(This field is not programmable.)* Identifies the current IP address of the network gateway. This value may change if using a DHCP or BOOTP server to obtain an IP address.

4.84 **Current WINS Server**: *(This field is not programmable.)* Identifies the current IP address of the Windows Internet Naming Service (WINS) server used for name resolution. This value may change if using a DHCP or BOOTP server to obtain an IP address.

4.85 **Hostname**: Enter the hostname, up to 15 characters, used to identify the CPC in the network. This allows you to access the IP port without having to enter the IP address. This name is also used to identify the CPC on the PC network.

NOTE: In general, you should not use special characters (e.g., asterisk, tilde, etc.) for hostnames.

4.86 **Static IP Address**: Enter the default IP address (do not include the periods) that is used if the card does not obtain an IP address from a DHCP or BOOTP server. By default, this is *192.168.200.201*.

4.87 **Static Subnet Mask**: Enter the subnet mask (do not include the periods) associated with the port's IP address. By default, this is *255.255.255.0*.

4.88 Static Gateway: Enter the default IP address (do not include the periods) of the gateway that is used to access the port. By default, this is *192.168.0.1*.

4.89 Static WINS Server: Enter the IP address (do not include the periods) of the WINS server that the system uses for name resolution. By default, this is *0.0.0.0*.

4.90 Telnet and Web Server: Enable these options to provide access to the Web and/or Telnet interfaces that are used for troubleshooting purposes. By default, these fields are both *enabled*.

NOTE: If using the IP port for the Board Utility, the Web Server option must be enabled.

4.91 Listening Port: Enter the port number that the CPC will use to process incoming socket requests (e.g., for voice mail, OAI, DB Programming, etc.). The range is *1-65535*, and the default is *4000*.

4.92 Web Listening Port: Enter the port number the CPC uses for Web access (i.e., for diagnostics). The range is *1-65535*, and the default is *80*.

NOTE: If using the IP port for the Board Utility, the Web Listening Port number must match the port number assigned in the Board Utility (see page 3-174).

4.93 IP Address Assignment: **BOOTP or DHCP**: Enable one or both of these options to assign a dynamic IP address to the CPC. If these flags are disabled, the CPC uses the IP address identified in the Default Static IP Address field. By default, both of these flags are *enabled*.

NOTE: Inter-Tel recommends using the static IP address for the CPC (and CP Server). If using a BOOTP or DHCP server, the IP address may change, dropping any CT Gateway and/or Voice Processing connections.

G. MODEMS

4.94 In the modem programming section, you can create off-node devices for modems on the other nodes and program individual modems on the Local node.

Off-Node Modems

NOTE: Programming off-node modems across an IPRC connection is not supported.

4.95 If a multi-node system is using more than one CPC, you can program the system to recognize the modems on the other CPCs. When programming modems, make sure you follow a universal numbering plan (i.e., the extensions must be unique). In addition, to program off-node modems on nodes running system versions prior to 8.0, you must assign the modem extension to an off-node single-line set. This provides modem access for nodes running earlier system versions.

NOTE: You must have a remote node programmed on the system before you can create an off-node modem extension.

4.96 To program off-node modem extensions:

1. Under System\Cabinets\Devices and Feature Codes\Modems\<*Node*>, right-click and select **Create Off-Node Modem**.
2. Select the starting extension and the number of modem extensions you want to create. The extension you assign to the off-node modem must match the extension programmed locally for the CPC on that node. For example, if the modem extension for the CPC on node 2 is 1502, the off-node modem extension for that CPC on other nodes must be 1502.

   NOTE: Do not create or delete more than 2000 off-node devices at a time. Batch creating more than 2000 off-node devices may cause problems with the system.
3. Program the Description (up to 20 characters) and Username (up to 10 characters) for each extension.

4.97 After programmed, you can use the off-node modem extension for the same functionality as the local modem extension.

Local Modems

4.98 When the system detects a CPC, a Modems folder is available under System\Devices and Feature Codes. You can also double-click the modem extension under the CPC folder to access this screen. You can then program the modem information as described below.

NOTICE
You cannot change the modem's extension if you are currently using the modem to connect to DB Programming.

4.99 Description and Username: All modems should have a description and a username. The description that appears can be up to 20 characters long. The username, that will appear on display phones, can have up to 10 characters. To program the names, select the desired text box and type the entry. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

4.100 **Address**: *(This field is not programmable.)* Indicates the circuit address assigned to the modem. All modems are assigned the fifth port. For example, a modem on a CPC located in slot 8 of Node 1 would be assigned address 1:08.05.

4.101 When you expand or double-click a modem extension, you can program the following fields:

- **Modem Enabled**: Allows the modem to accept incoming calls, if enabled. If disabled, the modem does not accept incoming calls, and INVALID EXTENSION appears on display phones. By default, this is *enabled*.

> **NOTE:** System Administrators can also enable/disable and reset the modem using the Administrator phone. See page 5-299 for details.

- **Minimum Bit Rate**: Select the minimum bit rate the modem will use for its connection. The range is *9600-33600*, and the default is *9600*.
- **Force Minimum Bit Rate**: Forces the modem to connect at the Minimum Bit Rate only, if enabled. If disabled, the modem can connect at bit rates higher than the programmed minimum. By default, this is *disabled*.

4.102 After you have programmed the modem information, you can assign the modem extension as the following:

- **Trunk Group Ring-In Destinations**: *(System\Devices and Feature Codes\CO Trunk Groups)* Allows remote access of the modem using a trunk call.
- **Call Routing Table Ring-In Destinations**: *(System\Trunk-Related Information\Call Routing Tables)* Allows modem access using one or more specific phone numbers.
- **Voice Mail Extension ID for Auto Attendant and Digit Translation**: *(Voice Processor\Devices\Extension ID)* Allows modem access through the Auto Attendant or Voice Mail extension.

H. NETWORK GROUPS

NOTE: To have IP endpoints communicate via P2P to MGCP/SIP gateways, place the IP endpoints and MGCP/SIP trunks (gateway and endpoints) in the same Network Group.

4.103 Network Groups define the IP devices that can communicate via P2P audio (see page 6-183 for details). When you assign an IP device (including SIP endpoints and MGCP/SIP gateways and endpoints) to a circuit, the device is automatically added to the default Network Group (PP029). This Network Group is not programmed for P2P audio and is not configurable. To use P2P audio, you must assign the IP devices to a Network Group that is programmed to support the feature. Before you assign IP devices to a Network Group, however, you must make sure the hardware is properly upgraded.

NOTE: You cannot delete the default Network Group.

Hardware Upgrade

CAUTION

Inter-Tel recommends that you run the Network Qualifier (for at least 24 hours) to determine if the network meets the minimum requirements identified for P2P audio. These requirements are the same as those identified for IP private networking, which is described on page 2-94.

4.104 P2P audio requires that you upgrade the IPRC and IP devices (IP phones and IP SLAs) to the latest version of firmware (v8.1). In addition, each node that will use P2P audio must be running software version 8.1 or later.

4.105 To upgrade the hardware:

1. If using more than 16 devices, install the 32-device daughter card on the IPRC.
2. Run the Upload Utility and upload the latest firmware to the IPRC.
3. Use the Upload Utility to upload the latest firmware to all IP devices that will use P2P audio.

NOTE: For details about using the Upload Utility to upgrade IP devices, see page 3-178.

Network Group Assignment

4.106 When you double-click Network Groups, the default Network Group (PP029) and any other programmed groups are displayed. To create a Network Group, right-click and select **Create Network Group**. Then select a P7XXX extension and the number of groups to create. Then click **OK**.

NOTE: Each Network Group on a node must have a unique extension.

4.107 Description and Username: All Network Groups should have a description and a username. The description that appears can be up to 20 characters long. The username, that will appear on display phones, can have up to eight characters. To program the names, select the desired text box and type the entry. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

4.108 Use Peer-To-Peer Audio: You must enable the Use Peer-To-Peer Audio flag to allow the devices in the Network Group to use P2P audio. If this flag is disabled, the devices in the group will use the system vocoders for all calls. Enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check, select the check box again.) By default, this flag is disabled.

4.109 After you have created the Network Group, you must assign stations and trunks to the group. The stations and/or trunks in this group will then use P2P audio or the TDM highway based on the **Use Peer-To-Peer Audio** flag. To access the programming options for assigning IP stations and trunks, double-click the desired Network Group.

4.110 IP Stations: IP stations include IP endpoints (except the IP SoftPhone), IP SLAs, and multi-protocol (SIP or IP) endpoints. To view a list of IP stations that are currently assigned to the Network Group, double-click **IP Stations**.

4.111 To add stations to the list:

1. Right-click and select **Move To IP Stations List**.
2. Select IP Phone, IP Single-Line Adapter, or IP softphone and click **Next**.
3. Highlight the devices to add to the list and click **Move To List**.
4. Click **Finish** when all of the desired devices have been moved.

4.112 IP Trunks: IP trunks include MGCP gateways and endpoints, and SIP trunks. To view a list of IP trunks that are currently assigned to the Network Group, double-click **IP Trunks**.

4.113 To add trunks to the list:

1. Right-click and select **Move To IP Trunks List**.
2. Select MGCP Endpoint, MGCP Gateway and Endpoint, or SIP Trunk and click **Next**.
3. Highlight the devices to add to the list and click **Move To List**.
4. Click **Finish** when all of the desired devices have been moved.

4.114 You can also program Network Groups for P2P audio across nodes, as described below, or you can use the Export/Import function if running system software v9.0 or later.

4.115 To program Network Groups for P2P audio across nodes:

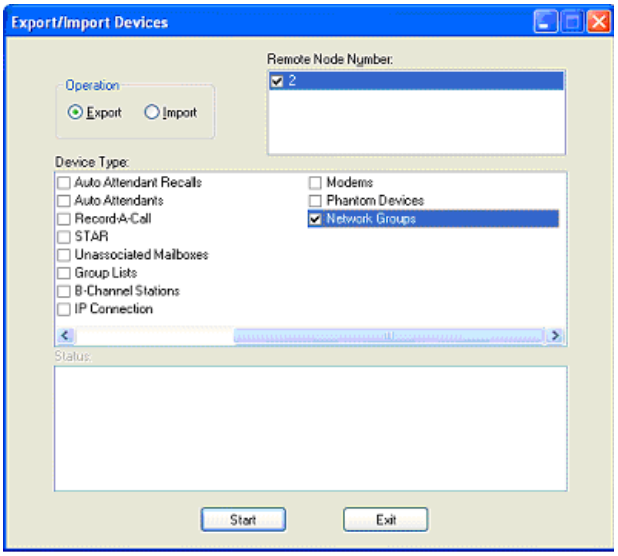
1. Create a Network Group on each node and assign each group the same extension number. For example, if the devices that will use P2P audio reside on nodes 2 and 3, program a Network Group with the same extension number (e.g., P7000) on both nodes.
2. Assign the appropriate IP endpoints and IP trunks to the Network Group on each node. For example, to use P2P audio between device 2000 on node 2 and device 3000 on node 3, Network Group P7000 on node 2 must contain device 2000, and Network Group P7000 on node 3 must contain device 3000.
3. Enable the **Use Peer-To-Peer Audio** flag for the Network Group on each node.

NOTE: IP softphones that are used as mobile devices cannot be part of a Network Group configured for Peer-To-Peer Audio. If you attempt to add an IP softphone to the Network Group list or change the Use Peer-To-Peer Audio setting of a Network Group to Yes, a warning message appears.

4.116 The Export/Import function (see page 5-171) located on the Operations menu can be used to export/import Network Groups information to/from another node.

4.117 To export\import Network Groups:

1. In DB Studio, select **Export\Import** from the Operations menu. The dialog box on the right appears.
2. Select **Export** or **Import** for the Operation.
3. Select the Remote Node Number to export to or import from.
4. Select Network Groups as the Device Type.



NOTE: Use the horizontal scroll bar to locate the Network Groups option.

5. Click **Start**. The Export/Import status appears in the Status box.

4.118 Note that the export/import may fail due to one of the following:

- There is an active programming session on the selected remote node.
- The remote node is unable to communicate with its Voice Processor port.
- The remote node or its links are down.

4.119 If the export/import fails, the new Network Group extension will not be added to or changed on that node, and you must manually add or change the new Network Group extension in the appropriate node's database or try the export\import operation later.

NOTE: If Message Print is enabled, an error message will indicate an unsuccessful broadcast when the node is logged in to DB Studio.

4.120 For Network Groups troubleshooting information, see page 7-65.

I. NODE IP CONNECTION GROUPS

4.121 The system automatically creates a node IP connection group for each remote node added. This node IP connection group, which corresponds to an IP network connection between the remote node and the local node, can then be programmed for each off-node IP connection. Like other devices, DB Programming displays off-node IP connections within a folder corresponding to the node on which the off-node IP connection resides.

4.122 When you access the IP Connections folder, the right side of the screen displays the extension, description, and username of the IP connection group.

4.123 Description and Username: All IP connection groups should have a description and a username. The description that appears in all IP connection group lists in the database can be up to 20 characters long. The username, that will appear on display phones, can have up to 10 characters. To program the names, select the desired text box and type the entry. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

4.124 The following information can be programmed for each node IP connection group:

- IP Call Configuration (see below)
- Off-Node IP Connections (see page 6-187)
- Emergency Outgoing Access (see page 6-188)
- Outgoing Access (see page 6-188)
- Remote Node (see page 6-189)
- Camp-Ons Allowed (see page 6-189)

IP Call Configuration

4.125 The IP Call Configuration field provides a link for programming the following call configuration parameters for the node IP connection group. Note that changes to these parameters do not take effect until the next call is made/received. Changes do not affect existing calls. When you double-click on this field, the following parameters are displayed:

- **Audio Frames/IP Packet**: Indicates the number of audio frames the system inserts into each packet. The fewer audio frames you program, the less latency in the signal, however, the larger bandwidth consumption increases the chance of jitter and network congestion. The more audio frames you program, the more latency in the signal, but the reduced bandwidth lowers the chance of jitter and network congestion. Use the scroll box to make your selection. Then press **ENTER** or click on another field to save your change. The default is *3* packets, and the range is *1* to *8* packets.

- **Average In Time Frame Percentage Threshold and Average In Time Frame Timer**: Indicate when network characteristics inhibit voice connections. When the average number of In Time frames falls below the Average In Time Frame Percentage Threshold, and the average number of In Time frames stays below that threshold for the time given by the Average In Time Frame Timer (in seconds), the system displays a warning about insufficient bandwidth (see page 5-167). Use the scroll box to make your selection. Then press **ENTER** or click on another field to save your change. The default for Average In Time Frame Percentage Threshold is *60%,* and the range is *0* to *100*. The default for Average In Time Frame Timer is *5* seconds*,* and the range is *0* to *255*.

- **DTMF Encoding Setting**: Defines the vocoder that the system uses when the system believes it is transmitting DTMF digits. Use the drop-down box to make your selection. Then press **ENTER** or click on another field to save your change. The options are G.711 Mu-Law, G.711 A-Law, G.729, and RFC 2833. The default is *G.711 Mu-Law [or A-Law for Europe]*.

NOTICE

Inter-Tel currently supports T.38 Fax over IP (FoIP) only. If you change any of the fax settings to a value other than the current default setting of T.38, a warning message appears. If you click **OK** when this message is displayed and continue to program the fax settings, you do so at your own risk.

- **Fax Detection Sensitivity**: Allows you to make false fax detection more or less likely. The higher this number, the less likely the system is to falsely detect fax transmission, but the more likely the system is to fail to correctly detect fax transmission. The lower the number, the more likely the system is to falsely detect fax transmission, but the less likely the system is to fail to correctly detect fax transmission. Use the scroll box to make your selection. Then press **ENTER** or click on another field to save your change. The range is 0 to 100. By default, this is 0, indicating that faxing is disabled.

- **Fax Encoding Setting (Fax Detection)**: Defines the vocoder the system uses to tell the other end of a fax transmission that it has detected fax. Use the drop-down box to make your selection. Then press **ENTER** or click on another field to save your change. The options are G.711 Mu-Law, G.711 A-Law, and RFC 2833. By default, this is *RFC 2833*.

NOTE: If a red "X" appears in this field, you must change the Fax Detection Sensitivity field to any value other than 0.

- **Fax Encoding Setting (Fax Transmission)**: Defines the vocoder that the system uses when the system believes it is transmitting fax data. Use the drop-down box to make your selection. Then press **ENTER** or click on another field to save your change. The options are G.711 Mu-Law and G.711 A-Law. By default, this is *G-711 Mu-Law.*

NOTE: If a red "X" appears in this field, you must change the Fax Detection Sensitivity field to any value other than 0.

- **Local Music Source**: Is meaningful only when you set the Music-On-Hold vocoder to Local Music Source. If set to Local Music Source, the system does not transmit and receive Music-On-Hold between the local node and the remote node. Rather, the local music source field defines the music source the caller on the local node hears when he would otherwise be listening to Music-On-Hold. The Music-On-Hold source can be programmed to Ringback, Silence, Tick Tone, or Board. If set to *Board*, you can program any on-node music source as the extended value. Use the drop-down box to make your selection. Then press **ENTER** or click on another field to save your change. The default is *Board*.
- **Minimum Playback Time**: Defines the minimum time, in milliseconds, that packets wait in the receive buffer before the system plays the audio. The higher this number, the more latency in the signal; however, it is less likely that network problems like jitter will cause lost or late audio packets. The lower the minimum playback time, the less latency there is in the signal; however, there is a greater chance of jitter. Use the scroll box to make your selection. Then press **ENTER** or click on another field to save your change. The default is *80 ms,* and the range is *1* to *320 ms*.

  NOTE: The Minimum Playback Time is supposed to adjust the jitter buffer size on the IP devices. The newer IP devices (e.g., multi-protocol endpoints) have a dynamic jitter buffer and ignore this setting. The only devices that use this setting are the Model 8660, IP Phone Plus, IP-SLA, and IPRC.

- **Music-On-Hold Encoding Setting**: Determines whether Music-On-Hold is transmitted across a network connection. You can program this field to either *Use Speech Encoding Setting* or *Use Local Music Source.* If set to *Use Speech Encoding Setting,* the system uses the vocoder specified by the speech encoding setting when transmitting Music-On-Hold across the network connection. If set to *Use Local Music Source*, the system connects the caller to the music source given by the local music field rather than transmitting Music-On-Hold across the network. Use the drop-down box to make your selection. Then press **ENTER** or click on another field to save your change. The default is *Use Local Music Source*.
- **RTP Profile**: Leave this field set to the default value of *Inter-Tel RTP Version 0 Format* for optimal performance. The system may use other options to interface to devices that use standard Real-time Transport Protocol (RTP). The available values are *Inter-Tel RTP Version 0 Format* and *Standard A/V RTP Profile (RFC)*.
- **Speech Encoding Setting**: Defines the vocoder that the system uses when transmitting speech data. The options are *G.711 Mu-Law*, *G.711 A-Law*, *G.729*, and *G.729B (VAD)*. Use the drop-down box to make your selection. Then press **ENTER** or click on another field to save your change. The default is *G.729*.
- **Transmit DTMF Level**: Defines the level at which the system injects tone onto the backplane when receiving an out-of-band DTMF tone. Use the drop-down box to select the country where the node resides (U.K., U.S.A., or Japan). Then press **ENTER** or click on another field to save your change.

Off-Node IP Connections

4.126 When you double-click on the Off-Node IP Connections field, a list of the off-node IP connections for this node IP connection group appears.

Day/Night Emergency Outgoing Access

4.127 There are separate lists for stations with emergency outgoing access in day and night modes. By default, the automatic station list (*Auto: All Stations*) is assigned to Day/Night Emergency Outgoing Access. To program the stations that will have emergency outgoing access for the node trunk group, select **Emergency Outgoing Access** and then Day Mode or Night Mode and do the following:

4.128 To add extensions to the list:

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To List**. A window appears that asks you to select the type of device to include.
3. Select the device type(s) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

> **NOTE:** To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate items and then select **Add Items**. When you have added all the desired devices select **Finish**. Your selections appear in the list.

> **NOTE:** If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

4.129 To delete destinations from the list: To delete an item from the list, highlight them and then right-click. Select **Remove Selected Items**.

NOTICE

It shall be the responsibility of the entity or person(s) completing installation and maintenance of hardware or software described herein to research, comply with and be responsible for the specific governmental rules and regulations regarding Emergency Outgoing Access 911 [999 in Europe] of the geographic location in which such functions are performed.

Day/Night Outgoing Access

4.130 There are separate lists for stations with outgoing access in day and night modes. By default, the automatic station list (*Auto: All Stations*) is assigned to Day/Night Outgoing Access. To program the stations that will have outgoing access for the node trunk group, select Outgoing Access and then Day Mode or Night Mode and do the following:

4.131 To add extensions to the list:

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To List**. A window appears that asks you to select the type of device to include.
3. Select the device type(s) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

> **NOTE:** To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate items and then select **Add Items**. When you have added all the desired devices select **Finish**. Your selections appear in the list.

NOTE: If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

4.132 To delete destinations from the list: To delete an item from the list, highlight them and then right-click. Select **Remove Selected Items**.

Remote Node

4.133 The Remote Node field provides a link to the node to which the connection group corresponds.

Camp-Ons Allowed

4.134 The Camp-On feature can be enabled or disabled for each node IP connection group. If disabled, users placing outgoing calls will hear busy signals when all connections in the group are in use or unavailable. If enabled, users will be able to camp on and wait for an available connection. To enable this flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

J. NODES

4.135 For a detailed explanation of nodes and networking, see Appendix A — Networking.

Local Node

4.136 For the local node, you can only assign the description, username, and node number.

Remote Nodes

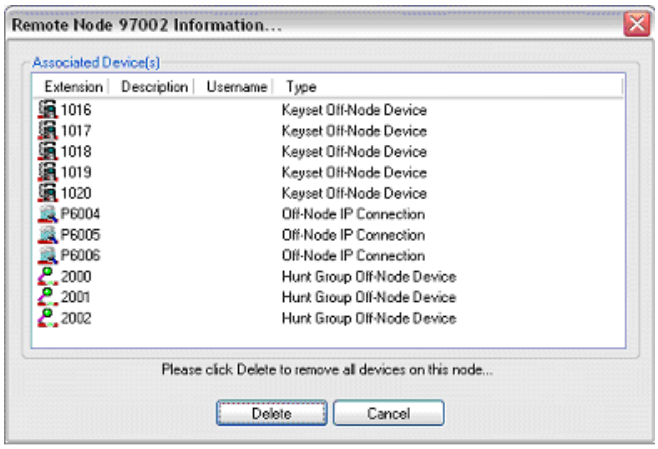
4.137 Each remote node must be programmed with a node number, node trunk groups, and a search algorithm. You can also program a description and a username for the node. To program a remote node's node trunk groups and search algorithm, select the specific node to display the programming options.

4.138 Node Number: You can select node numbers 1-99. If you change the node number, the system will reset after you exit. To change the current node number, select the current Value and enter the new value in the text box. Then press **ENTER** or select another field to save the change. If the number you entered conflicts with an existing node number, you will see an error message and must change the number to a valid value.

4.139 Description and Username: You can enter a new description or username for the node by selecting the appropriate field and then entering the new information in the text box. Then press **ENTER** or select another field to save the change. Descriptions can contain up to 20 characters and usernames can contain up to 10 characters. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

4.140 Deleting a Node and its Devices: Prior to v9.0, when deleting a node, you first had to first manually search for and remove all of the node's devices. In system software v9.0 and later, you can delete the node, and all of the node's devices in one step.

4.141 To delete a node, right-click the node under System\Devices and Feature Codes\Nodes, and select **Delete**. A dialog box, like the one shown below, opens displaying all of the node's devices.



4.142 In this example, the dialog box displays all of the off-node devices associated with node 97002. To delete the node and the associated devices, click **Delete**. To close the dialog box without deleting the node, click **Cancel**.

NOTE: The dialog box does not allow you to select a single device and delete it from the node when you click **Delete**, instead the entire node is deleted along with all of the devices.

Node Trunk/IP Connection Groups

4.143 Each node has a list of node trunk or node IP connection groups that are used to access the remote node. The node IP connection groups correspond to an IP connection between a remote node and a local node. For each node in the network, you must define the routes to every other node. For example, in a network with four nodes, you would define three routes for each node (one to each of the other three nodes). See Appendix A — Networking for a sample of node programming. To program the node trunk or node IP connection groups included in this node, double-click on **Node Trunk/IP Connection Groups** and then do the following:

4.144 To add trunk or IP connection groups to the list: Note that this is an *ordered list*. Place the trunk or IP connection groups in the list in the order you want them to be accessed when the hunt group receives a call.

1. *To add to the bottom of the list:* Do not highlight any existing trunk or IP connection groups and right-click anywhere in the right side of the window. An option box appears.

   To add to the list above an existing trunk group: Highlight the trunk or IP connection group and then right-click. An option box appears.

2. Select **Add To Node Trunk/IP Connection Groups List**. A window appears that asks you to select the type of device to include.

3. Select **Node Trunk Group** or **Node IP Connection Group** and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

   NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate items and then select **Add Items**. When you have added all the node trunk groups necessary, select **Finish**. Your selections appear in the list.

   NOTE: If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

4.145 To move a trunk or IP connection group to another location in the list: Drag and drop the trunk or IP connection group to the new position. Or, highlight the trunk or IP connection group to move and press **CTRL** + the up/down arrow to move the trunk or IP connection group up or down in the list.

4.146 To delete a trunk group from the list: Highlight the trunk or IP connection group(s) and then right-click. Select **Remove Selected Items**.

Search Algorithm

4.147 A search algorithm determines whether the node trunk groups are accessed in linear or distributed order. Set the search algorithm by selecting current Value and then using the drop-down list box to scroll to the desired option. The options are:

- **Linear**: The system first attempts to route a call through the first node trunk group listed. If it is unable to route through that node trunk group, it attempts to route the call through the second node trunk group in the node. The system continues to attempt to route the call through the subsequent node trunk groups listed in the node until it successfully routes the call or exhausts all node trunk groups in the list.
- **Distributed**: The system shifts the first node trunk group it attempts to use with each call. For example, if the system routed the first call through the first node trunk group in the node, it routes the second call through the second node trunk group in the node.

NOTE: Determining in what order to list the Node Trunk or IP Connection Groups in a node and when to use Linear or Distributed search type depends on the system configuration and traffic. For example, if the Node Trunk or IP Connection Group List has more than one trunk group that connects to the same node, you should use the Linear instead of Distributed search type. See Appendix A — Networking for details.

Audio for Calls Camped onto this Device

4.148 The Audio for Calls Camped onto this Device field defines the audio that a caller hears when camped-on to this node trunk or IP connection group. To program this field, use the drop-down list to select the type of audio and press **ENTER**. Your options are Silence, Ticktone, Ringback, or Board. If you select **Board**, specify the music source in the Extended Value field. The default is *Board*, and the default extended value is *01:08 Central Processor*.

K. PAGE PORTS

4.149 When you select Page Ports, a list including the local node and any remote nodes appears. Page port programming differs for remote and local nodes, as described below.

Local Page Ports

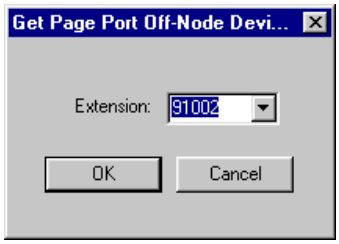
4.150 Local page ports are created in CPC or CPU Card programming. When you double-click on the Local node, a list of the existing local page ports appears. You can enter descriptions and usernames.

Remote (Off-Node) Page Ports

4.151 Off-node page ports allow paging access to the ports located on other nodes. When you double-click on a remote node, a list of its existing off-node page ports appears. You can create or delete off-node page ports and/or enter descriptions and usernames.

4.152 To create Off-Node Page Ports:

1. Right-click anywhere in the right half of the window. An option box appears.
2. Select **Create Off-Node Page Port**. The following window appears. Enter the desired extension number in the text box or scroll to the desired number. Then select **OK** to continue. (Or select **Cancel** to exit without creating the page port.)



The new page port appears in the list without a description or username.

4.153 Deleting Off-Node Page Port: To delete a page port, highlight it and press **DELETE** on your keyboard, or right-click and select **Delete**. The page port is automatically removed from the list.

4.154 Descriptions and Usernames: You can enter a new description or username for the page ports by selecting the appropriate field and then entering the new information in the text box. Then press **ENTER** or select another field to save the change. Descriptions can contain up to 20 characters and usernames can contain up to 10 characters. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

L. PAGE ZONES

4.155 When you select System\Devices and Feature Codes\Page Zones, a list including the local node and any remote nodes appears. Page zone programming differs for Local Page Zones and Remote Node Page Zones.

4.156 There can be up to 10 page zones (50 if using a CP Server). The default database contains no page zones. Page zones may be equipped as needed through DB Programming. They may be removed by right-clicking and choosing Delete.

4.157 To prepare for page zone programming, make a list of the phones, trunks, and/or the external paging port(s) included in local paging zones. Devices can be in more than one page zone. In the default state, all phones are assigned to page zone 1.

NOTE: Placing a large number of phones in a paging zone may affect system performance. If system operation is affected when a page is placed to a particular page zone, remove some phones from that zone or change to external paging for the area served by that page zone.

4.158 All phones and trunks within a page zone must reside on the same node as the page zone. However, a page zone can contain external page ports that are located on other nodes, if they are programmed as off-node devices on the local node (see page 6-198).

Local Page Zones

4.159 When you select the Local node, the page zones are listed. On this level you can change the description and username of the page zone. If you select a specific page zone, you can program the phones, trunks, and page ports that are included in the zone.

4.160 Extension: To change the page zone's extension, select its current Extension and enter the new number in the text box (or scroll to an available number).

4.161 Batch Extension Change: To change several page zone extension numbers at once, highlight the zones you want to change. Then right-click and select **Batch Extension Change**. When the Get Extension screen appears, select the number you want to assign to the first highlighted zone (the other highlighted zones will be numbered consecutively after this number). When you select **OK**, the page zones are automatically renumbered and re-sorted in the list.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items. (You may have to continue to hold **SHIFT** or **CTRL** while right-clicking to display the option without changing the highlighted selections.)

4.162 Descriptions and Usernames: You can enter a new description or username for the page zones by selecting the appropriate field and then entering the new information in the text box. Then press **ENTER** or select another field to save the change. Descriptions can contain up to 20 characters and usernames can contain up to 10 characters. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

4.163 Keyset, Trunk, and Page Port Lists: To assign the phones, trunks and external paging ports to a page zone, double-click on the page zone you want to program. Then double-click on **Keysets**, **Trunks**, or **Page Ports** to add devices to the page zone lists or remove items from the lists, as follows:

4.164 To add devices to the lists:

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To List**. A window appears that asks you to select the type of device to include.
3. Select the device type(s) and then select **Next**. A list of the selected devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.) To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

NOTE: IP Trunks are not supported as configurable options for Page Zones.

4. Highlight the appropriate items and then select **Add Items**. When you have added all the desired devices select **Finish**. Your selections appear in the list.

NOTE: If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

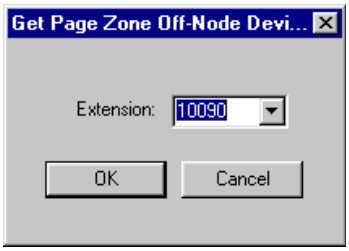
4.165 To delete an item from one of the lists: Highlight the list item(s) and then right-click. Select **Remove Selected Items**.

Remote Page Zones

4.166 Off-node page zones allow paging access to paging zones located on other nodes. When you double-click on a remote node, a list of its existing off-node page zones appears. You can create or delete off-node page zones and/or enter descriptions and usernames.

4.167 To create Off-Node Page Zones:

1. Right-click anywhere in the right half of the window. An option box appears.
2. Select **Create Off-Node Page Zone**. The following window appears. Enter the desired extension number in the text box or scroll to the desired number. Then select **OK** to continue. (Or select **Cancel** to exit without creating the page zone.) The new page zone appears in the list without a description or username.



4.168 To delete Off-Node Page Zones: Highlight the page zone(s) and right-click. Then select **Delete**. The page zone is automatically removed from the list.

4.169 Descriptions and Usernames: You can enter a new description or username for the page zones by selecting the appropriate field and then entering the new information in the text box. Then press **ENTER** or select another field to save the change. Descriptions can contain up to 20 characters and usernames can contain up to 10 characters. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

M. PHANTOM DEVICES

4.170 For complete information about Phantom Devices enhancements, see page 5-120.

4.171 To create a phantom device:

1. Go to System – Devices and Feature Codes – **Phantom Devices**.
2. Right-click in any area of the right panel and select **Create Phantom**.
3. In the Create Phantom Device Extension dialog box, enter the desired extension number or scroll to the desired number and also enter the number of extensions.
4. Click **OK** to continue. This is the same method used to create other off-node device types.

4.172 After you create the phantom device, double-click the device to program the following associated fields:

- Associated Extensions (see below)
- Class of Service (see below)
- Flags (see page 6-197)
- Forwarding Paths (see page 6-197)
- Mailboxes (see page 6-197)
- Password (see page 6-197)
- Audio for Calls Camped onto this Device (see page 6-197)
- Audio for Calls Holding for this Device (see page 6-197)
- Audio for Calls Ringing this Device (see page 6-197)

Associated Extensions

4.173 Double-click on Associated Extensions to view the list of extensions used by the phantom being programmed. Allows you to view the list of extensions used by the phantom being programmed. The following Associated Extensions fields apply to phantoms:

- Attendant
- Message Center
- Alternate Message Source
- Transfer Recall Destination
- Voice Mail Extension
- Outgoing Extension

Class of Service

4.174 Each phantom device has Classes of Service (COS), which restrict or allow certain dialing patterns from being dialed on a call. To program the toll restrictions double-click on Classes of Service, you have the option of choosing Day or Night. Double-click on the desired time period to view a list of current classes of service.

Flags

4.175 You can enable a flag by selecting the current value and then selecting the check box to place a check in it. Some flags require you to choose between two settings (such as Hunt Group Remove/Replace). For those flags, select the current value and use the scroll box to select the desired value. Press **ENTER** or select another field to save your changes. The phantom flags that can be programmed include:

- Alternate Hold Timer
- Attendant
- DID/E&M Receive Busy Instead of Camp-On
- Do-Not-Disturb Allowed
- Hunt Group Remove/Replace
- Manual Forward to Public Network
- Receive Busy Instead of DND
- System Forwarding

Forwarding Paths

4.176 When created, phantoms are automatically added to the All Stations (PP051) extension list. Phantoms can be forwarded and follow the rules for station manual and system forwarding. The forwarding displays and rules are the same as in previous versions.

Mailboxes

4.177 If the phantom being programmed serves as the message notification station for one or more mailboxes, the mailbox(es) will appear in the Mailboxes list. Select Mailboxes to see the list of mailboxes assigned to this phantom. You cannot program the mailbox here, but if you double-click on the mailbox, the screen will jump to the Mailbox programming area.

Password

4.178 This password is used for the Remote Programming feature. The Phantom password can be up to eight digits in length. The default password is the extension number of the Phantom. The password can also be changed by entering the Program Endpoint Password feature code at the endpoint or when using the Remote Programming feature.

Audio for Calls Camped onto this Device

4.179 The Audio for Calls Camped onto this Device field defines the audio that a caller hears when camped-on to this node trunk or IP connection group. To program this field, use the drop-down list to select the type of audio and press **ENTER**. Your options are Silence, Ticktone, Ringback, or Board. If you select **Board**, specify the music source in the Extended Value field. The default is *Board*, and the default extended value is *01:08 Central Processor.*

Audio for Calls Holding for this Device

4.180 This field defines the audio that a caller hears when holding for the device. The default is *Board*. Other options include Silence, Tick Tone, and Ringback.

Audio for Calls Ringing this Device

4.181 This field defines the audio that a caller hears when ringing the device. The default is *Ringback*. Other options include Silence, Tick Tone, and Board.

N. STATIONS

4.182 In the station programming section, you can create off-node devices for stations on the other nodes and program individual stations on the Local node. Remote Node programming is located on this page. Local station programming information begins on the next page.

Off-Node Devices on Remote Nodes

NOTE: Do not create or delete more than 2000 off-node devices at a time. Batch creating more than 2000 off-node devices may cause problems with the system.

4.183 Programming off-node devices (extensions from other nodes) gives the users on this node access to devices on other nodes. Each off-node device has an extension number, username, description, device type (station, hunt group, application, or page port), and node identification number. The username and description are used for directory features only. If you create an extension conflict between an extension on the node you are programming and an off-node device, DB Programming does not allow you to exit the screen until you correct the conflict.

NOTE: Only the local directory features are affected when you change the username and description of an off-node device. In other words, remote nodes do not send updates to other nodes when off-node device information is changed. Only the local node for that device sends out any updates. For example, if the device is programmed as John Doe on node 1, but you change the associated off-node device to Jane Smith on node 2, the node 2 IC directory reflects Jane Smith. The device on node 1, however, still displays John Smith.

4.184 You can use the wildcard character X in extension numbers to represent “any digit.” This allows you to include a range of extensions as one off-node device entry. Wildcard extensions can only be used for off-node devices.

4.185 Wildcard extensions are made up of digits (1-9), followed by wildcard digit X. Examples of valid wildcard extensions are 1XXX (range of 1000 to 1999), 14XX (range of 1400 to 1499), 7X (range of 70 to 79). For example, if there is a 14XX wildcard, users can dial 1433 and be connected to that off-node device.

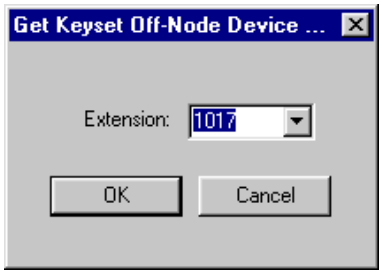
4.186 There are some limitations to using wildcard extensions:

- There will be no entry for wildcard extensions in the intercom directory. (DSS buttons can be programmed for the individual devices the wildcard extension is representing.)
- Voice Processing cannot create associated mailboxes for stations included in wildcard ranges. To have an associated mailbox, the off-node station must have its own off-node device entry on the node where the Voice Processing Unit is located.

4.187 To select the node to program: Double-click on the node number. The current list of off-node devices for that node appears.

4.188 To create off-node devices:

1. Right-click anywhere in the right half of the window. An option box appears.
2. Select **Create Off-Node Keyset** or **Create Off-Node Single-Line**. The following window appears. Enter the extension number or wildcard extension for the device(s). If desired, you can scroll through the list of available numbers and select your extension from the list. Then select **OK** If you entered the same number as an existing extension, you will receive an error message and must enter a new number. The new off-node device appears in the list without a description or username.



4.189 Description and Username: The description is used in the intercom directory and the username appears on phone displays. You can enter a new description or username by selecting the appropriate field and then entering the new information in the text box. Then press **ENTER** or select another field to save the change. Descriptions can contain up to 20 characters and usernames can contain up to 10 characters. Do not use slash (/), backslash (\), vertical slash (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

4.190 Intelligent Directory Search (IDS): Because of the IDS feature introduced in v9.1, the information you enter in any endpoint Description and Username field will have an impact on how users search for information in the system. The Description and Username fields are located in System\Devices and Feature Codes\Local\<*station*>. See page 5-244 for details about IDS.

4.191 To enter a description to support IDS:

1. Select the Descriptions field and then enter the full name, up to 20 characters, in the text box. The full name should be entered in the form: Last name, First name as much as possible. Note that the field is case-sensitive, therefore, IDS will display "Bond, James" rather than "BOND, JAMES" if that is what you programmed into the database. When using this method, the algorithm considers everything before the comma as the last name and everything after the comma as the first name. For example, a user named "Billy Bob Via Linda" should have his description programmed as "Via Linda, Billy Bob." Be careful with descriptions like "Smith, George IP" to indicate that this extension is George Smith's IP phone because IDS will think the name is "George IP Smith."
2. Press **ENTER** or select another field to save the change.

4.192 Hidden Entries in the Voice Mail Directory: There is a Voice Mail feature that allows you to program a description for a user in DB Programming but hides the description in the Voice Mail directory. For example, a user may have a primary extension (for example, at the office) programmed as "BOND, JAMES." However, the same user may have a secondary extension—for example, at home—that is not published from the Voice Mail directory.

4.193 To hide an entry in the Voice Mail directory, enter a tilde (~) immediately before the last name (for example, "~BOND, JAMES") in the Description field. See page 5-249 for details.

Stations on the Local Node

NOTE: For information about programming IP endpoints and IP SLAs, see page 6-148.

4.194 To view the list of stations on the Local node, select **Local**. A list of current stations appears. At this level, you can change the description or username and view the station type and circuit number (address).

4.195 Extension: To change the station's extension, select its current Extension and enter the new number in the text box (or scroll to an available number).

NOTE: If you change a station's extension to match a previously existing unassociated mailbox, a warning window will appear that tells you that the new extension is the same as an unassociated voice mailbox. Choosing **OK** will associate the mailbox with the current device. Choosing Cancel will leave the mailbox unassociated.

4.196 Batch Extension Change: To change several extension numbers at once, highlight the stations you want to change. Then right-click and select **Batch Extension Change**. When the Get Extension screen appears, select the number you want to assign to the first highlighted station (the other stations will be numbered consecutively after this number). When you select **OK**, the stations are automatically renumbered and re-sorted in the station list.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items. (You may have to continue to hold **SHIFT** or **CTRL** while right-clicking to display the option without changing the highlighted selections.)

4.197 Associated Devices and References: The Associated Devices and References feature (see page 5-31) allows you to see the associated devices for a station, mailbox, and hunt group. This feature also allows you to query various groups in the database to locate the associated references to the extension.

4.198 To search for devices associated with the station, right-click the station and select **Associated Devices and References**. A dialog box, containing associated device(s) and references for the mailbox appears.

4.199 Description and Username: All stations have a description and a username. The description of the station that appears in all station lists in the database can be up to 20 characters long. The username, that will appear on display phones in the "idle" display and in several call processing displays, can have up to 10 characters. To program the names, select the desired text box and type the entry.

NOTE: To ensure proper operation of Voice Processor directories, the full name should be entered in the form: *Last name, First name*. Do not use slash (/), backslash (\), vertical bar (|), or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames. Also, if you do not program the username and description, the local stations will not display in the intercom directory.

4.200 Intelligent Directory Search (IDS): Because of the IDS feature introduced in v9.1, the information you enter in any endpoint Description and Username field will have an impact on how users search for information in the system. The Description and Username fields are located in System\Devices and Feature Codes\Local\<*station*>. See page 5-244 for details about IDS.

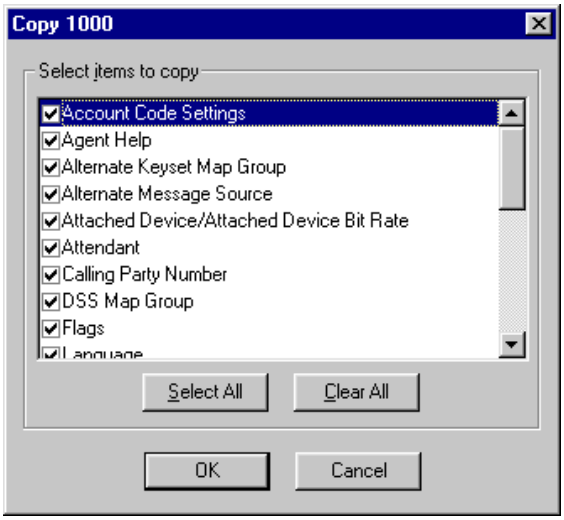
4.201 To enter a description to support IDS:

1. Select the Descriptions field and then enter the full name, up to 20 characters, in the text box. The full name should be entered in the form: Last name, First name as much as possible. Note that the field is case-sensitive, therefore, IDS will display "Bond, James" rather than "BOND, JAMES" if that is what you programmed into the database. When using this method, the algorithm considers everything before the comma as the last name and everything after the comma as the first name. For example, a user named "Billy Bob Via Linda" should have his description programmed as "Via Linda, Billy Bob." Be careful with descriptions like "Smith, George IP" to indicate that this extension is George Smith's IP phone because IDS will think the name is "George IP Smith."
2. Press **ENTER** or select another field to save the change.

4.202 Hidden Entries in the Voice Mail Directory: There is a Voice Mail feature that allows you to program a description for a user in DB Programming but hides the description in the Voice Mail directory. For example, a user may have a primary extension (for example, at the office) programmed as "BOND, JAMES." However, the same user may have a secondary extension—for example, at home—that is not published from the Voice Mail directory.

4.203 To hide an entry in the Voice Mail directory, enter a tilde (~) immediately before the last name (for example, "~BOND, JAMES") in the Description field. See page 5-249 for details.

4.204 Copying Station Programming: To copy a station, right-click on the extension number and then select **Copy**. To paste the programming information into another station, right-click on the station where you want the information pasted. Then select **Paste**. You will see a screen, like the one shown below, that allows you to select the attributes you want to copy.



4.205 To program a specific station, select it from the list of stations to view the station programming fields. The fields that are programmed for local stations include the following:

- Account Code Type (see page 6-203)
- Associated Extensions (see page 6-203):
- Attached Device (see page 6-205)
- Attached Device Bit Rate (see page 6-205)
- Associated Outside Number—B-channel stations only (see page 6-205)
- Audio for Calls Camped onto this Device (see page 6-206)
- Audio for Calls Holding for this Device (see page 6-206)
- Audio for Calls Ringing this Device (see page 6-206)
- Call Logging (see page 6-206)
- Calling Party Name (see page 6-207)
- Calling Party Number (see page 6-207)
- Day and Night Classes of Service (see page 6-208)
- DSS Consoles—Executone phones only (see page 6-208)
- Emergency Calling Party Number (see page 6-209)
- Station Flags (see page 6-209)
- Forwarding Paths (see page 6-214)
- IP Settings—IP endpoints and IP SLAs only (see page 6-215)
- House Phone (see page 6-219)
- House Phone Day/Night number (see page 6-219)
- Keymaps—standard, alternate, and DSS (see page 6-220)
- Mailboxes (see page 6-220)
- Language (see page 6-220)
- Password (see page 6-221)
- Programmable Keys (see page 6-221)
- Record-A-Call (see page 6-222)
- Timers—European systems only (see page 6-224)
- Secondary Language (see page 6-225)
- Speakerphone Type (see page 6-226)
- Validated Flag (see page 6-226)

Account Code Type

4.206 Select **Account Code** and scroll to the desired option (None, Standard, Forced) to determine the type of account code (if any) this station will use. Then press **ENTER** or click outside the field to save your change. The actual account code assignments are programmed in Station-Related Information—Account Codes, as described on page 6-301.

NOTE: Each station can be assigned *either* a forced or standard account code (or no account code). If you select a station for standard or forced account code that was previously programmed to require an account code, the previous account code type is replaced by the new code type.

Associated Extensions

4.207 Double-click on **Associated Extensions** to view the list of extensions used by the station being programmed. The Associated Extensions include the following:

- **Agent Help and Agent Help User-Keyed Extension**: The Agent Help feature allows a station user to request help from a supervisor during a call. The Agent Help Extension is the number that is called whenever the station user enters the Agent Help feature code. The Agent Help Extension can be a single station, or a hunt group, or it can be set to "None." This can be a device located on another node if it is programmed as an off-node device (see page 6-198). If desired, the station user can be allowed (or required) to enter the Agent Help Extension number, by setting the Agent Help User-Keyed Extension flag described on the next page.
- **Alternate Message Source**: If a station or application has an alternate message source and leaves a message at another station, the alternate message source will be called (instead of the station or application) when the message recipient responds to the message waiting indications. The alternate message source can be a hunt group, station, or application. This can be a device located on another node if it is programmed as an off-node device (see page 6-198).
- **Attendant**: This is the station, application, or hunt group (if any) that will serve as the attendant for this station. This can be a device located on another node if it is programmed as an off-node device (see page 6-198).
- **Data Port Extension**: The data port extension determines which data port the phone will use. This is the extension number of the data port itself, not the phone to which it is attached. The data port may be attached to one station and used by several other stations. The data port can be a device located on another node if it is programmed as an off-node device (see page 6-198). If the station has a DATA button, it shows the status of its associated data port. (Note that the phone does not have a DATA button by default. DATA buttons must be programmed in the database or by the user.) See page 5-269 in FEATURES for details.
- **Message Center**: The message center will receive messages after the Message Wait timer expires. This can be a device located on another node if it is programmed as an off-node device (see page 6-198).
- **Emergency Extension**: The emergency extension determines which trunk access (trunk, trunk group, or ARS) this station will use when the emergency number, 911 [999 in Europe] is dialed. This field defaults to Trunk Group 1 (92001). Stations cannot have direct emergency access to trunks on other nodes; they must use ARS to access off-node trunks.

NOTE: If using the DB Converter program to convert to a 7.0 or later system, the global Emergency Extension (System\Numbering Plan\Emergency\Emergency Extension) is automatically assigned as each station's Emergency Extension.

NOTICE

It shall be the responsibility of the entity or person(s) completing installation and maintenance of hardware or software described herein to research, comply with and be responsible for the specific governmental rules and regulations regarding Emergency Outgoing Access 911 [999] of the geographic location in which such functions are performed.

- **Outgoing Extension**: Determine which trunk access (trunk, trunk group, or ARS) this station will use when an idle CALL button or the OUTGOING button is pressed or when a system speed-dial number is selected for dialing before a trunk is selected. Defaults to Trunk Group 1 (92001). Stations cannot have direct outgoing access to trunks on other nodes; they must use ARS to access off-node trunks.
- **Transfer Recall Destination**: The transfer recall destination will receive this station's transfer recalls. This can be a device located on another node if it is programmed as an off-node device (see page 6-198).
- **Voice Mail**: This is the voice mail destination that this station will use for forwarding calls to voice mail. If this field is not programmed for an Executive Display, Professional Display, or Model 8560 Phone, the phone will not have the voice mail-related options on the feature button menu display.

4.208 To change the assignment of one of the Associated Stations, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the existing value. An option box appears.
2. Select **Change**. A window appears that asks you to select the type of device to include.
3. Select device type and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the device you want to assign as the associated extension and then select **Finish**. Your selection appears in the appropriate field.

NOTE: If the system administrator adds or changes ring-in devices using the administrator's phone, the system automatically changes the ring-in type to Multiple (even when only one device is selected). This occurs because the administrator can add multiple devices, which is prevented if the ring-in type is Single.

4.209 User-Keyed Extension: If desired, the station user can be allowed (or required) to enter the Agent Help Extension number. Enable this flag to allow the station user to enter the desired extension number. Or, disable the flag to automatically dial the Agent Help Extension programmed above (or disable the Agent Help feature for the station if there is no Agent Help Extension). Enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.) The operation of the feature is determined by the programmed combination of these flags, as shown in the chart below.

Table 6-22: *User-Keyed Extension Flag*

Agent Help	User-keyed Extension	Result
Extension number	Disabled	The Agent Help Extension is called when the feature code is entered.
Extension number	Enabled	The user is prompted to enter the desired number after the feature code is entered. If a number is not entered before the dialing timer expires, the programmed Agent Help Extension number is dialed automatically.
None	Enabled	The user is prompted to enter the desired number after the feature code is entered.
None	Disabled	The user hears reorder tone if the feature code is entered.

Attached Device

4.210 *(This field is used for digital phones only.)* Indicate the type of attached device if a DSS, Desktop Interface, or Desktop OAI IP is connected to the phone. Or, if nothing is connected, select **None**. To set this field, select the Attached Device current Value and use the drop-down list to scroll to the desired option. Then press **ENTER** or click outside the field to save your change.

Attached Device Bit Rate

4.211 *(This field is used for digital phones only.)* If the Attached Device is set for "Desktop Interface", you can set the bit rate of the PCDPM attached to the digital display phone. Valid rates are 300, 600, 1200, 2400, and 4800 (1200, 2400, and 4800 for a system serial port). This can be re-programmed by the phone user. To set this field, select the Attached Device Bit Rate current Value and use the drop-down list to scroll to the desired option. Then press **ENTER** or click outside the field to save your change.

Associated Outside Number

4.212 *(This field is used for B-channel stations only.)* This field is used to override the dialed outside number in an ISDN setup request message to a B-channel station. Some B-channel stations expect the same dialed outside number, or it will not accept the ISDN call.

4.213 With a U.S. system, on a BRI trunk using DNIS, the ISDN information in the System sends the dialed number DN (Directory Number) or DNIS to the BRI station port, then the BRI station port sends this DN or DNIS number to the ISDN device. The ISDN device must have the matching number programmed in its DN table or directory table, or the call will be rejected.

4.214 With a European system, on an ISDN trunk using DDI or DNIS, the ISDN information in the System sends the dialed number (DDI or DNIS) to the BRI station port, then the BRI station port sends this DDI or DNIS number to the ISDN device. The ISDN device must have the matching number programmed in its DDI table or directory table, or the call will be rejected.

4.215 The DN [DDI] or DNIS number that was sent from the BRI station port to the ISDN device can be substituted by using the Associated Outside Number field. This feature will take the incoming DN [DDI] or DNIS and remove it and then send the number programmed in this field. Normally, this field should be left blank. If for some reason the ISDN device that is connected to a BRI station port needs a different DN [DDI] or DNIS number other than the one that is sent from the ISDN message on the D-channel, then this is where the number needs to be programmed.

Audio for Calls Camped onto this Device

4.216 Defines the audio that a caller hears when camped-on to the station. To program this field, use the drop-down list to select the type of audio and press **ENTER**. Your options are Silence, Ticktone, Ringback, or Board. If you select **Board**, specify the music source in the Extended Value field. The default is *Board*, and the default extended value is *01:08 Central Processor*.

Audio for Calls Holding for this Device

4.217 Defines the audio that a caller hears when placed on hold at the station. To program this field, use the drop-down list to select the type of audio and press **ENTER**. Your options are Silence, Ticktone, Ringback, or Board. If you select **Board**, specify the music source in the Extended Value field. The default is *Board*, and the default extended value is *01:08 Central Processor*.

Audio for Calls Ringing this Device

4.218 Defines the audio that a caller hears when ringing the station. To program this field, use the drop-down list to select the type of audio and press **ENTER**. Your options are Silence, Ticktone, Ringback, or Board. If you select **Board**, specify the music source in the Extended Value field. The default is *Ringback*.

Call Logging

4.219 Call Logging is available on Axxess systems running version 9.1 or later. See page 5-180 for feature descriptions.

4.220 To program Call Logging:

1. In System\Devices and Feature Codes\Stations\Local, double-click the desired endpoint. Executone phones and single-line phones are not supported.
2. Double-click **Call Logging**. Program the following flags:
   - **Allow Duplicate Entries**: Allows you to store the duplicated calls in a call logs list. If the flag is enabled, all duplicates are allowed. If it is disabled, only the most recent entry is saved in the call logs list. For example, if a user dialed 1000, then 1001, and then 1000 again, if the duplicates flag was turned off you would see two entries in the dialed call logs list when you access it (1000 and 1001). However, if allow duplicates is turned on, you would see (1000, 1001, 1000) in the dialed call logs list for that endpoint. The Allow Duplicate Entries is set to *Yes* by default.

— **Enable Call Logging**: Allows you to enable the Call Logging feature. It is set to *Yes* by default. When this field is changed to No, the other two Call Logging fields display a red "X." This indicates all features associated with Call Logging are also disabled. The screen to the right shows the Call Logging options for a digital endpoint.

— **Maximum Number of Entries**: Allows you to set the maximum number of entries that can be stored in a call logs list. The valid range is 0 - 20 entries (the default value is *10*).

4.221 The corresponding feature code called **Call Logging (feature code 333)** is located under System\Devices and Feature Codes\Feature Codes. This feature allows digital/IP endpoints and IP softphone users to view the missed, received, and dialed call logs on their endpoint. If the endpoint does not support Call Logging or if the Enable Call Logging flag is not enabled (see above), when the endpoint enters this feature code, the endpoint displays an error message and receives a corresponding error tone. Non-display endpoints will only receive the error tone.

NOTE: For U.S. systems, a 1 is pre-pended to return CO calls by default. If needed, be sure to allow local area codes in User Group 1. This default action of the Call Logging feature may cause local calls that do not require 1+ dialing to be denied at the CO. To prevent such local area codes from being denied, program them in the Allowed Area Codes field under System\Numbering Plan\User Groups.

Calling Party Name

4.222 This field is similar to the existing "Calling Party Number" field, shown below. It is used only for ISDN calls to the public network (non-private networking). If this field is programmed, the system may use this information for the outgoing ISDN setup request message. You can program up to 20 alphanumeric characters in the box. This field is not used when the "Send Station Caller ID to Attached PBX," "Propagate Original Caller ID," and/or "Propagate Original Caller ID on Transfer (for voice mail)" flags are enabled. See "Caller ID Forwarding" on page 5-77 for additional information.

Calling Party Number

4.223 Each station can be programmed to send an identifying number when a call is placed. This is called the "Calling Party Number." (In the U.S., this information is required for emergency 911 calls in some states.) You can program any number up to 48 digits in the Calling Party Number field. However, check with your service provider to determine their specific requirements for this field. This number will be sent in the ISDN setup message in the Calling Party Number Information Element. In addition, the System will also send the extension number of the station in the Calling Party Number Subaddress Information Element. The CO should ignore this information element if it does not support it. To enter a number, select the current Value and enter the number in the text box. Then press **ENTER** or select another field to save the change.

NOTE: There is no default number for this field. It is up to you to supply the correct Emergency Calling Party Number for each station.

Day and Night Classes of Service

4.224 Each station has Classes of Service (COS) which restrict or allow certain dialing patterns from being dialed on a call. See page 6-281 for a complete explanation of the Classes of Service. To program the toll restrictions double-click on **Classes of Service**, you have the option of choosing **Day** or **Night**. Double-click on the desired time period to view a list of current classes of service.

4.225 To add a class of service:

1. Right-click in the window and select **Add To List**.
2. A window appears that allows you to select the types of toll restrictions you want to program. Select **ARS Only & Deny Area/Office** and/or **Classes of service** and then select **Next**.
3. Another window appears that shows you a list of available classes of service. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the desired class(es) of service and then select **Add Items**. The selected classes of service appear in the list. Select **Finish** to exit.
5. *(U.S. only) If you selected Deny Area/Office class of service,* the station must also be assigned to a User Group. To change the user group, right-click on **User Group** and select **Change User Group**. In the first window that appears, select **User Groups** and then **Next**. In the next window, select the desired user group and then select **Finish** to exit and save the change.

4.226 To delete one or more class of service: Highlight the item(s). Then right-click and select **Remove Selected Items**.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items.

DSS Consoles

4.227 *(Executone phones only)* You can associate up to three DSS Consoles with Executone phones. Double-click on DSS Consoles to view the current DSS Console list.

4.228 To add DSS Consoles:

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To DSS Consoles List**. A window appears that asks you to select the type of device to include.
3. Select **Next**. A list appears. (You can view them in a list by selecting **List** or view details by selecting **Details**.)

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the DSS Consoles you want to use and then select **Add Items**. When you have added all the Consoles necessary (up to 3), select **Finish**. Your selections appear in the list.

NOTE: If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

4.229 To delete DSS Consoles: To delete one or more item from the list, highlight them and right-click. Select **Remove Selected Items**.

Emergency Calling Party Number

4.230 This field allows subscribers to specify the Caller ID [CLID] to use when an endpoint makes an outgoing emergency call. Call Processing propagates the appropriate calling party number/name to the PSTN.

4.231 To program the Emergency Calling Party Number for a station:

1. Select the current Value in the Emergency Calling Party Number field, and then type the number (up to 48 numeric characters) in the text box.
2. Press **ENTER** or select another field to save the change.

Flags

4.232 You can enable a flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.) Some flags require you to choose between two setting (such as Hunt Group Remove/Replace). For those flags, select the current Value and use the scroll box to select the desired value. Then press **ENTER** or select another field to save your change. The station flags that can be programmed include the following:

- **Administrator**: *(Inter-Tel phones only)* A station can be assigned as an administrator to permit use of the administrator feature codes at that station. To assign the station as an administrator, enable this flag.
- **All Transient Displays**: *(Inter-Tel phones only)* When this flag is enabled, it will allow all transient call handling displays to appear on the phone display. If it is disabled, the CALL TRANSFERRED TO XXX and MESSAGE LEFT FOR XXX displays will not appear. In the default state it is *enabled.*
- **Alternate Hold Timer**: For users' convenience, the system has two Hold timers: *Hold* and *Hold - Alternate*. In the default state, the Alternate timer is set for a longer time period than the Hold timer. However, both timers are programmable. The "Alternate Hold Timer" station flag determines which timer each station will use. If the flag is disabled, the station uses the Hold timer. If it is enabled, the station uses the Alternate timer. In the default state, the flag is *disabled.*
- **Alternate Keymap**: *(Inter-Tel phones only)* If an alternate keymap will be used on the phone, it can be enabled through programming or by the phone user. This flag shows the current state of the alternate keymap (enabled or disabled). In the default state, the alternate keymap is *disabled* and the standard keymap is used. (The phone user can switch keymaps at any time by entering a feature code.)
- **Attendant**: A station can be designated as an attendant. It will then appear in the list of available stations when you are assigning Associated Extensions for other stations. *If you attempt to remove attendant status from the primary attendant,* you will see a warning message. (The primary attendant can only be changed using the programming fields described on page 6-315.)

NOTE: Message centers and Attendant stations can serve stations located on other nodes. However, because there is not a common database for all nodes, the message center and attendant assignments for the off-node stations must be performed using Associated Extensions for the individual stations on the remote node(s). Note that because the local node cannot see the databases of the other nodes, it is up to you to keep track of which attendants and message centers are serving off-node devices. Be careful not to remove those special purpose stations, leaving the off-node devices without an attendant or message center.

- **Audio Diagnostics**: The Audio Diagnostics flag enables/disables the Audio Diagnostics feature (see page 5-318) for a station. By default, this flag is disabled. To enable the Audio Diagnostics feature for the station, set this flag to **Yes**.
- **Automatic Answer CO or Intercom**: *(Inter-Tel phones only)* The Automatic Answer features can be enabled so that ringing outside and/or intercom calls are answered by simply lifting the handset. If automatic answer is disabled, the user must press a button to answer the call. (This can also be programmed by the station user.) In the default state, automatic access for both IC and CO are *enabled.*
- **Caller ID [CLIP] CO or Intercom**: *(Single-line sets only)* If the Caller ID [CLIP] IC and/or CO flags are enabled and there are available transmitters on the DKSC-16+ or OPC DSPs, Caller ID [CLIP] information is sent to single-line sets for intercom and outside calls. By default, these flags are *disabled*. (See page 5-73.)

NOTE: Because some single-line devices require an extended ring cadence to receive caller ID [CLIP] information, you must first enable the Extended Ring Cadence flag (see page 6-211) to configure caller ID for IC calls. Without the Extended Ring Cadence set to **Yes**, the Caller ID - IC flag cannot be set.

- **Camp-On Indications**: The tone and display that signal a user when a call has camped on can be enabled or disabled. (However, the lamp in the CALL or trunk buttons cannot be disabled.) In the default state, the tones are *enabled.*

NOTE: Hunt group camp-on tones can be disabled without disabling station camp-on tones. See the Hunt Group Camp-On Audio Indications flag.

- **Camp-On to ARS**: The station can be permitted to camp on to a trunk when using Automatic Route Selection (ARS), or camp on to ARS can be disabled. In the default state, camp on to ARS is *enabled.*
- **CO/IC Reseize**: *(Inter-Tel phones only)* The ability to immediately reseize a trunk or intercom channel, after disconnecting from a call, can be enabled. In the default state, it is *disabled.*
- **DID/E&M [DDI in Europe] Receive Busy Instead Of Camp-On**: E&M and DID callers can receive busy signal, or receive ringback and camp on, when calling a busy station. If this feature is enabled, the caller will hear busy signals when the called station is busy, and will not camp on. In the default state, busy tones are *disabled* and the callers will hear ringback while camped on to the called station.
- **Different Alerting Cadence Intercom/CO Call**: To allow users to easily differentiate between ringing intercom and outside calls, separate cadences can be enabled. In the default state, the different cadences are *enabled.*
- **Different Ringback Cadence Intercom/CO Call**: The ringback heard when a user places a call can be different for intercom and outside calls, or both types of calls can use the CO ringback cadence. In the default state, the different cadences are *enabled.*

- **Display Outside Party Name**: *(Inter-Tel phones only)* If this flag is enabled, the phone user can toggle the outside party name and number when connected to a CO call with outside party information. The phone user enters the Display Outside Party Name On/Off feature code (379). In addition, the enhanced ring-in displays will provide the user with more information such as both Caller ID [CLIP] name and number if available, or tell the user if a Caller ID [CLIP] number is blocked or out-of-area. This is a keyset-only station flag. In the default state, this flag is *enabled*. See page 5-74 for details about advanced ring-in display.
- **DKTS Alternate Transient Display Timer**: *(Inter-Tel phones only)* This flag, if enabled, allows the length of time that transient displays will remain visible to be controlled by the "DKTS Alternate Transient Display" timer. If this flag is disabled, transient displays will remain visible for 5 seconds. In the default state, it is *disabled.*
- **Do-Not-Disturb Allowed**: This option designates whether the station can be placed in do-not-disturb. In the default state, it is *enabled.*
- **Do-Not-Disturb Override**: *(Inter-Tel phones only)* The station can be given do-not-disturb override privilege that allows the user to place an intercom call to a station in do-not-disturb. In the default state, it is *disabled.*
- **Expanded CO Call Information On Displays**: *(Inter-Tel phones only)* This flag determines whether call information (trunk name or caller information) appears on the phone. If it is enabled, the Outside Call Party Information Has Priority flag (described on the next page) determines what appears. If it is disabled, the programmed trunk username will appear on the display. In the default state, this flag is *enabled.*
- **Extended Ring Cadence**: *(Single-line station only)* Extended ring cadence can be enabled or disabled. The extended setting lengthens the duration of the ring signal to meet the requirements of OPX and/or repeater applications. Because some single-line devices require an extended ring cadence to receive caller ID [CLIP] information, you must first enable the Extended Ring Cadence flag to configure caller ID [CLIP] for IC calls. Without the Extended Ring Cadence set to Yes, the Caller ID - IC flag cannot be set. The default setting is *disabled*. (See also FEATURES, page 5-149.)
- **Handsfree On/Off**: *(Inter-Tel phones only)* The station can be programmed to receive intercom calls handsfree, or handsfree answering can be disabled. (This can also be programmed at the phone.) In the default state, handsfree answering is *enabled.*

NOTE: This flag must be disabled for all SIP endpoints.

- **Headset Connect Tone**: Some headsets have a power-saver mode that disables the headphone after a period of silence. To prevent these headsets from missing portions of incoming calls, enabling this flag allows the phone to generate a tone that activates the headset before connecting to an incoming call. In the default state, it is disabled.
- **Headset On/Off**: *(Inter-Tel phones only)* If a headset will be used on the phone, it can be enabled through programming or by the phone user. In the default state, it is *disabled.*
- **Hunt Group Remove/Replace**: The station Hunt Group Remove/Replace feature can be programmed using this window or using the feature code at the station. In the default state, this feature is set for *replace* which allows the station to receive hunt group calls.
- **Hunt Group Camp-On Audio Indications**: The tone that signals a hunt group member when a call has camped on to the hunt group can be enabled or disabled. (However, the camp-on display and lamp in the CALL or trunk buttons cannot be disabled.) In the default state, the tones are *enabled.*

NOTE: If all station camp-on tones are disabled using the Camp-On Indications flag, hunt group camp-on tone will be disabled regardless of the Hunt Group Camp-On Audio Indications flag setting.

- **Immediate OHVA Transmit**: *(Inter-Tel phones only)* Determine if the station will have immediate off-hook voice announce capability enabled. If it is not enabled, the OHVA call does not go through until the Off-Hook Voice Announce Screening timer expires. In the default state, this option is *disabled*.
- **Manual Forward to Public Network**: When disabled, this flag prevents the station from manually forwarding calls to the public network. This prevents users from calling in to a local station and reaching toll numbers via forwarding. In the default state it is *enabled*.
- **Message Lamp**: *(Single-line station only)* The SLCs will support message lamps on single-line sets if the optional Lamp Option Card is installed on the SLC. To enable a message waiting lamp on a specific single-line station, the station must have the Message Lamp station flag enabled. In the default state it is *disabled*.
- **OHVA Receive/Transmit**: *(The OHVA Receive prompt appears for Inter-Tel phones only.)* Any station can be programmed to place off-hook voice announcements. Inter-Tel phones can also be programmed to receive off-hook voice announce calls. In the default state, receive and transmit are *disabled.* Note that the system-wide flag must also be enabled to allow the feature to be used.
- **OPX Gain Levels**: *(T1 OPX and Single-line station only)* When a single-line circuit is used as an off-premises extension, this option can be enabled to set receive and transmit gain levels to better suit the OPX application. In the default state, OPX gain levels are *disabled* (the gain levels are set for on-premises use). The OPX Gain increases the db level by 3db on both transmit and receive.
- **Outside Party Call Information Has Priority**: *(Inter-Tel phones only)* If the Expanded CO Call Information flag is enabled (as described on the previous page), this flag determines what information appears at the phone. If enabled, any call that is received on a trunk that provides outside caller information, Caller ID [CLIP] or ANI, will be identified on the phone display with the caller information. If disabled, the display will show the DID or DNIS information for the call (if available). In the default state, is it *enabled.*
- **Page Mode**: *(Inter-Tel phones only)* If desired, pages to the station can be enabled or disabled through programming or by the phone user (page remove/replace). In the default state, pages are *enabled*.
- **Propagate Original Caller ID on Transfer** (for endpoints): When this flag is enabled in the Flags folder, if the endpoint is on a call with an outside trunk that had caller ID information, when the endpoint performs a transfer back to the PSTN, the endpoint will forward the caller ID. Note that to forward the caller ID to the PSTN, the eventual trunk must be ISDN and the "Propagate Original Caller ID" flag in its CO trunk group must be set to **Yes**. The default of this flag is set to *No* (which means do not forward caller ID on transfer PSTN calls). For complete information about Caller ID Forwarding, see page 5-77.
- **Receive Busy Instead Of Do-Not-Disturb**: The tone that a caller hears when a station is in do-not-disturb can be a busy signal or a do-not-disturb signal. If this feature is enabled, the caller will hear busy signals when the station is in do-not-disturb, but the caller cannot camp on. In the default state, busy tones are *disabled.*

- **Redial Mode**: Station users can use *last number dialed* or *last number saved* redial modes. This option can be programmed using this window only, it cannot be changed by the station user. In the default state, the stations are set for *dialed* mode.
- **Ring Intercom Always**: When this feature is enabled, all calls placed to phones from this station are received as private calls. The feature can be enabled or disabled through programming or by the phone user. In the default state, this feature is *disabled.*
- **Send Alert Burst To Headset**: *(Inter-Tel phones only)* This flag, when enabled, will send a single tone to a headset instead of normal phone ring tones when a call is ringing at the station (non-handsfree intercom call, outside call, queue callback, or reminder message). By default the flag is disabled.
- **Send T1 OPX Disconnect Flash**: *(T1 OPX Single-line station only)* This flag, when enabled, sends a proprietary disconnect signal from the T1 OPX to the loop start trunk. (The "A" bit is toggled high for the duration of the SL Disconnect Flash Duration timer.) At default, this flag is *disabled.* To remind you, a warning will be displayed in DB Programming whenever the flag is changed. The warning states that the flag will have no effect unless the correct T1 firmware is installed on the card. If your T1 card does not have the proper firmware, there is an upgrade kit available. Although you can enable this flag for all single lines, the flag is ignored unless the single-line set is a T1 OPX.

  NOTE: This flag does not affect T1 channels configured for Loop Start. Therefore, an Inter-Tel system can be on the receiving end with T1 Loop Start channels and will recognize the disconnect. However, it cannot ***send*** the disconnect.

- **Standard Tone Frequencies**: The busy tones and dial tones can be pure system tones (as described in FEATURES, page 5-172) or they can be changed to more closely match the telephone company's standard tones. In the default state, standard tones are *enabled* (allowing standard telephone company type tones).
- **System Forwarding**: When this option is enabled, calls to the station follow the system forwarding path that is programmed for the station (if any). This option can be enabled is the database or at the station. In the default state, it is *enabled.*
- **Transfer To Connect Allowed**: *(Inter-Tel phones only)* If this flag is enabled, an announced transfer call (including an announced transfer from voice mail) connects immediately with the destination once the transferring party hangs up. (If the flag is disabled, the recipient must press a CALL button to answer the transferred call). If the user has a headset enabled, or if the station user picks up the handset for the initial call and the transferring party disconnects, the transferred party will hear a single tone and then be immediately connected without any further action from the station user. Calls will not automatically connect in handsfree speakerphone mode; the user must press a CALL button to answer the call. This flag is *disabled* by default.
- **Transient Call Indication On Call Answer**: *(Inter-Tel phones only)* This flag determines whether the station user will see a call indication display when answering a call by pressing a secondary extension button or by reverse transferring. If enabled, the display will tell if the call was ringing, recalling, transferred, or holding at the other station. In the default state, it is *enabled.*

Forwarding Paths

4.233 See FEATURES, page 5-223 for more details on system forwarding. Each station can have up to three forwarding paths and there can be 200 different programmed paths in the system, numbered 001-200. Path 000 (No Forwarding Path) can be assigned to disable system forwarding for the station. Add or remove forwarding path as follows:

4.234 To add a forwarding path:

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To Forwarding Paths List**. A window appears that asks you to select the type of device to include.
3. Select **Forwarding Paths** and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

> **NOTE:** To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate paths and then select **Add Items**. When you have added all the paths you want select **Finish**. Your selections appear in the list.

4.235 To delete a path: To delete one or more path from the list, highlight them and then right-click. Select **Remove Selected Items**.

4.236 To program a specific Forwarding Path: Double-click on the forwarding path to view the following options:

- **Forwarding Path**: This information is shown for reference. If you would like to view the programming for the path, double-click on **Forwarding Paths** to jump to Station-Related programming.
- **Ring Principal Once**: If the system-forwarded station is a phone and the user wants to be notified each time a call is sent to the system forwarding path, this option can be enabled. The station user will hear a single ring tone each time a call is system forwarded and display phone will show CALL SENT TO FORWARD PATH. (In the default state, this is disabled.) Enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)
- **Station Conditions (Busy, DND, No Answer, and Immediate)**: The station can be programmed to use the path to forward calls when the station condition is busy, in do-not-disturb, and/or no answer. Or, calls can be forwarded immediately after they are received at the station. (In the default state, all but Immediate are selected.) Determine which station conditions will cause the calls to follow the forwarding path and enable them. Note that if any condition other than Immediate is selected, Immediate will appear dimmed and cannot be selected. To enable a flag select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

- **System Conditions (Day and Night Mode)**: The station can be programmed to use the forwarding path when the system is in day mode and/or night mode. (In the default state, both day and night mode are selected.) Indicate whether calls will follow the forwarding path in day and/or night mode by enabling the desired system condition(s). To enable a flag select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)
- **Forwarding Call Types**: The types of calls that can be forwarded are listed below. The forwarding station can be programmed to forward any or all of these call types. (In the default state, call routing, ring-in, DISA, and transferred calls are enabled.) Determine the types of calls that will follow the forwarding path by enabling each desired call type. To enable a flag select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)
  - **Call Routing Table**: Outside calls received through a call routing table, including DID and E&M [DDI in Europe] calls, but not including DISA calls
  - **CO Ring-In**: Ringing outside calls (including calls received through network connections)
  - **CO Transfer/AA/VM**: Transferred outside calls (including automated attendant and voice mail transfers and transfers from other nodes)
  - **CO Recall**: Recalling outside calls (including calls from other nodes)
  - **DISA**: DISA calls (including DISA calls received through a call routing table)
  - **IC Calls**: Intercom calls (including calls from other nodes)

NOTE: Call forwarding paths are checked in numerical order when they apply to the same call type (for example, multiple forwarding paths for outside calls). If one forwarding path is for the "No Answer" call condition, it should be last on the list of forwarding paths for the station. If No Answer is the condition in the first path, the system will always wait for the System Forward Initiate timer to expire before forwarding, even if the station is in do-not-disturb mode or busy.

IP Settings

4.237 *(IP endpoints and IP SLAs only)* After you have programmed the circuits and voice channels for the IP devices (including IP SLAs), you must program the IP settings. The IP Settings folder allows you to program the IP connection information for each device. The available fields are as follows:

4.238 Network Configuration (see below)

- Network Group (see page 6-182)
- Call Configuration (see page 6-185)
- NAT Address Type (see page 6-218)

4.239 Network Configuration: To program the IP settings, double click **Network Configuration** and configure the following fields:

NOTE: Those fields marked with an asterisk (*) indicates that the system does not set the information for the endpoints, even though the fields exist in DB Programming.

- ***MAC Address (Device ID for IP softphone)**: The MAC Address/Devices ID field displays the MAC address/Devices ID that was programmed when the device was assigned a circuit. If desired, you can edit the address. To edit the address, click the existing address. The Edit MAC Address/Devices ID dialog box appears. Enter the new address (do not include the colons) and click **OK**.

  NOTE: The Device ID functions similarly to the MAC address, but because the ID of the IP Softphone is not tied to a specific Network Interface Card (NIC) on the user's PC, the ID will not be a MAC address.

- ***Hostname**: The hostname is used to identify the IP device on the network. Enter a hostname, up to 15 characters, in the text box provided. Then press **ENTER** or click another field to save the changes. By default, this is blank.
- ***Static IP Address**: If the IP device will not receive an address from a BOOTP or DHCP server, you must identify a static IP address. To enter the address, click the current Value. The Edit Static IP Address dialog box appears. Enter the IP address (do not include the periods) and click **OK**. By default, this is 192.168.200.201.
- ***Static Gateway**: The Static Gateway field identifies the IP address of the gateway that is used to access the port. To enter the address, click the current Value. The Edit Static Gateway Address dialog box appears. Enter the IP address (do not include the periods) and click **OK**. By default, this is 192.168.200.201.
- ***Static Subnet Mask**: The Static Subnet Mask identifies the subnet where the IP device resides. To enter the address, click the current Value. The Edit Static Subnet Mask dialog box appears. Enter the address (do not include the periods) and click **OK**. By default, this is 255.255.255.0.

  NOTE: If the IP device is programmed to receive an IP address from a BOOTP or DHCP server, the Static IP Address, Static Gateway, and Static Subnet fields display a red X and do not affect device programming.

- ***Static WINS Server**: The Static WINS Server field identifies the IP address of the Windows Internet Naming Service (WINS) server that the device uses for name resolution. To enter the address, click the current Value. The Edit Static WINS Server dialog box appears. Enter the address (do not include the periods) and click **OK**. By default, this is 0.0.0.0.
- ***IP Address Assignment**: **BOOTP or DHCP**: The BOOTP and DHCP flags determine whether the system will allocate a dynamic IP address to the IP device. If these flags are enabled, the **Static IP Address**, **Static Gateway**, and **Static Subnet** fields are ignored, and the BOOTP or DHCP server is used to assign the IP information. If disabled, the device uses the IP address identified in the **Static IP Address** field. To disable a flag, select the current Value and then select the check box to remove the check mark. Then press **ENTER** or select another field to save the change. (To return the check mark and enable a flag, select the check box again.) By default, both of these flags are *enabled*.

NOTE: The v8.1 IPRC firmware does not have a BOOTP server. You can, however, use a third-party BOOTP server to obtain IP addresses.

- ***Remote Server IP Address**: The Remote Server IP Address identifies the IP address of the IPRC to which the device is connected. The IPRC that you program here must have the IP device mapped to one of its circuits; otherwise, the device will not be able to connect to the card. To enter the address, click the current Value. The Edit Remote Server IP Address dialog box appears. Enter the address (do not include the periods) and click **OK**. By default, this is 192.168.200.201.

NOTE: Although you can use this field to assign the IP device to a different IPRC, the device will not come online unless it is programmed as an IP device for that card (see page 6-148).

- ***Overwrite Self Programming**: The Overwrite Self Programming flag determines whether the device uses the settings in DB Programming or those programmed on the phone itself. If this option is enabled, the device uses the settings programmed in DB Programming. If disabled, the device uses the settings programmed at the device and ignores the settings in DB Programming. To disable the flag, select the current Value and then select the check box to remove the check mark. Then press **ENTER** or select another field to save the change. (To return the check mark and enable the flag, select the check box again.) By default, this is *enabled*.

NOTE: If the IP endpoint resides behind a firewall, enable this flag. If enabled, the system sends the IP information to the endpoint, making it easier for the phone to reconnect to the system in the event of a power failure, reset, etc.

- ***Telnet Server**: The Telnet Server flag determines whether you can connect to the IP device using a telnet connection. If enabled, the telnet session is available. If disabled, you will not be able to access the device via a telnet session. To disable the flag, select the current Value and then select the check box to remove the check mark. Then press **ENTER** or select another field to save the change. (To return the check mark and enable the flag, select the check box again.) By default, this is *enabled*.
- ***Web Server**: The Web Server flag determines whether you can connect to the IP device using a Web server. If enabled, the Web server is available. If disabled, you will not be able to access the device via a Web server. To disable the flag, select the current Value and then select the check box to remove the check mark. Then press **ENTER** or select another field to save the change. (To return the check mark and enable the flag, select the check box again.) By default, this is *enabled*.

NOTE: To upload software to the IP device using the Upload Utility, you must enable the Web Server option.

- **Audio RTP Type of Service**: The Audio RTP Type of Service field determines the amount of audio that the device inserts into the RTP Type of Service (ToS) field. The lower the number, the less latency there is. To change this value, use the scroll box to select the new value. Then press **ENTER** or click another field to save your change. The valid range is *0-255*, and the default is *165*. Inter-Tel recommends that you do not change this value.

- **Audio Stream Receive Port**: The Audio Stream Receive Port defines the first of 33 even port numbers that off-node IP devices use to transmit audio data to the IPRC. You must choose an even-numbered value. The base audio port number, plus 64 additional port numbers, cannot conflict with other port numbers the device uses (e.g., call control port). For example, if the base audio port number is 5000, no other port numbers on the IP device can fall in the range from 5000 to 5064. Changing the Audio Stream Receive Port resets the IP device. To change the port number, click the current Value and enter a new value in the text box. Then press **ENTER** or click another field to save your change. The valid range is *1026-65468*, and the default is *5004*.
- **Password**: You should program a password that is require for connecting to the IP device using a telnet or Web interface. The device also uses this password to connect to the IPRC. To enter or change a password, right-click the current Value and select **Edit Password**. The Edit Password dialog box appears. Enter the password, up to nine characters, and then confirm the password in the available text boxes. The password appears as asterisks (***). Then click **OK** to save the password. The password is case-sensitive.

  NOTICE
  For security purposes, Inter-Tel recommends that you program a password for all IP devices.

- **Call Control Timeout**: The Call Control Timeout field determines how long the system will wait (in seconds) between heartbeats before it determines that the IP device is offline. To change the number of seconds, use the scroll box to select the new value. Then press **ENTER** or click another field to save the changes. The valid range is *1-300* seconds, and the default is *15* seconds.
- **IP Terminal TCP Call Control Port**: The TCP Call Control Port identifies the port number that the device uses for call control. To change the port number, click the current Value and enter a new value in the text box. Then press **ENTER** or click another field to save your change. The valid range is *1024-65535*, and the default is *5004*.
- **IP Terminal General Purpose UDP Port**: The General Purpose UDP Port identifies the port number that the device uses for general purpose and broadcast messages. To change the port number, click the current Value and enter a new value in the text box. Then press **ENTER** or click another field to save your change. The range is *1024-65535*, and the default is *5567*.

4.240 **NAT Address Type**: The Axxess system supports traversing near-end NAT using statically assigned NAT addresses. This capability allows an Axxess system to be placed inside a NAT or firewall and still communicate with IP endpoints outside the NAT/firewall.

4.241 System v9.1 or later software can be programmed so that each IP endpoint could use either the Native address or the NAT address. In other words, the system could be programmed with corresponding IP addresses for communicating with each IP endpoint, whether the endpoint was located inside or outside the NAT/firewall. You can choose one of the following options:

- **Native**: Used when the endpoint is located inside the NAT (those that do not pass through near-end NAT).
- **NAT**: Used when the endpoint is located outside the NAT (those that do pass through near-end NAT)
- **Auto**: Allows the IP endpoints to be moved inside or outside the firewall (NAT) without programming intervention. This is the default value. See below for details.

NOTE: The “Auto” option is not applicable to SIP Trunks, MGCP Gateways, and MGCP Endpoints. You must manually select the NAT Type setting (Native or NAT).

NOTE: IP endpoints retain the NAT Address Type settings after an upgrade.

4.242 Automatic Detection for NAT Traversal: The Automatic NAT Detection feature takes near-end NAT traversal one step further. An IP endpoint designated with the **Auto** NAT Type in DB Programming automatically selects the appropriate NAT or Native IP address. This feature allows the IP endpoint to be moved from inside a NAT to outside and back again without programming intervention.

4.243 If the default **Auto** NAT Type is designated for an IP endpoint in DB Programming, the Auto NAT Detection feature determines the correct IP address to use when the endpoint comes online. After the NAT type is determined, the IP endpoint uses the setting for all calls. However, the NAT IP address associated with the system must be programmed manually in order for the NAT detection test to run.

4.244 With applicable licensing, Auto NAT Detection supports multi-protocol endpoint Models 8600, 8620, 8622, 8662, and 8690; hard IP endpoints, including Model 8660, Axxess/ Eclipse IP PhonePlus, and IP Single-Line Adapter (IP SLA); and, IP softphones, including the Model 8602 IP softphone.

4.245 Auto NAT Detection does not support the following endpoints. The NAT setting must be set manually for:

- Inter-Tel IP endpoints in SIP mode
- Inter-Tel wireless IP endpoint Models 8664, 8665, and 8668
- IP gateway trunks (SIP and MGCP)
- Model 8601 SoftPhone for Pocket PC
- Non-Inter-Tel SIP phones

4.246 See "Configuring Near-End NAT Traversal for IP Endpoints" on page A-14 for more information about NAT traversal for IP endpoints and Automatic NAT Detection. For programming information, see page A-15.

House Phone

4.247 The station can be designated as a house phone that automatically dials a predetermined number when the handset is lifted. To enable this flag and assign the station as a house phone, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

House Phone Day/Night Numbers

4.248 The assigned day number is dialed when the system is in day mode and the assigned night number is dialed when the system is in night mode. The day/night house phone numbers can also be programmed using the house phone station speed-dial locations. Select **Day** and/or **Night** and enter the extension or outside number (up to 16 digits, including pauses and hook-flashes/recalls) that will be dialed when the house phone is used during the day and/or night mode. Then press **ENTER** or select another field to save the change. You can use the extension number of an off-node device, if desired. These numbers are also station speed-dial locations codes 0 (day) and 1 (night).

Keymaps

4.249 The database contains maps of default values for button layouts. (These maps are programmed in Station-Related Information, as described on page 6-306.) Keyset and DSS keymaps are user-equippable and may be equipped as needed through DB Programming.

NOTE: The user cannot change the number of keymaps or remove keymap # 1 in the Keyset keymaps and DSS keymaps. This ensures that the system always has at least one usable keymap.

4.250 The default database contains only the non-removable keymaps (keymap #1) as mentioned above. Keymaps can be batch created as needed through DB Programming. Unnecessary keymaps may be removed by right-clicking and choosing **Delete**.

4.251 Determine the preferred keymap for each phone (maps 01-250) and each DSS/BLF-equipped station (maps 01-20). To determine the Standard, Alternate, and DSS keymaps that will be used by this station, double-click on **Keymaps**. Then do the following:

- Right-click on the Keymap value for the map you want to change. An option box appears.
- Select **Change Map Group**. A window appears that asks you to select the type of device to include.
- Select the appropriate map and then select **Next**. The list of maps appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
- Highlight the map you want to use and then select **Finish**. Your selection appears in the Keymap field.

NOTE: If using the new Model 8500, 8520, or 8560 Phones on a system running 3.0-6.x software, you must use a Standard or Executive keymap. Because the newer phones have more buttons than the keymap, some button functionality will be lost. See page 5-135 for details.

4.252 To see the programming for a specific keymap, double-click on it to jump to the Station-Related Information\Key Assignment programming area.

Language

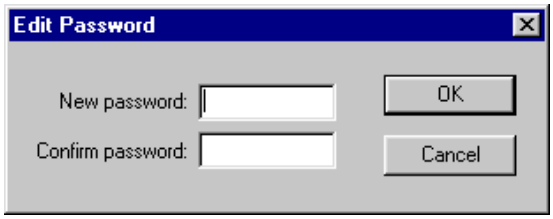
4.253 Determine which language will be used for the voice prompts and displays at this station. This field now can be set to any specific language so that the system can support more than two languages. The language choices are Use Primary Language, Use Secondary Language, American English, British English, Japanese, or Spanish. Note that Spanish voice prompts use Mexico City Spanish Dialect for VPU and EVMC, and American Spanish Dialect for EM. Select **Language** and scroll to the desired option. Then press **ENTER** or click outside the field to save your change.

Mailboxes

4.254 If the station being programmed serves as the message notification station for one or more mailboxes, the mailbox(es) will appear in the Mailboxes list. Select **Mailboxes** to see the list of mailboxes assigned to this station. You cannot program the mailbox here, but if you double-click on the mailbox, the screen will jump to the Mailbox programming area.

Password

4.255 This password is used for the Remote Programming feature. The station password can be up to 8 digits in length. The default password is the extension number of the station. The password can also be changed by entering the Program Station Password feature code at the station or when using the Remote Programming feature (described in detail on page 5-262). To program a password for the station double-click on **Password**. A window appears as shown below. Enter the password in the New Password box. The digits will not appear on the screen when typed; they will appear as asterisks (***). Then retype the password exactly as before in the Confirm Password box. Then select **OK** to exit and save the password. If the entered passwords match, you will return to the Password field. If not, you must re-enter the new password and verify it again. If you make a mistake while entering the password or want to leave it unchanged, select **Cancel**.



NOTE: To prevent unauthorized use of call forward to the public network, ***all*** stations using Remote Programming should have a password. To make the passwords difficult to guess, they should not match the extension number or consist of one digit repeated several times.

Programmable Keys

4.256 You determine the default values of the user-programmable buttons for each station. Select **Programmable Keys** to see a list of programmable buttons 1-35 and their current Values. To change the Key Type, Selection, or Ring When field for a programmable button at this station, do the following:

- **Key Type**: Select the current Key Type for the button and scroll through the drop-down list box until the desired key type is highlighted.
- **Selection**: The Selection programming required for each of the key types are as follows:
  - — **DSS/BLF Key**: Select the circuit or extension number of the station that will appear under the selected DSS/BLF button.
  - — **Feature Key**: Select the feature code that will be entered when this button is pressed.
  - — **Hunt Group Key**: Select the pilot number of the desired hunt group.
  - — **Intercom (IC) Key**: No further programming is necessary. This button will be used for seizing an intercom channel or answering a waiting intercom call.
  - — **Page Zone Key**: Select the page zone 0-9 or 0-49 that will be used by selecting the page zone access code 9600-9609 or 9600-9649 (respectively).
  - — **Secondary Extension Key**: Select the primary station that will be associated with this secondary extension button.
  - — **Station Speed Dial Key**: Select the station speed-dial location (0-9) that will be assigned.

— **System Speed Dial Key**: Select the system speed-dial location (000-999 or 0000-4999) that will be assigned.

— **Trunk Key**: Select the individual trunk that will be selected when this button is pressed.

— **Trunk Group Key**: Select the trunk group that will be selected when this button is pressed.

— **Undefined Key**: No further programming is necessary. This button can be programmed by the user.

To program the selection, right-click the Selection value and do the following:

a. Select **Change Selection**. A window appears that asks you to select the type to include.

b. Select the type (feature code, station, etc.) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

c. Highlight the appropriate item and then select **Finish**. Your selection appears in the Selection field.

- **Ring When**: This field allows the station to receive a burst of ringing when a certain number of calls are present at a primary station, and at least one call is ringing or camped on. The ring burst will repeat periodically as long as the programmed number of calls are present at a primary station. (This period is determined by the DKTS Secondary Extension Key Alerting Tone timer.) If this field is set to 0, the secondary station will never receive the burst of ringing. To set this number, select the current Value and scroll to or enter the new value in the box. Then press **ENTER** or select another field to save the change.

Record-A-Call

4.257 *The Record-A-Call software license is required, and a Record-A-Call application must be created in the Voice Processor database.* There are three fields for the Record-A-Call feature: Mailbox, Mailbox User-Keyed Extension Flag, and Application. Programming for each is described below. The application and mailbox might not be on the same node as the station. If so, the Voice Processor applications must be programmed as off-node devices on the local node and the station must be an off-node device on the Voice Processor's node (see page 6-198).

4.258 Determine which mailbox, if any, will be dialed automatically when the Record-A-Call feature is used. The Record-A-Call Mailbox can be set to "This Station's Associated Mailbox" to call the mailbox assigned to that station, or it can be set to any valid mailbox number. If you do not want a mailbox number dialed automatically when the Record-A-Call feature is used at this station, enable the User-Keyed Mailbox flag. This overrides the automatic entry and allows the station user to enter the desired mailbox number. The operation of the feature is determined by the programmed combination of these flags, as shown in the chart below.

Table 6-23: *Record-A-Call Flag*

Record-a-call Mailbox	User-keyed Mailbox	Result
"Associated" or mailbox number	Disabled	The Record-A-Call Mailbox is automatically called when the feature code is entered.
"Associated" or mailbox number	Enabled	The user is prompted to enter the desired mailbox number after the Record-A-Call feature code is entered. If a number is not entered before the dialing timer expires, the programmed Record-A-Call Mail box number is dialed automatically.

- **Mailbox**: To determine the mailbox, use one of the following methods:

  Method A

  a. Select the current Value and enter the new value in the text box.

  b. Press **ENTER**. A screen appears showing you what is associated with the number you entered.

  c. Select **OK**. The new number appears in the field.

  Method B

  a. Right-click on the current Value. An option box appears.

  b. Select **Change Mailbox**. A window appears that asks you to select the type of mailbox to include.

  c. Select **Mailbox**, **This Station's Associated Mailbox**, **Off-Node Mailbox,** or **Unassociated Mailbox Off-Node Device** and then click **Next**. The list of mailboxes appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

  d. Highlight the desired mailbox and then select **Finish**. Your selection appears in the Mailbox field.

- **Mailbox User-Keyed Extension**: To set the Mailbox User-Keyed Extension flag, select its current Value. To enable the flag, select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

- **Record-A-Call Application**: The Voice Processor must have one or more applications created for the Record-A-Call feature. In this programming field, you can choose the application that will be used by the station. (If you choose **None**, the station will not have access to the Record-A-Call feature.) To set the Application, use one of the following methods:

Method A

a. Select the current Value and enter the new value in the text box.

b. Press **ENTER**. A screen appears showing you what is associated with the number you entered.

c. Select **OK**. The new number appears in the field.

Method B

a. Right-click on the current Value. An option box appears.

b. Select **Change Application**. A window appears that asks you to select the type of application to include.

c. Select **None**, **Record-A-Call**, or **Off-Node Record-A-Call** and then select **Next**. The list of applications appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

d. Highlight the desired application and then select **Finish**. Your selection appears in the Application field.

Timers

4.259 *(European systems only)* The phone system now supports the transmission of CLIP to single-line sets in Europe. This feature uses the calling party information that the system cabinet receives from the local network provider. After programmed, on-hook single-line sets display the calling party's information when receiving an incoming outside call. The CLIP information is also displayed if the single-line set receives a transferred call from another station that has calling party information.

NOTE: CLIP is not transmitted to single-line adapters (SLAs).

4.260 Because Caller ID transmitters are required to send the information to the single-line set, you must upgrade the firmware on a 16-port Digital Keyset Card (DKSC-16+) that has an available Digital Signal Processor (DSP). Each DSP on a DKSC-16+ provides four Caller ID transmitters. (The Call Processing Card does not support Caller ID transmitters.) In addition, you must install the new SLC-16+. If you do not install the new SLC-16+, single-line sets will not support CLIP, even if there are available DSP resources.

4.261 To use CLIP for single-line sets, you must enable the Caller ID flag (System\Devices and Feature Codes\Stations\Local\<*Single-Line extension*>\Flags). You must also program the following timers:

- **Caller ID Line Reversal Timer**: Enter the number of milliseconds that the system reverses the line polarity so that the single-line set can receive Caller ID information. This value may differ for various phones. The valid range is 0 - 200 ms, and the default is *30 ms*.
- **Caller ID Alerting Tone Timer**: Enter the number of milliseconds that the system will send Caller ID alert tones to the single-line set. These tones are not audible to the end user and are used strictly to notify the phone that information will be transmitted. This value may differ for various phones. The valid range is 88 - 110 ms, and the default is *100 ms*.

Secondary Language

4.262 There is a new field added to each station in the system called "Secondary Language." This field corresponds to the Change Language feature (301). This feature used to toggle between the System Primary Language and the System Secondary language. It now toggles between the System Primary Language and the station's Secondary Language, or can specify a language. This allows any station in the system to have its own secondary language or use the System Secondary Language, giving the system the ability to support more than just two languages.

4.263 If a station's Secondary Language field is programmed to be Use Primary Language the Change Language feature will do nothing because the station will toggle between the System Primary Language and the station's Secondary Language, which is the System Primary Language.

4.264 If a station's Secondary Language field is programmed to be Use Secondary Language the Change Language feature will act like version 5.2 and toggle between the System Primary Language and the station's Secondary Language, which is the System Secondary Language. This is the system default.

4.265 If a station's Secondary Language field is programmed to be Japanese the Change Language feature will toggle between the System Primary Language and the station's Secondary Language, which is Japanese.

4.266 The Language field for stations indicates what language the station is currently set to. This field used to toggle between the Use Primary Language and Use Secondary Language. It can be set to any specific language along with the Use Primary Language and Use Secondary Language. The side-effect of changing the Language field is that if you change it to Japanese and the station's Secondary Language field is set to Spanish (Mexico City Spanish Dialect for VPU and EVMC, American Spanish Dialect for EM), then the user will have no way to get back to Japanese if they enter the Change Language feature code. This is because the first time the user enters the feature code, the system will toggle the station to the Use Primary Language. The next time the user enters the feature code, the system will toggle the language to the station's Secondary Language, which is Spanish. To avoid this situation, simply change the station's Secondary Language to Japanese and the station's Language field to Japanese.

NOTE: Setting a station's Secondary Language field to a specific language will not work across nodes unless all nodes in the system are at least 5.2F6 or newer since the older systems do not understand specific languages and expect to receive only Primary Language or Secondary Language values.

Speakerphone Type

4.267 This flag affects Digital Phones differently:

- Executive Display or Professional Display Phone: To enable the keyset's integrated speakerphone, to be used in standard and/or enhanced mode, select either of the speakerphone options (Enhanced or Standard). Selecting Standard does not prevent the Executive Display Phone user from using the enhanced speakerphone. To disable the Executive Display Phone's speakerphone, select **None**.
- Standard Display, Basic Digital, Associate Display, Basic, Model 8560, Model 8520, or Model 8500 Phone: If the station will have access to shared standard speakerphone resources, select Standard. For access to enhanced and standard speakerphone resources, select Enhanced. To prevent a Standard Display, Basic Digital, Model 8500, Model 8520, or Model 8560 Phone station from using shared speakerphone resources, select **Disabled**.

Speakerphone Type

4.268 *(This field is used for digital phones only; analog phones do not use the shared speakerphone resources.)* This flag affects digital phones differently:

- **Executive or Professional Display Phone**: To enable the phone's integrated speakerphone, to be used in standard and/or enhanced mode, select either of the speakerphone options (Enhanced or Standard). Selecting Standard does not prevent the Executive or Professional Display Phone user from using the enhanced speakerphone. To disable the Executive or Professional Display Phone's speakerphone, select **None**.
- **Other Digital Phones**: If the station will have access to shared standard speakerphone resources, select **Standard**. For access to enhanced and standard speakerphone resources, select **Enhanced**. To prevent any of these phone stations from using shared speakerphone resources, select **Disabled**.

Validated Flag

4.269 If a type of forced account code was assigned in the Account Code Type field, you can set the Validated flag to check the code entered (flag enabled) or accept any code (flag disabled). Enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

O. SYSTEM SERIAL PORTS

4.270 This section is used for determining the functions of the system serial ports. Serial ports can be used for the following purposes:

- Voice Processor port
- System OAI ports
- SMDR port and backup port
- Message printing port and backup port

4.271 Serial ports are located on the CPCs and CPU cards. They can also be configured on the OPC and on digital display phones with PCDPMs. There can be up to 32 system serial ports.

4.272 Each serial port has the following programmable fields:

- Description
- System OAI Ports
- Bit Rate
- Hardware and Software Handshaking
- Autobaud

4.273 The Port number and Address are not programmable. They are shown for your reference. Program the fields as follows:

4.274 Description: If desired, you can give the port a description to designate its assigned connection. To enter the description, select the current Value and enter the information in the text box. Then press **ENTER** or select another field to save the change.

4.275 System OAI Ports: This field has three possible settings: Level 2, Level 2 w/EFP, and None. The serial ports in the "None" category are available for use by the other serial port functions. The other two levels determine the system OAI protocol level used by the serial ports, as described below. Refer to the *System OAI Protocol Manual* for further information.

NOTE: If available, a socket connection is recommended for System OAI Level 2. (System versions 6.0 and later no longer support System OAI Level 1.)

4.276 To set the System OAI Port level, select the current Value and use the drop-down list to scroll to the desired setting. Then press **ENTER** or select another field to save the change.

- **Level 2**: The bidirectional support enhanced protocol is running, using the RS232 interface.
- **Level 2 w/EFP**: The TSAPI support enhanced protocol is running, with Inter-Tel EFP as the second layer protocol instead of RS232.

NOTE: These ports cannot be used as system programming serial ports even though you may be able to log on to them as such (the monitor displays "Awaiting ASAI Connection" and eventually times out). Also, ***do not*** assign all available ports to System OAI (leave at least one port for system programming).

4.277 Bit Rate: This field is used to set the bit per second (bps) rate of the serial port to match the configuration of external devices to which they will be attached (such as printers, PCs, etc.). To set the bit rate, select the current Value and use the drop-down list to scroll to the desired setting. Then press **ENTER** or select another field to save the change.

4.278 Hardware Handshaking: If the device that is connected to the selected port supports handshaking on the RS232 connector, enable the Hardware Handshaking flag. See Appendix B — RS232, for information about handshaking. Enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

4.279 Software Handshaking: If the device that is connected to the selected port supports software handshaking, enable the Software Handshaking flag. See Appendix B — RS232 for more information. Enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

4.280 Autobaud: To allow the bit rate to be automatically adjusted, enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

P. CO TRUNK GROUPS

4.281 When you select CO Trunk Groups, the right side of the screen displays the list of trunk groups on the Local node. At this level, you can program the description and username for each trunk group. To program the other fields for an individual trunk group, double-click on it to view these options:

- Trunks (see page 6-230)
- Multiple Ring-In (see page 6-230)
- Answer Access (see page 6-230)
- Emergency Outgoing Access (see page 6-231)
- Outgoing Access (see page 6-232)
- Toll Restriction (see page 6-232)
- Search Algorithm (see page 6-234)
- Audio for Calls Camped onto this Device (see page 6-234)
- Music-On-Hold (see page 6-234)
- Audio on Transfer to Ring (see page 6-235)
- Audio on Transfer to Hold (see page 6-235)
- Audio On Hold For Transfer Announcement (see page 6-235)
- PRI Call By Call (see page 6-235)
- One-Way Incoming Only (see page 6-236)
- Echo Trunk Number (see page 6-237)
- Enable Hookflash (see page 6-237)
- Camp-Ons Allowed (see page 6-237)
- ISDN Data Calls Allowed (see page 6-237)
- Day and Night Ring-In Type (see page 6-237)
- Send Station Caller ID to Attached PBX (see page 6-239)
- Propagate Original Caller ID (see page 6-240)

- Calling Party Name (see page 6-241)
- Calling Party Number (see page 6-241)
- Force Trunk Group Calling Party Name and Number (see page 6-241)
- Wait for ISDN Caller ID Information (see page 6-241)

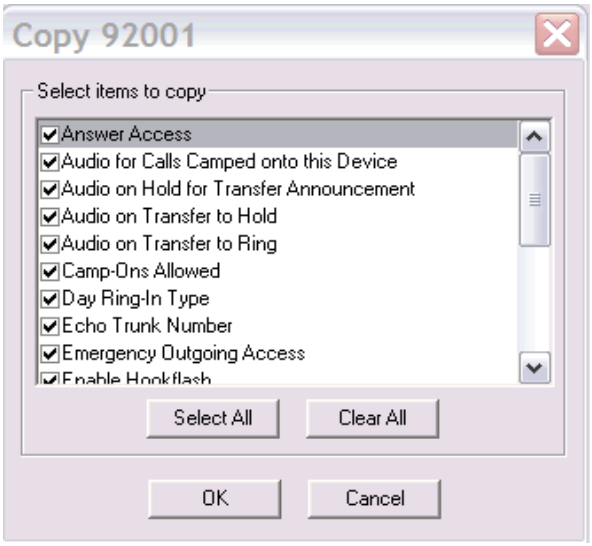
4.282 Trunk group programming is performed as described in the following paragraphs.

4.283 Extension: To change the trunk group's extension, select its current Extension and enter the new number in the text box (or scroll to an available number).

4.284 Batch Extension Change: To change several trunk group extension numbers at once, highlight the trunk groups you want to change. Then right-click and select **Batch Extension Change**. When the Get Extension screen appears, select the number you want to assign to the first highlighted trunk group (the other trunk groups will be numbered consecutively after this number). When you select **OK**, the trunk groups are automatically renumbered and re-sorted in the station list.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items. (You may have to continue to hold **SHIFT** or **CTRL** while right-clicking to display the option without changing the highlighted selections.)

4.285 Copy CO Trunk Group: To copy a CO trunk group, right-click on the trunk group extension and then select **Copy**. To paste the programming information into another trunk group, right-click the trunk group where you want the information pasted. Then select **Paste**. The Copy dialog box (shown below) allows you to select the attributes you want to copy. Select the desired attributes and click **OK**.



4.286 Description and Username: All trunk groups should have a description and a username. The description that appears in all trunk group lists in the database can be up to 20 characters long. The username, that will appear on display phones, can have up to 10 characters. To program the names, select the desired text box and type the entry.

Trunk List

4.287 Each trunk is assigned to a trunk group using this trunk list or during individual trunk programming. To view the list of trunks that have been assigned to this trunk group, double-click on **Trunks**.

- **To move trunks to this list from another list:** Right-click in any blank area on the right side of the screen (if there is no blank space, press **CTRL+N**). Then choose **Move To Trunks List**. A window appears that asks you to select the type of device to include. Select the trunk type(s) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.) Highlight the appropriate trunks and then select **Add Items**. When you have added all the desired trunks select **Finish**. Your selections appear in the list.
- **To move trunks out of this list and into another list:** Select the trunk(s) to be moved and drag and drop them into the trunk list of the new trunk group. (You can use the **CTRL** or **SHIFT** keys to select several trunks at a time.)

Day/Night Multiple Ring-In

4.288 If the Ring-In Type (see page 6-237) is set to Multiple, use this field to determine the ring-in destinations for day and night modes. The trunk group can ring in to a list of stations, extension lists, and/or applications (but not hunt groups). The list can include local or off-node device extension numbers.

NOTE: When using an extension list for ring-in, do not exceed 30 stations per list.

4.289 To set the ring-in destination, select **Multiple Ring-In** and then select **Day Mode** or **Night Mode** and do the following:

4.290 To add destinations to the list:

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To List**. A window appears that asks you to select the type of device to include.
3. Select the device type(s) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

   NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate items and then select **Add Items**. When you have added all the desired destinations select **Finish**. Your selections appear in the list.

   NOTE: If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

4.291 To delete destinations from the list: Highlight them and then right-click. Select **Remove Selected Items**.

Day/Night Answer Access

4.292 *(Not available for DISA trunks)* To determine which extensions will have allowed-answer permission for the trunk group in day or night mode, select **Answer Access** and then **Day Mode** or **Night Mode** and do the following:

4.293 To add extensions to the list:

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To List**. A window appears that asks you to select the type of device to include.
3. Select the device type(s) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

> **NOTE:** To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate items and then select **Add Items**. When you have added all the desired devices select **Finish**. Your selections appear in the list.

> **NOTE:** If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

4.294 To delete destinations from the list: To delete an item from the list, highlight them and then right-click. Select **Remove Selected Items**.

Day/Night Emergency Outgoing Access

4.295 There are separate lists for stations with emergency outgoing access in day and night modes. (On loop start trunks that are connected to paging equipment, the emergency outgoing access designation does not affect stations' ability to seize the trunk for paging. Trunks used for paging should not allow any station to have emergency outgoing access.) By default, the automatic station list (*Auto: All Stations*) is assigned to Day/Night Emergency Outgoing Access. To program the stations that will have emergency outgoing access for the trunk group, select **Emergency Outgoing Access** and then **Day Mode** or **Night Mode** and do the following:

4.296 To add extensions to the list:

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To List**. A window appears that asks you to select the type of device to include.
3. Select the device type(s) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

> **NOTE:** To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate items and then select **Add Items**. When you have added all the desired devices select **Finish**. Your selections appear in the list.

> **NOTE:** If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

4.297 To delete destinations from the list: To delete an item from the list, highlight them and then right-click. Select **Remove Selected Items**.

NOTICE

It shall be the responsibility of the entity or person(s) completing installation and maintenance of hardware or software described herein to research, comply with and be responsible for the specific governmental rules and regulations regarding Emergency Outgoing Access 911 [999] of the geographic location in which such functions are performed.

Day/Night Outgoing Access

4.298 There are separate lists for stations with outgoing access in day and night modes. (On loop start trunks that are connected to paging equipment, the outgoing access designation does not affect stations' ability to seize the trunk for paging. Trunks used for paging should not allow any station to have outgoing access for placing outside calls.) By default, the automatic station list (*Auto: All Stations*) is assigned to Day/Night Outgoing Access when a CO Trunk Group is created. To program the stations that will have outgoing access for the trunk group, select **Outgoing Access** and then **Day Mode** or **Night Mode** and do the following:

4.299 To add extensions to the list:

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To List**. A window appears that asks you to select the type of device to include.
3. Select the device type(s) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

   NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate items and then select **Add Items**. When you have added all the desired devices select **Finish**. Your selections appear in the list.

   NOTE: If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

4.300 To delete destinations from the list: To delete an item from the list, highlight them and then right-click. Select **Remove Selected Items**.

Toll Restriction

4.301 To program toll restriction for the trunk group, double-click on **Toll Restriction**. Each trunk group can have the following toll restriction parameters programmed.

NOTE: On loop start trunks that are connected to paging equipment, toll restriction does not affect stations' ability to seize the trunk for paging.

- **Class Of Service**: Each trunk group has Classes of Service (COS) which restrict or allow certain dialing patterns from being dialed on a call.

NOTE: When determining toll restriction for an ARS outgoing call, the network only checks the station toll restriction for the node on which the calling station resides. The system does not check the trunk class of service for ARS calls. See page 6-281 for a complete explanation of the Classes of Service. To program the toll restrictions double-click on **Classes of Service**, you then have the option of choosing Day or Night. Double-click on the desired time period to view a list of current classes of service.

To add a class of service:

a. Right-click in the window and select **Add To List**.

b. A window appears that allows you to select the types of toll restrictions you want to program. Select **ARS Only & Deny Area/Office** and/or **Classes of service** and then select **Next**.

c. Another window appears that shows you a list of available classes of service. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

d. Highlight the desired class(es) of service and then select Αδδ Ιτεμσ. The selected classes of service appear in the list. Select **Finish** to exit.

e. *(U.S. only) If you selected Deny Area/Office class of service,* the trunk group must also be assigned to a User Group. To change the user group, right-click on **User Group** and select **Change User Group**. In the first window that appears, select **User Groups** and then **Next**. In the next window, select the desired user group and then select **Finish** to exit and save the change.

To delete one or more classes of service: Highlight the item(s). Then right-click and select **Remove Selected Items**.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items.

- **Subject to toll restriction**: If the trunk group is not subject to toll restriction, the station and trunk classes of service (except ARS-Only restriction) are not checked when a trunk in the group is used. To enable this flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)
- **Exempt from ARS only**: The trunk group can be exempt from, or subject to, the ARS-only restriction. If exempt from ARS-only, stations with that restriction can directly select the trunks. If subject to ARS-only, ARS-restricted stations can use the trunk group only if it is part of an ARS route group. To enable this flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)
- **Call Cost**: For trunk groups that are not subject to toll restriction, the call cost rate to be used for calls placed on the trunks can be free, local, toll local, toll long distance, operator-assisted, or international rate. Select the Assigned Call Cost value and then scroll to the desired setting. Then press **ENTER** or click outside the field to save your change.
- **Absorbed Digits**: Trunk groups can be programmed to absorb digits in areas where the first digit(s) of the office code are absorbed. There can be up to 50 patterns with up to 48 digits each. To enable this flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

- **Absorbed Patterns**: If the Absorbed Digits and PBX Line flags are enabled, use this field to enter the absorbed digit patterns. There can be up to 50 patterns with up to 48 digits each. The only characters allowed in the pattern are digits (0-9), pound [hash] sign (#), parentheses ( ), brackets ([ ]), greater than (>) and less than (<) symbols, and the wildcards. (See page 5-111 for wildcard information.) Double-click on **Absorbed Patterns** to view the current list of patterns, if any. To add a pattern, do the following:

  a. Right-click anywhere in the right side of the screen and select **Add To Absorbed Patterns List**. A new, blank pattern appears on the list.

  b. Select the pattern to display the text box and enter the desired pattern. Then press **ENTER** or click outside the field to save the change.

  c. Repeat these steps if you need additional patterns.

- **PBX Line**: Trunk groups can be programmed to absorb a digit string (defined above) for PBX installations. To enable this flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)
- **Absorbed Digit String**: If the PBX Line flag, described above, is enabled, use this field to enter the absorbed digit string. Click on the current Value and enter the new digit string, up to 32 digits. Then press **ENTER** or select another field to save the change. The only characters allowed in the string are digits (0-9), pound [hash] sign (#), parentheses ( ), brackets ([ ]), greater than (>) and less than (<) symbols, and the wildcards. (See page 5-111 for wildcard information.).

Search Algorithm

4.302 The trunk group can be programmed for linear or distributed trunk searching, as described below. Select the Search Algorithm value and then scroll to the desired setting. Then press **ENTER** or click outside the field to save your change.

- **Linear Trunk Group**: Requests for an outgoing trunk are always processed beginning with the highest numbered trunk on the list and moving down the list until an available trunk is found.
- **Distributed Trunk Group**: The first request will be processed beginning with the highest numbered trunk on the list. The next request will begin with the second trunk, and each subsequent request will begin one trunk lower on the list. When the end of the list is reached, requests begin again with the highest numbered trunk on the list.

Audio for Calls Camped onto this Device

4.303 The Audio for Calls Camped onto this Device field defines the audio that a caller hears when camped-on to this node trunk or IP connection group. To program this field, use the drop-down list to select the type of audio and press **ENTER**. Your options are Silence, Tick-tone, Ringback, or Board. If you select **Board**, specify the music source in the Extended Value field. The default is *Board*, and the default extended value is *01:08 Central Processor.*

Music-On-Hold

4.304 The music-on-hold option that will be heard by outside callers on the trunks in the trunk group can be set for tick tones, ringback, board (music), silence, or to use the next device's audio source. Select the current Music-On-Hold value and scroll to the desired setting (Silence, Tick Tone, Ringback, Board, or Use Next Device's Audio Source). To use the audio programmed at individual stations, select **Use Next Device's Audio Source**. If you select **Board**, select the Extended Value and scroll to the desired board where the music source is connected. When you have made your selection, press **ENTER** or click outside the field to save your change.

Audio On Transfer To Ring

4.305 To select what outside callers (except those using DISA) will hear while their call is being transferred, the Audio on Transfer To Ring option can be set for Music-On-Hold, Ringback, Silence, Tick Tone, Board, or Use Next Device's Audio Source. (The feedback that DISA callers hear is determined by the DISA Transfer tone system flag that is programmed on page 6-259.) To use the audio programmed at individual stations, select **Use Next Device's Audio Source**. To use the audio programmed for the trunk's music-on-hold, select **Music-On-Hold.** If you select **Board**, select the Extended Value and scroll to the desired board where the music source is connected. When you have made your selection, press **ENTER** or click outside the field to save your change.

Audio On Transfer To Hold

4.306 This field specifies what callers hear when they are transferred to and held at a different extension. The Audio On Transfer To Hold option can be set for Music-On-Hold, Ringback, Silence, Tick Tone, Board, or Use Next Device's Audio Source. (The feedback that DISA callers hear is determined by the DISA Transfer tone system flag that is programmed on page 6-259.) To use the audio programmed at individual stations, select **Use Next Device's Audio Source**. To use the audio programmed for the trunk's music-on-hold, select **Music-On-Hold.** If you select **Board**, select the Extended Value and scroll to the desired board where the music source is connected. When you have made your selection, press **ENTER** or click outside the field to save your change.

Audio On Hold For Transfer Announcement

4.307 This field determines what outside callers (except those using DISA) will hear while their call is on transfer hold. After the transfer is completed, the caller will hear the Audio On Hold For Transfer Announcement selection (programmed above). The Audio on Hold For Transfer Announcement option can be set for Music-On-Hold, Ringback, Silence, Tick Tone, Board, or Use Next Device's Audio Source. (The feedback that DISA callers hear is determined by the DISA Transfer tone system flag that is programmed on page 6-259.) To use the audio programmed at individual stations, select **Use Next Device's Audio Source**. To use the audio programmed for the trunk's music-on-hold, select **Music-On-Hold.** If you select **Board**, select the Extended Value and scroll to the desired board where the music source is connected. When you have made your selection, press **ENTER** or click outside the field to save your change.

NOTE: If the trunk group's audio field, including Music-On-Hold, is set to **Use Next Device's Audio Source**, the system uses the application's programming. If the field is set to any other option, such as Board, the system uses the trunk group's audio source, overriding station programming.

PRI Call By Call

4.308 *(U.S. Only)* Trunk groups that contain B-channels can be assigned a PRI Call By Call feature to use for outgoing calls. The Trunk Group PRI Call By Call feature supports the following services:

- Call By Call Inactive
- AT&T WATS Band
- AT&T Banded OUTWATS
- AT&T Foreign Exchange
- AT&T TIE Trunk

- AT&T Local Operator
- AT&T Carrier Operator
- AT&T Virtual Private Network
- AT&T MEGACOM 800
- AT&T MEGACOM
- AT&T INWATS
- AT&T WATS Maximal Band
- AT&T ACCUNET Switched Digital
- AT&T International LDS
- AT&T International 800
- AT&T Electronic Tandem Network
- AT&T Private Virtual Network
- AT&T DIAL-IT 900 / MultiQ
- AT&T National ISDN INWATS
- AT&T Unbanded OUTWATS
- Nortel Private Network
- Nortel INWATS
- Nortel OUTWATS
- Nortel Foreign Exchange
- Nortel TIE trunk

One-Way Incoming Only

4.309 Certain types of trunks, such as incoming WATS, need to be marked as "incoming-only" so that the system will recognize the lines during power-up and/or testing. (This is because, on these types of lines, battery is not returned when the line is seized and the system cannot power up the trunk unless this option is enabled.) If the trunk group contains incoming-only lines, enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.). When you place the check mark in the check box, the system checks whether any stations have outgoing access for that trunk group. *If there are stations with outgoing access,* a window appears that asks if you want to delete the outgoing-access assignments for those stations. Select **Yes** to remove the outgoing-access assignments, or select **No** to allow outgoing access.

4.310 Echo Trunk Number: When this flag is enabled, the system will echo the trunk number, DNIS, DID [DDI in Europe], etc., if the call is being routed out a trunk. The base digits are combined with the collected trunk number to create an outside number. If the trunk rings into another trunk in the system via individual trunk, trunk group, or ARS, the trunk number is dialed as the outgoing number.

NOTE: Note that if the collected trunk number is incomplete or invalid, the collected digits are not used for the outside number, to avoid the system from dialing the wrong outside number.

Echo Trunk Number

4.311 When this flag is enabled, the system will echo the trunk number {DNIS, DID (or DDI for Europe), etc.}, if the call is being routed out a trunk. The base digits are combined with the collected trunk number to create an outside number. If the trunk rings into another trunk in the system via individual trunk, trunk group, or ARS, the trunk number is dialed as the outgoing number.

4.312 Note that if the collected trunk number is incomplete or invalid, the collected digits are not used for the outside number, to avoid the system from dialing the wrong outside number.

Enable Hookflash

4.313 The Hookflash (Recall) feature can be enabled or disabled for each trunk group. If disabled, users cannot use the Hookflash feature code (330) while using the trunks in the trunk group. Hookflashes (Recalls) dialed through ARS are ignored by the trunk groups with hookflash (recall) disabled. To enable this flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

Camp-Ons Allowed

4.314 The Camp-On feature can be enabled or disabled for each trunk group. If disabled, users placing outgoing calls will hear busy signals when all trunks in the group are in use or unavailable. If enabled, users will be able to camp on and wait for an available trunk. To enable this flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

ISDN Data Calls Allowed

4.315 This is a BRI-only trunk. This flag allows you to prevent ISDN data calls from being routed to trunk groups containing trunks that cannot support data calls (e.g., loop start trunks). When set to **Yes**, the system can route ISDN data calls to this trunk group, and you are allowed to add B-channel trunks only to the group. If the flag is set to **No**, the system will not route ISDN data calls to this trunk group, and you can add any type of trunk, except private networking B-channel trunks, to the group. Any non-B-channel trunks must be removed from the CO trunk group before the flag can be set to **Yes**. If you move multiple trunks into a CO trunk group that has this flag enabled, you will be warned that none of the trunks will be added to the group if any of the trunks to be added are non-B-channel trunks.

Day/Night Ring-In Type

4.316 Ring-in type is determined separately for day and night modes. To set the Ring-In Type, select the current Value and scroll to the new setting. Then press **ENTER** or select another field to save the change.

NOTE: When using an extension list for ring-in or hunt groups, do not exceed 30 stations per list. The system can send ring signal to up to 30 stations simultaneously.

4.317 The Ring-In Type can be the same for both modes, or any combination of the following can be used:

- **Single**: The trunk group can ring at a single extension number or a hunt group. This can be a local or off-node extension number. When you select Single, you must also enter the destination extension in the Ring-In column. Use one of the following methods:

Method A

a. Select the current Value and enter the new value in the text box.

b. Press **ENTER**. A screen appears showing you what is associated with the number you entered.

c. Select **OK**. The new number appears in the field.

Method B

a. Right-click on the existing value. An option box appears.

b. Select **Change Ring-In**. A window appears that asks you to select the type of device to include.

c. Select the appropriate device type and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

d. Highlight the device you want to use and then select **Finish**. Your selection appears in the Ring-In field.

- **Multiple**: The trunk group can ring in to a list of stations, extension lists, and/or applications (but not hunt groups). The list can include local or off-node device extension numbers. Set the ring-in destination as described on page 6-230.

- **DISA**: If the trunk group is to be used for DISA, a security code can be assigned by selecting the Ring-In field and entering the code in the text box. For example, a trunk group can use DISA to forward a station/endpoint to an outside number. If a trunk group has DISA access, an outside caller can call into the corresponding trunk and perform various "Remote Programming" features. After the outside caller gets past the DISA security code (if one exists), the user can enter the "359" Remote Programming feature code. At this point, the user will enter the station/endpoint extension and station password they want to forward. After getting past the station password, the outside caller can then use the "355" (manual forward) feature code to forward to an outside number (or inside number for that matter).

NOTE: To prevent unauthorized access to the public network, ***all*** trunk groups using DISA should have a security code.

- **Call Routing Table**: If the trunk is used for Caller ID [CLIP], DID/DNIS, or ANI, the calls can be routed according to the information sent by the central office. See page 5-89 in FEATURES for complete information on Call Routing Tables. When you select Call Routing Table, you must also enter the destination table in the Ring-In column, using one of the following methods:

Method A

a. Select the current Value and enter the new value in the text box.

b. Press **ENTER**. A screen appears showing you what is associated with the number you entered.

c. Select **OK**. The new number appears in the field.

Method B

a. Right-click on the existing value. An option box appears.

b. Select **Change Ring-In**. A window appears that asks you to select the type of device to include.

c. Select **Call Routing Table** and then select **Next**. The list of tables appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

d. Highlight the table you want to use and then select **Finish**. Your selection appears in the Ring-In field.

NOTE: Loop start trunks that are connected to paging equipment should not have any ring-in designations. They should be reserved for internal use only.

- **Collected Digits**: This ring-in type indicates that the collected DID [DDI in Europe] or DNIS digits (plus the base digits) should be used as the destination extension. This helps to keep the number of call routing table entries down to a minimum when using a 3200D to route calls to extensions.

  Both trunk groups and call routing table entries can use the new ring-in type. When the new Ring-In Type of Collected Digits is selected, the Ring-In Destination field is empty.

  If the collected digits plus the base digits do not make up a valid ring-in destination, the call is routed to the primary attendant. Valid ring-in destinations include on- or off-node stations, on-or off-node hunt groups, trunk groups, individual trunks, voice mail applications, automated attendants, and ARS.

Send Station Caller ID to Attached PBX

4.318 Allows the user name and extension in the ISDN setup request message to be sent on an outgoing ISDN call. All intercom calls that route externally from the system through an ISDN circuit will send the user name and extension for the caller ID name and number instead of the Calling Party Number or Name information. This field is intended for Axxess endpoint calls routed to an Inter-Tel 7000 system. The default value is set to *No*. Consider the following when using this flag:

- This flag is enabled on a per trunk group basis.
- Only ISDN trunks (non-private networking) use this field.
- The user name and extension that are sent on the outgoing call are always the extension that has originally placed the call. For example, a voice mail application calls an extension that is forwarded to an outside number. The system sends the user name and extension from the voice mail application and not the extension that was forwarded to the outside number. If the voice mail application answers and transfers an incoming call to an extension that is forwarded to an outside number, the system sends the user name and extension from the voice mail application.
- Hunt group calls that are processed by an extension or open architecture interface (OAI) are governed by the same conditions as listed in the chart on page 5-79. If the user name has not been programmed, the system will not include the name information on the outgoing call.
- The user name may not be processed on an outgoing call across nodes if the outgoing PRI is using “display IE” instead of “facility IE” for caller ID name. The user name on the Inter-Tel system is not passed immediately between nodes, which is why the “display IE” does not contain the user name. The “display IE” on an outgoing call is sent immediately where a “facility IE” is delayed.
- All System and Manual Forward types apply to this feature.
- This flag applies to both local and remote node calls.

- When set to **No**, the system refers to the endpoint Calling Party Number or Calling Party Name field in System\Devices and Feature Codes\Stations\Local\<*extension*>. When the endpoint's Calling Party Number or Calling Party Name field is programmed with a number or name, the programmed information is sent. If the field is blank, no caller ID information is sent.
- This flag has no effect on incoming trunk calls being forwarded or redirected out another trunk.

4.319 To forward a calling station user name and extension to an attached PBX, such as the Inter-Tel 5000 system:

1. Set the **Send Station Caller ID to Attached PBX** flag to **Yes**.
2. Press **ENTER** or click another field to save the change.

4.320 See "Caller ID Forwarding" on page 5-77 for related information.

Propagate Original Caller ID

4.321 Allows the system to pass the caller ID name or number on an outgoing ISDN call if the call has not been answered by the system (extension, voice mail, hunt groups, or OAI application) or for transfer announcement calls. This field is intended for customers who want to route incoming calls from the Axxess system back to the PSTN through ISDN lines. By default this flag is set to *No*. See "Caller ID Forwarding" on page 5-77 for related information.

4.322 This flag applies to three scenarios: 1) incoming trunk calls forwarded/deflected back to the PSTN through ISDN lines; 2) incoming trunk calls answered by a voice mail CRA and transferred back to the PSTN through ISDN lines; 3) incoming trunk calls answered by an operator and transferred back to the PSTN through ISDN lines. Consider the following when using this flag:

- This flag is enabled on a per trunk group basis.
- Only ISDN trunks (non-private networking) use this field for the outgoing call. Do not mix trunk types in the trunk groups when using this feature.
- This flag must be enabled on the outgoing trunk group for the caller ID information to be processed.
- All Manual Forward types apply to this feature when used to redirect the call to an outside number.
- OAI Deflect applies to this feature.
- If the incoming caller ID information only has number and no name, the system only sends the number that was received.
- Incoming caller ID information that is blocked on the incoming call is not sent on the outgoing call. If the incoming call is blocked, the system only passes the generic information received for blocked call. Example of these generic blocked messages are: Private, Out of Area, Restricted, Unassigned Number, or Blocked Caller. The system does not send information when no caller ID information was received.
- The outgoing trunk group must be ISDN for the Caller ID information to be processed on the outgoing call.
- The incoming call cannot be answered by the system. If the call is answered and transferred, this flag is ignored.
- Calls to a hunt group that have not been answered are considered unanswered calls.

- Calls that have been played an announcement through a hunt group announcement are still considered an unanswered call.
- Enabling this flag applies to local node and remote node calls that route through the trunk group that has this flag enabled.
- The public telco may remove the caller ID name that is sent by the system and only pass the caller ID number.

NOTE: The system or network administrator must confirm with the CO as to whether or not the Forward Original Call-ID for Unanswered Calls feature is supported, as the Public Switched Telephone Network (PSTN) may reject this call that is "masquerading caller ID."

4.323 To route incoming calls from the Axxess system back to the PSTN through ISDN lines:

1. Set the **Propagate Original Caller ID** flag to **Yes**.
2. Press **ENTER** or click another field to save the change.

Calling Party Name

4.324 The Calling Party Name field allows Caller ID information to be used on outbound calls through trunk groups. Type the name (up to 16 alphanumeric characters) in the text box.

Calling Party Number

4.325 The Party Number field allows Caller ID information to be used on outbound calls through trunk groups. Type the number (up to 48 numbers) in the text box.

Force Trunk Group Calling Party Name and Number

4.326 When selected, the system uses the trunk group Calling Party Name and Calling Party Number. When cleared, the system follows Caller ID forwarding as described in the next section. This allows the Caller ID to be the callback number (for example, a main number, cell number, etc.) without losing the ability to identify the location of the emergency call.

Wait for ISDN Caller ID Information

4.327 This timer is used to control the amount of time the system waits for the incoming ISDN Facility message that contains the caller ID name before routing the call to the ring-in destination. This timer only applies to incoming ISDN calls that use Facility messages instead of Display messages for providing caller ID name.

4.328 The default value of this timer is *0* seconds, which is a no wait delay. This timer should only be adjusted if the system is receiving Facility messages from the telco where the name is not appearing on the forwarded or redirected calls. Contact your ISDN provider to verify if you are receiving Facility messages for caller ID name. Or, use the diagnostic port on the T1/E1 board and enable "DON ALL" and review the incoming ISDN messages for Facility or Display messages (remember to disable the DON information after reviewing the information by typing `DOFF`).

4.329 Some ISDN central offices use a Facility IE message to transfer a calling party name to the system. This is necessary to allow the PSTN to look-up the calling party's name. The Facility message containing the caller ID name may be delayed for several seconds after receiving the original incoming setup request. This timer provides the delay mechanism that allows the incoming call information to be processed so that the caller ID name can be forwarded on the outgoing call. When this timer is set to a value greater than 0, the system waits for the Facility message or the timer to expire before processing the call. When the timer is set to 0, the system processes ISDN calls, as it did in versions earlier than v11.0, not waiting for the Facility message with name.

4.330 If you want to route incoming calls back to the PSTN, the outgoing call should not be sent until the system has received the "facility IE" message. After the system has the calling party name, the system can then forward the calling party name back to the PSTN. An ideal value for this timer is about 5 seconds.

4.331 To set the amount of time the system will wait for the incoming ISDN Facility message that contains a caller ID name:

1. Set the value for the **Wait for ISDN Caller ID Information** timer.
2. Press **ENTER** or click another field to save the change.

4.332 See "Caller ID Forwarding" on page 5-77 for related information.

Q. NODE TRUNK GROUPS

4.333 When Node Trunk Groups is selected, it will show any node trunk groups that were created when the T1/PRI, E1/PRI, or T1/E1/PRI card Switch Type is set to Private Networking.

4.334 At the main level, you can program the description and username for each node trunk group. You can also see which card slot contains the circuits used for this network connection.

4.335 To program the other fields for a node trunk group, double-click on it to view these options:

- Trunks (see page 6-244)
- Emergency Outgoing Access (see page 6-231)
- Outgoing Access (see page 6-232)
- Search Algorithm (see page 6-234)
- Camp-Ons Allowed (see page 6-237)
- ISDN Data Calls Allowed (see page 6-237)

4.336 Node trunk group programming is performed as described in the following paragraphs.

4.337 Description and Username: All node trunk groups should have a description and a username. The description that appears in all node trunk group lists in the database can be up to 20 characters long. The username, that will appear on display phones, can have up to 10 characters. To program the names, select the desired text box and type the entry.

4.338 Copy Node Trunk Group: To copy a node trunk group, right-click on the trunk group extension and then select **Copy**. To paste the programming information into another node trunk group, right-click the trunk group where you want the information pasted. Then select **Paste**. The Copy dialog box allows you to select the attributes you want to copy. Select the desired attributes and click **OK**.

4.339 Associated Slot Number: This is the slot number where the card associated with this node trunk group is located. This field is for reference only and cannot be changed.

R. TRUNKS

4.340 When "Trunks" is selected, the right side of the screen will show all trunks on the local node. (T1 or E1 trunks with a switch type of "Private Networking" will appear in the trunk list, but their programming fields cannot be changed.)

4.341 At this level, you can see the type of trunk and its circuit number (address). You can perform the following options by right-clicking the extension of the desired trunk:

- Change Extension (see page 6-245)
- Batch Extension Change (see page 6-245)
- Copy a Trunk (see page 6-245)
- Assign CO Trunk Groups (see page 6-245)
- Run a Hybrid Balance Test (see page 6-246)

4.342 To program a specific trunk, double-click on the desired trunk. The following information can be programmed:

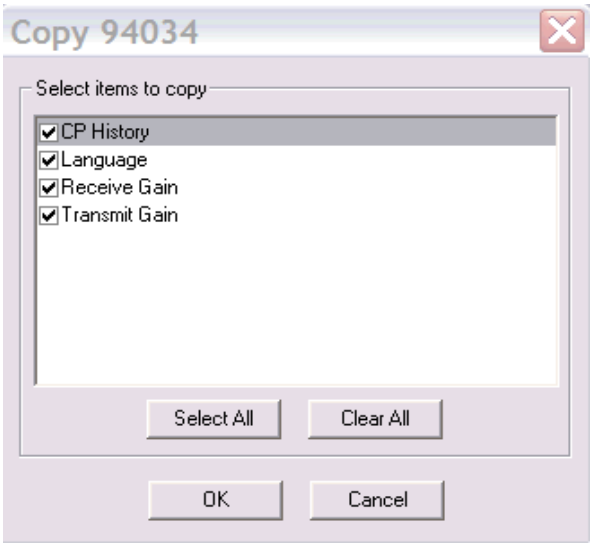
- CO Trunk Group (see page 6-251)
- Call Configuration—MGCP gateways and endpoints and SIP trunks only (see page 6-252)
- Network Group—MGCP gateways and endpoints and SIP trunks only (see page 6-252)
- SIP Gateway—SIP trunks only (see page 6-252)
- CP History—SIP trunks only (see page 6-252)
- MGCP Gateway IP Address—MGCP gateways only (see page 6-252)
- MGCP Gateway Port—MGCP gateways only (see page 6-252)
- Communication Timeout—MGCP gateways only (see page 6-253)
- Gateway Name—MGCP gateways only (see page 6-253)
- Associated Gateway—MGCP endpoints only (see page 6-253)
- Endpoint Name—MGCP gateways and endpoints only (see page 6-253)
- Service Type (see page 6-253)
- DTMF Signaling In Band (see page 6-254)
- Answer Supervision (see page 6-255)
- Connect Trunk-to-Trunk Call on Polarity Reversal (see page 6-255)
- Hybrid Balance (see page 6-256)
- Measured Echo Return Loss—ERL (see page 6-256)
- Last Hybrid Balance Test Timestamp (see page 6-256)
- Service Base Number (see page 6-256)
- Language (see page 6-257)
- Transmit Gain (see page 6-257)
- Receive Gain (see page 6-257)
- Send Digits En Bloc—loop start and MGCP trunks only (see page 6-257)

4.343 Change Extension: To change the extension of the trunk, select the current value and enter the new extension (or scroll to the desired number).

4.344 Batch Extension Change: To change several trunk extension numbers at once, highlight the trunks you want to change. Then right-click and select **Batch Extension Change**. When the Get Extension screen appears, select the number you want to assign to the first highlighted trunk (the other trunks will be numbered consecutively after this number). When you select **OK**, the trunks are automatically renumbered and re-sorted in the station list.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items. (You may have to continue to hold **SHIFT** or **CTRL** while right-clicking to display the option without changing the highlighted selections.)

4.345 Copy Trunk: To copy a trunk, right-click on the trunk extension and then select **Copy**. To paste the programming information into another trunk, right-click the trunk where you want the information pasted. Then select **Paste**. The Copy dialog box (shown below) allows you to select the attributes you want to copy. Select the desired attributes and click **OK**.



4.346 CO Trunk Group Assignment: Trunks can be assigned to trunk groups on an individual basis. To place the trunk in a trunk group, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the current CO Trunk Group setting and select **Change CO Trunk Group**. A window appears that asks you to select the type of device to include.
2. Select CO Trunk Group and then select **Next**. The list of trunk groups appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
3. Highlight the desired trunk group and then select **Finish**. Your selection appears in the Trunk Group field.

NOTE: You can also move trunks between trunk groups as described on page 6-228.

4.347 Run a Hybrid Balance Test: The Hybrid Balance Test provides data that allows you to optimize the Hybrid Balance Setting on loop/ground start trunk ports connected to loop/ground start trunks.

4.348 The Hybrid Balance Test generates 20 measurements of Echo Return Loss (ERL) at the trunk-port interface, one for each possible Hybrid Balance setting. By entering in DB Programming the optimal Hybrid Balance setting for each trunk, you can minimize echo from the CO interface into the system. And, if needed, you can provide actual ERL data in terms of dB loss for each loop/group start trunk attached to the Axxess system.

- **Definitions**: Echo Return Loss (ERL) refers to the difference between the transmitted and reflected audio levels. The higher the ERL, the lower the level of echo. The term "hybrid balance" describes the relationship of the impedances of the local loop start or ground start trunks on the Axxess system. The Hybrid Balance Setting for a trunk port can be adjusted to achieve a better match of impedances.
- **Background**: In general, echo occurs in IP audio because of inherent delays associated with packet switching through a network. On the Axxess system, echo in IP audio is associated with the increased use both of IP networking and of IP endpoints in the marketplace. Initially, the Hybrid Balance setting is programmed in accordance with a region-specific default value. This default value, however, may not be optimum for all loop/ground start trunks, and an adjustment to the Hybrid Balance setting is required. The Hybrid Balance Test provides real-time data you can use to reprogram the Hybrid Balance setting.Using data provided by the Hybrid Balance Test, you can minimize echo on trunk interfaces by programming the Hybrid Balance setting to a recommended optimal setting for each interface. As a result, any IP-based connection utilizing a trunk may have noticeably reduced echo. For programming instructions, see page 6-247.
- **Source of Hybrid Balance Assessment Data**: Assessment data for adjusting the Hybrid Balance setting is generated from the Digital Signal Processor (DSP) Echo Diagnostic and the Call Processing (CP) Hybrid Balance Diagnostic. The DSP Echo Diagnostic feature provides a means for quantifying the level of echo present on a given trunk. One function on the DSP plays a set of tones out on the transmit line of the trunk, and a second function listens on the receiving line of the trunk to determine the level of the reflected audio. The DSP performs this test for a specified duration and sends the measurement result to CP. The CP Hybrid Balance Diagnostic feature makes requests of the DSP Echo Diagnostic and searches the measurement results to determine the optimal setting for each of the loop start trunks. CP reports all results and a test summary using free-form Message Print.

4.349 Hybrid Balance Test Process: With the Hybrid Balance Test process you can assess the degree of impedance mismatching present between a loop/ground start trunk and a trunk port connected to it. With accurate measurements of echo return loss (ERL), you can optimize the loop/ground start trunk interface to minimize echo. IP endpoints that connect to the Public Switched Telephone Network (PSTN) across loop/ground start trunks may be noticeably reduced echo after carrying out the hybrid balance test process.

4.350 In v9.115 and later, the "Measured Echo Return Loss (ERL)" and "Last Hybrid Balance Test Timestamp" flags have been added to the Trunks folder in DB Programming. See page 6-256 for details about these flags.

4.351 To run a Hybrid Balance Test: Use the Hybrid Balance Test feature to verify loop/ground start trunk characteristics and to determine the optimal Hybrid Balance setting for each CO trunk port interface. To establish optimal Hybrid Balance setting for each loop/ground start trunk, you will need to carry out the following process and applicable procedures in sequence.

NOTICE

Exceptions to the Rule: For the performance of the following procedures only, you may enter the On-Line Mode (OLM) mode without the supervision of Inter-Tel Technical Support personnel. Any further or other programming in OLM mode is strictly prohibited without specific guidance from Inter-Tel Technical Support personnel.

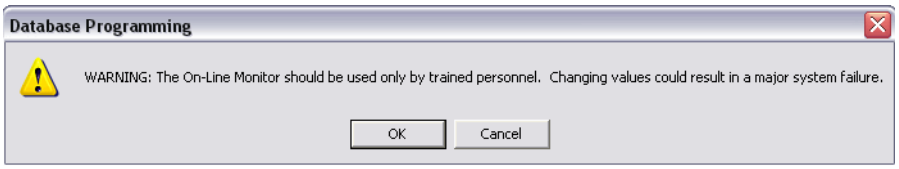
1. Set up Expanded Message Print capability in On-Line Monitor (OLM) programming (see page 6-248).
2. Verify that the "CO Timeout in seconds" timer is set to 4 seconds in OLM programming (see page 6-249).
3. Set up the Message Print client for Expanded Message Print (see page 6-249).
4. Run the Hybrid Balance Test for the loop/ground start trunks connected to the system (see page 6-250).

4.352 To obtain and maintain optimal Hybrid Balance settings for individual loop/ground start trunk ports, adhere to the following guidelines:

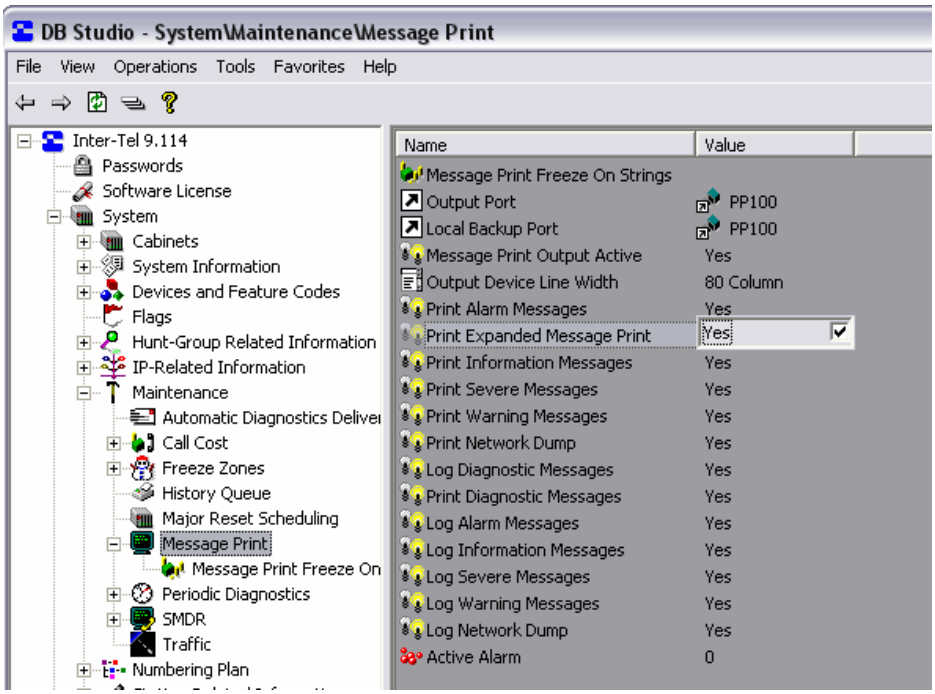
- Perform the test on each port at initial installation of the system.
- Perform the test whenever a loop/ground start trunk connection is added or changed.
- Run the test on a system that is idle or nearly idle.
- Run the test until the greatest Echo Return Loss (ERL) measurement is within ±2 dB twice in a row, as shown in Expanded Message Print. If the trunk port interface is optimally balanced, successive ERL readings should not vary a great deal.

4.353 To set up Expanded Message Print capability:

1. On an idle or nearly idle system, start Session Manager.
2. From DB Studio, select **View** and then click **On-Line Monitor**. The OLM warning message displays.



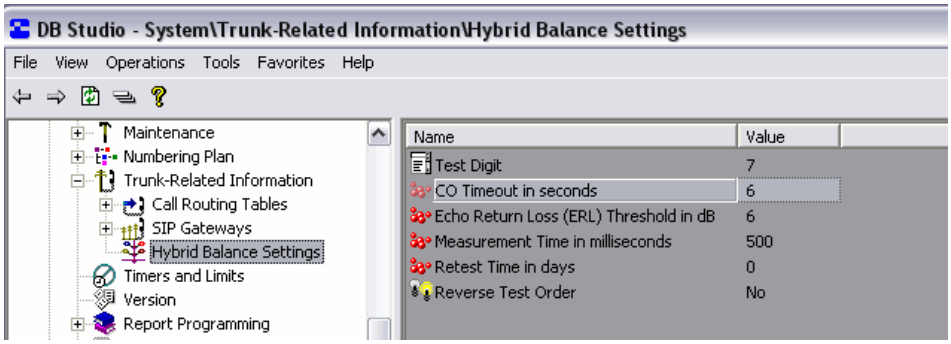
3. Click **OK** and a checkmark appears next to the On-Line Monitor option. The OLM view of DB Programming displays, as indicated by a darker background in the right pane and the appearance of previously hidden folders in the left pane.
4. From the left pane, select System – Maintenance – Message Print – **Print Expanded Message Print**. Make sure the value is set to **Yes**. The default is **No**.



4.354 To verify the "CO Timeout in seconds" timer value:

NOTE: Proceed *only* if the "Print Expanded Message Print" capability has been enabled in DB Programming. See page 6-248 for details.

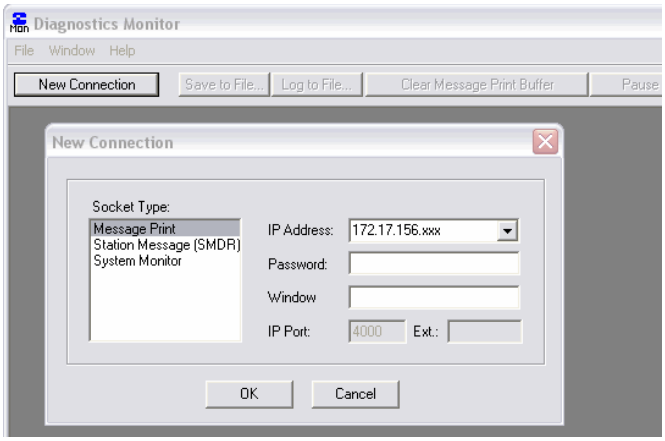
1. If you have not already done so, start DB Programming in OLM mode as described in steps 1. through 3. on page 6-248.
2. From the left pane, select System – Trunk-Related Information – **Hybrid Balance Settings**. Change the CO Timeout in seconds value to 4, if it is set to any other value. The default is 6 seconds.



3. After validating that the Print Expanded Message Print option is enabled and the CO Timeout in seconds value is set to 4, click **View** and then click **On-Line Monitor** to disable the option. You exit OLM mode and the regular DB Programming window displays. As appropriate, proceed either to the procedure to run a Hybrid Balance Test on a single trunk or on all trunks. See page 6-250.

4.355 To activate the Message Print client:

1. From your Windows desktop, select Start – Inter-Tel DB Programming – **Diagnostics Monitor**. The Diagnostics Monitor and New Connection windows display.

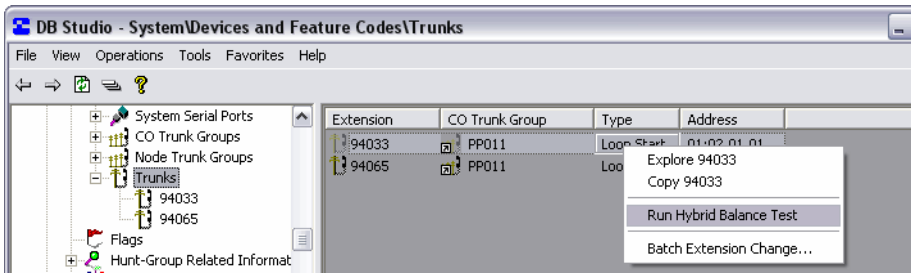


2. Select **Message Print** and click **OK**. Scrolling Message Print output from the system appears in the Diagnostics Monitor window.
3. Minimize the Diagnostics Monitor window until needed.

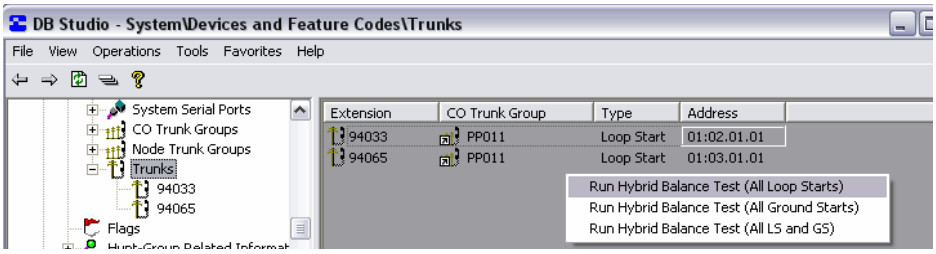
4.356 To run the Hybrid Balance Test:

NOTE: Proceed *only* if the "Print Expanded Message Print" capability has been enabled (page 6-248) and the "CO Timeout in seconds" timer is set to 4 (page 6-249).

1. From the left pane of DB Studio, select System – Devices and Feature Codes – **Trunks**. A list of trunks appears in the right pane, identified by Extension, CO Trunk Group, Type, and Address.
2. *For a single trunk,* right-click the extension in the Extension, Type, or Address column of the desired trunk. On the menu, select **Run Hybrid Balance Test**.

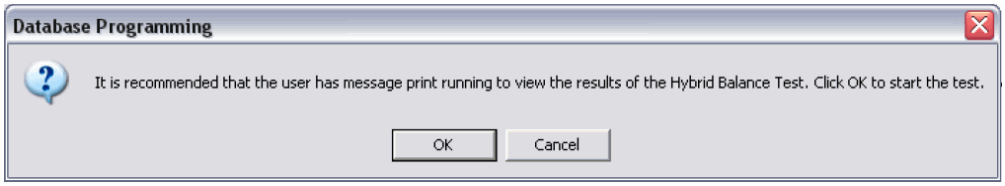


For all trunks, right-click the background area of the Trunks window, and select **Run Hybrid Balance Test (All Loop Starts)**, **Run Hybrid Balance Test (All Ground Starts)**, or **Run Hybrid Balance Test (All LS and GS)**.

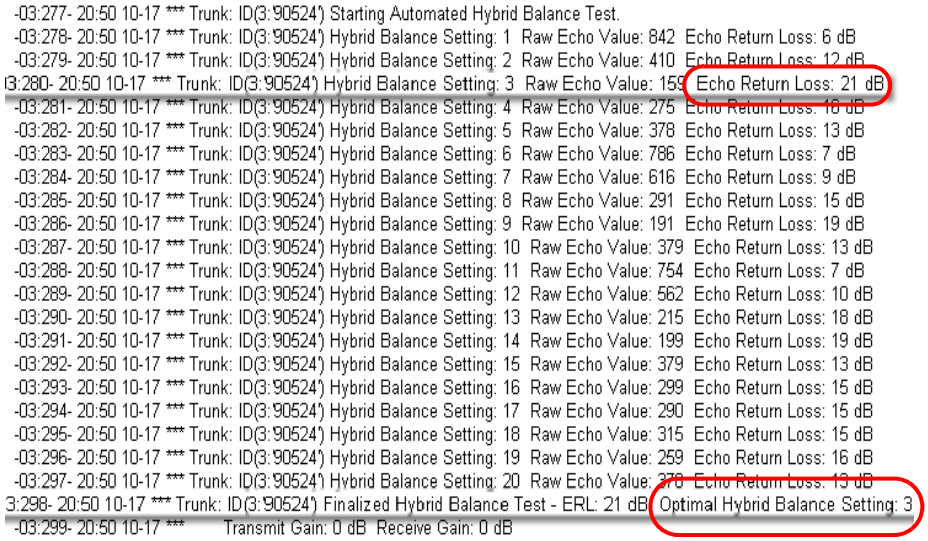


NOTE: While the Hybrid Balance Test runs, the selected trunk(s) are placed in a Busy condition.

3. When the following window displays, click **OK**.



4. Maximize the Diagnostics Monitor window displaying the Expanded Message Print output. For the selected trunk, a free-form Message Print output displays the Echo Return Loss (ERL) for each of the 20 possible Hybrid Balance Settings and—based on the greatest ERL value of the 20 settings—recommends the Optimal Hybrid Balance Setting. In the following example of Expanded Message Print output, the greatest ERL recorded for the is 21 dB and the recommended Optimal Hybrid Balance Setting is 3.



5. Record the pertinent results printed in the Message Print client.

6. Return to the DB Programming session. In the left pane of DB Studio, select System – Devices and Feature Codes – Trunks – <***the trunk extension you tested***>. For the Hybrid Balance option, select **Short** or **Normal** based on the value recommended for the Optimal Hybrid Balance Setting (see page 6-256).

7. Repeat this procedure and compare the greatest ERL values. Acceptable tolerance between the first and second runs is ±2 dB. If the second Hybrid Balance Test results are within tolerance, the process is complete and you can proceed to step 8. of this procedure.

   If the greatest ERL in second run is more than the allowed tolerance, run the test and set the Hybrid Balance value until you obtain consecutive ERL results within the ±2 dB tolerance.

8. Change the System – Message Print – Print Expanded Message Print option value to **No**.

9. Exit OLM mode by un-selecting the **On-Line Monitor** option in the View menu of DB Studio.

CO Trunk Groups

4.357 Allows you to program a specific trunk. See page 6-228 for details.

Call Configuration

4.358 *(MGCP gateways and endpoints and SIP trunks only)* The Call Configuration information is only required if using peer-to-peer (P2P) audio for the selected MGCP gateway, endpoint, or SIP gateway. The Call Configuration defines the setting that the gateways and endpoints use when connected to a call. After you program a Call Configuration you can assign it to multiple devices. By default, all IP devices are placed in Call Configuration 1, which is programmable. To assign the MGCP gateway, MGCP endpoint, or SIP gateway a call configuration, double-click **Call Configuration**. Then double-click the desired Call Configuration ID and add the gateway or endpoint to the IP Trunks list. This option is also located under System\IP-Related Information\Call Configurations.

Network Group

4.359 *(MGCP gateways and endpoints and SIP trunks only)* The Network Group information is only required if using P2P audio for the selected MGCP gateway, MGCP endpoint, or SIP gateway. To assign the gateway or endpoint to a network group, double-click **Network Group**. Then double-click the desired Network Group extension and add the gateway or endpoint to the IP Trunks list. This option is also located under System\Devices and Feature Codes\Network Groups.

SIP Gateway

4.360 (*SIP trunks only*) At least one SIP gateway must exist at all times (even in a default system) because it follows the model of a CO trunk group. To communicate with a SIP gateway, the following fields must be programmed (this option is also located under System\Trunk-Related Information\SIP Gateways.):

- **SIP Trunks**: Indicates the SIP gateway where SIP trunks are placed.
- **IP Address**: Indicates the IP address of the SIP gateway.
- **Listening Port**: Indicates the port on the CPC that the system listens to for SIP messages. By default, this port is set to 5060 (standard SIP port).

CP History

4.361 (*SIP trunks only*) This field, used for diagnostic purposes only, enables SIP message output in the CP history file. the CP history file allows you to trace SIP trunk calls. The available output options are Headers Only, No History, or Full History. By default, this field is set to Headers Only.

NOTE: The system-wide SIP format is set to No Output by default. To change this setting, use the Diagnostic-SIP View (9987) feature. Options include No Output, Headers and Full Output.

MGCP Gateway IP Address

4.362 *(MGCP gateways only)* The MGCP Gateway IP Address field identifies the IP address used to access the MGCP gateway. To enter an IP address, click the current Value. The Edit MGCP Gateway IP Address dialog box appears. Enter the IP address (do not include periods) and click **OK** when finished. By default, this is *192.168.200.201*.

MGCP Gateway Port

4.363 *(MGCP gateways only)* The MGCP Gateway Port field defines the port number used to access the MGCP gateway. To change the port number, click the current Value and enter a new number. Then press **ENTER** or click another field do save the changes. The valid range is *1024-65535*, and the default is *2427*.

Communication Timeout

4.364 *(MGCP gateways only)* The Communication Timeout field specifies the number of seconds that the gateway will wait for an acknowledgement from an associated endpoint before it considers the endpoint offline. To change the number of seconds, click the current Value and use the scroll box to select the new value. Then press **ENTER** or click another field to save the change. The valid range is *3-300*, and the default is *15*.

Manufacturer

4.365 *(MGCP gateways only)* The Manufacturer setting identifies the brand of the MGCP gateway that is connected to the phone system. Your options are *None* and *Audiocodes MP10X*. To change the brand, click the current Value and use the drop-down box to make your selection. Then press **ENTER** or click another field to save the change. By default, this is *Audiocodes MP10X*.

Gateway Name and Endpoint Name

4.366 *(MGCP gateways and endpoints only)* These fields display the names that were assigned to the gateway and associated endpoint when the circuits were programmed (see page 6-151). If desired, you can change the names here. To change a name, click the current Value. The Edit Gateway (or Endpoint) Name dialog box appears. Enter a new name, up to 18 characters, that identifies the gateway or endpoint on the network. Then click **OK**.

Associated Gateway

4.367 *(MGCP endpoints only)* The Associated Gateway field displays the MGCP gateway to which the MGCP endpoint is connected. If a gateway was not assigned when the circuit was programmed, this defaults to the first gateway on the IPRC. To change the associated gateway, right-click the current extension and select **Change Associated Gateway**. Then select MGCP Gateway and Endpoint and click **Next**. In the next dialog box, select the desired gateway and click **Finish**. If you know the trunk extension for the associated gateway, you can click the current Value and enter the extension instead of having to use the dialog boxes. Then press **ENTER** or click another field to save the change.

Service Type

4.368 *(For the U.S.) This cannot be programmed unless the Advanced CO Interfaces premium feature is licensed (see page 6-97). Also, caller information features require OPC programming, described on page 6-117.)* Available service types are determined by the type of trunk being programmed, as described below. To set the Service Type select the current Value and scroll to the desired option. Then press **ENTER** or click outside the field to save your change.

- **Loop Start**: Loop start trunks can use Caller ID [CLIP]. When selected, the Caller ID [CLIP] service type option indicates that the associated trunk will provide caller identification signals. Because the central office sends the information after the first cycle of ring voltage is applied, selecting this box will also prevent the system from signaling an incoming call to the station users until the system has had an opportunity to collect the caller information. (Note that the system will provide ring signaling on the second ring signal if no caller information has been collected.) Caller ID [CLIP] uses Caller ID [CLIP] receivers.
- **E&M and DID**: E&M and DID trunks can use DID, ANI, and DNIS. Because DTMF receivers are required to receive DID and ANI information, you should enable additional receivers.
- **Ground Start**: Ground start trunks do not support any of the caller information service types.

NOTE: Each individual trunk, not the trunk group, is programmed to collect digits using Caller ID [CLIP], DID, ANI, or DNIS. If the trunk is programmed to collect digits, but the trunk group does not use call routing tables, the system routes the call based on the trunk group programming and ignores the collected digits. (However, the collected digits will appear in SMDR and can be used by the Desktop Interface.) If the trunk is not set up to collect digits and the trunk group uses call routing tables, the system uses the call routing destination for "no-digit" calls.

Service Type

4.369 *(For Europe) This cannot be programmed unless the Advanced CO Interfaces premium feature is licensed (see page 6-97).* The Service Type field is available only for loop start trunks. When selected, the Caller ID [CLIP] service type option indicates that the associated trunk will provide caller identification signals when used with a loop start adapter (LSA) and an MGCP gateway (see the *IP Devices Installation Manual*, version 1.3 or later, for details). To set the Service Type, select the current Value and scroll to the desired option. Then press **ENTER** or click outside the field to save your change.

NOTE: Do not change the Service Type to Caller ID unless using IP LSAs.

Start Type

4.370 *(E&M and DID only)* E&M and DID trunks perform "handshaking" between the system and the central office to transfer call information. The three start types are described below. To set the Start Type select the current Value and scroll to the desired option. Then press **ENTER** or click outside the field to save your change.

- **Immediate**: If programmed for "Immediate" start type, the calling system will immediately begin sending the dialed digits to the receiving system.
- **Wink**: If programmed for "Wink" start type, the systems perform a "handshake" to allow the receiving system to signal that it is ready to receive the digits dialed by the other system. (This is the default start type.)
- **Delay Dial**: If programmed for "Delay Dial" start type, the calling system will wait until its E&M Dial Delay timer expires before sending any digits to the receiving system.
- **Dial Tone**: *(E&M trunks only)* If programmed for "Dial Tone" start type, the calling system will wait until it detects dial tone from the other system before sending digits.

DTMF Signaling In Band

4.371 *(With a European system, this is used for Loop Start Trunks)* Trunks can be set for DTMF or dial pulse signals. To set the signaling method select the current Value and scroll to the desired option. Then press **ENTER** or click outside the field to save your change.

Disconnect Timer

4.372 *(E&M and DID Only)* The trunk can be programmed to uses the Disconnect timer for disconnecting calls immediately whenever the inside party hangs up on a call. If the Disconnect timer is not enabled, it will wait up to 2 seconds for an on-hook signal from the central office. To set the Disconnect Timer option select the current Value and scroll to the desired option (Disconnect Timer or Wait for Remote On-Hook). Then press **ENTER** or click outside the field to save your change.

Answer Supervision Type

4.373 *(Non-T1 Loop Start and E&M only)* Answer supervision determines whether the system should consider a call valid (or disconnected). Available answer supervision types depend on whether a loop-start or E&M trunk is being programmed, as described below. To set the Answer Supervision Type select the current Value and scroll to the desired option. Then press **ENTER** or click outside the field to save your change.

- **Non-T1 Loop Start**:
  - — **Polarity Reversal**: A loop reversal must be received to consider the call valid. When the first loop reversal is received, the call is made valid immediately, and the phone display will begin call cost. When a second loop reversal is received, the system will terminate the call.

    NOTE: If a second loop reversal is not received, the system will not terminate the call unless the inside party hangs up or loss-of-loop is received from the CO [local exchange].

  - — **Valid Call Timer**: After the Valid Call Timer expires, the call is validated. All polarity reversals received before and after the Valid Call Timer are ignored.
  - — **Valid Call Timer with Polarity Reversal**: If a loop reversal is received before the Valid Call Timer expires, the call will be made valid immediately, and the phone display will begin call cost. When a second loop reversal is received, the system will terminate the call. If a loop reversal is not received before the Valid Call Timer expires, the call will be validated by the timer. If a loop reversal is received after the timer expires, the loop reversal is ignored, but the call cost is reset. If a second loop reversal is then received, the system will terminate the call.

    NOTE: If a second loop reversal is not received, the system will not terminate the call unless the inside party hangs up or loss-of-loop is received from the CO [local exchange].

- **E&M**:
  - — **Off-Hook Debounce Timer**: After the Off-Hook Debounce Timer expires, the call is validated. All polarity reversals received before and after the Off-Hook Debounce Timer duration are ignored.
  - — **Answer Recognition Timer**: After the Answer Recognition Timer expires, the call is validated. All polarity reversals received before and after the Answer Recognition Timer duration are ignored.

Connect Trunk-to-Trunk Call on Polarity Reversal

4.374 This is programmed for non-T1 Loop Start and B-channel trunks only. It is located in System\Devices and Feature Codes\Trunks\<*individual trunk*>.

- *If the flag is set to Yes*, when an ISDN incoming call is made to a loop start trunk (trunk-to-trunk call), the call will be validated when a polarity reversal is received. The Inter-Tel system will connect the voice path to hear progress tones and then will send the ISDN connect message when the loop start polarity reversal is detected.
- *If the flag is set to No*, when an ISDN incoming call is made to a loop start trunk (trunk-to-trunk call), the call will be validated after receiving an end of dialing handshake. After the handshake, the CO [local exchange] may provide progress tones, such as ring-back.

NOTE: For this feature to work properly, both the incoming (B-channel trunk) and outgoing (non-T1 Loop Start trunk) sides of the trunk-to-trunk call have to have this flag enabled (Yes). For non-T1 Loop Start trunks, this flag is available only when the answer supervision is set to "Polarity Reversal" or "Valid Call Timer with Polarity Reversal."

Hybrid Balance

4.375 *(LSC Analog Loop Start, LGC Ground Start, and MGCP trunks only)* Hybrid balance can be set to Normal or Short to allow the installer to adjust the hybrid impedance of the trunk based on local central office (or local exchange) conditions. To set the Hybrid Balance select the current Value and scroll to the desired option. Then press **ENTER** or click outside the field to save your change.

- **Short**: The short balance is used when trunks are supplied by other equipment located within the same building, using private, copper wire termination (e.g., an OPX supplied by another PBX that does not use the public network) or when the site is located close to the central office [local exchange].
- **Normal**: The normal balance can be used by most trunks.

NOTE: *(Loop/Ground Start trunks only)* In version 10.0 or later system you can use the Hybrid Balance Test feature to verify loop/ground start trunk characteristics and to determine the Optimal Hybrid Balance Setting for each CO trunk port interface. See "Run a Hybrid Balance Test" on page 6-246 for details.

Measured Echo Return Loss (ERL)

4.376 This read-only value is the "Finalized Hybrid Balance Test – ERL" measured in the most recent Hybrid Balance Test on a loop/ground start trunk. This value displays in Message Print output. See page 6-246 for details about running a Hybrid Balance test.

Last Hybrid Balance Test Timestamp

4.377 This read-only value indicates the month, day, year, and time of day of the most recent Hybrid Balance Test. If the test has never run on the selected trunk, the field displays "Untested" and the Measured Echo Return Loss (ERL) value is undefined. See page 6-246 for details about running a Hybrid Balance test.

NOTE: The Measured Echo Return Loss (ERL) and the Last Hybrid Balance Test Timestamp values are updated only with the start of each DB Programming session. If a test is run during an active session of DB Programming, the fields will *not* be updated with the latest test results until DB Programming is restarted. If Expanded Message Print is enabled, the test results appear in Message Print output. See the Expanded Message Print output example on page 6-251.

Service Base Number

4.378 *(Used for DID, DNIS, *DNIS*, and *ANI*DNIS*)* This is the base number for the trunk. Select the current Value and enter the digits in the text box (up to 32 digits). Then press **ENTER** or select another field to save the change.

NOTE: If using B-channel trunks, the central office will send fewer than 10 digits for the DNIS number on some ISDN calls. When this happens, the System needs to be able to construct a 10-digit number for the System OAI and Desktop Interface features. To do this it requires a base number for the trunk. for the DNIS number. When this happens, the System needs to be able to construct a 10-digit number for the System OAI and Desktop Interface features. To do this it requires a base number for the trunk. Enter the base number digits in the text box (up to 32 digits).

Number Of Digits To Receive

4.379 *(Used for DID, ANI, and DNIS)* This is the number of digits the System will use for directing the call. Select the current Value and enter or scroll to the correct amount (1-32) in the text box. Then press **ENTER** or select another field to save the change.

Service Base Number

4.380 *(Used for B-Channel trunks.)* This is the base number for the trunk. Select the current Value, and enter the digits in the text box (up to 48 digits). Then press **ENTER** or select another field to save the change.

4.381 Language: This field determines which language will be used for the voice prompts and displays when this trunk is used. This field can be set to any specific language so that the system can support more than two languages. The language choices are Use Primary Language, Use Secondary Language, American English, British English, Japanese, or Spanish. Note that Spanish voice prompts use Mexico City Spanish Dialect for VPU and EVMC, and American Spanish Dialect for EM. Select the current Value and use the drop-down list box to scroll to the appropriate choice). Then press **ENTER** or select another field to save the change.

NOTE: Setting a trunk Language field to a specific language will not work across nodes unless all the nodes in the system are at least 5.2F6 or newer since the older systems do not understand specific languages and expect to receive only Primary Language or Secondary Language values.

4.382 Loop Current Dial Tone Detection: *(Used on LGC ground-start trunks only.)* If this flag is enabled, the system processes an outgoing call as soon as dial tone is detected. If the flag is disabled, the system processes the call when the GS Dialing Wait After Connect timer expires. To enable this flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

4.383 Transmit and Receive Gain: These fields allow gain control programming for trunks on a T1/E1/PRI or on a T1, E1/PRI, or T1/PRI Card that has been modified using the T1 or E1 Gain Control Kit. All trunks, regardless of the board they are on, will have default transmit and receive gains of 0 dB. If the trunk resides on a board that does not support the gain control, the values are ignored. The range for the Transmit or Receive gain is -16 to +12dB. To select the gain setting, select its current Value and scroll to the desired setting. Then press **ENTER** or select another field to save the change. If gain has been adjusted using the hardware settings, be sure to check that the database settings match when upgrading to system version 5.1 or later software.

4.384 Send Digits En Bloc: *(Used for loop start and MGCP trunks only)* This field determines how the system sends digits to the trunk. If enabled (set to *Yes*), the system sends all of the available digits when the system seizes the trunk (e.g., when the user accesses ARS). If disabled (set to *No*), the system sends the digits to the trunk one at a time. By default, this is *disabled*, which is recommended for loop start trunks and MGCP gateways/endpoints.

5. SYSTEM FLAGS

5.1 Select Flags to view a list of all system-wide flags. You can enable a flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.) Some flags require you to choose between two setting (such as 12- or 24-Hour Time Display Format). For those flags, select the current Value and use the scroll box to select the desired value. Then press **ENTER** or select another field to save your change.

5.2 The system flags include the following:

- **Agent Help Display**: This flag determines whether phone users will see the Agent Help displays during an Agent Help call. In the default state, this flag is set to *Yes*.
- **Agent Help Tone**: This flag determines whether the parties on a call will hear a tone when a supervisor joins or monitors a call using the Agent Help feature. The frequency of the tones are determined by the Agent Help Tone Interval timer, described on page 6-329. In the default state, this is set to *Yes*.
- **Allow Green LEDs**: This flag gives the installer the option of having all red/green phones and red-only phones operate the same, or allowing the red/green phones to use the green LEDs. If this flag is enabled, the dual-color phones will use both the red and green LEDs. If it is disabled, the phones will operate in red-only mode, like the phones without green LEDs. In the default state, this is set to *No*.

  NOTE: This flag is not supported by multi-protocol endpoints.

- **Allow Immediate ACD Auto Connect After DND**: Enable this flag to allow calls to autoconnect to the ACD agent station immediately after the agent removes the station from DND. If this flag is disabled, the first call received after DND is disabled rings until the agent answers the phone (this is the current functionality). By default, this flag is set to *No*.

  NOTE: To use the autoconnect feature, ACD agents must answer calls using a headset.

- **Alternate Keyset Menu Display**: This flag determines the display that Executive Display, Professional Display, and Model 8560 Phone users will see during an outside call. In the default state, the main menu display includes the TRANSFER-TO-HOLD option. If alternate displays are enabled, the FLASH option appears in its place. In the default state, this flag is set to *No*.

  NOTE: Version 7.0 or later system software is required to support the menu display and buttons on the Model 8560.

- **Audible Message Indication for SL Sets**: If enabled, single-line sets will receive message waiting indications (six tones before dial tone) when a message is waiting. In the default state, this is *Yes*.
- **Audio Diagnostic Alarm Suppression**: This flag gives you the option of suppressing Alarm 128 (see page 7-20), which is generated when the Audio Diagnostics feature (see page 5-318) is used. By default, this flag is set to **No**, which means that the alarm is generated and displayed on the administrator's endpoint when a user access the Audio Diagnostics feature. If this flag is set to **Yes**, the alarm is suppressed.

- **Automatic Board Equip**: If this flag is enabled, all cards inserted into unequipped slots are automatically detected and the slot is equipped to reflect the inserted card when the system is powered up and a new programming session is started. (All ports on inserted cards are unequipped.) This simplifies initial configuration, because all cards can be installed into the equipment chassis and they will appear in DB Programming automatically, reflecting the actual configuration of the system. In the default state, this is *Yes*.

  NOTE: The Auto Board Detect function applies only to unequipped slots. Changes of card type are not automatically detected and previous programming of cards and ports is left intact when card types are changed in previously used slots. Therefore, you should use local programming mode and delete any cards that exist in slots you want automatically detected before beginning the direct or remote programming session.

  The Auto Board Detect feature can only function when a DB Programming session is not active.

  If a board that is automatically equipped requires a premium feature to be turned on, the system still equips the board. A Message Print error, however, is generated, indicating that the premium feature is required. If this occurs, you must upload a software license that has that premium feature enabled.

- **Barge-In Notification Display**: This flag controls whether the agent sees the **BARGE-IN IN PROGRESS** or **BARGE-IN TERMINATED** display whenever a supervisor barges-in or hangs up after barging-in. The default setting is **Yes**. To enable the Barge-In Notification Display, set this flag to **Yes**. To disable the Barge-In displays, set this flag to **No**.

- **Barge-In Notification Tone**: When this flag is enabled (**Yes**), each of the barged-in parties (not including the supervisor) can hear a tone on every barged-in call. The tone repeats at the frequency set by the Barge-In Notification Tone Frequency Timer (see below). The default setting is **No**.

  NOTE: If the supervisor attempts to steal the other party away from the agent, and the other party is on a node that is earlier than v9.0, the steal fails. The other party may be another station or a trunk on a different node. The supervisor will see **VERSION MISMATCH STEAL REJECTED** on his or her display.

- **Broadcast Alarms To All Administrators**: Determine whether system alarms will display on all administrator stations (set to "yes"). If not, they will display only at the primary attendant's station (set to "no"). In the default state, this is set to *Yes*.

- **Broadcast Station Off-Hook Alarms**: Determine whether off-hook alarms will be displayed on all administrator stations and the primary attendant's station (set to *Yes*). In the default state, this is set to *Yes*.

- **Companding Type**: Determine whether the system uses Mu-Law or A-Law companding.

- **Date Display Format**: This flag determines the order the Day-Month-Year appear in the system. This includes SMDR Header, Keyset Date Programming, and Message Print. In the default state, it is set to Month-Day-Year.

- **DISA Transfer Tone**: Select the ring or music option. If ring tones are enabled, callers will hear ringing instead of music when using DISA to call a station. In the default state, this is set to *Ring*.

- **Disable Confirmation Tone to Trunks**: When a Central Office (CO) trunk call is transferred to a Call Routing Announcement (CRA) application by Call Processing, a confirmation tone is played at the beginning of the CRA message. This flag disables the confirmation tone played at the beginning of a CRA message to a transferred CO trunk call. By default, this flag is disabled (set to *No*) and the user hears a confirmation tone.
- **Display Caller ID Name and Number**: Allows you to adjust the display on an endpoint for an incoming CO call to always show caller ID name and number. When this flag is enabled, the endpoint user will not see the elapsed call timer that shows the duration of the call. Instead, the outside caller's name appears on the first line and the caller's number appears on the second line of the endpoint display. See page 5-76 for details.
- **Drop Incomplete Outgoing Calls**: Occasionally, a station user will seize a trunk at the exact time a call is coming in on that trunk and the user will be connected to the incoming call instead of dial tone. (This is called "glare.") If the Drop Incomplete Outgoing Calls option is enabled (set to *Yes*), the resulting call on that trunk will be dropped, as it would be if the user did not dial a valid telephone number. If the option is disabled (set to *no*), the call will remain connected regardless of the number of digits dialed, if any. In the default state, this is set to *Yes*.
- **Enable SL Pulse Detection**: Enable this flag to detect pulse tone from single-line phones that terminate at the system. If this flag is disabled, the system will not be able to detect digits received from pulse-tone phones. By default, this flag is set to *No*.
- **Enable T1/E1/PRI DTMF Dialing**: If this flag is enabled, any T1/E1/PRI cards on the system will be able to use their DTMF dialing capability. If it is disabled, the DTMF dialing will be performed by the CPC or CPU, as usual.
- **Handsfree Announce On System Forward Transfer**: When a station user transfers a call to the principal station of a forwarding path, this option determines what the caller will hear. If enabled (yes), the caller will place a handsfree call to the principal station and can announce the transfer (the call does not enter the forwarding path). If the option is disabled, the transferring station user will hear ringing and can announce the call only if it is answered by the principal station or one of the forwarding points. This defaults to *No*.
- **House Phone Mode**: *(For single-line house phones only)* The House Phone Mode flag determines whether a single-line House Phone returns dial tone or does not return dial tone after the called party disconnects. At the System level of programming, all House Phones may be set in either Normal mode or Restricted mode. At the individual Endpoint level of programming, a House Phone must be programmed to dial specific digits as soon as the handset is taken off-hook. The programmed digits may ring a specific endpoint or a Hunt Group that rings multiple endpoints.

  Normal mode allows the user to enter a feature code or place a call after the automatically called number hangs up. Restricted mode prevents the user from performing any operation other than placing a House Phone call. The System default state is *Normal*.
- **Insufficient Bandwidth Alarm**: This flag indicates whether the administrator phone displays a message when alarm 032 is generated. Alarm 032 is generated whenever the IP network does not have enough bandwidth to support the IP call that is currently connected to the device. If enabled (set to Yes), SYS ALARM #32 X<*ext*> INSUF BAND, where <*ext*> is the extension number of the affected device, appears on the administrator phone. If disabled (set to No), the phone does not display any information when the alarm is generated. By default, this flag is *enabled*.

- **Music On Hold For IC Calls**: This option determines whether a user hears music when placed on hold by another station or when camped onto a resource. If set to *No*, IC callers never hear music, although they may hear other tones such as tick tone and ringback. In the default state, this is *Yes*.
- **Network with United States *(for Europe only)***: The U.K. allows the capability to network U.S. and U.K. systems together using the Inter-Tel IP Networking Module (previously called InterPrise 3200N) Products. In the default state, this is *No*.

  When this flag is set to No, no U.S. nodes can be added to the U.K. network.

  When this flag is set to Yes, aU.K. system can network with a U.S. system.

  NOTE: The U.S. and U.K. network must use the IP Networking Module products at v1.3 or higher.
- **OHVA Enable**: Determine whether the Off-Hook Voice Announce (OHVA) feature will be enabled system-wide. In the default state, this is *Yes*.
- **Play Pre-Record-A-Call Message**: This flag enables the Voice Processor to play a message before the Record-A-Call message.
- **Receive Network Alarms**: This flag determines whether the node will receive and display network-wide alarms sent by other nodes in the network. In the default state, it is *No*.
- **Record-A-Call Display**: This flag determines whether phone users will see the Record-A-Call displays when the feature is used. In the default state, this flag is set to *Yes*.
- **Record-A-Call Tone**: This flag determines whether the parties on a call will hear a tone when the Record-A-Call feature is in use. The timing of the tones are determined by the Record-A-Call Tone Interval timer, described on page 6-329. In the default state, this is *Yes*.
- **Ring Flash On DSS Lamps**: When enabled, this option allows DSS/BLF buttons to indicate a ringing call to a station by showing the ring flash on the associated button. If disabled, the button will show that the station is busy (solid lamp). In the default state, this is set to *Yes*.
- **Send Network Alarms**: This flag determines whether a node will broadcast alarms that occur on that node to the rest of the network. In the default state, this is set to *No*.
- **Single Idle Time for All Hunt Groups**: This flag invokes an Automatic Call Distribution (ACD) algorithm that sends an incoming call to the agent station with the longest idle time in all the hunt group queues to which that station belongs. For agent stations belonging to multiple hunt groups, this feature allows calls to be distributed to other stations having the longest idle time, regardless of the station idle time status in an individual hunt group. It is set to *No* by default.
- **SLC Ring Zones**: In some installations, where more than one single-line device is connected in series to an SLC circuit, zoned ringing may be necessary. The system can support devices with 3 Ringer Equivalences (REs) per circuit. However, the system can only ring eight REs at once. Therefore, if there are more than eight REs on any SLC, the system-wide SLC Ring Zones flag must be enabled to allow the single-line devices to ring in smaller groups (zones). In the default state this flag is *disabled*.

- **SPCL Key Required For Feature Code Entry**: Depending on the setting of this flag, users can either enter feature codes immediately after lifting the handset or while on hook, or they must press the SPCL button before entering the feature code. If this flag is enabled, phone users must always press the SPCL button before entering a feature code. In the default state, it is set to *no*.
- **System Speed Dial Override Toll Restriction**: If this option is enabled, the system speed-dial numbers can be speed dialed at any station regardless of toll restrictions. If disabled, all system speed-dial numbers are subject to toll restriction. In the default state, this is *No.*
- **Time Display Format**: This determines whether phone displays will show the time in 24-hour or 12-hour (AM/PM) format. In the default state, it is set to *12 hour.*
- **UCD Station-Monitor Indications**: This flag determines whether station monitoring tones are sent to a UCD hunt group member when the hunt group supervisor is monitoring the call. In the default state, this flag is *enabled.* (Note that call monitoring may be illegal in some locations. It is up to the end user to ensure that use of this feature is in compliance with local laws.) The frequency of the tone is determined by the UCD Station Monitor Indication Frequency timer.
- **Use Green LEDs for Direct Rings**: This flag determines whether direct ring-in calls cause a red or green lamp. If the flag is enabled, direct ring-in calls to your station will be green. If disabled, all direct ring-in calls will cause the LED to be red. In the default state, this flag is enabled. (The Allow Green LEDs must also be enabled for this feature to work.)
- **Validate Voice Mailbox Numbers**: When enabled, the system will check that a dialed mailbox number matches a programmed extension number when the record-a-call or transfer-to-voice mail features are used. In the default state, this is set to *Yes.*
- **Wrap-Up Mode For Holding ACD Calls**: If enabled, this flag prevents an agent from receiving additional ACD hunt group calls while a call the ACD agent placed on hold is holding and the agent's station is idle. (The station can still receive non-ACD calls, as usual.) If the flag is disabled, the agent will be available to receive additional ACD calls as soon as an ACD call is placed on hold. In the default state, this flag is *disabled.*

6. HUNT GROUP-RELATED INFORMATION: ACD AGENT IDS

NOTE: The ACD Hunt Group software license is required to use this feature.

6.1 There is an ACD hunt group option that allows the use of Agent ID numbers in place of station extensions in the hunt group list. (See page 6-160 for Hunt Group programming information.)

6.2 ACD hunt group members are referred to as "agents." Agents log in to the ACD hunt group to receive calls and log out to halt ACD hunt group calls. An ACD hunt group can be programmed to circulate calls to agents in two ways:

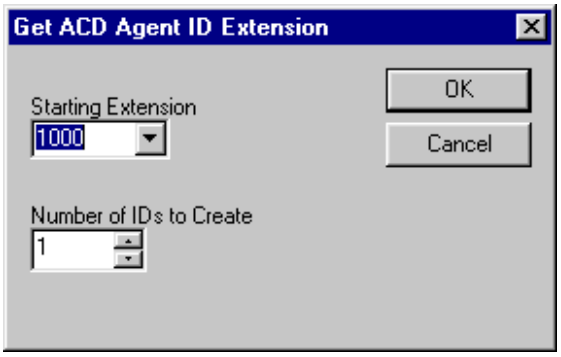
- **Agent IDs**: If the hunt group is programmed to use ACD Agent IDs, each agent is assigned an Agent ID number which the agent enters during the login procedure. The hunt group calls are routed to logged in agents, according to their Agent ID number instead of their station extension. Because the Agent ID is not associated with any station extension, the agent can use any station in the system to log in and does not have to use the same station every time.
- **Members**: If the hunt group ***is not*** programmed to use Agent IDs, it will have a list of stations and will send calls to the stations where agents are logged in.

6.3 When a call camps on to an ACD hunt group that uses Agent IDs, only the agents currently logged in to the hunt group will receive camp-on indications. ACD hunt group supervisors will receive visual camp-on displays if they are programmed as members of the hunt group and have the ACD Agent Logout feature enabled.

6.4 Select Hunt Groups and then double-click on **ACD Agent IDs** to view the current list of agent IDs, if any. Use the following procedures to create, program, or delete Agent IDs.

6.5 To create ACD Agent IDs:

1. Right-click anywhere in the right half of the window. An option box appears.
2. Select **Create ACD Agent ID**. The following window appears. Enter the desired extension number (or lowest extension number, if creating multiple IDs) in the text box or scroll to the desired number. Then enter the number of IDs you want to create and select **OK** to continue. (Or select **Cancel** to exit without creating the IDs.)



The new ACD Agent IDs appear in the list without a description or username.

6.6 Description: The ACD Agent ID description can be up to 20 characters long. To program the description, select the current description and type the new entry in the text box. Then press **ENTER** or select another field to save the change.

6.7 **Auto Connect**: When this flag is enabled, and headset mode is enabled, ACD hunt group calls will automatically be connected following a short ring burst. When the ACD agent logs in, however, the first call rings until the ACD agent answers it. For subsequent calls, the agent hears the ring burst on the phone, and the call is automatically connected. When the ACD agent removes the station from DND, the call may or may not ring until the agent answers it. This is dependent on the **Allow Immediate ACD Auto Connect after DND** flag (under System\Flags). If this flag is enabled, the agent is automatically connected to all calls, including the first one received after exiting DND. If this flag is disabled, the first call rings until the agent answers it, but subsequent calls are automatically connected. The Auto Connect flag overrides the station's Transfer-To-Connect station flag (see page 6-213) and is disabled by default. To enable it, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

6.8 **To delete ACD Agent IDs:** Highlight the agent ID(s) and then right-click. Select **Delete**.

NOTE: To highlight a series of IDs, hold down **SHIFT** while selecting the first and last IDs you want to delete. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

7. IP-RELATED INFORMATION

7.1 The IP-related information flags include the following:

- Call Configurations (see below)
- SIP UDP Listening Port Enabled (see page 6-267)
- SIP UDP Listening Port (see page 6-267)
- System NAT IP Address (see page 6-267)

Call Configurations

7.2 The IP-Related Information field also contains the Call Configurations field. This field defines the settings that IP phones and SLAs, SIP endpoints, and MGCP gateways and endpoints will use when connected to a call. After programmed, you assign each device a specific Call Configuration. This allows you to program one call configuration for multiple devices.

7.3 By default, all IP devices are placed in Call Configuration 1, which is programmable. If desired, you can program up to 25 different Call Configurations (250 if using a CPS1024 or ATM System).

NOTE: It is not necessary to program the same Call Configuration for IP devices that communicate using P2P. When the system detects that the IP devices have different settings, it uses the setting that consumes the least amount of bandwidth.

ALSO: You cannot delete the default Call Configuration.

7.4 **To add a Call Configuration,** right-click and select **Add To Call Configurations List**. Enter the starting ID and the number of call configurations to program. Then click **OK**.

7.5 **Description**: The Call Configuration description can be up to 20 characters long. To program the description, select the current description and type the new entry in the text box. Then press **ENTER** or select another field to save the change.

7.6 To program the Call Configuration information, double-click the Call Configuration ID and configure the following fields:

- **Audio Frames/IP Packet**: Enter the number of audio frames the system inserts into each packet. The lower this value, the less latency in the signal, but the higher the bandwidth consumption, which increases the chance of jitter and network congestion. The higher this value, the more latency, but the lower the bandwidth consumption, which lowers the chance of jitter and network congestion. To change the number of frames, click the current Value and use the scroll box to select the new value. Then press **ENTER** or click another field to save the change. The valid range is *1-8* packets, and the default is *3* packets.
- **Average In Time Frame Percentage Threshold and Average In Time Frame Timer**: Indicate when network characteristics inhibit voice connections. When the average number of In Time frames falls below the Average In Time Frame Percentage Threshold, and the average number of In Time frames stays below that threshold for the time given by the Average In Time Frame Timer (in seconds), the system displays a warning about insufficient bandwidth. To change the percentage, click the current Value and use the scroll box to select the new value. Then press **ENTER** or click another field to save the change. The range for Average In Time Frame Percentage Threshold is *0-100%*, and the default is *60%*. The range for Average In Time Frame Timer is *0-255* seconds, and the default is *5* seconds.

- **Echo Suppression Sensitivity Level**: Select the threshold at which the system suppresses the echo from the far end at the beginning of a conversation. A value closer to zero results in very little echo suppression (full duplex), and a value closer to 100% results in a large amount of echo suppression (half duplex). To change the suppression sensitivity level, click the current Value and use the scroll box to select the new value. Then press **ENTER** or click another field to save the change. The range is *0-100%*, and the default is *60%*.
- **Minimum Playback Time**: Enter the minimum time, in milliseconds, that packets wait in the receive buffer before the system plays the audio. The higher this number, the more latency in the signal; however, it is less likely that network problems like jitter would cause lost or late audio packets. The lower the minimum playback time, the less latency there is in the signal; however, there is a greater chance of jitter. To change the playback time, click the current Value and use the scroll box to select the new value. Then press **ENTER** or click another field to save the change. The range is *1-320* ms, and the default is *80* ms.
- **Transmit DTMF Level**: Defines the level at which the system injects tone onto the backplane when receiving an out-of-band DTMF tone. Your options are *U.S.A.*, *U.K.*, or *Japan*. To change the country, click the current Value and use the drop-down box to select the new value. Then press **ENTER** or click another field to save the change. The Extended Value and dB columns automatically update to reflect the values associated with the country. By default, this is the country associated with the language selected in the Session Manager.
- **DTMF Encoding Setting**: Select the vocoder type used to send DTMF. Your options are *G-711 Mu-Law*, *G-711 A-Law*, *G-729*, *G-729B (VAD)*, and *RFC 2833*. For MGCP gateways and endpoints, this value must either match the Speech Encoding Setting field or be set to RFC 2833. By default, this is *G.711 Mu-Law*.
- **Speech Encoding Setting**: Select the vocoder that the system uses when transmitting speech data. The options are *G.711 Mu-Law*, *G.711 A-Law*, *G.729*, and *G.729B (VAD)*. For MGCP gateways and endpoints, if the DTMF Encoding Setting is set to RFC 2833, set this value to either G.729 or G.711; if the DTMF Encoding Setting is set to G.729 or G.711, this value must match (i.e., be set to G.729 or G.711). To change the vocoder, click the current Value and use the drop-down box to select the new vocoder. Then press **ENTER** or click another field to save the change. The default is *G.729*.

NOTE: For MGCP gateways and endpoints, the DTMF Encoding Setting cannot be set to G.711 while the Speech Encoding Setting is set to G.729 (and vice versa). If set incorrectly, users will not be able to dial DTMF tones while on a call.

- **Echo Suppression**: Determines if callers hear echo while the canceller adapts. If enabled (set to *Yes*), the conversation is played in half-duplex mode for a short period of time while the system adapts the echo canceller based on audio characteristics. If disabled (set to *No*), callers may hear echo while the canceller adapts. To disable the flag, select the current Value and then click the check box to remove the check mark. Then press **ENTER** or select another field to save the change. (To add the check mark and enable the flag, select the check box again.) By default, this is *enabled*.
- **Echo Saturation Blocker**: Determines if the system switches to half-duplex mode when the volume becomes very loud. If enabled (set to *Yes*), the conversation is played in half-duplex mode when the volume increases to a certain level. If disabled (set to *No*), the system always plays the conversation in full-duplex mode. To disable the flag, select the current Value and then click the check box to remove the check mark. Then press **ENTER** or select another field to save the change. (To add the check mark and enable the flag, select the check box again.) By default, this is *enabled*.

NOTE: Full-duplex mode can lead to echo audio because the echo canceller cannot cancel echo caused by very loud or saturated audio.

7.7 **IP Stations**: IP stations include IP endpoints (except the IP SoftPhone), IP SLAs, and multi-protocol (SIP or IP) endpoints. To view a list of IP stations that are currently assigned to the Call Configuration, double-click **IP Stations**.

7.8 **To add stations to the list:**

1. Right-click and select **Move To IP Stations List**.
2. Select IP Keyset or IP Single-Line Adapter and click **Next**.
3. Highlight the devices to add to the list and click **Move To List**.
4. Click **Finish** when all of the desired devices have been moved.

7.9 **IP Trunks**: IP trunks include MGCP gateways and endpoints. To view a list of IP trunks that are currently assigned to the Call Configuration, double-click **IP Trunks**.

7.10 **To add trunks to the list:**

1. Right-click and select **Move To IP Trunks List**.
2. Select MGCP Endpoint or MGCP Gateway and Endpoint and click **Next**.
3. Highlight the devices to add to the list and click **Move To List**.
4. Click **Finish** when all of the desired devices have been moved.

SIP UDP Listening Port Enable

7.11 The SIP UDP Listening Port Enable flag is set to *Yes* by default. This port configures the receive call control communication path from external SIP gateways to the Axxess system. The transmit call control communication path resides in the SIP gateway programming. If your site is on the public Internet and you are not using SIP, you can disable the SIP UDP Listening Port Enable flag by selecting the current Value and removing the check mark from the check box. When you disable this flag, the SIP UDP Listening Port field is disabled and cannot be programmed. If you change the SIP UDP Listening Port Enable flag, the change does not take effect until the next system reset.

SIP UDP Listening Port

7.12 By default, the port number for the SIP UDP Listening Port is *5060* and the SIP UDP Listening Port Enable flag is set to *Yes*. If the site has a conflict with port 5060, the port number can be changed. The valid range of port numbers are 1000 through 65468. If the SIP UDP Listening Port Enable flag is set to No, the SIP UDP Listening Port is no longer programmable. If you change the port number for the SIP UDP Listening Port, the change does not take effect until the next system reset.

System NAT IP Address

7.13 The System NAT IP Address field provides the ability to put a SIP gateway behind a NAT. Enter the NAT (public) IP address that the system has on the NAT side of the firewall. To correctly inform the gateway where the SIP messages are originating from, you must also program the SIP Gateway Name field (see page 6-328). For complete NAT traversal and programming information, see NAT Traversal for IP Endpoints on page F-14.

8. SYSTEM MAINTENANCE

8.1 Use the Maintenance programming area to set the following parameters:

- Call Cost (see below)
- Freeze Zones (see page 6-270)
- Major Reset Scheduling (see page 6-271)
- Message Print (see page 6-274)
- SMDR (see page 6-276)

8.2 Each of these is discussed individually in the following pages.

8.3 In systems running v9.0 and later, when DB Studio is operating in OLM mode, you can clear the following Call Processing Logs on a CPC:

- History Queue
- Message Print Log
- Network Diagnostics Log
- Reset Log
- Software Performance Statistics Log

8.4 **To clear a log**, in OLM, under System\Maintenance\History Queue, right-click in the screen and select the log you want to clear.

NOTE: Do not use OLM mode unless you are instructed to do so by support personnel.

A. CALL COST

8.5 When you select Call Cost the current daytime rates and multiplication factors are displayed. To change a rate, select its current Value and enter the new value in the text box. Then press **ENTER** or select another field to save the change.

8.6 **To prepare to program call cost:**

1. Determine the daytime rates, in dollars [pounds in Europe] per minute, for the following types of calls (all default values are 0.00). Use several telephone bills from months with typical usage to calculate the average cost per minute of each type of call. Record the charges in dollars (pounds) and cents (pence) from 00.00 to 99.99. You may need to adjust the calculations later to more accurately estimate actual call costs. (This program is to be used as an estimate only; see page 5-305 in FEATURES for details.)

   For U.S. Systems:

   — Local calls
   — Toll – local calls
   — Toll – long distance calls
   — Operator-assisted calls
   — International calls
   — Incoming calls
   — Free calls
   — Network calls

For European Systems:

— Local calls

— Toll (National) calls

— Operator calls

— International calls

— Incoming calls

— Free calls

— Network calls

2. Determine the multiplication factors that adjust the daytime (peak) per-minute call cost for evening (standard) and weekend (cheap) rates.

   For U.S. Systems: The multiplicative factor adjusts the daytime per-minute call cost for evening and weekend rates of outgoing calls. For example, the evening call cost multiplier is 0.65 if calls are 35% less expensive after 5:00PM. The evening (E) multiplicative factor and night/weekend (N/W) multiplicative factors are used on the following schedule:

Table 6-24: *Multiplicative Factors Schedule (U.S.)*

	SUN	MON	TUE	WED	THU	FRI	SAT
8 AM to 5 PM	N/W	No Multiplication Factors Apply					N/W
5 PM to 11PM	E	E	E	E	E	E	N/W
11PM to 8 AM	N/W	N/W	N/W	N/W	N/W	N/W	N/W

For European Systems: The multiplicative factor adjusts the peak per-minute call cost for standard and cheap rates of outgoing calls. For example, the call cost multiplier is 0.65 if calls are 35% less expensive after 6:00PM. The standard (S) and cheap (C) multiplicative factors are used on the following schedule:

Table 6-25: *Multiplicative Factors Schedule (Europe)*

	SUN	MON	TUE	WED	THU	FRI	SAT
9 AM to 1 PM	C	No Multiplication Factors Apply					C
1 PM to 6 PM	C	S	S	S	S	S	C
6 PM to 9 AM	C	C	C	C	C	C	C

B. FREEZE ZONES

8.7 You can freeze the system to "lock" the current state of the system fault history queue, which is a sequential list of all system commands and inputs. This list, when decoded, can be used to determine a series of events which may have resulted in an error. Unfreezing the system "unlocks" the current state of the history queue, and resumes event collection.

8.8 Using freeze zones, you can determine which nodes in the network are frozen during each freeze request. There can be up to 10 freeze zones in the database.

8.9 To program a freeze zone, first select Freeze Zones to display the list. At this level you can select the description field and enter a descriptive name for the zone in the text box. Then, to program an individual zone, double-click on it. A list of the nodes in that freeze zone, if any, appears. To add or delete nodes, do the following:

8.10 **To add nodes to the freeze zone:** Use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To List.** A window appears that asks you to select the type of device to include.
3. Select the node type(s) and then select **Next**. A list of nodes appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

> **NOTE:** To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate nodes and then select **Add Items**. When you have added all the necessary nodes select **Finish**. Your selections appear in the list.

> **NOTE:** If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

8.11 **To delete nodes from the freeze zone:** To delete one or more node from the list, highlight them and then right-click. Select **Remove Selected Items**.

C. MAJOR RESET SCHEDULING

8.12 A system reset is required to ensure proper system operation after any cards or devices are equipped or unequipped, an extension list is changed, a trunk group list is changed, or an extension number is changed. A delayed reset can be enabled as described below.

NOTE: A major system reset will terminate all calls and will end the programming session. However, a delayed system reset will check system activity at the scheduled time and wait for the system to become idle and any active programming session to end before allowing the reset to occur. A system reset is no longer required for UCD or ACD hunt groups after the hunt group member list changes.

8.13 **System Requires Reset**: *(Read Only)* This field indicates if Call Processing requires a reset. If *Yes* is displayed, the system will reset at the time indicated in the System Delayed Major Reset field. If *No* is displayed, a delayed major reset will not occur, even if one is scheduled. (This field was previously called System Required Reset.)

8.14 **Scheduled Reset Time**: If the System Requires Reset flag is enabled, this field determines the default time for ***scheduled*** system resets. Scheduled major resets can be scheduled when system administrators program the database or when **Reset Now** is selected from the Operations menu. To program the default time for delayed resets of the system, select the current hours, minutes, or AM/PM value and then scroll to or enter new setting in the text box. Press **ENTER** or select another field to save the change. By default, this is *12:01 AM*.

8.15 In previous versions, a scheduled major reset was required when various programming operations were completed. As of version 8.224, however, the following operations no longer require a major reset:

- Changing ACD agent IDs.
- Changing extension numbers (this includes any extension number, not just UCD/ACD, such as station, trunk group, etc.)

8.16 Because the major reset is not required for these actions, administrators are not prompted to perform an immediate or scheduled reset if performed from an administrator's phone. Instead, these database changes take effect immediately.

8.17 **Force Reset If Not Idle**: Normally, the system will not perform a major reset if there are any active calls. However, if this flag is enabled, the system will force a major reset at the specified time. The time is either the Scheduled Reset Time or the Scheduled Reset Time programmed above. The major reset will cause all active calls on the system to be dropped. The flag should be used only on systems which are busy 24 hours each day and, therefore, do not have a consistent time when all resources are idle and a normal delayed major reset can be performed. This flag affects any request to perform a major reset, whether it be by the system itself for resource reconciliation, or requested through DB Programming. The default for this flag is *disabled.* (This field was previously called Forced Delayed Major Reset.) To enable this flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

NOTE: If this flag is enabled, the installer should inform the customer that enabling this flag will drop all active calls at the specified time, should a major reset be necessary. This will not happen every day, but it will happen occasionally. The customers should be aware of this so that they do not mistake the reset for a system failure.

8.18 **Days of the Week**: Select the days of the week on which you want automatic resets to occur. Resets occur on the days of the week that are enabled, provided the **Always Reset On Days Of Week** flag is enabled. By default, all days of the week are *disabled*.

8.19 **Always Reset On Days Of Week**: Set this option to *Yes* to have resets occur on the specified days of the week. If this option is disabled, resets will not occur on the specified days. By default, this is *disabled*.

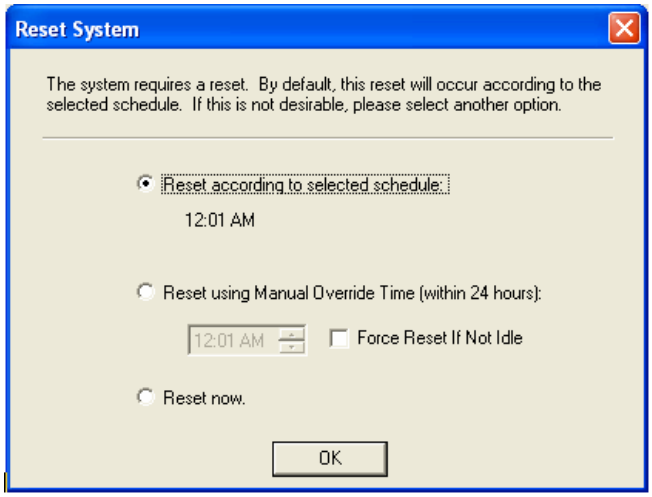
NOTE: If a red "x" is displayed next to **Always Reset On Days Of Week**, you have not enabled any of the days of the week (e.g., Monday, Tuesday, etc.). You must have at least one day of the week enabled; otherwise, resets will not occur.

8.20 When finished, the system reset as programmed.

NOTICE
A system reset terminates all calls in progress. Inter-Tel recommends that you schedule resets to occur after normal business hours.

8.21 **To perform an immediate reset:** Select **Reset Now** from the Operations menu.

8.22 **Reset System Dialog**: When you make a change that requires a reset, the Reset System dialog appears upon existing DB Programming.



8.23 This dialog has the following options:

- **Reset according to selected schedule**: Resets the system according to the schedule that the user has selected in the Major Reset Scheduling folder (System\Maintenance). The reset schedule that is currently selected is also displayed.

- **Reset using Manual Override Time (within 24 hours)**: If selected, you can use the scroll box to change the scheduled reset time to a different time (within 24 hours). For example, if the system is programmed to reset on Wednesdays at 11:30 PM, and you schedule a manual override time for 10:00 PM on a Wednesday, the system will reset at 10:00 PM and then again at 11:30 PM. By default, this is 12:01 AM. Note that the Manual Override Time is only used when the system requires a reset. This option can also be accessed from the Major Reset Scheduling folder in OLM mode. (This field was previously called Reset at Delayed Major Reset Time.)

NOTE: The Reset using Manual Override Time (within 24 hours) value must be at least 15 minutes before the programmed Periodic Backup Database Save Time value (under System); otherwise, the backup may not occur. If you select a value that is less than 15 minutes before the Periodic Backup Database Save Time, an error message appears, requesting a new value.

- **Force Reset If Not Idle**: Enable this option to force a reset, even if there are active calls. If disabled, the reset is not performed until the system is idle. By default, this is set to *No*. This option can also be accessed from the Major Reset Scheduling folder. (This field was previously called Forced Delayed Major Reset.)
- **Reset now**: Performs an immediate system reset. Do not select this option if the system is currently backing up the database; otherwise, the backup is aborted.

8.24 Click **OK** when finished.

D. MESSAGE PRINT

8.25 System messages can be printed to give service personnel and Inter-Tel engineers information about system status during troubleshooting. You may enable any combination of the error message types. The available message types are:

- **Alarm Messages**: Indicate that a minor alarm has occurred, but that general system operation was not affected.
- **Information Messages**: Provide information concerning system operation.
- **Severe Messages**: Indicate that a severe error has occurred in the system.
- **Warning Messages**: Indicate that a condition exists which may affect system performance.
- **Network Dump**: Provides information concerning network operation.

8.26 The fields you must program to set up message print include the following:

- Output Port
- Local Backup Port
- Message Print Output Active
- Output Device Line Width
- Print Alarm Messages
- Print Information Messages
- Print Severe Messages
- Print Warning Messages
- Print Network Dump

8.27 See FEATURES, page 5-313, or TROUBLESHOOTING, page 7-12, and the *Message Print Diagnostics Manual* (part number 550.8018—supplied on the software CD) for more information regarding system messages and alarms.

8.28 **Output Port And Local Backup Port**: Each node has its own message print programming, message print output port, and message print output port backup. There should be a message print terminal at each node to monitor node and network performance and aid in troubleshooting.

- If a node's Message Print output port is a node, the network sends Message Print records to the specified node.
- You cannot select a node as the backup message print output port.
- If Message Print output programming forms a loop, the system will send the output to the node's backup Message Print port. For example, if the Message Print port on Node 1 routes to Node 2 and the Message Print port on Node 2 routes to Node 1, the configuration will cause an infinite loop. Message Print reports for Node 1 would be printed to the backup serial port on Node 2 and vice versa.

8.29 Select the output port and backup port for the message print reports, using one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the existing port. An option box appears.
2. Select **Change Port**. A window appears that asks you to select the type of device to include.
3. Select **None**, **Remote Node**, or **Serial Port (local)** and then select **Next**. The list of ports or nodes appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the desired port and then select **Finish**. Your selection appears in the appropriate port field.

NOTE: Do not connect a Voice Processor to a serial port that is programmed for message print output (such as PP100, PP101, etc.). This may cause the serial port to lock up and the Voice Processor will not be connected to the port.

8.30 **Message Print Output Active**: To activate the error/message reporting feature, enable the Message Output Active flag. To enable the flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

8.31 **Output Device Line Width**: To indicate whether the output device has 64, 80 or 132 character columns, select current Value and scroll to the desired column width. Then press **ENTER** or click outside the field to save the change.

8.32 **Print Options**: These flags determine the types of error messages to be included in the error report. To enable a flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.) The message types are described on the preceding page.

E. SMDR

8.33 The station message detail reporting (SMDR) produces a record of calls and their costs. Report format is discussed in the FEATURES section, starting on page 5-308.

8.34 Each node in a network has its own SMDR programming. You can enable or disable network call records on each node. SMDR information is programmed as described in the following paragraphs.

8.35 **Devices**: To assign the stations and trunks that will be included in the SMDR output, double-click on **Devices**. A list of current devices, if any, appears. You can add or delete devices as follows:

8.36 **To add devices:**

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To Devices List**. A window appears that asks you to select the type of device to include.
3. Select the device type(s) and then select **Next**. A list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

> **NOTE:** To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate items and then select **Add Items**. When you have added all the desired devices select **Finish**. Your selections appear in the list.

> **NOTE:** If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

8.37 **To delete devices:** Highlight the item(s) in the list and then right-click. Select **Remove Selected Items**.

8.38 **Output Port and Local Output Port**: Each node has its own SMDR programming, SMDR output port, and a local (backup) SMDR output port. You can enable or disable network call records on each node. By default, the system suppresses network call records. However, when they are enabled, the following applies:

- If a node's SMDR output port is a node, the network sends SMDR records to the specified node.
- If SMDR output programming forms a loop, the system will send the SMDR output to the node's local SMDR port. For example, if the SMDR port on Node 1 routes to Node 2 and the SMDR port on Node 2 routes to Node 1, the configuration will cause an infinite loop of SMDR routing. SMDR reports for Node 1 would be printed to the local serial port on Node 2 and vice versa.

8.39 Select the output port and local port for the message print reports as follows:

1. Right-click on the existing port. An option box appears.
2. Select **Change Port**. A window appears that asks you to select the type of device to include.

3. Select **None**, **Remote Node**, or **Serial Port (local)** and then select **Next**. The list of ports or nodes appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the desired port and then select **Finish**. Your selection appears in the appropriate port field.

NOTE: Do not connect a Voice Processor to a serial port that is programmed for SMDR (such as PP100, PP101, etc.). This may cause the serial port to lock up and the Voice Processor will not be connected to the port.

8.40 SMDR Output Active: To activate the SMDR reporting feature, enable the SMDR Output Active flag. To enable the flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

8.41 Output to System Manager: If disabled, SMDR information is not available to System Manager users. If enabled, System Manager users can run call reports for this node. To enable the flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

8.42 Display Elapsed Time in Seconds: To allow SMDR to give a more accurate representation of elapsed time, an option can be enabled that records the elapsed time of calls in seconds instead of minutes. For calls up to 999999 seconds long, the ELAPSED TIME field will show "S=XXXXXX" (XXXXXX represents the number of seconds). For calls lasting longer than 999999 seconds, ELAPSED TIME will show "HH:MM" (hours and minutes rounded up to the nearest minute). If you want to have call durations of fewer than 999999 seconds displayed in seconds, enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

8.43 Display "O/I" for Operator and International Calls: If enabled, operator and international calls are displayed in SMDR as one entry under the call-type abbreviation "O/I." (This is the functionality in software versions 6.0 and earlier.) If disabled, operator and international calls are displayed separately in SMDR: operator calls under "OP," and international calls under "INT." By default, this is *enabled*. To disable this flag, select the current Value and then select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To enable the flag, select the check box again.)

NOTE: If using Inside Track, this flag must be enabled.

8.44 Display Redirected Station: A station that transfers or manually forwards a call to the public network can be recorded in the SMDR report. To enable this option, go to System\Maintenance\SMDR, and select **Display Redirected Station**. Then, set the value to **Yes**. This flag is disabled by default. Then press ENTER or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

NOTE: To display redirected information for trunk-to-trunk calls, add the trunks and stations to the SMDR list. If the trunks and stations are not listed, the stations that transfer or manually forward CO calls will not be recorded in the SMDR report for redirected calls.

8.45 Record Calls: The Record Calls options determine the content of the SMDR output. To choose an option, select its current Value and place a check mark in the box. To remove an option, select remove the check mark. Options include the following:

- **Record All Incoming Calls**: Includes all incoming calls in the SMDR report.
- **Record All Local Calls**: Records all local, non-toll, valid calls.
- **Record All Free Calls**: Records all calls that use the "free" call cost.
- **Record All Ring-in Diagnostics**: A ring-in message is recorded for every incoming call (whether answered or unanswered) to indicate how long it rang. All incoming calls are recorded, even those involving stations not listed in the station list.
- **Record All Toll Local Calls** *(U.S. only)*: Records all valid local toll calls.
- **Record All Toll Long Distance Calls** *(U.S. only)*: Records all valid long distance toll calls.
- **Record All Toll (National) Calls** *(Europe only)*: Records all valid long distance toll calls.
- **Record All Operator Calls**: Records all operator-assisted calls.
- **Record All International Calls**: Records all international calls.
- **Record All DISA Calls**: Includes all DISA calls in the SMDR report.
- **Record All Conference Calls**: Includes all conference calls in the SMDR report.
- **Record All DID/DNIS Calls**: Records all calls received through DID and DNIS.
- **Record All Trunk To Trunk Calls**: Records all calls made from one outside caller to another.
- **Record All Network Calls**: The system generates SMDR records (labeled "NET") for node-to-node Private Networking calls on the nodes where each trunk used resides. For example, if a caller on Node A places a call using a trunk group on Node B, the SMDR report for Node A will show a NET call to Node B and the SMDR reports on Node B will show a NET call from Node A. (This flag has no effect on the Record Off-Node Devices flag described above.)
- **Record All Off-Node Devices**: You can determine whether calls placed by off-node devices, through this node, are included in the reports. (This flag has no effect on the Record All Network Calls flag described above.)

8.46 Suppress Digits: The Suppress Digits options determine which digits, if any, will be suppressed when the dialed digits are reported. To choose an option, select its current Value and place a check mark in the box. To remove an option, select remove the check mark. Options include the following:

- **Suppress Absorbed Digits**: Absorbed digits (on local or PBX lines) will not appear in the report if this option is selected.

  NOTE: If absorbed digits are repeatable on a local line, the absorbed digits will not appear in the SMDR report even when repeated.

- **Suppress Equal Access Digits** *(U.S. only)*: Equal access digits will not appear in the report if this option is selected.
- **Suppress Outside Party Number**: Caller information that is received through ANI or Caller ID [CLIP] will not appear if this option is selected.
- **Suppress Toll Digits**: When this option is selected, toll digits will not appear in the report.
- **Suppress Trunk Number**: Information received through DID or DNIS [DDI in Europe] will not be included in the report if this option is selected.

9. NUMBERING PLAN

9.1 The programming areas included in Numbering Plan include the following:

- Area Flags—*U.S. only* (see below)
- Classes Of Service (see page 6-281)
- Device Baseline Extensions (see page 6-285)
- Emergency (see page 6-286)
- Home Area Codes—*U.S. only* (see page 6-286)
- Route Groups (see page 6-291)
- Toll String Dialing Patterns (see page 6-287)
- User Groups—*U.S. only* (see page 6-289)
- Dial Rules, Facility Groups, and Route Groups (see page 6-291)—For a full description of ARS, see page 5-99 in FEATURES. Note that the ARS software license is required (see page 6-97).

9.2 The other Numbering Plan programming areas are described in the following pages.

A. AREA FLAGS (U.S. ONLY)

9.3 If the system is installed in an area where office and area codes overlap and/or where toll digits are allowed on toll calls or required on long distance calls, double-click on **Area Flags** to view the area and office code flags.

9.4 The following numbering plan area flags can be programmed. The first two flags determine in what way area and office codes overlap. The chart illustrates the difference between the standard North American Numbering Plan (NANP) and each of the overlap flags. In the chart, N=2-9, Z=0 or 1, and X=0-9.

Table 6-26: *Numbering Plan Area Flags*

Flag	Area Codes Can Be:	Office Codes Can Be:
NANP	NZX	NXX
Office Codes as Area Codes	NZX or NXX	NXX
Area Codes as Office Codes	NZX	NXX or NZX

- **Office Codes Used as Area Codes**: An area code in another location uses an NXX pattern that matches an office code within the system site's area code. Because the system cannot differentiate between an office code and an area code when the second digit dialed is 0-9, it will wait for the Interdigit timer to expire or another digit to be dialed before assuming that dialing is completed.
- **Area Codes Used as Office Codes**: One or more office codes within the system site's area code use an NZX pattern that is the same as an area code in another area. Because area codes do not resemble office codes (NXX), end-of-dialing detection is not affected by this flag.
- **Local 7/10 Digit Dialing**: When this flag is enabled, outgoing calls are identified as having reached the end of dialing if the first digits are not a toll field, equal access field, operator access field, or a local area code. This function speeds up placement of local seven-digit calls in an area where some local calls require ten digits.

NOTE: The ATM Switch cannot span area codes unless the area codes are considered local. You can, however, program an ATM node to require ten-digit dialing on all local calls. This makes ten-digit dialing optional for all dual chassis residing in the home area code. Metropolitan areas with overlapping area codes would, however, have to dial ten digits for home area code calls.

- **Toll Digit Allowed On Toll Local Calls**: This option applies only if the area and office codes overlap. Callers in the site's area code usually dial a 1 when placing a call within the local area code(s).
- **Toll Digit Required On Toll Long Distance Calls**: This option applies only if the area and office codes overlap. Callers in the site's area code must dial a 1 when placing a call outside of the local area code(s).

9.5 If any of the numbering plan flags apply to your system location, enable this flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.) Depending on the combination of flags that are selected, some flags cannot be selected.

B. CLASSES OF SERVICE (COS)

9.6 Classes of service are used for toll restriction purposes. COS designations (01-09 or 01-07) have default values and COS designations (10-16 or 08-16) are blank. Classes of service 02-16 have programmable dialing patterns and all have programmable day and night lists of devices. The default values of COS for U.S. and European systems are as follows (the toll string wild cards and non-programmable wildcards are shown on the previous page).

For U.S. Systems:

- **COS 01 – ARS Only**: *(This is a station COS only. It is not used for trunk groups. Also, it cannot be used unless the ARS premium feature is licensed.)* Calls can only be placed using the Automatic Route Selection (ARS) feature when this restriction is assigned. The user will hear reorder tones when attempting to place a call using any other method. A restricted user can still select individual trunks if the trunks are designated as "exempt from ARS Only" (as described on page 5-104), were transferred, were placed on hold, or are recalling or ringing. Trunk restriction determines which trunks in the ARS route group can be selected by the station or Voice Processing application.
- **COS 02 – Deny Area/Office**: This restriction is divided into eight user groups to allow the use of varying area/office code restriction tables. This is useful for reducing restrictions for some of the stations, Voice Processing applications, or trunk groups while increasing restrictions for others. Each station, application, and trunk group is assigned a day mode and a night mode user group in DB Programming. Within each user group, area codes can be designated as restricted, allowed, or extended. Restricting an area code prevents users from placing calls to that area code. Allowing an area code allows all office codes within that area code. Designating an area code as extended allows the programmer to determine which office codes (up to 800) are allowed or restricted within that area code. For each user group, 800 area codes can be marked as allowed or restricted in the database list, and up to six area codes can be marked as extended.
- **COS 03 – Deny Operator**: Calls to numbers that match the dial patterns for this class of service (defaults to [Q]RN+, [Q]R0, and [Q]RE) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.

- **COS 04 – Deny Toll Access**: Calls to numbers that match the dial patterns for this class of service (defaults to [Q]TN+ and [Q]TE) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 05 – Deny International**: Calls to numbers that match the dial patterns for this class of service (defaults to [Q]I+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 06 – Deny Equal Access**: Calls to numbers that match the dial patterns for this class of service (defaults to Q+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 07 – Deny Local Calls**: Calls to numbers that match the dial patterns for this class of service (defaults to N+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 08 – Denied Numbers**: Calls to numbers that match the dial patterns for this class of service (defaults to 1900NXXXXXX+ and 976XXXX+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 09 – Allowed Numbers**: Calls to numbers that match with the dial patterns for this class of service, defaults to 1(800, 888, 877, 866, 855, 844, 833, and 822)NXXXXXX+, are allowed, even if number also matches a dial pattern in a restricted class of service that is assigned to the station, Voice Processing application, or trunk group being used.

For European Systems:

- **COS 01 – ARS Only**: *(This is a station COS only. It is not used for trunk groups. Also, it cannot be used unless the ARS premium feature is licensed.)* Calls can only be placed using the Automatic Route Selection (ARS) feature when this restriction is assigned. The user will hear reorder tones when attempting to place a call using any other method. A restricted user can still select individual trunks if the trunks are designated as "exempt from ARS Only" (as described on page 5-104), were transferred, were placed on hold, or are recalling or ringing. Trunk restriction determines which trunks in the ARS route group can be selected by the station or Voice Processing application.
- **COS 02 – Deny Operator**: Calls to numbers that match the dial patterns for this class of service (defaults to R+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing Unit application, or trunk group being used.
- **COS 03 – Deny Toll Access**: Calls to numbers that match the dial patterns for this class of service (defaults to TN+, TE, 010+, and T1+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing Unit application, or trunk group being used.
- **COS 04 – Deny International**: Calls to numbers that match the dial patterns for this class of service (defaults to I+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.

- **COS 05 – Deny Local Calls**: Calls to numbers that match the dial patterns for this class of service (defaults to N+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 06 – Denied Numbers**: Calls to numbers that match the dial patterns for this class of service (defaults to 0891+ and 0898+) are restricted, unless the number also matches a dial pattern in an "allowed" class of service that is assigned to the station, Voice Processing application, or trunk group being used.
- **COS 07 – Allowed Numbers**: Calls to numbers that match with the dial patterns for this class of service, defaults to 0345+, 0500+, 0645+, and 0800+, are allowed, even if number also matches a dial pattern in a restricted class of service that is assigned to the station, Voice Processing application, or trunk group being used.

9.7 Calls are only restricted if the dialed patterns match the denied pattern exactly and that is the only class of service you have. Allowed patterns, however, always override denied patterns, even if the numbers are similar.

9.8 For example, with a U.S. system, the following table demonstrates what is allowed or denied when the restricted pattern is 976+ and the allowed pattern is 976-1111.

Table 6-27: *COS Examples (U.S.)*

Number Dialed	Allowed or Denied?
97 (partial match)	Allowed
9761234 (exact match with "denied" pattern)	Denied
9761111 (exact match with "allowed" pattern)	Allowed

9.9 With a European system, the following table demonstrates what is allowed or denied when the restricted pattern is 619+ and the allowed pattern is 619000.

Table 6-28: *COS Examples (Europe)*

Number Dialed	Allowed or Denied?
61 (partial match)	Allowed
619123 (exact match with "denied" pattern)	Denied
619000 (exact match with "allowed" pattern)	Allowed

9.10 Program the Classes of service as described in the following paragraphs.

9.11 **Description**: Select the current description and enter the new description in the text box. Then press **ENTER** or select another field to save the change.

9.12 **Day or Night List of devices that use the COS**: Double-click on the COS to be programmed. Then double-click on either **Day List** or **Night List** and do the following:

9.13 **To add a device to the list:**

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To List**. A window appears that asks you to select the type of device to include.
3. Select the device type(s) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate items and then select **Add Items**. When you have added all the devices select **Finish**. Your selections appear in the list.

NOTE: If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

9.14 To delete: To delete one or more item from the list, highlight them and then right-click. Select **Remove Selected Items**.

9.15 Dialing Patterns: *(COS 02-16 only)* To change the dialing patterns for the class of service, double-click on **Dialing Patterns** to view the current list. Then do the following:

9.16 To add a dialing pattern to the class of service:

1. *To add to the bottom of the list:* Do not highlight any existing patterns and right-click anywhere in the right side of the window and select **Add To List** (if there is no blank space, press **CTRL+N**). A blank pattern appears at the bottom of the list.
2. *To add to the list above an existing device:* Highlight the device below the location where you want the new entry and then right-click. Then select **Add To List**. A blank pattern appears above the pattern you highlighted.

9.17 To delete a dial pattern from the class of service: To delete one or more pattern from the list, highlight them and then right-click. Select **Remove Selected Items**.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

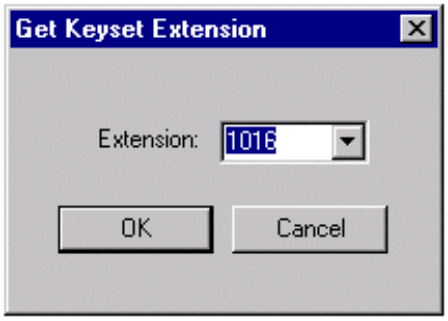
9.18 To move a pattern to another location in the list: Drag and drop the pattern to the new position. Or, highlight the pattern to move and press **CTRL** + the up/down arrow to move the pattern up or down in the list.

9.19 To program the digits for a pattern: Select the current Value for the pattern and enter the new digits in the text box. Then press **ENTER** or select another field to save the change.

9.20 Allow Dialing Patterns: If you want the dialing patterns (programmed above) to be "allowed" patterns, enable this flag. If you want them to be "restricted" patterns, disable the flag. You can enable the flag by selecting the current Value and then selecting the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

C. DEVICE BASELINE EXTENSIONS

9.21 The Device Baseline Extensions feature is located under System\Numbering Plan. The user can define the base extensions of all equippable devices in the system. This allows networked systems to manage extension ranges with less effort from the user. When the system equips one of the devices listed below, it checks the Base Extension Table for the starting extension for that device. For example, the user equips a new phone and DB programming pops up a message box, as shown below.



9.22 In this case, the base extension for a phone is 1000; however, the system already has extensions 1000-1015 allocated so DB programming picks the first extension that is available starting with the base extension.

NOTE: The number of digits in the base extension is maintained. For example, if the base extension were 10000 instead, DB programming will pick the first free extension after 10000. Note that wildcard extensions (e.g., 12XXX) are not valid.

ALSO: Although the "P" prefix is not displayed, any new extension generated for "P" type extensions will be a "P" extension.

9.23 The base extension numbers for all devices are as follow:

Table 6-29: *Base Extension Numbers*

Device	Base Extension
ACD Agent ID	1000
Auto Attendant	2500
Auto Attendant Recall	2500
B-Channel Station	1000
Call Routing Announcement	2500
CO Trunk Group	92001
Group List	1000
Hunt Group	2000
IP Connection	6000
Keyset Base Extension	1000
Message Notification/Retrieval	2500
Node Trunk Group	97501
Non-Associated Mailbox	1000
Off-Node IP Connection	6000
Page Port	91000 [99000 in Europe]

Table 6-29: *Base Extension Numbers*

Device	Base Extension
Page Zone	9600
Record-A-Call	2500
Remote Node	97001
Single-Line	1000
STAR	2500
Trunk	94000
Voice Mail	2500

D. EMERGENCY

9.24 The System\Numbering Plan\Emergency programming area is used for programming the emergency number(s) that will be used when the Emergency Call feature code is entered by a station user.

9.25 Emergency calls, by default, use the first local trunk group and will not be sent through other nodes using node trunk groups. However, when ARS is used to place an emergency call, Route Group 1 is used even if it contains nodes. This means that the network ***can*** access a trunk on a node other than the user's node if the user accesses ARS and dials the emergency number. ***Inter-Tel highly recommends that local trunks be installed and used for emergency number trunk access and that nodes are not used in Route Group 1.*** When a user places an emergency call, every administrator in the network receives an emergency alarm.

9.26 The Emergency Numbers folder stores up to ten emergency numbers that are dialed when the Emergency Call feature is used. For details about the emergency call feature, see page 5-112.

9.27 To program or change emergency numbers, double-click on **Emergency Numbers**. Then select the current value (or blank field) of the number you want to program. Enter the desired number, up to 48 digits, in the text box. Then press **ENTER** or select another field to save the entry. Only the first emergency number has a default value 911 [999].

NOTICE

It shall be the responsibility of the entity or person(s) completing installation and maintenance of hardware or software described herein to research, comply with and be responsible for the specific governmental rules and regulations regarding Emergency Outgoing Access 911 [999] of the geographic location in which such functions are performed.

E. HOME AREA CODES (U.S. ONLY)

9.28 The database requires a "home" area code for identifying local calls. It also allows up to 15 additional home area codes. The system refers to these area codes for toll restriction and call cost. The first area code programmed behaves differently from the next 15 home area codes. For the first area code programmed, the area code digits are stripped from the dialed outgoing number. To program the home area code(s) for the system site, select the current Digits and enter the new area code in the text box. Then press **ENTER** or click outside the field to save the change.

F. TOLL STRINGS

9.29 Toll strings are dialing patterns which are abbreviated to single character "wildcards." These wildcards are used in Class of Service and Automatic Route Selection programming. Toll strings can contain any digit 0-9 including the keypad special characters # and *. In addition, a variety of special characters may be entered into the database to reflect particular digit strings, hookflashes/recalls, or special digit strings. Each of the following toll string wildcards can be reprogrammed and/or renamed to meet the customer's requirements. ***However, note that changing these dialing patterns will affect all of the other parts of the system where they are used.*** To change the description of a pattern, select the current description and then enter the new description in the text box. Then press **ENTER** or select another field to save the change. To change the patterns within the toll string, double-click on the toll string to view the current pattern(s). Then do the following:

9.30 **To add a dial pattern to the toll string:** Note that this is an *ordered list*. Place the devices in the list in the order you want them to be accessed.

- **To add to the bottom of the list:** Do not highlight any existing patterns and right-click anywhere in the right side of the window and select **Add To List** (if there is no blank space, press **CTRL+N**). A blank pattern appears at the bottom of the list.
- **To add to the list above an existing device:** Highlight the device below the location where you want the new entry and then right-click. Then select **Add To List**. A blank pattern appears above the pattern you highlighted.

9.31 **To delete a dial pattern from the toll string:** To delete one or more patterns from the list, highlight them and then right-click. Select **Remove Selected Items**. To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

9.32 **To move a pattern to another location in the list:** Drag and drop the pattern to the new position. Or, highlight the pattern to move and press **CTRL** + the up/down arrow to move the pattern up or down in the list.

9.33 **To program the digits for a pattern:** Select the current Value for the pattern and enter the new digits in the text box. Then press **ENTER** or select another field to save the change.

9.34 The first table on the next page shows the programmable toll string wildcards. The second table on the next page shows the special characters that may be used when specifying dialing patterns; these characters are not programmable.

Toll String Wild Cards for the U.S.

Table 6-30: *Toll String Wild Cards (U.S.)*

Character	Meaning
S	**Any Toll String**: Any of the designated patterns can be dialed at that point in the telephone number. (In the default state, the dialing patterns for this toll string are “I,” “R,” and “T.”)
R	**Operator Access**: This represents the digit string that is required to call the telephone company's operator. (In the default state, this dialing pattern is 0)
T	**Toll Access**: Represents the digit(s) required for using long distance service. (In the default state, this dialing string is 1.)
I	**International**: Represents the digits required for international dialing. (In the default state, the dialing patterns for this toll string are 011 and 01.)
Q	**Equal Access**: Represents the digits required to select secondary carriers for equal access dialing. (In the default state, the dialing patterns for this toll string are 101XXXX, 10NXX, and 100XX.)

Toll String Wild Cards for Europe

Table 6-31: *Toll String Wild Cards (Europe)*

Character	Meaning
S	**Any Toll String:** Any of the designated patterns can be dialed at that point in the telephone number. (In the default state, the dialing patterns for this toll string are “I,” “R,” and “T.”)
R	**Operator Access:** This represents the digit string that is required to call the network provider's operator. (In the default state, this dialing pattern is 1XX.)
T	**Toll Access:** Represents the digit(s) required for using long distance service. (In the default state, this dialing strings are 0 and 01.)
I	**International:** Represents the digits required for international dialing. (In the default state, the dialing pattern for this toll string is 00.)

Special Characters For Dialing Patterns

Table 6-32: *Special Characters for Dialing Patterns*

Character	Meaning
X	Any digit 0-9
A	Any keypad entry (0-9, #, *)
N	Any digit 2-9
Z	Any digit 0 or 1
B	# or *
H	A hookflash (recall)
E	End of dialing; the pattern will not match if any other digits are dialed beyond this point
+	Any additional dialing will be accepted, from that point in the string, with no further checking for a match. This also means that no further dialing is required beyond this point.

Table 6-32: *Special Characters for Dialing Patterns (Continued)*

Character	Meaning
[*x*]	Indicates an optional pattern within another pattern. For example, with a U.S. system, the International Access character (I) could be defined as 01[1]. (The 01 is followed by an *optional* 1.)
(*x*-*x*)	Indicates a range of digit strings within a pattern. The strings on either side of the hyphen and all strings that fall within the numerical range are included in the match. The strings on either side of the hyphen must be the same length, and the only digits that may appear in the range are 0-9 (#, *, pauses, and flashes are not allowed).
<*x*>	Indicates repeatable patterns within patterns. In other words, no matter how many times the digit string within the brackets is dialed, the system will consider the dialed digits to match the pattern. Note that a repeatable pattern is an entire pattern; no other characters are allowed before or after a repeatable pattern. In other words, a repeatable pattern cannot be included within any other pattern.

G. USER GROUPS (U.S. ONLY)

9.35 This section is used for programming the various aspects of area and office code restriction used for the Deny Area/Office class of service.

9.36 You can set up area and office code tables of up to eight user groups to allow different area/office code restriction to be used. This is useful for reducing restrictions for some users while increasing restrictions for others. Each station, application, and trunk group is assigned to a user group.

9.37 Within each user group, area codes can be restricted, allowed, or extended:

- Restricting an area code prevents users from placing calls to that area code and all of its office codes.
- Allowing an area code allows all office codes within that area code.
- Designating an area code as extended allows you to determine which office codes within that area code are allowed or restricted. There can be up to six extended area codes per user group.

9.38 To prepare for programming, list the area codes that are allowed, restricted, and extended. Also, list the office codes within the extended area code(s) that are allowed and restricted.

9.39 Select **User Groups** to view a list of the eight user groups. The list shows the user group numbers and their descriptions (if programmed). At this level, you can program the description. To program the area codes, office codes, day list, and night list for a user group double-click on the desired user group. Then follow the programming instructions below.

9.40 **Description**: Select the current description and enter a name for the user group, up to 20 characters, in the text box. Then press **ENTER** or select another field to save the change.

Area Code Programming (U.S. Only)

9.41 Area code programming uses a drag and drop method to move the area codes from one location to another. This ensures that each area code is include in a list and none are skipped.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

9.42 **Allowed Area Codes**: To allow users in the user group access to area codes, the codes must be placed in the Allowed list. To place an area code in the Allowed list, double-click on **Extended** or **Restricted Area Codes** and locate the area code you want to restrict. Then drag and drop that area code to the Allowed Area Codes directory in the left portion of the window. The next time you open Allowed Area Codes, you will see the code(s) you dropped into the directory. To remove an area code from the Allowed list, move it to the Restricted or Extended list in this User Group.

9.43 **Extended Area Codes**: There can be up to six extended area codes per user group, each with an individual list of allowed and restricted office codes. To create an extended area code, double-click on **Allowed** or **Restricted Area Codes** and locate the area code you want to extend. Then drag and drop that area code to the Extended Area Codes directory in the left portion of the window. The next time you open Extended Area Codes, you will see the code(s) you dropped into the directory. Office codes can then be programmed as allowed or restricted for that area code by dragging and dropping them into the appropriate categories. To remove an area code from the Extended list, move it to the Restricted or Allowed list in this User Group.

9.44 **Restricted Area Codes**: To prevent users from placing calls to certain area codes, place them in the Restricted list. Double-click on **Allowed** or **Extended Area Codes** and locate the area code you want to restrict. Then drag and drop that area code to the Restricted Area Codes directory in the left portion of the window. The next time you open Restricted Area codes, you will see the code(s) you dropped into the directory. To remove an area code from the Restricted list, move it to the Allowed or Extended list in this User Group.

9.45 **Day/Night List**: ***Note that only extensions with COS 02 can be placed in a user group day or night list.*** To program the list of stations, applications, and trunk groups that are included in a specific user group in day or night mode, select Day List or Night List. Then do the following:

9.46 **To add a device to the list**:

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To List**. A window appears that asks you to select the type of device to include.
3. Select the device type(s) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

   NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate items and then select **Add Items**. When you have added all the devices select **Finish**. Your selections appear in the list.

   NOTE: If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

9.47 **To delete:** To delete one or more item from the list, highlight them and then right-click. Select **Remove Selected Items**.

H. ARS DIAL RULES, ROUTE GROUPS, AND FACILITY GROUPS

Overview of ARS

9.48 Each node has its own ARS programming. ARS is programmed using route groups and facility groups:

9.49 **Route Groups**: A route group contains dialing patterns and facility groups.

- The dialing patterns are used to determine the calls that will be routed through the route group. For example, the default dial pattern for Route Group 1 is N+, any number of digits beginning with digit 2-9 for the U.S. [digit 2-9, 345+, 0500+, 0645+, and 0800+ for Europe]. If a number is dialed that begins with 1, it will not be routed through this route group.
- Each route group has an ordered list of facility groups that contain lists of local trunk groups and/or nodes. There can be 100 facility groups in the system. The facility groups should be programmed so that the least-expensive route is checked and, if available, is selected first. If the least-expensive facility group is not available the system checks the other groups in the list until it finds an available trunk.

9.50 **Facility Groups**: A facility group contains trunk group/node lists and dial rules.

- The list can include local trunk groups or nodes. (They cannot contain node trunk groups.)
- The dial rules tell the system what to dial. (The system can have up to 32 dial rules, 26 of which are programmable. Each facility group can use up to 32 dial rules.) For example, if the selected route group requires that the number contain "1" but no area code [national dialing in Europe], the dial rules include the 1 and drop the area code (national dialing). The modified telephone number can contain up to 32 digits. (If SMDR is enabled, the modified telephone number, not the digits dialed, will appear in the SMDR call record.) When programming ARS, you can use preset dial rules or create new dial rules that add up to 16 digits each. The pre-programmed dial rules are listed on page 6-296.

9.51 Because users do not have direct access to trunks on other node, ARS is the only way users can place calls using the other nodes' trunks.

9.52 ARS calls are limited to one "hop" to another node. If the system routes an outgoing call to another node, the other node cannot route the outgoing call to any other node. This eliminates the system from getting into an infinite loop searching for a node to route the outgoing call.

Planning ARS Programming

9.53 Programming ARS follows this general outline. (Detailed programming instructions are given on the following several pages.)

1. **Plan**: Determine how ARS is going to be used by the customer:

   a. What types of calls will the users be making (local, long distance, international, etc.)? Where will the call destinations be? There will be a route group and a facility group for each type of call.

   b. What dialing patterns identify each type of call? For example, local calls begin with any digit 2-9 (but not a toll digit) and operator assisted calls begin with 0 [1XX in Europe]. These will be the dialing patterns that tell the system which route group to select.

   c. *With a U.S. system*, what are the best routes for each type of call? If you have nodes in Phoenix and Los Angeles, would it be better to route calls from Phoenix to Southern California through a node or over your long distance service? This will determine the trunk groups and nodes included in each facility group.

      With a European system, what are the best routes for each type of call? If you have nodes in London and Kettering, would it be better to route calls from London to Glasgow through a node or over your long distance service? This will determine the trunk groups and nodes included in each facility group.

   d. Which trunks (on this node or other nodes) would be the best route for each type of call? This determines the order in which you want the facility groups accessed.

   e. What special characters, if any, need to be added or removed from the dialed number when a call is placed? Determine this for each trunk you will be using.

      For example:

      With a U.S. system, if a caller in Phoenix dials 1-714-XXX-XXXX, the system will need to remove the toll digit and possibly the area code if the call is sent out over local trunks on the Los Angeles node. However, if the call is routed through the long distance provider in Phoenix, you will probably need to include (echo) the toll and area code digits or even add other digits as required by the long distance service. These will be the facility groups' dial rules.

      With a European system, if a caller in London dials 020-8335XXX, the system will need to remove the toll digit and possibly the national dialing code if the call is sent out over local trunks on the Kettering node. However, if the call is routed through the long distance provider in London, you will probably need to include (echo) the toll and national dialing code digits or even add other digits as required by the long distance service. These will be the facility groups' dial rules.

2. **Program the dial rules**: Program dial rules you determined were needed in step 1e above.

3. **Program the facility groups**: Do the following:

   a. Determine what types of calls will be processed by each facility group and program names for those groups.

   b. For each facility group, determine which trunk groups and nodes will be used and program the list. Make sure that the trunk groups and nodes contain the proper type of trunks (local, FX, WATS, long distance, etc.) for the calls that will be routed. (Use the information from steps 1c and 1d) Also ensure the stations that will be using ARS have outgoing access for the trunk groups and nodes (see page 6-228 for trunk group programming and page 6-243 for node trunk group programming).

   c. For each facility group, program the list of dial rules that will be used so that the outgoing numbers include all of the necessary digits. See the information from step 1e to determine which dial rules each facility group uses.

4. **Program the route groups**:

   a. Determine what types of calls will be processed by each route group and program names for those groups. (See step 1a.)

   b. The dialing patterns tell the system which route group to select. For each route group, program its dialing patterns. (See step 1b) When checking the route group, the system will look at the first dial pattern. If it does not match, the system will continue checking the other patterns in the route group's list. If a match is found, the route group is selected. If not, the next route group is checked.

   c. For each route group, select the facility group(s) to be used to route the calls. See the information in step 1c and the facility group programming done in step 3.)

Route Group Default Values

9.54 The default values of the pre-programmed route groups are as follows. All route groups are programmable.

For U.S. Systems:

- **Local (P1000)**: This route group is used for all calls that do not begin with a toll digit (1) or operator digit (0) and for calls to 911 (its default dial pattern is N+). It uses the Local Facility Group.

  NOTE: When ARS is used to place an emergency call (by entering the Emergency Call feature code or dialing 911), Route Group 1 is used even if it contains nodes. This means that the network ***can*** access a trunk on a node other than the user's node if the user accesses ARS and dials the emergency number. ***Inter-Tel highly recommends that local trunks be installed and used for emergency number trunk access and that nodes are not used in Route Group 1.***

- **Toll Local (P1003)**: This route group is used for calls with a toll digit (1) and a seven-digit number, and for calls to 1N11 (N=Any digit 2-9). Its default dial patterns are TN11 and TXXXXXXX. It uses the Local Facility Group.

- **Toll Long Distance (P1011)**: This route group processes calls with a toll digit (1) and a 10-digit number. The numbers may also include equal access digits. The default dial pattern is [Q]TNXXXXXX+. This Route Group uses the Toll Long Distance Facility Group.

- **Operator (P1013)**: This route group processes calls that begin with an operator digit (0) but ***do not*** begin with an international access code (01 or 011). The numbers may also include equal access digits. Its default dial patterns are [Q]RN+, [Q]RR, and [Q]R. This Route Group uses the Operator Facility Group.
- **International Station-to-Station (P1014)**: This route group is for calls that begin with an international station-to-station access code (011) but ***do not*** begin with the international operator access code (01). The numbers may also include equal access digits. The default dial pattern is [Q]011+. This Route Group uses the International Station Facility Group.
- **International Operator (P1015)**: This route group is for calls that begin with an international operator access code (01). The numbers may also include equal access digits. The default dial pattern is [Q]01+. This Route Group uses the International Operator Facility Group.

For European Systems:

- **Local (P1000)**: Used for all calls that do not begin with a toll digit (01 or 0) or operator digit (1XX) and for calls to emergency services.

NOTE: When ARS is used to place an emergency call (by entering the Emergency Call feature code or dialing 999 or any of the programmed emergency numbers), Route Group 1 is used even if it contains nodes. This means that the network ***can*** access a trunk on a node other than the user's node if the user accesses ARS and dials an emergency number. ***Inter-Tel highly recommends that local trunks be installed and used for emergency number trunk access and that nodes are not used in Route Group 1.***

- **Toll (National) (P1011)**: Processes calls with a toll digit (0 or 01). This Route Group uses the National Facility Group.
- **Operator (P1013)**: Processes calls that begin with an operator digit (1XX) but ***do not*** begin with an international access code (00).

9.55 **International (P1014)**: Used for calls that begin with an international access code (00). Dial patterns are assigned to the route group in the order they will be used. When checking the route group, the system will look at the first dial pattern. If it does not match, the system will continue checking the other patterns in the route group's list. If a match is found, the route group is selected. If not, the next route group is checked.

9.56 Only the default route groups have default dial patterns. Their defaults are as follows. A complete list of special characters and toll strings is shown on page 6-287.

For U.S. Systems:

Table 6-33: *Default Dial Patterns (U.S.)*

Route Group #	Dial Pattern(s)
Local (1001)	N+
Toll Local (1004)	TN11, TXXXXXXX
Toll Long Distance (1011)	[Q]TNXXXXXX+
Operator (1013)	[Q]RN+, [Q]RR, [Q]R
International Station-to-Station (1014)	[Q]011+
International Operator (1015)	[Q]01+

For European Systems:

Table 6-34: *Default Dial Patterns (Europe)*

Route Group #	Dial Pattern(s)
Local (1001)	N+
Toll Local (1004)	TN11, TXXXXXXX
Toll Long Distance (1011)	[Q]TNXXXXXX+
Operator (1013)	[Q]RN+, [Q]RR, [Q]R
International Station-to-Station (1014)	[Q]011+
International Operator (1015)	[Q]01+

Facility Group Default Values

9.57 Facility groups determine the trunk groups that will be used to route calls and the dial rules that are needed with its trunks. The default values of the pre-programmed facility groups are as follows. All facility groups are programmable.

For U.S. Systems:

- **Local (P1500)**: This group uses Trunk Group 1 and Dial Rules 3 and 4.
- **Toll Local (P1501)**: This facility group uses Trunk Group 1 and Dial Rules 2 and 4.
- **Toll Long Distance (P1502)**: This facility group uses Trunk Group 1 and Dial Rules 1, 2, 3, and 4.
- **Operator (P1503)**: This facility group uses Trunk Group 1 and Dial Rules 1, 2, 3, and 4.
- **International Station-to-Station (P1504)**: This facility group uses Trunk Group 1 and Dial Rules 1, 2, 3, 4, and 7.
- **International Operator (P1505)**: This facility group uses Trunk Group 1 and Dial Rules 1, 2, 3, 4, and 7.

For European Systems:

- **Local (P1500)**: This group uses Trunk Group 1 and Dial Rules 1 and 2.
- **National (P1510)**: This facility group uses Trunk Group 1 and Dial Rules 1 and 2.
- **Operator (P1512)**: This facility group uses Trunk Group 1 and Dial Rules 1 and 2.
- **International (P1513)**: This facility group uses Trunk Group 1 and Dial Rules 1 and 2.

9.58 Each facility group has a list of trunk groups and/or node trunk groups that will be used for routing calls. For example, the “Local” facility group would contain trunk groups that include local trunks, but the “Los Angeles” facility group might have node trunks groups that include local trunks on a node in Los Angeles.

9.59 Facility groups must be placed in the Ordered List Box in the order that they should be selected. Make sure that the facility groups being used contain the proper type of trunks (local, FX, WATS, long distance, etc.) for the calls that will be routed.

Dial Rule Default Values

9.60 Each facility group has a programmed set of dial rules that tell the system what to dial. (The system can have up to 32 dial rules, 26 of which are programmable. Each facility group can use up to 32 dial rules.) For example, if the selected facility group requires that the number contain "1" but no area code [national dialing in Europe], the dial rules include (echo) the toll field, but would not echo the area code. The modified telephone number can contain up to 32 digits. When programming ARS, you can use preset dial rules or create new dial rules. Pre-programmed dial rules include the following:

For U.S. Systems:

- **Dial Rule 1 – ECHO Equal Access**: This dial rule includes the equal access digits (wildcard Q, which defaults to 10XXX and 101XXX) in the number, if dialed.
- **Dial Rule 2 – ECHO Toll Field**: The toll field is included in the number, if dialed.
- **Dial Rule 3 – ECHO 3 Digits After Toll Field**: With this dial rule, the three digits after the toll field are included in the number. (These digits are usually the area code.)
- **Dial Rule 4 – ECHO Local Address**: This dial rule allows ARS to dial the rest of the digits that were dialed by the station user.
- **Dial Rule 5 – ECHO Account Code**: This dial rule causes the system to dial the account code that is associated with the call, if available before end of dialing. The account code can be entered using any of the account code types, including All Calls Following, as long as the system receives the account code before end of dialing.
- **Dial Rule 6 – ECHO Extension Number**: This dial rule tells the system to include the extension number of the station being used to place the call.
- **Dial Rule 7 – ADD #**: This dial rule adds a pound [hash] (#) to the dialed number. This pre-programmed dial rule is programmable.

9.61 Dial rules can contain any digit (0-9, *, #) and/or hookflashes and pauses.

For European Systems:

- **Dial Rule 1 – ECHO Toll Field**: Includes the toll field in the number, if dialed.
- **Dial Rule 2 – ECHO Local Address**: Allows ARS to dial the rest of the digits that were dialed by the station user.
- **Dial Rule 3 – ECHO Extension Number**: Tells the system to include the last three digits of the extension number of the station being used to place the call.
- **Dial Rule 4 – ECHO Account Code**: Causes the system to dial the last three digits of account code that is associated with the call, if available, before end of dialing. The account code can be entered using any of the account code types, including All Calls Following, as long as the system receives the account code before end of dialing.
- **Dial Rule 5 – ECHO Serial Number**: Tells the system to dial the system-assigned serial number of an ARS call. The serial number range is 000-998, excluding 112. The serial number can be reset in dial rule programming.
- **Dial Rule 6 – ADD #**: Adds a hash (#) to the dialed number. This pre-programmed dial rule is programmable.

NOTE: Dial rules 3 (echo extension), 4 (echo account code), and 5 (echo serial number) will always come after dial rules 1 and 2. If 3, 4, or 5 precedes dial rules 1 or 2, the SMDR Call Type and Call Cost will be affected.

9.62 Dial rules can contain any digit (0-9, *, #) and/or hookflashes/recalls and pauses.

Programming Dial Rules

9.63 To program ARS dial rules, select Dial Rules from the Numbering Plan programming area. When you double-click on **Dial Rules**, the list of rules 1-32 appears. Rules 7-32 are programmable to allow you to add digits to a dialed number. To enter the digits for a dial rule, select its current Value and enter the new value in the text box. Then press **ENTER** or select another field to save the change.

NOTE: You cannot edit the dial rule description (Dial Rule field).

9.64 Dial rules can contain any digit (0-9, *, #) and/or hookflashes/recalls and pauses.

Programming Facility Groups

9.65 To program ARS facility groups, select Facility Groups from the Numbering Plan programming area. When you double-click on **Facility Groups**, the current list of facility groups displays. At this level you can change the description of the facility group by selecting its current description and then entering the new information in the text box.

9.66 Each facility group has a list of trunk groups and/or node trunk groups that will be used for routing calls. For example, the "Local" facility group would contain trunk groups that include local trunks, but the "Branch" facility group might have node trunks groups that include local trunks on a node in another city.

9.67 All facility groups are programmable. The default values of the pre-programmed facility groups are as follows.

- **Local**: This group uses Trunk Group 1 and Dial Rules 3 and 4.
- **Toll Local**: This facility group uses Trunk Group 1 and Dial Rules 2 and 4.
- **Toll Long Distance**: This facility group uses Trunk Group 1 and Dial Rules 1, 2, 3, and 4.
- **Operator**: This facility group uses Trunk Group 1 and Dial Rules 1, 2, 3, and 4.
- **International Station**: This facility group uses Trunk Group 1 and Dial Rules 1, 2, 3, 4, and 7.
- **International Operator**: This facility group uses Trunk Group 1 and Dial Rules 1, 2, 3, 4, and 7.

9.68 Then press **ENTER** or select another field to save the change.

9.69 **To create a facility group:** Right-click in any open area in the right side of the screen. Select **Create Facility Group**. The new facility group appears in the list and can be programmed as described below.

9.70 **To delete a facility group:** Highlight it and right-click. Then select **Delete**.

9.71 If you double-click on a facility group, you will see the Dial Rules and Trunk Groups/Nodes options.

9.72 **Dial Rules**: To assign the dial rules that will be used by this facility group, double-click on Dial Rules. The current list of dial rules, if any, appears. You can then do the following:

9.73 **To add dial rules to the facility group:** Note that this is an *ordered list*. Place the dial rules in the list in the order you want them to be used.

9.74 **To add to the bottom of the list:** Do not highlight any existing dial rules and right-click anywhere in the right side of the window. An option box appears.

9.75 To add to the list above an existing dial rule: Highlight the dial rule below the location where you want the new entry and then right-click. An option box appears.

1. Select **Add To Dial Rules List**. A window appears that asks you to select the type of device to include.
2. Select **Dial Rules** and then select **Next**. The list of dial rules appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

> **NOTE:** To highlight a series of items, hold down **SHIFT** while selecting the first and last item you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

3. Highlight the dial rule(s) you want to add and then select **Add Items**. (Note that if you select more than one, they will be placed in the list in numerical order, not in the order you select them. To keep the list in the proper order, you may have to add dial rules one at a time.) When you have added all the dial rules you need, select **Finish**. Your selections appear in the list.
4. If you want to see the programming for a specific dial rule, double-click on its number to jump to its programming area.

9.76 To move a dial rule to another location in the list: Drag and drop the dial rule to the new position. Or, highlight the dial rule to move and press **CTRL** + the up/down arrow to move the dial rule up or down in the list.

9.77 To delete a dial rule from the list: To delete one or more dial rule from the list, highlight them and then right-click. Select **Remove Selected Items**.

9.78 Trunk Groups/Nodes: Trunk groups and nodes must be placed in the list in the order that they should be selected. Make sure that the trunk groups or nodes being used in the facility group contain the proper type of trunks (local, FX, WATS, long distance, etc.) for the calls that will be routed. Trunk group toll restrictions will not apply when ARS is used; only station toll restrictions are checked. To determine the trunk groups and/or nodes that will be used by this facility group, double-click on **Trunk Groups/Nodes**. The current list of trunks/nodes, if any, appears. You can then do the following:

9.79 To add trunk groups or nodes to the facility group: Note that this is an *ordered list*. Place the trunks and nodes in the list in the order you want them to be used.

1. **To add to the bottom of the list:** Do not highlight any existing entry and right-click anywhere in the right side of the window. An option box appears.
2. **To add to the list above an existing device:** Highlight the entry below the location where you want the new entry and then right-click. An option box appears.
3. Select **Add To Trunk Groups/Nodes List**. A window appears that asks you to select the type of device to include.
4. Select **CO Trunk Group** and/or **Remote Node** and then select **Next**. The list of available devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

> **NOTE:** To highlight a series of items, hold down **SHIFT** while selecting the first and last item you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

5. Highlight the trunk group(s) and/or node(s) you want to add and then select **Add Items**. (Note that if you select more than one, they will be placed in the list in numerical order, not in the order you select them. To keep the list in the proper order, you may have to add one at a time.) When you have added all the trunk groups and nodes you need, select **Finish**. Your selections appear in the list.

> **NOTE:** If you want to see the programming for a specific trunk group or node, double-click on its number to jump to its programming area.

9.80 **To move a trunk group or node to another location in the list:** Drag and drop the trunk group or node to the new position. Or, highlight the trunk group to move and press **CTRL** + the up/down arrow to move the trunk group up or down in the list.

9.81 **To delete a trunk group or node from the list:** To delete entries from the list, highlight them and then right-click. Select **Remove Selected Items**.

9.82 Also ensure the stations that will be using ARS have outgoing access for the trunk groups and nodes (see page 6-228 for trunk group programming and page 6-243 for node trunk group programming).

Programming Route Groups

9.83 To program ARS route groups, select Route Groups from the Numbering Plan programming area. When you double-click on **Route Groups**, the current list of route groups displays. At this level you can change the description of the route group by selecting its current description and then entering the new information in the text box. Then press **ENTER** or select another field to save the change.

9.84 **To create a route group:** *(This can be done only if there are fewer than 32 route groups)* Right-click in any open area in the right side of the screen. Select **Create Route Group**. The new route group appears at the end of the list and can be programmed as described below. Then enter the description as described above.

9.85 **To move a route group to another location in the list:** The route groups are in an ordered list and the Order column shows you the number associated with each route group. You can drag and drop a route group to a new position, if necessary. Or, you can highlight the route group to move and press **CTRL** + the up/down arrow to move the route group up or down in the list.

9.86 **To delete a route group:** Highlight it and right-click. Then select **Delete**.

9.87 If you double-click on a route group, you will see the following programming options:

- Dial Patterns
- Facility Groups
- Audio for Calls Camped onto this Device

9.88 **Dial Patterns**: The dialing patterns tell the system which route group to select. When checking the route group, the system will look at the first dial pattern. If it does not match, the system will continue checking the other patterns in the route group's list. If a match is found, the route group is selected. If not, the next route group is checked. To assign the dial patterns that will be used by this route group, double-click on **Dial Patterns**. The current list of dial patterns, if any, appears. You can then do the following:

9.89 To add dial patterns to the route group: Right-click anywhere in the right side of the window (if there is no blank space, press **CTRL+N**). An option box appears.

1. Select **Add To Dial Patterns List**. A blank dial pattern is added to the bottom of the list.
2. Select the new dial pattern and enter its digits in the text box. Then press **ENTER** or select another field to save the change.

9.90 To move a dial pattern to another location in the list: Drag and drop the pattern to the new position. Or, highlight the dial pattern to move and press **CTRL** + the up/down arrow to move the dial pattern up or down in the list.

9.91 To delete a dial pattern from the list: To delete one or more dial pattern from the list, highlight them and then right-click. Select **Remove Selected Items**.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

9.92 Facility Groups: For each route group, select the facility group(s) to be used to route the calls. Facility groups must be placed in the list in the order that you want them to be selected. To determine the facility groups that will be used by this route group, double-click on **Facility Groups**. The current list of facility groups, if any, appears. You can then do the following:

9.93 To add facility groups to the route group: Note that this is an *ordered list*. Place the facility groups in the list in the order you want them to be used.

1. **To add to the bottom of the list:** Do not highlight any existing entry and right-click anywhere in the right side of the window. An option box appears.

   To add to the list above an existing device: Highlight the entry below the location where you want the new entry and then right-click. An option box appears.
2. Select **Add To Facility Groups List**. A window appears that asks you to select the type of device to include.
3. Select **Facility Group** and then select **Next**. The list of available facility groups appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

   NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.
4. Highlight the facility group(s) you want to add and then select **Add Items**. (Note that if you select more than one, they will be placed in the list in numerical order, not in the order you select them. To keep the list in the proper order, you may have to add one at a time.) When you have added all the facility groups you need, select **Finish**. Your selections appear in the list.

   NOTE: If you want to see the programming for a specific facility group, double-click on its number to jump to its programming area.

9.94 To move a facility group to another location in the list: Drag and drop the facility group to the new position. Or, highlight the facility group to move and press **CTRL** + the up/down arrow to move the facility group up or down in the list.

9.95 **To delete a facility group from the list:** To delete entries from the list, highlight them and then right-click. Select **Remove Selected Items**.

9.96 **Audio for Calls Camped onto this Device**: Defines the audio that a caller hears when camped-on to the route group. To program this field, use the drop-down list to select the type of audio and press **ENTER**. Your options are Silence, Ticktone, Ringback, or Board. If you select **Board**, specify the music source in the Extended Value field. The default is *Board*, and the default extended value is *01:08 Central Processor*.

10. STATION-RELATED INFORMATION

10.1 Station-related programming includes the following areas:

- Account Codes (see below)
- Attendants (see page 6-303)
- Audio Diagnostics (see page 6-304)
- Flags (see page 6-305)
- Key Assignments (see page 6-306)
- Message Centers (see page 6-314)
- Messages (see page 6-314)
- Primary Attendants (see page 6-315)
- System Forwarding Paths (see page 6-316)
- System Speed Dial (see page 6-317)
- Keyset DB Programming Password (see page 6-318)

10.2 Each of these programming areas is described in detail in the following pages.

A. ACCOUNT CODES

10.3 There can be up to 500 account codes (5000 if using a CP Server). The default database does not contain any account codes. Forced and Standard account codes may be individually or batch created in DB Programming. Unneeded codes may be removed by right-clicking and choosing Delete.

10.4 To program account codes, select Account Codes from the Station-Related Information programming area. When you double-click on **Account Codes**, the following options appear:

- Forced
- None
- Standard
- Maximum Account Code Length

10.5 There can be up to 256 of each type of account code (512 in a quad-chassis system).

10.6 Maximum account code length can be up to 12 digits. Account codes can be of varying lengths as long as they do not exceed the maximum. For example, if maximum length is four digits, 1, 11, 111, and 1111 can be entered as separate account codes.

10.7 In a network, the system validates account codes against the account code table on the user's node. The account code follows the call as it moves from node to node and appears on every SMDR record associated with the call.

10.8 Each station can be assigned a standard account code or one of six [four in Europe] types of forced account codes, four [two in Europe] are ARS dependent. Or, if desired, the station can have no associated account code. When a standard account code is assigned, a specific code must be designated (001-256 in a single- or dual-chassis system, 001-512 in a triple- or quad-chassis system). The account code types are:

For U.S. Systems:

- **Standard**: Automatically appear in the SMDR printout to identify calls from this station. The station user is not required to enter any digits during the call.
- **Forced – All Calls Validated**: Requires the user to enter a forced account code before an outside call can be placed. If the code matches one of the pre-programmed forced account codes in the database, the call is allowed. If the code does not match, the call is blocked.
- **Forced – All Calls Non-Validated**: Requires the user to enter an account code before placing any outside call. The code is not checked against any lists and the call is allowed as soon as the code is entered.
- **Forced – Local Toll Calls Validated**: Requires the user to enter an account code after dialing a local toll call number when using ARS. If the code matches one of the pre-programmed forced account codes in the database, the call is allowed. If the code does not match, the call is blocked.
- **Forced – Local Toll Calls Non-Validated**: Requires the user to enter an account code after dialing a local toll call or long distance number when ARS is used. The code is not checked against any lists and the call is allowed as soon as the code is entered.
- **Forced – Long-Distance Toll Calls Validated**: Requires the user to enter an account code if the system detects that a long distance call has been dialed when ARS is used. If the code matches one of the pre-programmed forced account codes in the database, the call is allowed. If the code does not match, the call is blocked.
- **Forced – Long-Distance Toll Calls Non-Validated**: Requires the user to enter an account code if the system detects that a long distance call has been dialed when ARS is used. The code is not checked against any lists and the call is allowed as soon as the code is entered.
- **None**: Does not require the user to enter an account code. The user may still use optional account codes or enable the Account Code For All Calls Following feature, as desired.

For European Systems:

- **Standard**: Automatically appears in the SMDR printout to identify calls from this station. The station user is not required to enter any digits during the call.
- **Forced – All Calls Validated**: Requires the user to enter a forced account code before an outside call can be placed. If the code matches one of the pre-programmed forced account codes in the database, the call is allowed. If the code does not match, the call is blocked.
- **Forced – All Calls Non-Validated**: Requires the user to enter an account code before placing any outside call. The code is not checked against any lists, and the call is allowed as soon as the code is entered.
- **Forced – Toll (National) Calls Validated**: Requires the user to enter an account code after dialing a national call number when using ARS. If the code matches one of the pre-programmed forced account codes in the database, the call is allowed. If the code does not match, the call is blocked.

- **Forced – Toll (National) Calls Non-Validated**: Requires the user to enter an account code after dialing a national call when ARS is used. The code is not checked against any lists, and the call is allowed as soon as the code is entered.
- **None**: Does not require the user to enter an account code. The user may still use optional account codes or enable the Account Code For All Calls Following feature, as desired.

10.9 Forced Account Codes: When you double-click on **Forced** account codes you have the following programming options:

- **Non-Validated and Validated**: When you select either of these options you see a list of account code types. When you select one of the account code types, you will see a list of stations that have been assigned to that account code list.
- **Account Codes 001-512**: To enter a new account code, select the current Value for the code and enter the new digits in the text box. The press **ENTER** or select another field to save the change. If you attempt to add an account code that has already been assigned, an error message tells you that every account code must be a unique number. You must enter a different number.

10.10 To add a device to an account code list:

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To List**. A window appears that asks you to select the type of device to include.
3. Select the device type(s) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate items and then select **Add Items**. When you have added all the devices select **Finish**. Your selections appear in the list.

NOTE: If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

10.11 To delete a device from a list: To delete one or more item from the list, highlight them and then right-click. Select **Remove Selected Items**.

B. ATTENDANTS

10.12 A station can be designated as an attendant and assigned a list of stations that it will serve. When you select Attendants from the Station-Related Programming area, you can view a list of the existing attendants.

10.13 If you double-click on a specific attendant, you can view the list of stations it serves.

10.14 To create an attendant, enable a station's Attendant flag (as described on page 6-209).

10.15 To assign the stations served by the attendant:

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Add To List**. A window appears that asks you to select the type of device to include.

3. Select the device type(s) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate items and then select **Add Items**. When you have added all the devices select **Finish**. Your selections appear in the list.

NOTE: If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

10.16 To delete a device from a list: To delete one or more item from the list, highlight them and then right-click. Select **Remove Selected Items**.

10.17 You can also assign attendants in station programming by selecting the station's Associated Extensions and Attendant options and selecting the station that will serve as the attendant (see page 6-203).

C. AUDIO DIAGNOSTICS

10.18 This folder contains 12 flags listing audio diagnostics problems. You can enable only four of the 12 audio problems for the system. In v9.115 or later system, the first four audio problem numbers are selected by default. After the selected audio problems are enabled (set to **Yes**), the problems are displayed as options on the endpoint when the station user accesses the Audio Diagnostics (see page 5-318) feature code (320). Each audio problem has a unique number (01-12) assigned so non-display users can access the feature. To see the same displays across all nodes, the Audio Diagnostics settings must be programmed the same in each node in DB Studio.

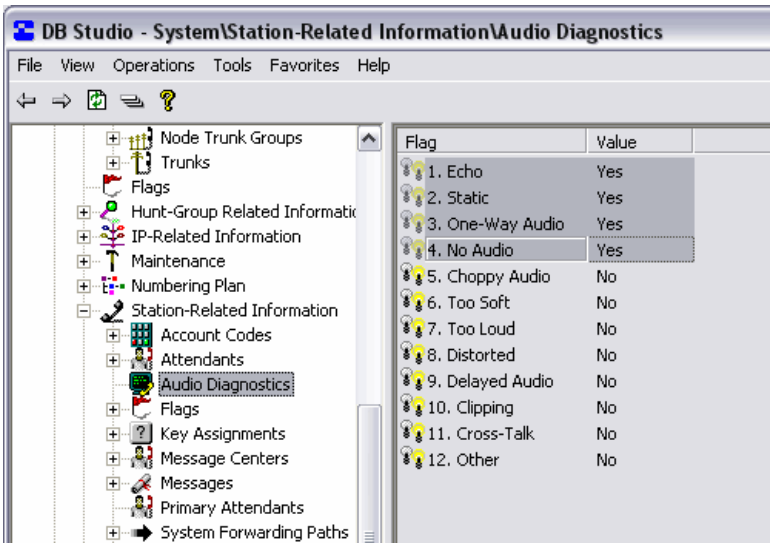
10.19 Select four of the following 12 flags to enable Audio Diagnostics for the system:.

Table 6-35: *Audio Diagnostics*

AUDIO PROBLEM	NUMBER	AUDIO PROBLEM	NUMBER
Echo	01*	Too Loud	07
Static	02*	Distorted	08
One-Way Audio	03*	Delayed Audio	09
No Audio	04*	Clipping	10
Choppy Audio	05	Cross-Talk	11
Too Soft	06	Other	12

* Enabled by default.

10.20 To replace one or more of the default problem numbers, clear one of the default numbers, and then select an option (5-12) to replace it. See the following example.



D. FLAGS

10.21 You can batch program the stations that have a specific station flag enabled by selecting the flag. Then do the following:

1. Right-click anywhere in the right side of the window (if there is no blank space, press **CTRL+N**). An option box appears.
2. Select **Add To List**. A window appears that asks you to select the type of device to include.
3. Select **Keyset** and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the desired phone(s) and then select **Finish**. Your selection appears in the appropriate Attendant Station field.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

10.22 For a complete list of station flags and their definitions, see page 6-209

E. KEY ASSIGNMENTS

10.23 The Key Assignments programming area is used for performing the following programming tasks for keyset map groups DSS/BLF map groups:

- Naming the keymaps.
- Determining the layout of the buttons for the station instrument(s) that will use each keymap. Note that when a button is designated on one type of phone, it will appear on the corresponding button on all stations in that keymap.
- Assigning the stations that will use each map as their standard and alternate keymaps.
- Determining the default values for the user-programmable buttons and, if desired, updating selected stations.

NOTE: The volume adjustment button on the digital phones also can be used for scrolling through displays. However, the analog phone volume buttons are used ***only*** for adjusting volume; analog phones need to have Previous/Next buttons programmed in their keymaps to scroll through displays. They also should have an Accept button to accept entries when in numeric programming mode.

10.24 Key type options and the selections that are required are described below.

- **Accept Key**: *(This button type is not required for digital phones.)* No further programming is necessary. This button will be used to accept an entry when using numeric mode. (Digital phones use the VOLUME button, by pressing both arrows at once.)
- **Call Key**: The next CALL button number is automatically selected.
- **Cancel Key**: No further programming is necessary. This button will be used to cancel a feature or cancel an entry when in numeric mode.
- **Down Key**: *(This button type is not required for digital phones.)* No further programming is necessary. This button can be used scrolling backward through displays. When the phone is idle, it can be used for adjusting volume, in place of the VOLUME button. (Digital phones use the VOLUME DOWN button.)
- **DSS/BLF Key**: Select the circuit or extension number of the station that will appear under the selected DSS/BLF button. If this is assigned to an off-node device, the button can be used for dialing the station, but the lamp will not show station status (no BLF).
- **Feature Key**: Select the feature code that will be entered when this button is pressed.
- **Forward Key**: Select the desired Call Forward feature code.
- **Hunt Group Key**: Select the pilot number of the desired hunt group.
- **Intercom (IC) Key**: No further programming is necessary. This button will be used for seizing an intercom channel or answering a waiting intercom call.
- **Next Key**: *(This button type is not required for digital phones. It is intended for analog phone use.)* No further programming is necessary. This button will be used for scrolling forward through displays. (Digital phones use the VOLUME UP button.)
- **Page Zone Key**: Select the page zone 0-9 or 0-49 that will be used by selecting the page zone access code 9600-9609 or 9600-9649 (respectively).
- **Previous Key**: *(This button type is not required for digital phones. It is intended for analog phone use.)* No further programming is necessary. This button will be used scrolling backward through displays. (Digital phones use the VOLUME DOWN button.)

- **Programmable Key**: Select the desired programmable button number (1-20).
- **Save Key**: *(This button type is not required for digital phones. It is intended for analog phone use.)* No further programming is necessary. This button will be used for saving volume levels or programmed phone features. (Digital phones use the VOLUME button, by pressing both arrows at once.)
- **Secondary Extension Key**: Two fields are programmed for this type of button:
  - — **Selection**: Select the primary station that will be associated with this secondary extension button. The primary station must be on the same node as the station being programmed.
  - — **Ring When __ Calls At Extension**: This field allows the station to receive a burst of ringing when "n" number of calls are present at a primary station, and at least one call is ringing or camped on. (That is, a setting of 2 will cause the station to ring when the primary station receives a third call, while two other calls are ringing.) The ring burst will repeat periodically as long as "n" number of calls are present at a primary station. (This period is determined by the DKTS Secondary Extension Key Alerting Tone timer.) If this field is set to 0, the secondary station will never receive the burst of ringing.
- **SPCL Key**: No further programming is required.
- **SPKR Key**: *(This button type is not required for digital phones. It is intended for analog phone use.)* No further programming is necessary. This button will be used for turning the phone speaker or headset on and off. (Digital phones already have a non-programmable SPKR button and do not need this button in their keymap.)
- **Station Speed Dial Key**: Select the station speed-dial location (0-9) that will be assigned.
- **System Speed Dial Key**: Select the system speed-dial location (000-999 or 0000-4999) that will be assigned.
- **Trunk Group Key**: Select the trunk group that will be selected when this button is pressed.
- **Trunk Key**: Select the individual trunk that will be selected when this button is pressed.
- **Undefined Key**: No further programming is necessary. This button can be programmed by the user.
- **Up Key**: *(This button type is not required for digital phones. It is intended for analog phone use.)* No further programming is necessary. This button can be used for scrolling forward through displays. When the phone is idle, it can be used for adjusting volume, in place of the VOLUME button. (Digital phones use the VOLUME UP button.)

10.25 Three things you should keep in mind when programming keymaps are:

- Key assignments within a keymap are common to all phone types in that map. Modifying a button changes that button assignment for **ALL** phones assigned to that map, regardless of phone type. For example, if the top left button is changed to a user-programmable button for Executive Display Phones in Keyset Map Group 1, the corresponding button will also be user-programmable on all other phones in Keyset Map Group 1.

- Users who have Standard and Alternate keymaps will want to have their user-programmable buttons in the same positions in both keymaps. That way their user-programmable button locations do not change when they switch keymaps. Also, if a fixed feature button is created for the Switch Keymap feature code, it should be in the same position in both keymaps.
- The volume adjustment button on the digital phones also can be used for scrolling through displays. The analog phone volume buttons are used ***only*** for adjusting volume; they need to have Up/Down or Previous/Next buttons programmed in their keymaps to scroll through displays. They also should have an Accept button to accept entries when in numeric programming mode.

10.26 Key Assignment Programming: When you select Key Assignments, the right screen shows the DSS and Keyset Options.

- When you double-click on **DSS**, a list that includes keymap #1 appears. Up to 20 keymaps can be programmed for DSS Units.
- When you double-click on **Keysets**, a list that includes keymap #1 appears. Up to 250 keymaps can be programmed for phones.

NOTE: A CP Server (1024) or ATM system is required to support 250 keyset keymaps.

- When you double-click **IDS DSS**, a list that includes keymap #1 appears. Up to 30 keymaps can be programmed for Executone IDS DSS Units.

10.27 Keyset keymap #1 is used as a template for any new keyset keymaps that are created. This keymap is used regardless of the type of keymap selected. For example, if keymap #1 is an Executive Display keymap with the default Hold button programmed as a Forward button, any newly created keymaps that have a Hold button, even Standard Display keymaps, will reflect this change. Because keymap #1 is used as a template, do not change this keymap. The DSS keymaps do not use a template.

10.28 To add a keymap to the list, right-click on the list and select **Add To DSS (or Keyset) List**. A dialog box appears. Select the Starting ID and the number of keymaps (IDs) to add. When you click **OK**, the keymaps are automatically added.

NOTE: If the Number of IDs value results in exceeding the maximum number of keymaps, an error message appears, and the requested keymaps are not added.

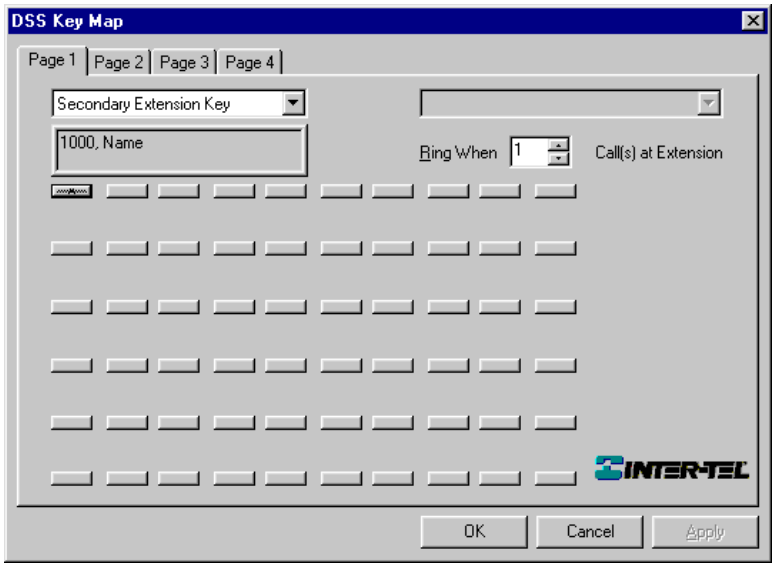
10.29 After you have added keymaps, you can assign a new description to any of the keymaps by selecting the current description and typing the new information (up to 20 characters) in the text box. Then press **ENTER** or select another field to save the change.

10.30 DSS or IDS DSS: When you double-click a specific DSS or IDS DSS keymap, you have the following options:

- **DSS Map**: If programming a DSS Unit, there are three DSS maps available. The 8450 DSS map corresponds to the 50-key DSS Unit. The 60-key DSS Keymap 6-row and 12-row Model options correspond to the 60-key DSS Unit. Use the drop-down box to make your selection. Then press **ENTER** or click on another field to save your change. If programming an IDS DSS Unit, the only available option is the 48-key DSS Unit. By default, there are no stations associated with the DSS or IDS DSS keymap. To define the keymap, you can either manually assign the buttons or use the Populate feature.

  To automatically populate the DSS or IDS DSS keymap, right-click the DSS or IDS DSS Keymap and select **Populate DSS Keymap**. A warning message will display informing you that DB Programming will automatically populate the keymap with the current system's phones. To do this, the system starts at the lowest phone extension value and populates every button in ascending order. If there are more DSS buttons than extensions, any buttons that are not assigned an extension are changed to "Undefined." If you populate the keymap when there are no extensions available, all buttons are changed to "Undefined."

  To manually populate the keymap, double-click **DSS Map** to view an illustration of the device. If programming DSS keymap, four "pages" of buttons are available; one for each device that can be connected to the phone. To see the buttons on a particular page, select the tab at the top of the page. If programming an IDS DSS keymap, however, only one page is available.



  — To change a button requiring additional information (feature code, extension number, etc.), click on the button for which you want to assign a new value. A list window appears. Select the device type from the drop-down box, and select **Next**. Highlight the desired value from the available list, and select **Finish**. The new value will appear in the box below the Key Type drop-down box.

  — To change all other buttons, use the arrow keys on your keyboard to move around the keymap. After you have selected the desired button (selected buttons will have a line with an "x" in the center, click on **Key Type** to scroll to the new key type. If you change the button type to one requiring additional assignments (e.g., extension, hunt group, etc.), any previously programmed information for that button type will remain. To change this information, you must click on the button again and follow the directions detailed above.

NOTE: Secondary extension buttons also have an additional field, "Ring When _ Calls At Extension, as shown above. This field allows the station to receive a burst of ringing when a certain number of calls are present at a primary station, and at least one call is ringing or camped on. If this field is set to 0, the secondary station will never receive the burst of ringing. To set this number, select the current Value and scroll to or enter the new value in the box. Then press **ENTER** or select another field to save the change.

- **DSS Station List**: When you double-click on **Station List**, you see the stations that are currently assigned to the DSS or IDS DSS keymap.

  To add to the list:

  a. Right-click on the list. An option box appears.

  b. Select **Move to Station List**. A window appears that asks you to select the type of device to include.

  c. Select the desired device and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

  d. Highlight the desired device(s) and then select **Move Items.** When you have selected all the devices necessary, click **Finish.** Your selection appears in the station list.

  NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

  To remove a station from the list: Highlight it and press **DELETE**. You will be asked to confirm that you want to delete the station. Select **Yes**.

10.31 Keyset: There are several phone types that are associated with a keymap in DB Programming. These phone types are as follows:

- Model 8600 Phone
- Model 8662, 8660, 8620, 8560, and 8520 Phones
- Model 8500 Phone
- Executive Display Phone
- Standard Display Phone
- Axxess Basic Phone
- Professional Display and Associate Display Phones
- Eclipse™ Basic Phone
- 8-, 12-, and 24-Line AIM Analog Phones
- 8-, 12-, and 24-Line DVK Analog Phones
- 12-, and 24-Line GMX Analog Phones
- 8-, 12-, and 24-Line IMX Analog Phones

NOTE: There are certain limitations associated with the Model 8500, 8520, and 8560 Phones when used with versions 3.0-6.x system software. See page 5-135 for details.

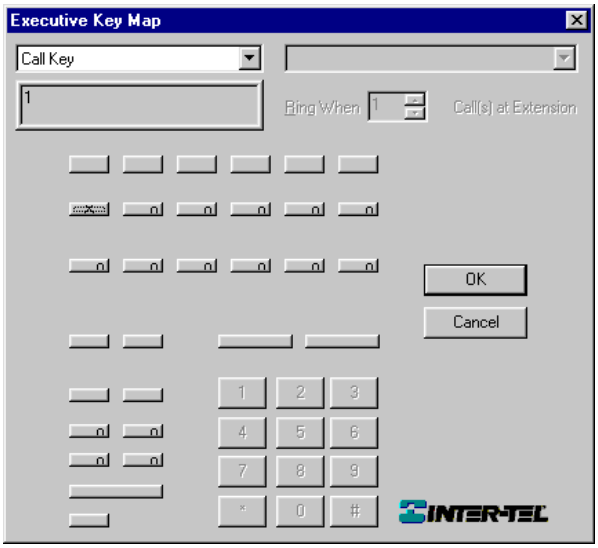
10.32 When you double-click on a specific Keyset keymap, you have the following options:

- **Keyset Map**: Double-click on **Keyset Map** to view an illustration of the selected phone type. The Executive Display Phone is shown in the example below.

  — To change a button requiring additional information (feature code, extension number, etc.), click on the button for which you want to assign a new value. Then use the Key Type drop-down box to scroll to the desired key type. To assign the additional information (e.g., extension, hunt group, etc.), click on the button again. A window with a list of device types appears. Select the device type from the drop-down box, and select **Next**. Highlight the desired value from the available list, and select **Finish**. The new value will appear in the box below the Key Type drop-down box.

  — To change all other buttons, click on the button for which you want to assign a new value. Then use the Key Type drop-down box to scroll to the desired button type. If you change the button type to one requiring additional assignments (e.g., extension, hunt group, etc.), any previously programmed information for that button type will remain. To change this information, you must click on the button again and follow the directions detailed above.

NOTE: Secondary extension buttons also have an additional field, “Ring When _ Calls At Extension, as shown above. This field allows the station to receive a burst of ringing when a certain number of calls are present at a primary station, and at least one call is ringing or camped on. If this field is set to 0, the secondary station will never receive the burst of ringing. To set this number, select the current Value and scroll to or enter the new value in the box. Then press **ENTER** or select another field to save the change.



- **Standard List or Alternate List**: When you double-click on **Standard List** or **Alternate List**, you see the stations that were assigned to the keymap during station programming (see page 6-220). You cannot change these lists here, they are shown for reference only. However, if user-programmable button values have been changed, you can use the Default User Keys option to update stations in the selected keymap (or you can allow the users to update their own stations by using the Feature Key Default feature code – 395). To default user-programmable buttons, do the following:

  — To default ALL phones in the keymap, do not highlight any stations. Then right-click and select **Default All User Keys**. All of the stations on the list will have their user-programmable buttons defaulted.

  — To default one or more phones in the list, highlight the selected stations. (To highlight a series of stations, hold down **SHIFT** while selecting the first and last stations you want. To highlight two or more stations that are not consecutive, hold down **CTRL** while selecting the desired stations.) Then right-click and select **Default Selected User Keys**. Only the stations you select will have their user-programmable buttons defaulted.

- **User Programmable Keys**: Select **User Programmable Keys** to determine the default values of the user-programmable buttons for the keymap and see a list of programmable buttons 1-35 and their current Values. To change the Key Type, Selection, or Ring When field for a programmable button at this station, do the following:

  — **Value**: Select the current Value for the button and scroll through the drop-down list until the desired button type is highlighted.

  — **Selection**: The Selection programming required for each of the key types are as follows:

    - *DSS/BLF Key:* Select the circuit or extension number of the station that will appear under the selected DSS/BLF button. This defaults to the first valid hunt group or extension programmed in the system.
    - *Feature Key:* Select the feature code that will be entered when this button is pressed.
    - *Hunt Group Key:* Select the pilot number of the desired hunt group.
    - *Intercom (IC) Key:* No further programming is necessary. This button will be used for seizing an intercom channel or answering a waiting intercom call.
    - *Page Zone Key:* Select the page zone 0-9 or 0-49 that will be used by selecting the page zone access code 9600-9609 or 9600-9649 (respectively).
    - *Secondary Extension Key:* Select the primary station that will be associated with this secondary extension button.
    - *Station Speed Dial Key:* Select the station speed-dial location (0-9) that will be assigned.
    - *System Speed Dial Key:* Select the system speed-dial location (000-999 or 0000-4999) that will be assigned.
    - *Trunk Key:* Select the individual trunk that will be selected when this button is pressed.
    - *Trunk Group Key:* Select the trunk group that will be selected when this button is pressed.
    - *Undefined Key:* No further programming is necessary. This button can be programmed by the user.

  To program the selection: Right-click the Selection value and do one of the following:

Method A

1.) Select the current Value and enter the new value in the text box.

2.) Press **ENTER**. A screen appears showing you what is associated with the number you entered.

3.) Select **OK**. The new number appears in the field.

Method B

1.) Select **Change Selection**. A window appears that asks you to select the type to include.

2.) Select the type (feature code, station, etc.) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

3.) Highlight the appropriate item and then select **Finish**. Your selection appears in the Selection field.

— **Ring When**: This field allows the station to receive a burst of ringing when a certain number of calls are present at a primary station, and at least one call is ringing or camped on. The ring burst will repeat periodically as long as the programmed number of calls are present at a primary station. (This period is determined by the DKTS Secondary Extension Key Alerting Tone timer.) If this field is set to 0, the secondary station will never receive the burst of ringing. To set this number, select the current Value and scroll to or enter the new value in the box. Then press **ENTER** or select another field to save the change.

10.33 The default values for the programmable buttons are shown in the chart below.

Table 6-36: *Default Programmable Buttons*

BTN #	Key Type	Code	Feature
01	Feature Key	372	Do-Not-Disturb
02-06	Undefined	None	None
07	Feature Key	5	Conference
08	Feature Key	336	Individual Hold
09	Feature Key	345	Transfer To Ring
10	Feature Key	381	System Speed Dial
11	Feature Key	380	Redial
12	Feature Key	7	Page
13	Feature Key	6	Queue Request
14-16	Undefined	None	None
17	Feature Key	313	Background Music
18	Undefined	None	None
19	Page Zone	9601	Zone 1
20	Feature Key	382	Station Speed Dial
21-35	Undefined	None	None

F. MESSAGE CENTERS

10.34 A station can be designated as a message center and assigned a list of stations that it will serve. When you select **Message Centers** from the Station-Related Programming area, you can view a list of the existing message centers.

10.35 If you double-click on a specific message center, you can view the list of stations it serves.

10.36 To assign the stations served by the message center:

1. Right-click anywhere in the right side of the window. An option box appears.
2. Select **Move To List**. A window appears that asks you to select the type of device to include.
3. Select the device type(s) and then select **Next**. A list appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

> **NOTE:** To highlight a series of items, hold down **SHIFT** while selecting the first and last items you want. To highlight two or more items that are not consecutive, hold down **CTRL** while selecting the desired items.

4. Highlight the appropriate items and then select **Move Items**. When you have added all the devices select **Finish**. Your selections appear in the list.

> **NOTE:** If you want to see the programming for a specific item, double-click on its extension number to jump to its programming area.

10.37 To delete a device from the list: Highlight the item(s) and then right-click. Select **Remove Selected Items**.

10.38 You can also assign message centers in station programming by selecting the station's Associated Extensions and Message Center options and selecting the station that will serve as the message center (see page 6-203).

G. MESSAGES

10.39 The system has default do-not-disturb and reminder messages in English/Spanish and Katakana (Japanese) characters. However, the Katakana messages can be reprogrammed only through an Administrator's station (see page 5-283). If you change any of the primary language messages through DB Programming, you should use an Administrator station to reprogram the corresponding secondary language messages to keep the meaning of each message the same for both languages. That is, if the English message 02 is changed to "PAGE ME," a similar message should be programmed for secondary language message 02.

10.40 The default do-not-disturb messages are as follows:

DO-NOT-DISTURB MESSAGES:

01 DO-NOT-DISTURB	11 OUT OF TOWN 'TIL
02 LEAVE A MESSAGE	12 OUT OF OFFICE
03 IN MEETING UNTIL	13 OUT UNTIL
04 IN MEETING	14 WITH A CLIENT
05 ON VACATION/HOLIDAY 'TIL	15 WITH A GUEST
06 ON VACATION/HOLIDAY	16 UNAVAILABLE
07 CALL ME AT	17 IN CONFERENCE
08 AT THE DOCTOR	18 AWAY FROM DESK
09 ON A TRIP	19 GONE HOME
10 ON BREAK	20 OUT TO LUNCH

10.41 The default reminder messages are as follows:

REMINDER MESSAGES:

01 MEETING	11 CALL ENGINEERING
02 STAFF MEETING	12 CALL MARKETING
03 SALES MEETING	13 CALL ACCOUNTING
04 CANCEL MEETING	14 CANCEL DND
05 APPOINTMENT	15 CANCEL CALL FWD
06 PLACE CALL	16 TAKE MEDICATION
07 CALL CLIENT	17 MAKE RESERVATION
08 CALL CUSTOMER	18 REVIEW SCHEDULE
09 CALL HOME	19 LUNCH
10 CALL CORPORATE	20 REMINDER

10.42 You can reprogram any or all of the reminder messages and/or do-not-disturb messages with messages of up to 16 characters each. When you select Messages from the Station-Related Programming list, you will see the option for programming in English.

10.43 When you double-click on **Primary Language** you see the DND and Reminder Message options. Double-click on the type of message you want to program to view the current list.

10.44 To program a specific message, select its current Value. Then enter the new information in the text box (up to 16 characters) and press **ENTER** or select another field to save your change.

H. PRIMARY ATTENDANTS

10.45 When you select Primary Attendants from the Station-Related Programming list, you see fields for programming the Primary Attendant Station and the Local Attendant Station. Those fields are programmed as described below.

- **Primary Attendant Station**: This is the network primary attendant.
- **Local Attendant Station**: This is the attendant for the node being programmed.

10.46 If an attendant is set to None, calls that would normally go to the attendant are handled as follows:

- If the system has seized the call, but it has not been sent to a station, the call is disconnected.
- If the call has been sent to a station, it remains at the station and rings until answered.
- If the call is not seized and not sent to a station, the caller hears ringing until the caller hangs up. The call will not ring at any station.

10.47 To program an attendant do one of the following:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the existing Primary Attendant Station or Local Attendant Station. An option box appears.
2. Select the Change Attendant Station option. A window appears that asks you to select the type of device to include.

3. Select the desired device or None and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the desired device and then select **Finish**. Your selection appears in the appropriate Attendant Station field.

I. SYSTEM FORWARDING PATHS

NOTE: This feature cannot be used unless the feature has been enabled as described on page 6-97.

10.48 Up to 200 unique forwarding paths (001-200) can be programmed. When System Forwarding is selected from the Station-Related Information programming area, the list of forwarding paths 1-200 appears.

10.49 This area is used for assigning forwarding points to the system forwarding paths. Stations can then be programmed to use the forwarding paths in station programming (see page 6-214).

10.50 Each forwarding path can have a distinctive description (of up to 20 characters) and four forwarding points. The forwarding points can be local stations, voice mail ports, or hunt groups, or they can be off-node devices. Program the fields for the forwarding paths as follows:

10.51 Description: Select the current description and enter a name for the forwarding path, up to 20 characters, in the text box. Then press **ENTER** or select another field to save the change.

10.52 Forwarding Point(s): Up to four forwarding points can be programmed for each path. To program a forwarding point, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the existing Forwarding Point. An option box appears.
2. Select **Change Forwarding Point**. A window appears that asks you to select the type of device to include.
3. Select the desired device and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the desired device and then select **Finish**. Your selection appears in the appropriate Forwarding Point field.

J. SYSTEM SPEED DIAL

10.53 There can be up to 1000 system speed-dial numbers (5000 if using a CP Server). The default database contains only system speed-dial bin #000, because this is used by the system to establish links to other speed-dial bins. You may create new bins, individually or in batches, as needed, through DB Programming.

10.54 System speed-dial bins may also be programmed from an Administrator's phone, using the system speed-dial feature code (9801). If the user enters a bin that is currently not equipped, one will be automatically equipped and ready for programming. This is transparent to the user at the phone.

10.55 When converting from an older software version to a newer version, the conversion program will automatically unequip all system speed-dial bins that have an empty name and number, with the exception of bin #000.

10.56 A user at a phone cannot program system speed-dial bins when there is an active DB Programming session, and no one can log into DB Programming when the system speed-dial bins are being programmed from a phone.

10.57 System speed-dial numbers can only be used on the node where they are programmed. Each node in the network must have its own system speed-dial numbers. For each speed-dial number, you can program the following information:

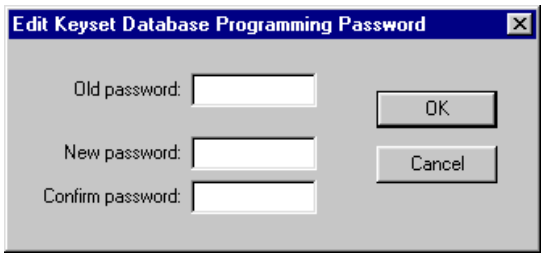
- **Name**: If desired, you can program a name for the speed-dial number. Each name can have up to 16 characters. To program the name, select the current Value and enter the new name in the text box. Then press **ENTER** or select another field to save the change.
- **Number**: Numbers can include up to 48 digits and can include digits (0-9, *, and #), timed pauses, or hookflashes/recalls. Timed pauses and/or hookflashes/recalls are used when entering a series of numbers, such as access codes, security codes, and telephone numbers for specialized common carrier (SCC) dialing (U.S. only). To include a pause in the number, enter the letter P for a pause. To include a hookflash/recall, enter F. The pause length represented by the P is determined by the Pause timer. Each pause or hookflash (recall) is considered one of the 32 digits. To program the number, select the current Value and enter the new digits in the text box. Then press **ENTER** or select another field to save the change.
- **Private Number**: Determine which system speed-dial number locations may be viewed on display phones. Non-display numbers do not appear on display phones when dialed and cannot be redialed at display phones. Non-display numbers will appear in the SMDR when dialed. To allow the speed-dial number to be displayed when dialed, enable the flag. Disable the flag to make it a non-display number. To enable the flag, select the current Value and select the check box to place a check in it. Then press **ENTER** or select another field to save the change. (To remove the check and disable the flag, select the check box again.)

K. KEYSET DB PROGRAMMING PASSWORD

10.58 This password is used by the Administrator stations when programming through a phone. It can consist of only digits. See FEATURES, page 5-276 for details.

10.59 To program the password right-click on **Keyset DB Programming Password**. A window appears as shown below. Then do the following:

1. Enter the old password, if one exists, in the Old Password box.
2. Enter the new password (up to 8 digits, using digits 0-9) in the New Password box. The digits will not appear on the screen when typed; they will appear as asterisks (***).
3. Retype the password exactly as before in the Confirm Password box.
4. Select **OK** to exit and save the password. If the entered passwords match, you will return to the Password field. If not, you must re-enter the new password and verify it again. If you make a mistake while entering the password or want to leave it unchanged, select **Cancel**.



NOTE: To provide system security and prevent unauthorized access to the system database, Keyset DB Programming should have a password. To make the passwords difficult to guess, they should not consist of one digit repeated several times. The password should be changed periodically.

11. TRUNK-RELATED INFORMATION

11.1 The Trunk-Related Information folder contains the following programmable areas:

- Call Routing Table (see below)
- Music-On-Hold Profiles
- SIP Gateways.

A. CALL ROUTING TABLES

NOTE: The Advanced CO Interfaces software license is required to use Call Routing Tables. See page 6-97.

11.2 When a trunk group is programmed to ring in to a call routing table, the system analyzes the digits received from DID, ANI, or DNIS [DDI in Europe] number strings, and Caller ID [CLIP] Information. It uses these digits to determine which route to use for the call, as determined by the call routing table. The caller information is passed along to the phone users' display along with other call routing table information about the callers (such as location, name, or which advertisement they saw). For a full description of how call routing tables work and sample applications, see page 5-89 in FEATURES.

11.3 A call routing table consists of fields for dialing patterns, ring-in types/destinations, and names. The fields are used for analyzing the source or destination of the call and directing it to the proper destination.

11.4 Each individual trunk, not the trunk group, is programmed to collect digits using Caller ID, DID, ANI, or DNIS [CLIP or DDI in Europe]. If the trunk is programmed to collect digits, but the trunk group does not use call routing tables, the system routes the call based on the trunk group programming and ignores the collected digits. If the trunk is not set up to collect digits and the trunk group uses call routing tables, the system uses the call routing destination for "no-digit" calls. (See page 6-244 for trunk programming information.)

11.5 Call Routing Tables allow patterns that route calls to destinations of individual trunks, trunk groups, and ARS anywhere single ring-in destinations are programmed.

11.6 If the caller is routed to a trunk or trunk group that is busy, the system will camp on to the trunk. The caller will hear busy tones, followed by music. When a trunk becomes available, the caller hears dial tone and can complete the call.

11.7 To provide security on outgoing calls made through direct trunk-to-trunk interfaces, the toll restriction of the selected ***outgoing*** trunk will be checked. However, if ARS is used, toll restriction is not checked. The toll restriction of the ***incoming*** trunk's group and that trunk's "Subject To Toll Restriction" flag are not checked. If a caller dials a number that is not allowed through toll restriction, the call will be routed to the primary attendant.

NOTE: Even though Primary Rate trunks appear in the selection lists, they can only be used by selecting ARS; individual B-channel trunks or trunk groups containing B-channel trunks will not function properly with this feature.

11.8 If you have numbers, such as unused DID [DDI in Europe] numbers, that you do not want callers to use, you can program them to ring in to an extension number that returns busy signal. To do this, locate a phone that does not have an attached MDPM. Then program that phone's circuit as a phone with an MDPM to create a "phantom" MDPM extension (see Keyset Card Configuration on page 6-118). In Individual Station programming, program the MDPM circuit with the "DID/E&M [DDI in Europe] Receive Busy Instead Of Camp-On" flag to cause it to always send busy signal (the system will consider it permanently off hook because no equipment is installed on the circuit). When creating the Call Routing Table for the trunk group, program the selected numbers to ring in to the MDPM that returns busy signal.

Programming Call Routing Tables

11.9 When you select Trunk-Related Programming, only the Call Routing Tables option appears in the right screen. When you double-click on **Call Routing Tables**, you see a list of call routing tables 1-15. At this level you can change the Description or Call Routing Key.

- **Description**: To change the description of the call routing table, select its current Value and enter the new description in the text box. Then press **ENTER** or select another field to save the change.
- **Call Routing Key**: This field determines which information the call routing table will use to direct the call. If you select Trunk number, the system will look for the dialed number provided by DID [DDI in Europe] or DNIS. If you choose **Outside Party Number**, the system checks for the ANI or Caller ID [CLIP] information that identifies the source of the call. Select the desired call routing button by selecting the desired option button. To change the Call Routing Key, select the current Value and scroll to the desired setting. Then press **ENTER** or select another field to save the change.

11.10 When you select a specific Call routing Table, the Patterns option appears. To program the patterns for the call routing table, double-click **Patterns**. A list of existing patterns, if any, appears.

11.11 To delete a pattern: Highlight the pattern to be deleted and right-click. Then select **Remove Selected Items**.

11.12 To add a single pattern: To create a single pattern, right-click anywhere in the right screen and select **Add To List** (if there is no blank space, press **CTRL+N**). A blank option appears in the list.

11.13 To copy and paste one or more patterns: Highlight and right-click the pattern you want to copy. The Copy Patterns dialog box opens showing the Current CRT. Select the Destination CRT using the drop-down list box. Click **Copy** to copy the pattern(s) from the Current CRT to the Destination CRT, or click **Cancel** to cancel the operation.

11.14 Batch Create Patterns: Right-click anywhere in the right screen and select the Batch Create Patterns option. In the first screen you must indicate whether you are creating Area Code or DID [DDI in Europe] patterns by selecting the appropriate option button. Then select **Next**. You can then program the fields for the patterns as follows:

- **Area Code Patterns** *(U.S. only)*:
  - **Pattern Name**: Enter the desired name in the Pattern Name text box. This name will be used for all area code patterns created using this batch command. The name field determines how calls will be identified at display phones. The name can be a word or number that identifies the call source. You can use any combination of letters and numbers.

— **Ring-in Type**: Select the desired ring-in type by selecting the desired option button. This ring-in type will be used for all area code patterns created using this batch command. The call can ring in to a single extension number (off-node device, station, hunt group, application, trunk group, individual trunk, or ARS), an extension list, as a DISA call (which allows the caller to select a destination), a destination based on collected digits, or as another call routing table. If Single, Extension List, or Call Routing Table is selected, you must also program the destination as described below. DISA ring-in requires no additional programming. Any DISA codes programmed for the trunk group will apply to callers.

— **Ring-in Destination**: To select the specific destination for the area codes, select the Change Ring-In Destination button and use the selection screens to choose desired ring-in destination and type. When you return to the Call Routing Name and Ring-In Information screen, select **Next** to continue.

NOTE: Even though Primary Rate trunks appear in the selection lists, they can only be used by selecting ARS; individual B-channel trunks or trunk groups containing B-channel trunks will not function properly with this feature.

— **Area Codes**: Select the area code(s) that will be included in the batch of patterns. Then you can select a series of area codes by pressing **SHIFT** and selecting the first and last in the series. Or you can select individual area codes by holding down **CTRL** while highlighting the desired area codes. When you have selected all the area code you need, select **Finish**.

- **DID Patterns** *(U.S. only)*:

  — **Name**: Each pattern is identified in the list by the digits that are sent after the handshake. The base number of the DID numbers is the name prefix. The prefix and the collected digits are combined to form a complete number for display (for example, a base number 9619 would combine with digits 000 to display 9619000). Enter the base number of the DID number block in the Name text box.

  — **Initial DID**: This is the first number in the series of DID numbers being programmed (without the base number). For example, if you were programming for DID numbers 9619000 through 9619019 (9619 is the base number), you would enter 000. Enter the first DID number in the Initial DID text box.

  — **Number of Patterns**: This is the number of DID numbers that will be included in the pattern list. In the example above (9619000 through 9619019) you would enter 20 because the range includes 20 DID numbers. Scroll to or enter the appropriate number (up to 999) for your DID list in the box.

- **DDI Patterns** *(Europe only)*:

  — **Name**: Each pattern is identified in the list by the digits that are sent after the handshake. The base number of the DDI numbers is the name prefix. The prefix and the collected digits are combined to form a complete number for display (for example, a base number 01162903 would combine with digits 000 to display 01162903000). Enter the base number of the DDI number block in the Name text box.

  — **Initial DDI**: This is the first number in the series of DDI numbers being programmed (without the base number). For example, if you were programming for DDI numbers 01162903000 to 01162903019 (01162903 is the base number), you would enter 000. Enter the first DDI number in the Initial DDI text box.

— **Number of Patterns**: This is the number of DDI numbers that will be included in the pattern list. In the example above (01162903000 to 01162903019), you would enter 20 because the range includes 20 DDI numbers. Scroll to or enter the appropriate number (up to 999) for your DDI list in the box.

11.15 Programming the patterns: You can then program the fields for the patterns as follows:

- **Pattern**: The patterns are listed in the call routing table in the order they are checked against the incoming digits received from the trunk interface for the incoming call. When the system finds a match with a pattern, it stops searching through the list and routes the call according to the programming for that pattern. If the call reaches the end of the list without matching any patterns, the call is automatically sent to the primary attendant. Pattern entries can include up to 32 digits. Valid characters are shown in the table below. To edit pattern, select the current Pattern value and enter the new digits in the text box. Then press **ENTER** or select another field to save the change.

Table 6-37: *Patterns*

Character	Matches
0-9,*, #	Specific digits
X	Any digit 0-9
A	Any digit 0-9, #, or *
N	Any digit 2-9
Z	Digits 0 or 1
[*x*]	Optional digits; for example, 2[13]2 matches 22 or 2132.
(*x*-*x*)	Range of digits; for example, with a U.S. system, (320-360)XXXX matches any 7-digit number with an office code between 320 and 360, inclusive; with a European system, (200-209)XXXX would catch all 7-digit DDI numbers in the range 200-209.
+	Matches 0 or more digits of any value. A “+” by itself will match any incoming information, including anonymous calls that send no digits.
E	End of dialing; the pattern will not match if any digits are dialed beyond this point. An “E” by itself in a pattern will match when no digits are received, such as anonymous Caller ID [CLIP] calls.

NOTE: To ensure all patterns are included in the Call Routing Table, always include + followed by E in the last two patterns. The + pattern allows the system to detect any digits that did not match the other patterns in the list. The E pattern allows the system to detect when no digits are received. This ensures the call is labeled as either ANY or EMPTY when the call is sent to the primary attendant. These labels notify the attendant that the telco did not properly route the DNIS digits.

- **Description**: This field determines how the call will be identified at display phones when received. It can contain up to 12 characters. The name can be any word or number that identifies the call source. (You can use any combination of letters and numbers.) To change the description of the pattern, select its current Value and enter the new description in the text box. Then press **ENTER** or select another field to save the change.
- **Ring-In Type**: To change the Ring-In Type, select the current Value and scroll to the desired setting. Then press **ENTER** or select another field to save the change. The following ring-in types are available:

  — **Single**: The destination can be a station, hunt group, application, trunk group, individual trunk, or ARS. These can be local or off-node devices.

— **DISA**: If the trunk group is to be used for DISA, a security code can be assigned by selecting the Ring-In field and entering the code in the text box.

— **Extension List**: You can choose from any extension list in the local node.

— **Call Routing Table**: When programming a call routing table that rings in to another table, make sure the two tables do not send calls to each other, creating a "loop." You can only select call routing tables located on the local node.

— **Collected Digits**: This ring-in type indicates that the collected DID [DDI in Europe] or DNIS digits (plus the base digits) should be used as the destination extension. This helps to keep the number of call routing table entries down to a minimum when using a 3200D to route calls to extensions. Both trunk groups and call routing table entries can use the new ring-in type. When the new Ring-In Type of Collected Digits is selected, the Ring-In Destination field is empty. If the collected digits plus the base digits do not make up a valid ring-in destination, the call is routed to the primary attendant. Valid ring-in destinations include on- or off-node stations, on-or off-node hunt groups, trunk groups, individual trunks, voice mail applications, automated attendants, and ARS.

NOTE: If Single, Extension List, or Call Routing Table is selected, you must also program the destination as described below. DISA ring-in requires no additional programming. Any DISA security codes programmed for the trunk group will apply to callers.

11.16 Ring-In Destination: To select the specific destination for the pattern, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the existing Ring In Destination. An option box appears.
2. Select the Change Ring-In Destination option. A window appears that asks you to select the type of device to include.
3. Select the appropriate destination type and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.) If the Ring-In Destination is set to "None" the call will be sent to the primary attendant. (Do not set this to None if there is not a primary attendant, because the call will not ring at any station.)

NOTE: Even though Primary Rate trunks appear in the selection list, they can only be used by selecting ARS; individual B-channel trunks or trunk groups containing B-channel trunks will not function properly with this feature.

4. Highlight the desired device and then select **Finish**. Your selection appears in the Ring-In Destination field.

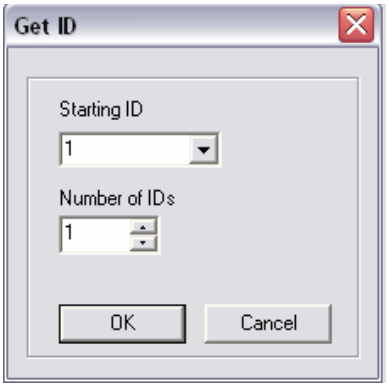
B. MUSIC-ON-HOLD PROFILES

11.17 This section provides the instructions for creating a Music-On-Hold Profile for a Call Routing Table (CRT). For complete information about this feature, see "Music-On-Hold for Call-Routing Tables" on page 5-98.

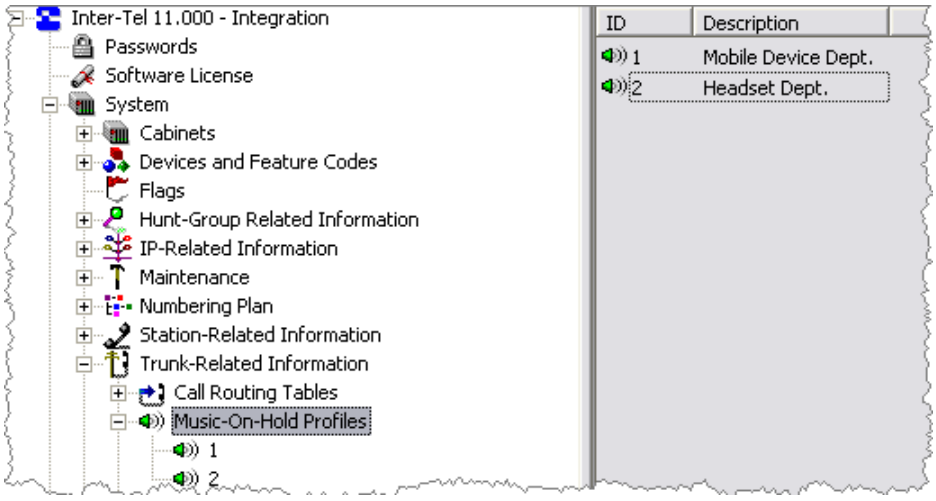
Programming the Music-On-Hold Profiles

11.18 To program the Music-On-Hold Profiles for trunk groups:

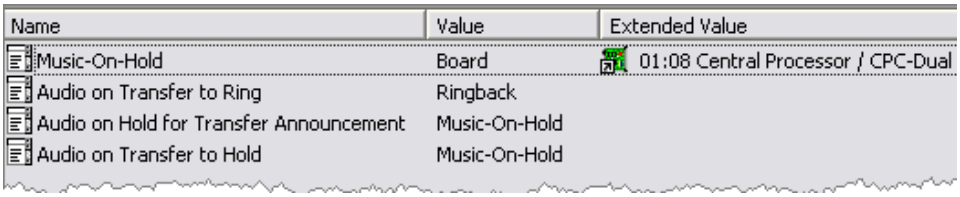
1. Go to System – Trunk-Related Information – **Music-On-Hold Profiles**.
2. Right-click **Add to Music-On-Hold Profiles List** in the right pane. The Get ID dialog box appears.
3. Select the starting ID and the number of IDs (up to 25) to create. For example, to create IDs 1–9, select **1** as the "Starting ID" and **9** as the "Number of IDs."
4. Click **OK**. The items are added to the list with default values.



5. Click the desired Description field, and then type the profile description.



6. Double-click one of the Music-On-Hold Profiles. The following screen appears.



7. Set the desired values for each field by selecting the value you want to change and scrolling to the desired setting. The available fields are as follows:

   - **Music-On-Hold**: Determines what outside callers will hear after their call is put on hold. This option can be set for Silence, Tick Tone, Ringback, Board, or Use Next Device's Audio Source. It is set to *Board* by default.
   - **Audio on Transfer to Ring**: Determines what outside callers, except those using Direct Inward System Access (DISA), will hear after their call is transferred. This option can be set for Music-On-Hold, Ringback, Silence, Tick Tone, Board, or Use Next Device's Audio Source. It is set to *Ringback* by default. (The feedback that DISA callers hear is determined by the DISA Transfer tone system flag.)
   - **Audio on Hold for Transfer Announcement**: Determines what outside callers (except those using DISA) will hear while their call is on transfer hold. After the transfer is completed, the caller will hear the Audio On Transfer to Ring (or Hold) selection. This option can be set for Ringback, Silence, Tick Tone, Music-On-Hold, Board, or Use Next Device's Audio Source. It is set to *Music-On-Hold* by default. (The feedback that DISA callers hear is determined by the DISA Transfer tone system flag.)
   - **Audio on Transfer to Hold**: Determines what outside callers (except those using DISA) will hear while their call is being transferred to hold. This option can be set for Ringback, Silence, Tick Tone, Music-On-Hold, Board, or Use Next Device's Audio Source. It is set to *Music-On-Hold* by default. (The feedback that DISA callers hear is determined by the DISA Transfer tone system flag.)

8. Press **ENTER** or click outside the field to save your change.

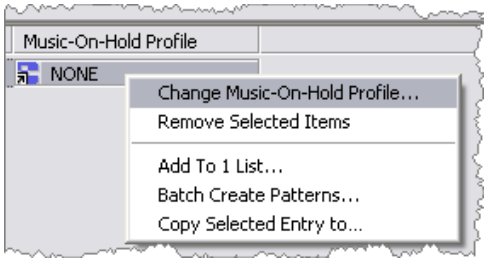
11.19 To program a Music-On-Hold Profile for a CRT:

1. Go to System – Trunk-Related Information – **Call Routing Tables**. A list of CRTs appears.
2. Click the desired CRT.
3. Right-click anywhere in the right screen, and then click **Add To List** (if there is no blank space, press CTRL-N). The Patterns list appears.

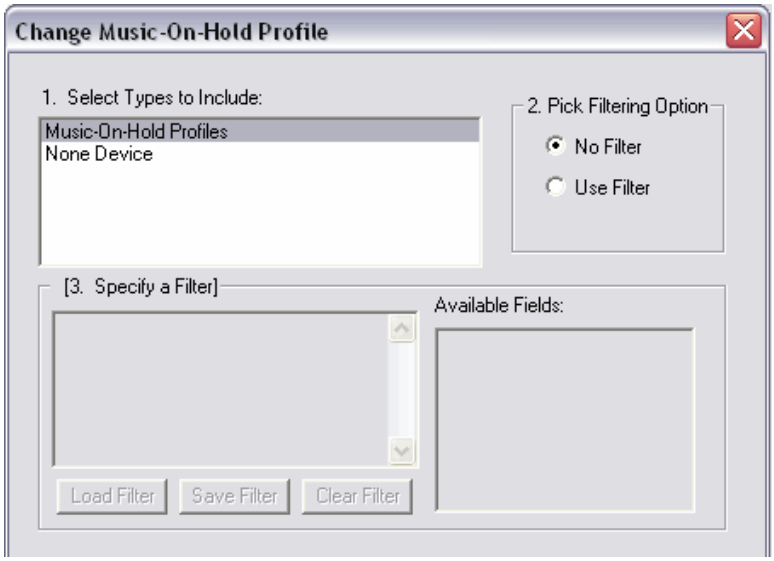
4. Do one of the following:

   - Type the desired profile ID in the **Music-On-Hold Profile** field.
   - Select the profile ID from the Change Music-On-Hold Profile dialog box:

     1.) Right-click the existing Music-On-Hold Profile value. A submenu appears.

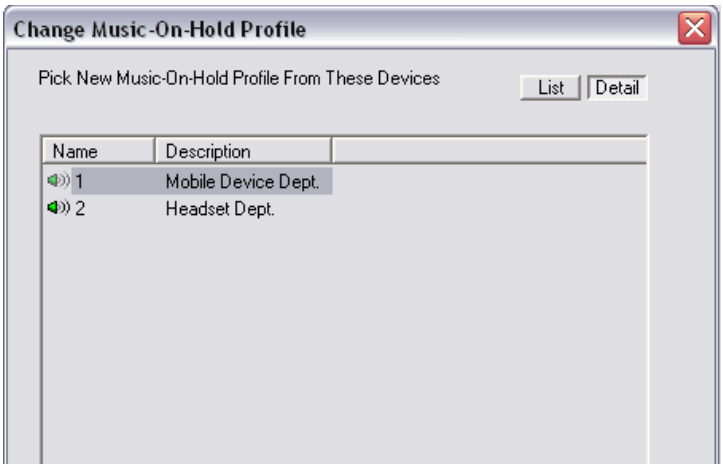


2.) Select **Change Music-On-Hold Profile**. The Change Music-On-Hold Profile dialog box appears.



3.) Select **Music-On-Hold Profiles**, and then click **Next**.

4.) Select the desired Music-On-Hold profile, and then click **Finish**.



C. SIP GATEWAYS

11.20 The system supports SIP (Session Initiated Protocol) trunks to reach the CO. SIP trunks allow the system to communicate with the CO via SIP-enabled gateways. As the SIP protocol becomes more and more popular, it is important to be able to communicate to SIP gateways in the IP-centric world. Inter-Tel currently supports:

- Quintum® AFT 400 SIP gateway
- AudioCodes™ MP-114 SIP gateway

11.21 For details about the SIP gateways, refer to the *Inter-Tel Axxess/5000 Quintum SIP Gateway Installation Manual* (document part no. 835.3123) or the *Inter-Tel Axxess/5000 AudioCodes MP-114 SIP Gateway Installation Manual* (document part no. 835.3202)

11.22 The SIP Gateways field (System\Trunk-Related Information\SIP Gateways) contains a name and description field for the gateway. When you double-click the gateway you jump to the programming fields for SIP trunks.

Programming SIP Trunks

11.23 At least one SIP gateway must exist at all times (even in a default system) because it follows the model of a CO trunk group. To allow the system to communicate with the CO through a SIP gateway, the following fields must be programmed:

11.24 **SIP Trunks**: SIP trunks allow the system to communicate with the CO via SIP-enabled gateways. When you access the SIP Trunks folder under System\Trunk-Related information\SIP Gateways\<*Gateway*>\SIP Trunks, the Extension, Value, Type, and Address fields for the SIP Trunk are displayed. SIP trunks have the following functionality:

- They are transparent to the system user because SIP trunks act like any other CO trunk in the system.
- They support transferring trunks, putting trunks on hold, and connecting trunks to conferences similar to other CO trunks in the system.
- They support making and receiving calls by any endpoint.
- They support peer-to-peer audio by IP endpoints.
- They reside in CO trunk groups just like other trunks so that SIP trunk calls can be routed using Automatic Route Selection (ARS)

11.25 If you double-click a SIP trunk, you jump to the SIP trunk programming fields. Program the following fields for the SIP trunk:

- **Call Configuration**: Defines the setting that SIP gateways will use when connected to a call. After programmed, you assign each SIP trunk a specific Call Configuration. This allows you to program one call configuration for multiple trunks. By default, all IP devices are placed in Call Configuration 1, which is programmable. This option is also located under System\IP-Related Information\Call Configurations.
- **CO Trunk Group**: Defines the CO trunk group to which SIP trunks can be moved. Because SIP trunks are considered CO trunks, this feature functions just like other trunks. By default, they are placed in PP011. This option is also located under System\Devices and Feature Codes\CO Trunk Groups.
- **Network Group**: Defines the network group to which SIP trunks are assigned. By default, they are assigned to the PP029 network group. This option is also located under System\Devices and Feature Codes\Network Groups.

- **SIP Gateway**: At least one SIP gateway must exist at all times (even in a default system) because it follows the model of a CO trunk group. To communicate with a SIP gateway, the following fields must be programmed (this option is also located under System\Trunk-Related Information\SIP Gateways.):
  - **SIP Trunks**: Indicates the SIP gateway where SIP trunks are placed.
  - **IP Address**: Indicates the IP address of the SIP gateway.
  - **Listening Port**: Indicates the port on the CPC that the system listens to for SIP messages. By default, this port is set to 5060 (standard SIP port).
  - **SIP Gateway Name**: The SIP Gateway Name field provides the ability to put a SIP gateway behind a NAT. Enter the gateway name that is programmed on the SIP gateway itself. The default value is *DEFAULT*. If you attempt to set the SIP Gateway Name to an empty string, the system displays an error message. To correctly inform the gateway where the SIP messages are originating from, you must also program the System NAT IP Address field (see page 6-267). For complete NAT traversal and programming information, see NAT Traversal for IP Endpoints on page F-14.
- **CP History** (Used for diagnostic purposes only): Enables SIP message output in the CP history file, which allows you to trace SIP trunk calls. The available output options are Headers Only, No History, or Full History. By default, this field is set to Headers Only.
- **Language**: Determines which language is used for the voice prompts and displays when this trunk is used.
- **Receive and Transmit Gain**: Allows gain control programming for trunks on an IPRC that has been modified using the Gain Control Kit. All trunks, regardless of the board they are on, will have default transmit and receive gains of 0 dB. If the trunk resides on a board that does not support the gain control, the values are ignored. The range for the Transmit or Receive gain is -16 dB to +12 dB.

12. SYSTEM TIMERS AND LIMITS

12.1 There are programmable timers that control various system functions. In the Timers and Limits programming area, you can change any or all of them.

12.2 The default values have been carefully selected to ensure proper system operation under most circumstances. Occasionally, one or more of the timers may need to be adjusted.

12.3 To change the value of a system timer, select its current Value and enter the new value in the text box. Then press **ENTER** or select another field to save the change.

12.4 The timer definitions are shown below and on the following pages. A "DID [DDI in Europe]" in the timer names indicates that the timer applies to DID [DDI] trunks only. "DKTS" refers to phones only and "SL" applies to single-line sets only. "E&M" timers apply only to E&M trunks (except E&M Disconnect Flash Duration). "LS" indicates a loop-start trunk timer, "GS" refers to ground-start trunks, and "LS/GS" applies to both types. The "UCD" timer applies only to UCD hunt groups.

Table 6-38: *System Timers*

Timer	Default Value	Range	Purpose
Abandoned Call	10	1-255 minutes	After a call has recalled to the last possible station, it will recall until this timer expires. If it remains unanswered, the system will disconnect the call. (This timer is started when the call begins recalling the first recall station.)
Agent Help Tone Interval	0	0-1000 seconds	The system can be programmed to send a tone to the call parties when a supervisor joins or monitors a call using the Agent Help feature. This timer determines how often the tone will be heard. If you set this timer to 0, there will be a single tone when the supervisor joins the call, if tones are enabled, but there will not be periodic tones through out the call.
Background Keyset Update Date and Time	360	60-360 minutes	This determines how often phone date and time displays are updated.
Barge-In Notification Tone Frequency Timer			Controls whether the agent sees the BARGE-IN IN PROGRESS or BARGE-IN TERMINATED display whenever a supervisor barges-in or hangs up after barging-in.
Busyout Switch Timeout	60	10-10000 seconds	When a T1/E1 circuit is made busy using the busyout switch or programming command, this timer starts. If it expires, the Make-Busy LED on the card flashes rapidly and the busyout process is canceled.
Camp-On	3	1-255 seconds	Length of time a caller hears busy tone before camping on.
Camp-On Tone	15	5-255 seconds	Length of time between camp-on tones.
CO Reseize	3	1-15 seconds	When a user reseizes a trunk, this timer determines the length of time the system will hold the trunk open.
Dial Initiation	15	5-30 seconds	Limits time a station can remain off hook without dialing before the system sends reorder tones.

Table 6-38: *System Timers (Continued)*

Timer	Default Value	Range	Purpose
DID Disconnect Recognition	150	2-500 millisec. (0.002-0.5 sec.)	This is the minimum amount of time the circuit must be on hook for the system to recognize that the call has been disconnected.
DID Inpulse-dial Interdigit Pause Recognition	240	2-500 millisec. (0.002-0.5 sec.)	This is the minimum pause allowed between in-coming pulse-dial signals sent from the remote circuit. It allows the system to recognize the separation between digits.
DID Off-hook Debounce	10	2-500 millisec. (0.002-0.5 sec.)	This is the minimum amount of time the circuit must be off hook before the system will recognize another on-hook/off-hook transition.
DID On-hook Debounce	12	2-500 millisec. (0.002-0.5 sec.)	This is the minimum amount of time the circuit must be on hook before the system will recognize another on-hook/off-hook transition
DID Post-Seize Delay	64	2-500 millisec. (0.002-0.5 sec.)	This is the minimum time allowed between the recognition of a seizure and the beginning of digit validation. Used only for Immediate-Dial circuits.
DID Post-Signal Delay	30	2-500 millisec. (0.002-0.5 sec.)	This is the minimum time allowed between the end of a handshake and the beginning of digit validation. Used only for Wink-Start and Dial-Delay circuits.
DID Pre-Signal Delay	100	2-500 millisec. (0.002-0.5 sec.)	This is the amount of time that must elapse between the recognition of a trunk seizure and the handshake.
DID Ready Timeout	4000	2-65000 millisec (0.002-65 sec.)	Used only for Wink-Start and Dial-Delay circuits. This is the maximum time the T1 or T1/E1/PRI Card or SLA will wait for the “Digit Register Ready” command. If the timer expires, the SLA or card will automatically initiate a handshake. Used only for Wink-Start and Dial-Delay circuits
DID Seizure Recognition	30	2-500 millisec. (0.002-0.5 sec.)	This is the minimum amount of time the circuit must be on hook for the system to recognize that the trunk has been seized.
DID Signal Hold	200	2-500 millisec. (0.002-0.5 sec.)	This is the maximum duration of a wink-start handshake or the minimum duration of a dial-delay handshake. Used only for Wink-Start and Dial-Delay circuits.
DISA Invalid Extension Failure Limit	3	0-9	This determines how many times a caller will be allowed to enter an incorrect DISA security code without being dropped. To program the failure limit, enter the desired number in the text box. Entering 0 will allow unlimited attempts.
DISA Security Code Failure Limit	3	0-9	This determines how many times a DISA caller will be allowed to dial an invalid extension number before being sent to the primary attendant. To program the limit, enter the desired number in the text box. Entering 0 will allow unlimited attempts.

Table 6-38: *System Timers (Continued)*

Timer	Default Value	Range	Purpose
Disconnect Wait After Dialing	20	1-60 seconds	Length of time the system waits after dialing an outside telephone number before checking the trunk for disconnect.
DKTS Alternate Transient Display	10	1-255 tenths (0.1-25.5 sec.)	This is the amount of time transient displays will appear on phones with the "Alternate Transient Display Timer" station flag enabled.
DKTS Secondary Extension Key Altering Tone	6	1-1000 seconds	This timer is in effect only if the station has a number greater than 1 in the "Ring When *n* Calls At Extension" field. It determines how often the station with the secondary extension button will hear an alerting tone while the primary station has *"n"* number of calls present.
DTMF Digit Duration/ Pause	60	30-255 millisec. (0.03-0.255 sec.)	Adjusts the duration of and pause between digits of DTMF tones sent by the system. Both the tone and the pause will use the assigned duration (for example, a 6/100 second tone has a 6/100 second pause between digits). **NOTE:** If using an IPRC for private IP networking, set this value to 100 ms or higher. If this value is lower than 100, the DTMF tones may not be detected.
E&M Answer Recognition	4500	2-20000 millisec. (0.002-20 sec.)	Determines the minimum amount of time the receiving PBX must be off-hook, when a station user places an outgoing call on an E&M trunk. This allows the system to recognize that the call has been answered.
E&M Dial Delay	70	2-500 millisec. (0.002-0.5 sec.)	This is the maximum amount of time the E&M circuit will wait before transmitting digits following a handshake. This timer goes in effect after the handshake on a Wink-Start or Dial-Delay circuit or after seizure on an Immediate-Dial circuit.
E&M Dial Delay Hold	140	2-500 millisec. (0.002-0.5 sec.)	This is the minimum length for a Dial-Delay handshake. Used only for Dial-Delay circuits.
E&M Dialing Wait After Hookflash	3000	2-20000 millisec. (0.002-20 sec.)	Determines how long the E&M circuit will wait when transmitting a hookflash (recall) before dialing additional digits or checking for disconnection.
E&M Disconnect Flash Duration	15000	2-40000 millisec. (0.002-40 sec.)	This is the minimum amount of time a T1 E&M or DID circuit will remain on hook to cause a disconnection from the remote circuit.
E&M Disconnect Recognition	1500	2-10000 millisec. (0.002-10 sec.)	This is the amount of time a circuit must be on hook before the E&M circuit recognizes a disconnection.
E&M False Signal Debounce	50	2-500 millisec. (0.002-0.5 sec.)	Determines the minimum length of a valid handshake signal that is received from a remote circuit. Used only on Wink-Start and Dial-Delay circuits.

Table 6-38: *System Timers (Continued)*

Timer	Default Value	Range	Purpose
E&M Handshake Timeout	5000	2-20000 millisec. (0.002-20 sec.)	Determines the maximum length of a valid handshake signal. Used only on Wink-Start and Dial-Delay circuits.
E&M Hookflash Duration	600	2-10000 millisec. (0.002-10 sec.)	Determines the length of hookflashes (recalls) sent to the remote circuits.
E&M Hookflash Recognition	300	2-10000 millisec. (0.002-10 sec.)	Determines the minimum length of recognizable hookflashes (recalls) from the remote circuit.
E&M Inpulse-Dial Interdigit Pause Recognition	300	2-10000 millisec.	This is the minimum pause allowed between incoming pulse-dial signals sent from the remote circuit. It allows the system to recognize the separation between digits.
E&M Off-Hook Debounce	10	2-500 millisec. (0.002-0.5 sec.)	This is the minimum amount of time the remote circuit must be off hook before the E&M circuit will recognize another on-hook/off-hook transition.
E&M On-Hook Debounce	10	2-500 millisec. (0.002-0.5 sec.)	This is the minimum amount of time the remote circuit must be on hook before the E&M circuit will recognize another off-hook/on-hook transition.
E&M Outpulse-Dial Interdigit Pause	700	2-10000 millisec. (0.002-10 sec.)	This is the minimum amount of time the E&M circuit will pause between pulse-dial digits when dialing.
E&M Outpulse-Dial Interpulse Pause	40	2-500 millisec. (0.002-0.5 sec.)	This is the length of time the E&M circuit will pause between pulse-dial digits when dialing.
E&M Post-Seize Delay	65	2-500 millisec. (0.002-0.5 sec.)	This is the minimum allowed time between the recognition of a seizure and the beginning of digit validation. Used only for Immediate-Dial circuits.
E&M Post-Signal Delay	30	2-500 millisec. (0.002-0.5 sec.)	This is the minimum allowed time between the end of a handshake and the beginning of digit validation. Used only for Wink-Start and Dial-Delay circuits.
E&M Pulse Hold	60	2-500 millisec. (0.002-0.5 sec.)	This is the maximum length of a pulse-dial signal that is dialed by the E&M circuit.
E&M Ready Timeout	4000	2-20000 millisec. (0.002-20 sec.)	**On a Wink-Start circuit:** This is the maximum time the E&M circuit will wait for a “Digit Register Ready” command before initiating the Wink-Start handshake. **On a Dial-Delay circuit:** This is the maximum time the E&M circuit will wait for a “Digit Register Ready” command before the Dial-Delay signal is terminated.
E&M Receive Handshake Delay	20	2-500 millisec. (0.002-0.5 sec.)	This is the minimum time required, after seizure, before an incoming handshake signal can be recognized. Used only for Wink-Start and Dial-Delay circuits. If a Wink or Dial Delay signal is detected within this time, the outgoing call is blocked.
E&M Seizure Debounce	2	2-500 millisec. (0.002-0.5 sec.)	Determines the minimum amount of time the remote circuit must remain off hook before the E&M circuit validates the incoming call.

Table 6-38: *System Timers (Continued)*

Timer	Default Value	Range	Purpose
E&M Transmit Handshake Delay	100	2-500 millisec. (0.002-0.5 sec.)	This is the minimum delay time between the recognition of a seizure and the beginning of a handshake signal. Used only for Wink-Start and Dial-Delay circuits.
E&M Wait for Dial Tone	1000	2-20000 millisec. (0.002-20 sec.)	This is how long the E&M circuit will wait for dial tone before dialing digits on an outgoing call. Only used on Immediate-Dial circuits with the Dial Tone Wait option enabled.
E&M Wink Hold	214	2-500 millisec. (0.002-0.5 sec.)	Determines the length of a Wink-Start handshake.
E&M Wink Timeout	350	2-500 millisec. (0.002-0.5 sec.)	Determines the maximum allowed length of wink signals that are received from the remote circuit. If the time limit is exceeded, the call will be blocked and the attempt terminated. Used only for Wink-Start circuits.
Forward No Answer	15	3-255 seconds	Length of time a call waits at an unavailable station before being forwarded. (Applies to manual call forwarding only, not system forwarding.)
GS Dialing Wait After Connect	30	1-50 tenths (0.1-5 sec.)	This is the length of time the system waits after a ground start trunk has been seized, to place an outgoing call, before dialing digits. (This timer is not used if the Loop Current Dialtone Detection option is selected, as described on page 6-257.)
GS Tip-Ground Debounce	50	10-500 millisec. (0.001-0.5 sec.)	This is the length of time the system will ignore subsequent attempts to seize a ground start trunk once it has been seized. **NOTE:** The database contains a station option that can be set to prevent users from reseizing a trunk. If selected, the station user cannot reseize a trunk until it is disconnected by replacing the handset, pressing the SPKR button (if off hook), or pressing another trunk button.
GS Transition Delay	10	10-500 millisec. (0.001-0.5 sec.)	This is the amount of time that must elapse after a disconnection before another call can ring in on that trunk.
Hold	60	0-255 seconds	Limits time a call remains on hold before recalling the station. If set to 0, the call will not recall.
Hold – Alternate	180	0-1000 seconds	If the station has the Alternate Hold Timer flag enabled, this limits time a call remains on hold before recalling the station. If set to 0, the call will not recall.
Inactivity Alarm	60	10-255 seconds	Limits the time a station can remain off hook and inactive (after first receiving reorder tones) before registering a system alarm.

Table 6-38: *System Timers (Continued)*

Timer	Default Value	Range	Purpose
Interdigit – Long Interdigit – Short	15 4	2-255 seconds 2-30 seconds	Used in determining end of dialing. Short timer is used after a valid number has been dialed. Long timer is used until digits form a valid number.
Loopback Timeout	600	10-10000 seconds	This is the maximum allowed length of a remote loopback test. If the timer expires, the test is automatically terminated.
LS Dialing Wait After Connect	15	1-50 tenths (0.1-5.0 sec.)	The length of time the system waits for outside dial tone before dialing or checking the trunk for a disconnect.
LS/GS Caller ID Relay Hold	3747	2-4000 millisec. (0.002-4.0 sec.)	Amount of time the system will look for Caller ID [CLIP] information when a call is received.
LS/GS Caller ID Ring Idle	128	2-4000 millisec. (0.002-4.0 sec.)	This sets the time between the end of first ring and the time at which the system begins to check for Caller ID [CLIP] information. **NOTE:** The Caller ID [CLIP] timer values combined must be shorter than the period of silence between CO rings.
LS/GS CO Hookflash	60	2-1000 hundredths (0.02-10.0 sec.)	Adjusts the duration of the timed hookflash (recall) that is sent over the trunk by the system when the Hookflash feature code is used.
LS/GS CO-CO Disconnect	35	2-1000 hundredths (0.02-10.0 sec.)	A call is disconnected by the system if it detects loss of loop current lasting longer than this timer during trunk-to-trunk calls.
LS/GS Dialing Disconnect	120	2-1000 hundredths (0.02-10.0 sec.)	The connection is dropped if the system detects loss of loop current lasting longer than this timer during dialing.
LS/GS Dialing Wait After Hookflash	30	2-199 tenths (0.2-19.9 sec.)	Delays dialing after a hookflash (recall) to allow the system and central office hardware to recover.
LS/GS IC-CO Disconnect	60	2-1000 hundredths (0.02-10.0 sec.)	A call is disconnected by the system if it detects loss of loop current (lasting longer than this timer) during station-to-trunk calls.
LS/GS Inter-Ring Silence	60	1-250 tenths (0.1-25.0 sec.)	Indicates the duration of the silence between rings on an incoming call to determine if the trunk has stopped ringing prior to being seized. In most areas, trunk ring pattern is 2 seconds on/ 4 seconds off. Check with the local telephone company for the ring pattern in your area. **NOTE:** This timer must always be set higher than the central office's ring off time.

Table 6-38: *System Timers (Continued)*

Timer	Default Value	Range	Purpose
LS/GS Loop Current Debounce	100	2-500 millisec. (0.002-0.5 sec.)	This is the minimum amount of time the system must detect loop current for it to recognize that a trunk is present when it is seized.
LS/GS Outpulse-Dial Inter-digit Pause	700	2-1500 millisec. (0.002-1.5 sec.)	This is the idle time between pulse-dial digits sent by the system.
LS/GS Outpulse-Dial Inter-pulse Pause	40	2-500 millisec. (0.002-0.5 sec.)	This is the amount of time between pulses when pulse-dial digits are sent by the system.
LS/GS Outpulse-Dial Pulse Hold Duration	60	2-500 millisec. (0.002-0.5 sec.)	This is the duration of a single pulse-dial digit sent by the system.
LS/GS Ring Frequency – High Boundary	100	max. 1000 Hertz	These parameters determine the valid range of ring frequencies that will be recognized by the system.
LS/GS Ring Frequency – Low Boundary	15	min. 4 Hertz	Any ring signal outside of this range is ignored. **NOTE:** The ranges for the Ring Frequency timers are interdependent. The minimum value for the high boundary is the current Value of the low boundary. The maximum for the low boundary is the current Value for the high boundary.
LS/GS Trunk Ring Detection	150	100–1024 ms (.1–2.5 sec)	This is a low-level timer that specifies the duration that continuous ring voltage must be detected on a trunk for the system to recognize a new incoming call. If this timer is too low, then false rings could be detected. If the timer is too high, then new incoming calls may not be detected at all. Common ring durations sent by central offices are 1-second and 2-second.
Message Wait	5	1-255 seconds	Length of time a caller waits after pressing the MSG button before being connected to the called party's message center.
Off-Hook Voice Announce Screening	5	1-255 seconds	After the Camp-On timer expires, length of time before an OHVA call can be completed.
Page	15	0-255 seconds	Limits duration of page. If set to 0, pages are unlimited in length.
Pause Dialing Digit Length	3	1-15 seconds	Length of timed pauses used in system and station speed-dial telephone numbers and in ARS dial rules.
Queue Callback	15	10-255 seconds	Time allowed for a station to respond to a queue callback before the queue is canceled.
Recall	60	10-255 seconds	Length of time a hold or transfer recall rings at a station before recalling that station's recall destination. If the station receiving the recall has no recall destination, the call remains at the station until the Abandoned Call timer expires.

Table 6-38: *System Timers (Continued)*

Timer	Default Value	Range	Purpose
Record-A-Call Tone Interval	0	0-255 seconds	The system can be programmed to send periodic tones when the Record-A-Call feature is used. This timer determines how often the tone will be sent. If you set this timer to 0, there will be a single tone when the Record-A-Call feature begins, if tones are enabled, but there will not be periodic tones through out the call.
Remote Programming Invalid Extension Failure Limit	3	0-9 failures	Determines how many times a caller will be allowed to enter an incorrect extension number while attempting remote programming, without being dropped. To program the failure limit, enter the desired number in the text box. Entering 0 will allow unlimited attempts.
Remote Programming Password Failed Limit	3	0-9 failures	Determines how many times a caller will be allowed to dial an invalid extension password while attempting remote programming, before being disconnected. To program the limit, enter the desired number in the text box. Entering 0 will allow unlimited attempts.
SL Disconnect Flash Duration	15	1-250 tenths (0.1-25.0 sec.)	When a single-line station is involved in a call that is dropped by the other party, and the single-line station remains off hook, the system turns off tip and ring battery for the duration of this timer. If there is peripheral equipment on the single-line circuit (such as a voice mail unit, page amplifier, or other phone system) then this loss of battery will signal a call disconnect and will usually cause the equipment to disconnect.
SL Hookflash Maximum	12	2-20 tenths (0.2-2.0 sec.)	The maximum amount of time a single-line set user can press the hookswitch before the system disconnects calls.
SL Hookflash Minimum	2	1-10 tenths (0.1-1.0 sec.)	The minimum length of time a single-line set user must press the hookswitch for the system to recognize a hookflash (recall).
SL Inpulse-Dial Inter-digit Pause	300	2-1500 millisec. (0.002-1.5 sec.)	The minimum pause allowed between incoming pulse-dial signals sent from the remote circuit. It allows the single-line device to recognize the separation between digits.
SL Wait For Disconnect	2	1-60 seconds	When a single-line station is involved in a call that is disconnected by the other party, and the single-line station remains off hook, this timer specifies the length of time between the disconnection and when the disconnect flash (which is set by the Disconnect Flash Duration timer) is transmitted to the single-line circuit. During this time, the station receives no audible signal.
System Forward Advance	15	2-255 seconds	Determines how long a call will ring (unanswered) at each forwarding point in a system forwarding path.

Table 6-38: *System Timers (Continued)*

Timer	Default Value	Range	Purpose
System Forward Initiate	15	2-255 seconds	Determines how long a call will ring (unanswered) at the principal station before advancing to the first forwarding point in a system forwarding path.
Transfer – Attendant	30	10-255 seconds	When an attendant transfers a call, this timer limits the time a transferred call rings unanswered before it recalls the attendant.
Transfer – Available	20	10-255 seconds	Limits time a transferred call rings unanswered before it recalls the transferring station. (Does not apply to calls transferred by attendant stations or the Voice Processor or calls transferred to a busy station.)
Transfer – Busy	24	10-255 seconds	Limits time a transferred call waits at a busy station before recalling the transferring station. (Does not apply to calls transferred by attendant stations or Voice Processor or to calls transferred to an idle station.)
Transfer – Voice Processor	20	10-255 seconds	Limits time a transferred call waits at a Voice Processor destination before recalling the programmed recall destination.
Trunk Busy Out	180	1-65535 seconds	When a trunk is made busy using the administrator DB Programming option (see page 5-289), this timer starts. When it expires (or the administrator cancels the busyout), the trunk is again made available.
Trunk Key Debounce	3	1-30 seconds	The length of time the system will ignore subsequent attempts to press a trunk button once it has been pressed. This prevents a phone user from accidentally disconnecting a call by re-selecting the trunk button if the button is pressed twice while answering a call. **NOTE:** The database contains a CO/IC Reseize station flag that can be set to prevent users from reseizing a trunk. If selected, the station user cannot reseize a trunk until it is disconnected by replacing the handset, pressing the SPKR button, or pressing another trunk button.
UCD Station Monitor Indication Frequency	15	1-255 seconds	This is the length of the pause between hunt group station monitoring tones.
Unsupervised CO	5	1-255 minutes	Limits duration of outside calls transferred or forwarded to outside telephone numbers before recalling the primary attendant.
Valid Call	15	1-60 seconds	Minimum duration of an outgoing call before it is recorded in SMDR. Calls placed on hold or transferred are not subject to this timer.

Table 6-38: *System Timers (Continued)*

Timer	Default Value	Range	Purpose
Voice Mail Dial Delay	5	1-250 tenths (0.1-2.5 sec.)	(Used for Analog Voice Mail Hunt Groups Only) When a call is answered by the voice mail unit, this indicates the amount of time the system waits before sending digits to the voice mail unit after it answers the call.

13. REPORT PROGRAMMING

13.1 Customized system programming reports can be created through the database. You can select the programming area to be reported and then select the information that will be included. The report types and their options include the following.

- **ARS Facility Group Report**: This report lists the facility groups and their dial rules and/or trunk groups/nodes.
- **ARS Route Group Report**: This route group list can include dial patterns and/or facility groups.
- **Call Routing Report**: This report shows the call routing tables.
- **Class of Service Report**: This report can include the day/night lists of stations and the dial patterns for each class of service.
- **CO Trunk Groups Report**: This trunk group list can include day/night answer access, day/night outgoing access, day/night emergency outgoing access, day/night ring in, toll restriction, and/or trunk lists.
- **Detailed Station Report**: Selected stations are listed with account codes, mailboxes, flags, port, station-related information, programmable buttons, special purpose station status, system forwarding, toll restriction, and/or voice mail information.
- **E1 Report**: This report shows information for selected E1/PRI Cards, including error statistics, error thresholds, flags, primary rate timers, and/or reference clock information.
- **General Station Report**: Selected stations can be listed by attendant, class of service, equipment, and/or special purpose station status.
- **Hunt Group Report**: Selected hunt groups are listed with their agent/member lists, supervisors, and/or timers.
- **Individual Trunk Reports**: Selected trunks are listed with their answer supervision setting, connect trunk-to-trunk call on polarity reversal, disconnect timer, DTMF signaling information, hybrid balance setting, language, loop current dial tone detection, number of digits to receive, base number, service type, start type, and/or trunk group.
- **Keymaps Report**: Selected keyset and DSS map groups and map diagrams are listed with their button assignments and/or station lists.
- **Off-Node Devices**: Lists of devices can include off-node applications, modems, IP connections, hunt groups, paging ports, paging zones, phones, and/or single-line sets.
- **Phone List**: Selected stations, hunt groups, and page zones are listed.
- **Station Flags**: Stations with the flag enabled are listed for selected station flags.
- **System Speed Dial Report**: Selected system speed-dial numbers are listed.
- **System Timers Report**: Selected system timers are listed.
- **T1 Report**: This report shows information for selected T1 Cards, including error statistics, error thresholds, flags, primary rate timers, and/or reference clock information.
- **User Group Report** *(U.S. only)*: Selected user groups are listed with day/night lists of stations, allowed area and office codes, restricted area and office codes, and/or extended area codes.

13.2 The language used to create the report depends on the country selected in Session Manager, as shown in the following table.

Table 6-39: *Report Languages*

Country	Report Language
U.S.A.	American English
U.K.	British English
Mexico	American English
Japan	Japanese English

13.3 Reports can be sent to files on the PC or to a printer port.

13.4 When extension numbers are listed in a report, they can be sorted by name, extension number or circuit number. Reports can also be assigned a Description.

13.5 To program a report:

1. Open **Report Programming** and select **Report Type** for the report you want to program.
2. Scroll to the type of report you want and press **ENTER** or select another field to save your change.
3. If desired, you can enter a title for the report in the Description field.
4. To program the report contents and options, double-click on the report. The options you have depends on the type of report being programmed. The possible fields are as follows:
   - **Devices**: Depending on the report type selected, the "devices" included in the report can be stations, hunt groups, trunks, call routing tables, ARS route groups etc. Selecting the device folder allows you to select the devices that will be included in the report.
   - **Options**: Select **Options** to view available options for each report type. For each option you want enabled, select its check box to place a check mark in it. Then press **ENTER** or select another field to save your change.
   - **Group Page Breaks**: Select this check box to place a check mark in it if you want to have each group (circuit board, map group, hunt group, etc.) on a separate page of the report. Then press **ENTER** or select another field to save your change.
   - **Print Format**: The devices included in the report can be sorted by extension number, circuit number, or name. Scroll to the desired sorting method and press **ENTER** or select another field to save your change.
5. To print the report, use the Print Report option in the Operations menu as described on page 6-82.

14. SYSTEM MANAGER PROGRAMMING

14.1 System Manager is a server-based application that centralizes management functions for the Axxess system. Each node (up to 99) within a system must physically connect to the System Manager server to allow remote management. After a connection is established, the node is considered an "agent" to System Manager. To configure the node to interface with System Manager, double-click **System Manager** (under System).

NOTE: If you have not selected a time zone yet, a warning message appears when you double-click System Manager. You must first program a time zone before any of the System Manager fields.

ALSO: System Manager's performance can suffer when connected to a network that includes 40 or more nodes. The performance degradation occurs when several systems send large amounts of Message Print data to System Manager. In an effort to eliminate this, if the system administrator plans on connecting 63 nodes or more to System Manager, a newer version of System Manager is needed. This new version supports the ability to enable or disable incoming Message Print messages.

14.2 **IP Address**: This field identifies the IP address of the System Manager server. To enter the IP address, click the current Value. The Edit IP Address dialog appears. Enter the correct IP address and click **OK**. By default, this value is *192.168.200.1*.

14.3 **Port**: This field identifies the port used to connect to the System Manager server. To change the port number, click the current Value and enter the new number. Then press **ENTER** or click another field to save your change. The valid range is 1025-65535, and the default is *3707*.

14.4 **Username**: This field indicates the username of the agent account that is programmed in System Manager. To enter the username, click the current Value and enter the username, up to 64 characters. Then press **ENTER** or click another field to save your change. This field is case-sensitive.

14.5 **Password**: This field identifies the password for the agent account that is programmed in System Manager. To enter the password, click the current Value and enter the password, up to 64 characters. Then press **ENTER** or click another field to save your change. This field is case-sensitive.

NOTE: The **Username** and **Password** fields must match the information that is programmed in System Manager. If these fields do not match the agent account information, the node will not connect to System Manager.

- **Enable Connection**: This flag determines whether the node will attempt to connect to System Manager. Set this flag to *Yes* to allow the node to connect to System Manager. If disabled, the node will not connect to System Manager. By default, this is *disabled*. To enable this flag, select its check box to place a check mark in it. Then press **ENTER** or select another field to save your change.
- **History Capacity Limit**: This field sets the size limit of the history message buffer for the CP agent before messages are sent to the CP snap-in. By default this field is set to 50.
- **Require System Manager ACK**: When this flag is set to *Yes*, the CP Snap-in sends an acknowledgement to the CP agent whenever the CP agent sends the CP Snap-in a history message. The default value for this flag is *No*.

- **Send CPH to System Manager**: When this flag is set to *Yes*, the CP agent sends history messages to the CP Snap-in. The default value for this flag is *No*.
- **Valid CA Certificate**: This field is a read-only. If set to *Yes*, this indicates that the system has a valid CA certificate loaded.

VOICE PROCESSOR PROGRAMMING

1. INTRODUCTION

1.1 When you click on **Voice Processor** in the left side of the programming window, the following programming areas are displayed on the window on the right:

- Voice Processor system-wide parameters (see page 6-344):
- Application-Related Information (see page 6-353)
- Devices (see page 6-356)
- E-Mail Gateway (see page 6-398)
- Fax-On-Demand (see page 6-401)
- Mailbox-Related Information (see page 6-406)
- Maintenance (see page 6-407)
- Networking (see page 6-411)
- Report Parameters (see page 6-418)
- SMDR Buffer (see page 6-422)
- Timers and Limits (see page 6-423)
- Unified Messaging (see page 6-430)

1.2 Each of these programming areas and fields are described in detail in the following pages.

2. VOICE PROCESSOR SYSTEM-WIDE INFORMATION

2.1 A program planning sheet that lists the Voice Processor system-wide information is included on the system software CD-ROM.

2.2 The following system-wide parameters are described in this section:

- Dial-0 Destinations (see page 6-344)
- Disk Usage Statistics (see page 6-345)
- System Administrator Mailbox (see page 6-345)
- VPIM Postmaster (see page 6-345)
- VPIM Home Domain (see page 6-346)
- Alternate Tone Detection (see page 6-346)
- Volume (see page 6-346)
- Save Message on Return Call (see page 6-346)
- Swap "7 For Save" and "9 For Delete" Message Keys (see page 6-347)
- Identification Prompt (see page 6-347)
- Enterprise Messaging System Fields (see page 6-348)
  - — Automatic Speech Recognition (ASR) Setting
  - — Automatic Speech Recognition (ASR) Enabled
- Monitor Password (see page 6-350)
- Windows Network Login (see page 6-351)

A. DIAL-0 DESTINATIONS

2.3 These fields are used for programming the dial-0 operator destination for the Voice Mail and Automated Attendant applications (where calls are sent if the caller presses 0).

2.4 **Day and Night Dial-0 Destination Type**: These drop-down list boxes are used for selecting the type of device that will be used for the day and night operator destinations. Select the desired drop-down list box and scroll to the desired type (Extension, Mailbox, or None). Then press **ENTER** or click on another field to save the change.

- If **Mailbox** is selected, the system will automatically transfer the caller to a the mailbox, specified in the Dial-0 Destination below, when 0 is pressed.
- If **None** is selected, operator access will be denied from Voice Mail and Auto Attendant applications.
- If **Extension** is selected, any station, application (including a STAR application), or hunt group extension number can be selected.

NOTE: Version 7.0 and later system software supports STAR applications as a valid Dial-0 Destination.

2.5 **Dial-0 Destination**: If day or night destination types are selected, you can select the specific device that will serve as the operator destination. To determine the Dial-0 Destination, do the following:

- Right-click on the existing value. An option box appears.
- Select **Change Dial-0 Destination**. A window appears that asks you to select the type of device to include.
- Select the desired type of device and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
- Highlight the desired destination and then select **Finish**. Your selection appears in the list.

B. DISK USAGE STATISTICS

2.6 These fields show the status of the Voice Processing PC disk storage. The information is presented for reference only and cannot be programmed.

2.7 The statistics include the following information:

- **Total Storage on the System**: This is the amount of disk space available on the Voice Processing PC.
- **Total Storage Used**: This is the disk space that has been used by the Voice Processing applications.
- **Total Storage Remaining**: This is the amount of disk space remaining in the Voice Processing PC. If the disk is nearly full, you can increase disk space by clearing the SMDR buffer or mailbox messages. If available disk space is frequently low, a larger-capacity disk drive should be installed in the Voice Processing PC.

C. SYSTEM ADMINISTRATOR MAILBOX

2.8 A System Administrator's mailbox is defined in order to record custom Audiotex recordings for Voice Processing applications created in Applications Programming, broadcast messages to all subscribers, and perform Mailbox and Group List maintenance. To assign the System Administrator Mailbox, do the following:

1. Right-click on the existing value. An option box appears.
2. Select **Change System Administrator Mailbox**. A window appears that asks you to select the type of device to include.
3. Select **Mailbox** and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the desired mailbox and then select **Finish**. Your selection appears in the System Administrator Mailbox field.

D. VPIM POSTMASTER

2.9 Type the configured mailbox on the local system. The Postmaster mailbox is similar to the Administrator mailbox. The difference is that the Administrator mailbox receives messages relative to delivery problems associated with normal, local message delivery and the VPIM (Voice Profile for Internet Mail) Postmaster mailbox receives messages relative to delivery problems associated with VPIM message delivery. The Administrator mailbox and the VPIM Postmaster mailbox can be configured to be the same mailbox. For complete information about VPIM, see page 4-71.

E. VPIM HOME DOMAIN

2.10 Type the domain that identifies the local system. For example, the following VPIM address "Doe, John <1000@localDomain.com>" has a VPIM Home Domain of "localDomain.com." When a VPIM message is sent out from the local system, the "From" address contains the VPIM Home Domain. When other VPIM systems send messages to the local system, the "To" address contains an address with the domain being the VPIM Home Domain on the local system. For complete information about VPIM, see page 4-71.

F. ALTERNATE TONE DETECTION

2.11 In some remote messaging situations, the device being called uses signaling tones that the Voice Processor does not recognize, at default. If remote messaging is not properly signaling the device, enable this flag to allow the Voice Processor to detect alternate tones.

2.12 To enable the Alternate Tone Detection option, select the current Value and place a check mark in it. Then press **ENTER** or click on another field to save the change. To disable the option, select the check box again to remove the check mark.

G. VOLUME

2.13 This field allows you to set the playback volume level for all prompts, voice mail messages, and Audiotex recordings simultaneously. These areas cannot be adjusted separately and only the playback value is affected, not the recording volume. The volume range is -8 for the softest setting to +8 for the loudest. The normal setting is 0.

NOTE: When a caller, using voice mail, increases or decreases the volume during the call, the system volume level currently programmed does not change. Only the voice channel being used by the caller is temporarily altered. When the user has completed the call, the system resets the volume of the voice channel used to the setting currently established in the Volume Level drop-down list box.

2.14 To set the volume level, click on the current Value and scroll to or enter the new value in the text box. Then press **ENTER** or click on another field to save your change.

H. SAVE MESSAGE ON RETURN CALL

2.15 The Return Call feature (described on page 4-46 in VOICE PROCESSING FEATURES) allows the voice mail user to place a return call to the message sender directly from the voice mailbox for a message that has a Caller ID or extension number.

2.16 The Save Message on Returned Call flag tells the Voice Processor whether to save the message that the user is listening to at the time the user chooses to make the return call.

- If the flag is set to "yes," the Voice Processor will automatically move the message to the user's saved message queue when the user makes the return call. If the message is already in the saved message queue when the user makes the return call, then it is unchanged. Note that this applies to both the new message queue and deleted messages. If the flag is enabled and the user makes the return call from a deleted message, that message will be moved to the saved message queue, effectively restoring the message.
- If the flag is set to "no," the Voice Processor will simply leave the message in the same queue it was in before the user made the return call. Note that if the flag is set to "no" and the message is in the new message queue, the Voice Processor will refresh the message lamp on the user phone.

2.17 To enable the Save Message On Return Call option, select the check box to place a check mark in it. Then press **ENTER** or click on another field to save the change. To disable the option, select the check box again to remove the check mark.

I. SWAP "7 FOR SAVE" AND "9 FOR DELETE" MESSAGE KEYS

2.18 Enterprise Messaging v2.1 or later provides a swap feature whereby the 7 and 9 digit operations in Message Options can be reversed. By default, the 7 digit is used for SAVE and the 9 digit is used for DELETE. The swap feature allows the 7 digit to be used for DELETE and the 9 digit to be used for SAVE, just like the functionality operated by the cell phone voice mail systems.

2.19 This helps users who are familiar with a cell phone functionality from accidentally deleting messages by pressing 9, intending to save messages. This feature can be programmed for individual mailboxes by the subscribers or for all mailboxes using this option. For complete information about 7-9 Key Swap, see page 4-70.

2.20 To swap the keys for all mailboxes:

1. Select **Voice Processor**.
2. Set the Swap "7 For Save" and "For Delete" Message Keys option to **Yes**.

NOTE: This field only applies for EM v2.1 or later only. For other types of Voice Processor, this field appears with a red "X."

J. IDENTIFICATION PROMPT

2.21 The Identification Prompt flag enables/disables a custom message that identifies the voice processing system as Inter-Tel's Voice Processing Unit. When *enabled*, this prompt is heard before the voice mail greeting and states, *"Your call is being handled by the Inter-Tel voice processing system."* This prompt will automatically play when **outside callers**:

- Are transferred to a voice mailbox. The ID prompt is played **BEFORE** the mailbox greeting (primary, alternate, or system) is heard.
- Are forwarded to a voice mailbox. The ID prompt is played **BEFORE** the mailbox greeting (primary, alternate, or system) is heard.

2.22 If *disabled*, the identification prompt is not played before the voice mailbox greeting when an **outside caller** accesses a mailbox.

2.23 By default, this flag is *enabled*. To disable the Identification Prompt, select the check box to remove the check mark. Then press **ENTER** or click on another field to save the change. To enable the option, select the check box again to place the check mark in the box.

K. ENTERPRISE MESSAGING SYSTEM FIELDS

2.24 The EM platform uses additional fields that are required for automatic speech recognition (ASR) and command/event information.

Programming the Automatic Speech Recognition (ASR) Settings

2.25 The new EM voice processing system supports Automatic Speech Recognition (ASR). This feature allows you to access voice processing applications, such as voice mail, by speaking menu options instead of dialing digits on your dialpad. Although you can enable this feature at a system, application, or mailbox level, if it is enabled at a top-level folder, that setting overrides any subfolder programming. For example, if you enable the feature at the system level, the feature is automatically enabled at the application and mailbox levels. If you enable the feature at the application level only, the feature is available for the programmed applications and mailboxes only. If you enable the feature at the mailbox level only, the feature is available for mailboxes only.

2.26 To enable ASR, program the following fields:

- **ASR Setting**: Determines whether the system will use the ASR Enabled setting in the current folder level or in the next folder level. Depending on the folder level, you may have one or more of the following options:
  - — *Active: (system, applications, and mailboxes)* Indicates that the system will use the ASR Enabled setting for the current folder. If selected, any subfolders that have the ASR options automatically use the ASR Enabled setting for this folder. This is the default for all folder levels.
  - — *Delegated: (system and applications only)* Indicates that the system will use the ASR Enabled setting for the next folder level.
  - — *Ignored: (applications and mailboxes only)* Indicates that the system will ignore the ASR Enabled setting for this folder. This folder will then use the settings programmed for the parent level. For example, mailboxes will use the setting specified for the application, and applications will use the setting specified for the system.
  - — *Ignored and Delegated: (applications only)* Indicates that the system will ignore the ASR Enabled setting for this folder and use the parent level folder (that is, the Voice Processor folder). If, however, the Voice Processor folder ASR Setting field is set to Delegated, this folder will use the settings programmed for the next folder level (that is, the mailbox folder).
- **ASR Enabled**: Determines whether or not ASR is enabled for the current folder (and any subfolders or parent folders based on the ASR Setting field). If enabled, the system, application, or mailbox supports voice recognition for accessing that particular feature. If disabled, the system, application, or mailbox will not support voice recognition.

NOTE: The ASR Enabled field is ignored (that is, displays a red “x”) if the ASR Setting is set to Delegated, Ignored, or Ignored and Delegated.

2.27 The following table outlines some of the possible combinations you can configure.

Table 6-40: *ASR Table*

Voice Processor	Application	Mailbox	Results
ASR Setting: Active **ASR Enabled:** Yes	N/A (settings are ignored)	N/A (settings are ignored)	ASR is enabled for all applications and mailboxes.
ASR Setting: Active **ASR Enabled:** No	N/A (settings are ignored)	N/A (settings are ignored)	ASR is disabled for all applications and mailboxes.
ASR Setting: Delegated **ASR Enabled:** N/A (ignored)	**ASR Setting:** Active **ASR Enabled:** Yes	N/A (settings are ignored)	ASR is enabled for all applications and mailboxes.
ASR Setting: Delegated **ASR Enabled:** N/A (ignored)	**ASR Setting:** Active **ASR Enabled:** No	N/A (settings are ignored)	ASR is disabled for all applications and mailboxes.
ASR Setting: Delegated **ASR Enabled:** N/A (ignored)	**ASR Setting:** Ignored **ASR Enabled:** N/A (ignored)	N/A (settings are ignored)	ASR is disabled for all applications and mailboxes.
ASR Setting: Delegated **ASR Enabled:** N/A (ignored)	**ASR Setting:** Delegated **ASR Enabled:** N/A (ignored)	**ASR Setting:** Active **ASR Enabled:** Yes	ASR is enabled for all applications and mailboxes.
ASR Setting: Delegated **ASR Enabled:** N/A (ignored)	**ASR Setting:** Delegated **ASR Enabled:** N/A (ignored)	**ASR Setting:** Active **ASR Enabled:** No	ASR is disabled for all applications and mailboxes.
ASR Setting: Delegated **ASR Enabled:** N/A (ignored)	**ASR Setting:** Ignored and Delegated **ASR Enabled:** N/A (ignored)	**ASR Setting:** Active **ASR Enabled:** Yes	ASR is enabled for mailboxes only.
ASR Setting: Delegated **ASR Enabled:** N/A (ignored)	**ASR Setting:** Ignored and Delegated **ASR Enabled:** N/A (ignored)	**ASR Setting:** Active **ASR Enabled:** No	ASR is disabled for all applications and mailboxes.

2.28 As the table above indicates, there are several ways to enable or disable ASR. The important thing to remember is that the system determines the setting in a top-down approach.

So, if the ASR Setting is set to Active for any of the upper-level folders, the ASR Enabled value is applied to that level and any subfolders below it.

Programming Command and Event Ports

2.29 The EM system supports an Operations, Administration, and Management (OAM) interface that allows you to send commands to and receive responses from the system. To use this interface, you must program the appropriate ports in DB Programming. These ports include the following, which are located under Voice Processor:

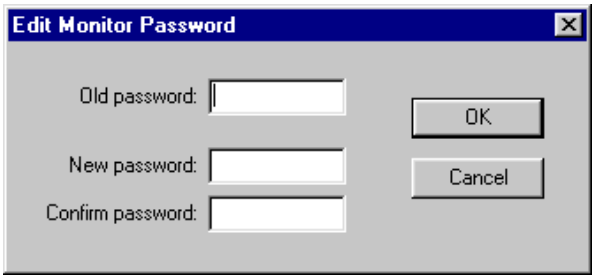
- **Management Command Port**: Enter the port number that the system will use for receiving commands and sending responses. The valid range is 0 - 65535, and the default is *4444*.
- **Management Event Port**: Enter the port number that the system will use for sending event messages to the connected console. The valid range is 0 - 65535, and the default is *4445*.

NOTE: Make sure these ports are open on firewalls/routers if the PC issuing these commands is located outside of the current subnet.

L. MONITOR PASSWORD

2.30 To protect the Voice Processor against unauthorized access, a password can be required for the AVDAPMon utility described on page 3-231. To program the password right-click on **Monitor Password** and select **Edit Password**. A window appears as shown below.

1. Enter the current password, if one exists.
2. Enter the new password (up to 40 characters) in the New Password box. The characters will not appear on the screen when typed; they will appear as asterisks (***).
3. Retype the password exactly as before in the Confirm Password box.
4. Select **OK** to exit and save the password. If the entered passwords match, you will return to the Password field. If not, you must re-enter the new password and verify it again. If you make a mistake while entering the password or want to leave it unchanged, select **Cancel**.



NOTE: To provide system security, the e-mail system must have a password. To make the passwords difficult to guess, they should not consist of predictable patterns, such as one digit repeated several times.

M. WINDOWS NETWORK LOGIN

2.31 The following information is used by the Voice Processor service when it logs on to the network. See page 3-230 for information about setting up the Avdap user account.

NOTE: Axxess version 4.3 used Windows NT-supplied encryption libraries for storing the network domain, username, and password. Unfortunately, Microsoft is not guaranteeing compatibility with new releases of their software. This renders information encrypted under one release unreadable under the next release. To avoid this problem, Inter-Tel has completely discarded Microsoft's encryption scheme in system versions 4.4 and later. Therefore, because version 4.3 Voice Processing databases use the Microsoft encryption scheme, any upgrade from 4.3 to a later version will require the installer to re-enter these three fields of the network account information. However, once the Voice Processing Unit is upgraded from 4.3 and the network information is re-entered and saved using Inter-Tel's encryption scheme, any further system upgrades will automatically update these fields.

ALSO: This section is not applicable to Enterprise Messaging.

Windows Networking Username

2.32 To change the username, click once on the current Value. When the text box appears, type the username that the Avdap service user account will use to log on to the network in the text box. Then press **ENTER** or click on another field to save the change.

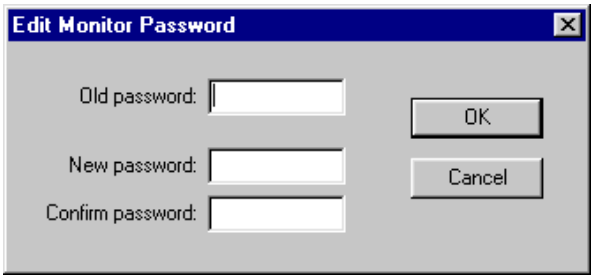
Windows Networking Domain

2.33 In this field, enter the name of the network domain in which the Voice Processing PC is a member. To change the domain name, click once on the current Value. When the text box appears, type the name in the text box. Then press **ENTER** or click on another field to save the change.

Windows Networking Password

2.34 This field specifies the network password for the Avdap service user account. Right-click on **Windows Networking Password** and select **Edit Password**. A window appears as shown below.

1. Enter the current password, if one exists.
2. Enter the new password (up to 40 characters) in the New Password box. The characters will not appear on the screen when typed; they will appear as asterisks (***).
3. Retype the password exactly as before in the Confirm Password box.
4. Select **OK** to exit and save the password. If the entered passwords match, you will return to the Password field. If not, you must re-enter the new password and verify it again. If you make a mistake while entering the password or want to leave it unchanged, select **Cancel**.



2.35 If Voice Processing is unable to log on to the network, a message is automatically entered into the Windows System Event Log. Because Voice Processing tries repeatedly to log in to restore a network connection, if the network is down for a long time (or Voice Processing is running in an environment without a network), many entries will be entered into the Event Log. Eventually, the log fills up and warning messages start appearing on the PC screen. Use the following procedure to change the setting of the Event Log.

1. Select Start - Programs - Administrative Tools (Common) - **Event Viewer**.
2. Choose **Log Settings** in the Log drop-down menu.
3. In the Event Log Settings screen, make sure that Event Log Wrapping is set to **Overwrite Events as Needed**.
4. Select **OK** to save your change and exit.

3. APPLICATION-RELATED INFORMATION

3.1 Voice Processor Application-Related Information includes the following areas:

- Audiotex Recordings (see below)
- Time Slot Groups (see page 6-354)
- Voice Mail Information (see page 6-354)
- Auto Attendant Information (see page 6-355)

3.2 A program planning sheet that lists the Application-Related fields is included on the software CD-ROM, Documentation folder.

A. AUDIOTEX RECORDINGS

3.3 *For VPUs,* this programming area allows you to assign a description to each Audiotex recording. It also allows you to view the length of each recording. *For EM units,* this programming area allows you to view the recording number and assign a description to each Audiotex recording that has been re-recorded. It also allows you to view the length of each recording. Note that DB Studio only displays recording information (recording number, description, and length) for those Audiotex recordings that have already been re-recorded. See page 4-58 for more information about creating and changing Audiotex recordings.

3.4 This programming area allows you to assign a description to each Audiotex recording. It also allows you to view the length of each recording.

3.5 To enter the Audiotex Recordings programming area, double-click on **Audiotex Recording** or right-click and then choose **Explore Audiotex Recordings**.

3.6 **Description**: To change the description of a recording, click once on the current description. When the text box appears, type the new description, up to 20 characters. Then press **ENTER** or click on another recording to save the change.

NOTE: If desired, you can use Cut, Paste, Copy, or Delete to modify the descriptions. When the current description is highlighted in the text box, simply right-click and select the desired operation.

3.7 **Recording Length**: Recording length is shown in seconds. This is presented for reference only. It cannot be changed here. It will change only if the recording is re-recorded by the System Administrator.

B. TIME SLOT GROUPS

3.8 Each application is assigned to a "Time Slot Group" which determines the maximum number of voice channels (up to 32) that will be used by the applications in that group. Voice channels are used for processing calls between the System and the applications.

3.9 The voice channel limit for the time slot group may exceed the number of voice channels actually provided by the hardware, due to programmed limits set higher than the hardware limits and/or heavy system traffic. For example, if an Automated Attendant application is assigned to a time slot group that has a programmed limit of five voice channels, it can normally support five simultaneous transfers of outside calls to extensions. However, if only four voice channels are available (because all other channels are in use by other applications or the hardware only supports four channels), a fifth call cannot be completed to that (or any other) application. When all voice channels are busy, intercom callers will hear reorder tone and see a CALL CANNOT BE COMPLETED display and outside callers will hear ringing but their calls will not be answered. (Call Routing Announcement digit translation nodes do no have assigned time slot groups.)

NOTE: Time slot group and voice channel programming should be completed before applications are created and programmed.

3.10 The list shows the time slot groups and the number of channels currently assigned to the groups.

3.11 **Description**: To change the description of a time slot group, click once on the current description. When the text box appears, type the new description, up to 20 characters. Then press **ENTER** or click on another time slot group to save the change.

3.12 **Maximum Channel Allocation**: This determines the maximum number of voice channels that will be used by the application(s) that will be assigned to this time slot group. By default, this text box will show the maximum available voice channels, as determined by the hardware configuration. Enter the maximum number of voice channels that will be used by the application(s) that will be assigned to this time slot group.

NOTE: The combined total of voice channels assigned to all time slot groups may exceed the number of voice channels provided by the hardware. This is allowed because it is not likely that all time slot groups will be busy at once. However, an individual time slot group is limited to the Maximum Channel Allocation. This parameter is used for setting the limit of voice channels that will be used by the associated application.

C. VOICE MAIL INFORMATION

3.13 Use these fields to enable or disable the voice mail directory and determine the directory sort order.

- **Enable Voice Mail Directory**: If this flag is enabled, the system allows anyone routed or answered by voice mail or any mailbox users while in their mailbox to be presented with the directory option. If the flag is disabled, the voice mail will not prompt the caller for the directory option while in the voice mail areas (non AA or CRA). Select **Enable Voice Mail Directory** and place a check mark in the box to enable the voice mail directory. Then press **ENTER** or click outside the field to save your change. To disable the directory, select the check box again to remove the check mark.

NOTE: If the voice mail mailbox directory is disabled, callers using the voice mail system will not receive a system prompt giving the option to search the directory for the person to whom they want to speak. If the dial pad button normally associated with access to the mailbox directory is selected, the caller will be informed that the selection is invalid.

- **Directory Sort Order**: This option is only for Voice Mail calls. Select **Directory Sort Order** and scroll to the desired option (last name or first name) to determine the directory sorting order. Then press **ENTER** or click outside the field to save your change. This parameter also defines which system voice prompt will play when directing callers to spell a first or last name.

D. AUTO ATTENDANT INFORMATION

3.14 You can enable or disable the automated attendant directory and determine whether the directory is sorted by first or last name using these fields. You can also determine whether the system automated attendant transfer prompt will play when calls are transferred to stations without mailboxes or extension IDs.

- **Enable Auto Attendant Directory**: If this flag is enabled, the system allows anyone routed to an Auto Attendant application to access the company directory. If the flag is disabled, the Auto Attendant will not prompt the caller for the directory option while in the Auto Attendant area. To enable the directory, select the value and place a check mark in the box. Then press **ENTER** or click outside the field to save your change. To disable the directory, select it again to remove the check mark.
- **Auto Attendant Transfer Prompt**: Determines if the transfer prompt ("Please hold while your call is being transferred to...") plays after a caller has entered an extension number that does not have an associated mailbox or extension ID. This applies to transfers from Call Routing Announcement applications that use the "Transfer To Extension" action (as described on page 6-361). To disable the prompt, select the value and click on the check box to remove the check mark. Then press **ENTER** or click outside the field to save your change. To enable the prompt, select the box again to replace the check mark.
- **Directory Sort Order (Last Name or First Name)**: This option is only for Auto Attendant calls. To determine whether the mailbox and extension ID descriptions in the directory will be sorted by first name or last name, select the drop-down list and scroll to the desired option. Then press **ENTER** or click outside the field to save your change.
- **Transfer Method (Unannounced, Announced-Only, or Screened)**: Determines how transfers will be made to other applications from Call Routing Announcement applications. The options are Announce Only, Screened, and Unannounced (defaults to Unannounced). See page 5-211 for a definition of these terms. Select the drop-down list and scroll to the desired transfer method. Then press **ENTER** or click outside the field to save your change.

4. DEVICES

4.1 Voice Processor Devices programming information includes the following. Each of these programming areas and their fields are described in the following pages:

- Applications (see page 6-356)
- Digit Translation Nodes (see page 6-371)
- Extension IDs (see page 6-373)
- Group Lists (see page 6-375)
- Mailboxes (Off-Node And Local) (see page 6-376)
- Voice Processor Nodes (see page 6-394)

A. APPLICATIONS

4.2 Applications can include the following (see VOICE PROCESSING FEATURES for more information regarding each application type):

- **Automated Attendant**: The automated attendant is a programmable feature that can be used to provide automated call answering service. Calls can be transferred, forwarded, or can directly ring in to an Automated Attendant application. When an automated attendant answers a call, it plays a recording that gives dialing instructions. After hearing the recording, the caller may then directly dial a station extension number, voice mail application number, or hunt group pilot number.
- **Automated Attendant Recall**: If a call that was transferred by the Automated Attendant application is not answered before the Transfer Voice Processor timer expires, the call recalls the Automated Attendant Recall application. The recall destination announces that the station is unavailable and allows the caller to choose to leave a message (if a mailbox is programmed for that station) or dial another extension number.
- **Call Routing Announcement**: Call Routing Announcements can be used two ways:
  - A Call Routing Announcement application can be used in place of a playback device. The playback device function is especially useful for programming hunt group announcement and overflow stations. When called, the Call Routing Announcement application will play a recording and then hang up.
  - The Call Routing Announcement application can use Digit Translation which allows the caller to press a single digit for access to a mailbox, a Fax-On-Demand function, or to a station or hunt group that has an associated mailbox or extension ID. Digit translation can be programmed for each digit 0-9, #, and *, plus a Timeout that is used when the caller does not enter a digit. Each digit can lead to a "digit translation node" that has its own digit translation values. This layered Call Routing Announcement digit translation creates a "tree" of programmable digit translation nodes. There can be up to 200 digit translation nodes in the system and up to 20 per Call Routing Announcement application.
- **Message Notification/Retrieval**: Only one Message Notification/Retrieval application may be created for the system and it ***must*** be created in order to allow voice mail message notification and quick mailbox access.

- **Record-A-Call**: This feature allows a station user to record an ongoing call and place it in a voice mailbox. When a station user enters the Record-A-Call feature code, the system places a call to the station's assigned Record-A-Call application using the station's Record-A-Call mailbox. When the application answers, the system sets up a conference call. If programmed, the mailbox plays a greeting to indicate that the recording is in progress. There can be separate greetings for day and night modes.
- **Scheduled Time-Based Application Routing (STAR)**: STAR allows you to have applications with alternate greetings and/or different programming set up for holidays, weekends, and other scheduled events.
- **Voice Mail**: This application handles all calls that are directed to voice mail (other than to the Message Notification/Retrieval application) by subscribers and non-subscribers. Callers will hear the main company greeting, followed by a menu of available options. Stations can forward or transfer calls directly to their mailbox using this application's extension number.

4.3 Up to 150 Voice Processor applications can be programmed (however, there can be only one Message Notification/Retrieval application). In the default database, extension numbers 2500-2649 are reserved for applications, but any available extension number can be used.

NOTE: Time slot group and voice channel programming should be completed before applications are created and programmed.

4.4 A program planning sheet that lists the Voice Processor Devices—Applications fields in included on the system software CD-ROM, Documentation folder.

4.5 When you select Applications, a list of the currently existing applications appears in the right side of the window.

4.6 **To create a new application:** Right-click in any blank area of the right side of the window (if there is no blank space, press **CTRL+N**). You will see a list of options that allow you to create applications.

4.7 After you create an application, you cannot change the application type. You can, however, delete it by highlighting it, right-clicking, and selecting **DELETE**.

4.8 **You can program the following for each application**:

- **Extension**: You can change the extension number of the application by highlighting the current number and either scrolling to a new number or type the new number in the text box. Then press **ENTER** or click outside the field to save your change.
- **Description**: The description for an application appears wherever the application is listed, as well as in the system directory. You can enter a description of up to 20 characters in this field. Then press **ENTER** or click outside the field to save your change.
- **Username**: The username for the application will appear on phone displays when the application is used. You can enter a username of up to 10 characters in this field. Then press **ENTER** or click outside the field to save your change.

4.9 Batch Extension Change: To change several extensions at once, highlight the applications you want to change. Then right-click and select **Batch Extension Change**. When the Get Extension screen appears, select the number you want to assign to the first highlighted feature code (the other highlighted codes will be numbered consecutively after this number). When you select **OK**, the extensions are automatically renumbered and re-sorted in the list.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items. (You may have to continue to hold **SHIFT** or **CTRL** while right-clicking to display the option without changing the highlighted selections.)

4.10 Copy and Paste Application Attributes: The copy and paste operations in DB Studio allow you to copy certain attributes from one application and paste the attributes into several other like applications. This allows you to quickly program several like-devices at one time. To copy an application, right-click on the application extension and then select **Copy**. To paste the programming information into another application, right-click the application where you want the information pasted. Then select **Paste**. The Copy dialog box allows you to select the attributes you want to copy. Select the desired attributes and click **OK**. The copy and paste operation is supported for the following applications:

- Auto Attendant
- Auto Attendant Recall
- Message Notification Retrieval
- Call Routing Announcement
- Record-A-Call
- STAR
- Voice Mail

4.11 To program the parameters for an application: Double-click on the application. The fields that are displayed depend on the type of application being programmed. The chart on the next page shows a summary of the programming required for each application type. See page 6-356 for programming details.

Table 6-41: *Application Parameters*

Application Type	Required Programming
Automated Attendant (AA)	Needs a description and username, attendant, time slot, transfer recall destination (usually the Automated attendant Recall application), music-on-hold information, and ASR settings. The greetings can be changed or you can use the default greetings. (May also require Extension ID programming for stations that do not have mailboxes.)
Automated Attendant Recall (AAR)	*Used by the Automated Attendant application(s).* Needs a description and username, attendant, time slot, transfer recall destination, music-on-hold information, and ASR settings. Does not use a greeting.

Table 6-41: *Application Parameters (Continued)*

Application Type	Required Programming
Call Routing Announcements (CRA)	Needs a description and username, attendant, time slot, transfer recall destination (can be a station, hunt group, or another local or off-node Call Routing Announcement application), music-on-hold information, and ASR settings. It can also have digit translation, including digit translation nodes. By default it uses the Voice Mail day and night greetings, but these can be changed. The greeting can include pauses and can announce queue position and/or time to wait. The digit translations may also require Audiotex recordings (greetings). If using Fax-On-Demand the application requires a fax delivery destination. If in a network, it can be programmed to accept AMIS calls.
Message Notification/ Retrieval (MNR) (see page 6-369 for programming details)	*Required for message notification, Remote Messaging, and Fax-On-Demand.* Needs a description and username, attendant, time slot, transfer recall destination, toll restriction class of service, music-on-hold information, and ASR settings. Does not use a greeting. Also needs to be programmed for outgoing access in Trunk Group Programming.
Record-A-Call (RAC) (see page 6-369 for programming details)	Needs a description, username, time slot, music-on-hold information, and ASR settings. By default it uses the Voice Mail day and night greetings, but these can be changed.
STAR (see page 6-369 for programming details)	Needs a description, username, default application, time slot, music-on-hold information, and ASR settings. Each of the schedules within the STAR application require: description, application, specific date(s), start and stop date, days of the week, specific time(s), start and stop time, and day/night mode.
Voice Mail (VM)	*Required for subscriber and non-subscriber access to voice mailboxes.* Needs a description and username, attendant, time slot, transfer recall destination, music-on-hold information, and ASR settings. The day and night greetings can be changed or you can use the default greetings. Additional programming required includes: System Administrator mailbox and Subscriber mailboxes.

Programming VM Applications

4.12 Double-click on the application you want to program and its programming options appear. Each Attendant requires the following programming.

Table 6-42: *Voice Mail Application Programming Fields*

Field Names	Ref.	AA[1]	AAR	CRA	MNR	RAC	STAR	VM
Digit Translation	6-360			✓				
Day and Night Greetings	6-364	✓		✓		✓		✓
Accept AMIS Calls	6-365			✓				
Class of Service	6-365				✓			
Attendant	6-366	✓	✓					✓
Fax Delivery Destination	6-366			✓				
Fax Delivery E-mail Address	6-366			✓				
Default Application	6-366						✓	
Audio for Calls Camped onto this Device	6-367	✓	✓	✓	✓	✓	✓	✓
Audio for Calls Holding for this Device		✓	✓	✓	✓	✓	✓	✓
Audio for Calls Ringing this Device		✓	✓	✓	✓	✓	✓	✓
Time Slot Group	6-367	✓	✓	✓	✓	✓	✓	✓
Transfer Recall Destination	6-368	✓	✓	✓	✓			✓
Automatic Speech Recognition (ASR) Setting	6-368	✓	✓	✓	✓	✓	✓	✓
ASR Enabled	6-368	✓	✓	✓	✓	✓	✓	✓
Propagate Original Caller ID on Transfer	6-369	✓	✓	✓	✓	✓	✓	✓
Calling Party Name	6-369	✓	✓	✓	✓	✓	✓	✓
Calling Party Number	6-369	✓	✓	✓	✓	✓	✓	✓

1. Your Automated Attendant application may also require Extension ID programming for stations that do not have mailboxes.

4.13 For application-specific programming information, see the following pages:

- MNR (see page 6-369)
- RAC (see page 6-369)
- STAR (see page 6-369)

Digit Translation

4.14 Digit translation allows callers to dial a single digit to select a designated extension number, modem, mailbox, or hunt group pilot number. Digit translation can be programmed for each digit 0-9, #, and *, plus a Timeout that is used when the caller does not enter a digit. Each digit can lead to a "digit translation node" that has its own digit translation actions. This layered Call Routing Announcement digit translation creates a "tree" of programmable digit translation nodes. For a detailed explanation of Call Routing Announcement digit translation nodes and a sample tree, see page 4-21 in VOICE PROCESSING FEATURES.

4.15 After a digit translation node is created, it can be assigned to more than one Call Routing Announcement application. This allows entire node hierarchies to be shared or moved without reprogramming. When using a Call Routing Announcement application with digit translation, the individual Voice Processing applications assigned to the digits can be programmed to override the device language and provide prompts in one language only.

4.16 To program digit translations for the Call Routing Application, double-click on **Digit Translations**. A list of the digits and their actions and destinations will appear. To program each digit use the following fields:

- **Name**: This is the digit that the caller presses. Timeout represents no digits being entered.
- **Action**: Scroll through the drop-down list box to the desired Action for the digit being programmed. The Actions that can be selected for the digits (*, #, 0-9) and for the Timeout option are as follows:
  - — **Cancel Fax Selections**: This is usually assigned to the digit *. It is used for canceling fax document selections that a caller has entered.
  - — **Company Directory - First Name**: This sends the caller to the directory prompt that asks the caller to enter the first name of the desired party.
  - — **Company Directory - Last Name**: This sends the caller to the directory prompt that asks the caller to enter the last name of the desired party.
  - — **End Fax Selections**: This is usually assigned to the digit #. It is required for signaling the system that the caller has entered all desired fax document selections (either through a Select Fax Document By Number or a Select Fax Document Action). You will see a warning message when you exit if you have programmed a Select Fax without creating an End Fax Selections action.
  - — **Invalid**: (Not available for Timeout.) The digit will not be used. Callers who press this digit hear a recording that tells them that it is invalid.
  - — **Select Fax Document**: (Not available for Timeout.) This selects a specific document for faxing. Callers should be prompted to select this digit with a recording that tells them the name of the document. For example, "Press 2 for a price list." This must be accompanied by an End Fax Selection and should also be accompanied by a Cancel Fax Selection Action.
  - — **Select Fax By Document Number**: (Not available for Timeout, or digits * or #.) This is like the Transfer to Collected Extension described below. It allows the caller to enter the exact four-digit number(s) of the desired document(s). Assign this Action to the digit(s) that correspond to the first digits of the document numbers. For example, if you have documents numbered 0001-1000, digits 0 and 1 should have a Select Fax By Document Number Action. Callers can then dial four-digit document numbers that begin with 0 or 1.

    NOTE: When document numbers begin with 0, the caller can select documents by entering the number without 0 and pressing # (but the Select Fax By Document Number action must be assigned to the digit 0). For example, the caller can enter 99# for document number 0099. This must be accompanied by an End Fax Selection and should also be accompanied by a Cancel Fax Selection Action.

  - — **Subscriber Access**: This sends the caller to the Voice Mail application that prompts the caller to enter a mailbox number.

— **Transfer To Collected Extension**: (Cannot be used for Timeout, *, #, or 0) To allow callers to dial extension numbers of stations and hunt groups (including off-node devices) that have a mailbox or extension ID, use this Action for digits that correspond to the first digits of extension numbers. For example, if digit 1 is "Transfer To Collected Extension," callers can dial extension numbers that begin with 1. (However, if digit 1 is "Transfer to Extension 2000," as described on the next page, a caller attempting to dial a station extension number that begins with 1 will instead be transferred to 2000.)

NOTE: If a caller dials an extension number of a mailbox that does not have an associated station, the call will be sent to the mailbox instead of an extension.

— **Transfer To Extension**: This sends the call to the extension (station, hunt group, application, or off-node device) that appears in the Transfer Destination field.

— **Transfer To Mailbox**: This sends the call to the designated mailbox.

— **Transfer To Node**: (Not available for Timeout.) This sends the call to a digit translation node that allows access to further digit translation options.

— **Transfer To Operator**: This transfers the call to the programmed Dial-0 Destination. (See page 6-344.)

NOTE: If you select Transfer To Operator and there is not a programmed Dial-0 Destination (see page 6-344), you will see a warning message and must program the Dial-0 Destination.

— **Hang Up**: (This is used for Timeout only.) This disconnects from the call if the user does not enter a digit.

- **Transfer Destination**: The actions Select Fax Document, Transfer To Extension, Transfer To Mailbox, and Transfer To Node each require a Transfer Destination. See the chart below to determine the type of destination, if any, to be programmed for the digit translation.

Table 6-43: *Transfer Destination*

Action	Transfer Destination
Cancel Fax Selections	None
Company Directory - First Name	None
Company Directory - Last Name	None
End Fax Selections	None
Invalid	None
Select Fax By Document Number	None
Select Fax Document	Document number
Subscriber Access	None
Transfer To Collected Extension	None
Transfer To Extension	Extension number
Transfer To Mailbox	Mailbox number
Transfer To Node	Digit translation node number
Transfer To Operator	None
Hang Up	None

- **Override Language**: If this flag is enabled, the Call Routing Announcement application will use the application's language (indicated below) for voice prompts. If the flag is disabled, the calling station's or trunk's programmed language will be used whenever the application receives a call. By default, this flag is disabled. To enable it, select it to place a check in the check box. To disable it, select the check box again.
- **Language**: If Override Language is enabled, you can select this drop-down list box to scroll to the desired language. This field now can be set to any specific language so that the system can support more than two languages. The language choices are Use Primary Language, Use Secondary Language, American English, British English, Japanese, or Spanish (Mexico City Spanish Dialect for VPU and EVMC, American Spanish Dialect for EM).

NOTE: Setting a digit translation Language field to a specific language will not work across nodes unless all the nodes in the system are at least 5.2F6 or newer since the older systems do not understand specific languages and expect to receive only Primary Language or Secondary Language values.

4.17 The length of the pauses in the greetings determine the "Timeout" for digit translation. Short, intermediate, and long pauses can be used in any combination to create a pause of any desired length. You can combine several recordings and special characters to provide information to the caller. For example you could arrange a series as follows:

1. **Recording 005**: *"Thank you for calling Inter-Tel. All of our agents are busy. To reach the operator, press 0."*
2. **Long Pause (L)**: This allows the caller to respond to Recording 005, if desired.
3. **Queue system-generated recording (Q)**: *"Your call is X in line."*
4. Time-To-Wait system-generated recording (T): *"Your call should be answered within X minutes."*
5. **Recording 006**: *"Our busiest hours are from 10 am to 1 pm. Our business hours are 8 am to 5 pm."*

Day and Night Greetings

4.18 When the application receives a call, a custom Audiotex Recording (made using the System Administrator's mailbox, as described in VOICE PROCESSING FEATURES, page 4-58) will play. You can select any Greeting Number for the Day and/or Night Greeting. The Day message is played when the System is in day mode, and the night message is played during night mode. The greetings can be changed or you can use the default greetings.

4.19 When used on Automated Attendants and Call Routing Announcements, the greeting can include pauses and can announce queue position and/or time to wait using special Audiotex selections. The special characters that can be used for programming greetings include the following:

- **Short Pause**: Inserts a 1-second pause.
- **Intermediate Pause**: Inserts a 5-second pause.
- **Long Pause**: Inserts a 10-second pause.
- **Position in Queue**: *(Used for hunt group Call Routing Announcements only.)* Tells the caller how many calls are waiting ahead of his or her call.
- **Time To Wait**: *(Used for hunt group Call Routing Announcements only.)* Tells the callers how long they can expect to wait, based on the number of calls waiting and the Average Call Length programmed for the hunt group.

4.20 To add or change a greeting, double-click on **Day Greeting** or **Night Greeting** to view the current list of greetings.

4.21 To add a greeting:

1. To add above an existing greeting, highlight that greeting and right-click. To add to the end of the list, right-click in any blank area of the screen (if there is no blank space, press **CTRL+N**). An option box appears.
2. Select **Add To Greeting List**. A window appears that asks you to select the type to include.
3. Select **Audiotex Recording** and then select **Next**. The list of Audiotex Recordings appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the recording(s) and/or special characters or pauses you want to use. Then select **Add Items**. Your selections appear in the list. When done, select **Finish**.

4.22 To move a greeting to another location in the list: Drag and drop the greeting to the new position. Or, highlight the greeting to move and press **CTRL** + the up/down arrow to move the greeting up or down in the list.

4.23 To delete one or more greeting: Highlight the greeting(s) and right-click. Then select **Remove Selected Items**.

NOTE: To highlight a series of recordings, hold down **SHIFT** while selecting the first and last recordings you want. To highlight two or more recordings that are not consecutive, hold down **CTRL** while selecting the desired greetings.

Accept AMIS Calls

4.24 This flag determines whether the Call Routing Application will accept incoming AMIS (Audio Messaging Interchange Specification) calls. If this flag is enabled, when the Call Routing Application answers an incoming CO call, in addition to detecting the normal DTMF tones (0-9, *, and #), it will also look for the DTMF tone "C." If it detects a C, the application will begin processing the AMIS call. If this flag is disabled, the application will not look for the DTMF tone C. In this case, if the application answers an incoming AMIS call, it will treat the call as a normal call (which means it will eventually time out). The default value for this field is disabled. To enable it, select the check box to place a check in it. To disable it again, select the checkbox again to remove the check.

NOTE: Call Routing Announcement applications programmed for AMIS cannot have greetings programmed, because the AMIS protocol will cause the destination system to disconnect if voice is detected.

Day and Night Classes of Service

4.25 *(For MNR Only)* Because the Message Notification/Retrieval application places outgoing calls for remote notification and Fax-On-Demand, it can have toll restriction classes of service. To program the toll restrictions double-click on **Classes of Service**, you have the option of choosing Day or Night. double-click on the desired time period to view a list of current classes of service.

4.26 To add a class of service:

1. Right-click in the window and select **Add To List**.
2. A window appears that allows you to select the types of toll restrictions you want to program. Select **ARS Only & Deny Area/Office** and/or **Classes of service** and then select **Next**.
3. Another window appears that shows you a list of available classes of service. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the desired class(es) of service and then select **Add Items**. The selected classes of service appear in the list. Select **Finish** to exit.
5. *(U.S. only) If you selected Deny Area/Office class of service,* the application must also be assigned to a user group. To change the user group, right-click on **User Group** and select **Change User Group**. In the first window that appears, select **User Groups** and then **Next**. In the next window, select the desired user group and then select **Finish** to exit and save the change.

4.27 To delete a class of service: Highlight the item(s). Then right-click and select **Remove Selected Items**.

NOTE: To highlight a series of items, hold down **SHIFT** while selecting the first and last item. To highlight two or more that are not consecutive, hold down **CTRL** while selecting the desired items.

Attendant

4.28 A station, hunt group, or other application can serve as the attendant station for an application. This attendant will receive recalls when the Transfer Recall Destination does not answer or is unavailable. To program the attendant for the application, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on **Attendant**. An options box appears. Select **Change Attendant**. A window appears that asks you to select the type to include.
2. Use the drop-down list box to scroll to Keyset and select **Next**. A list of phones appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
3. Select the desired phone and then select **Finish**. The new attendant appears in the Attendant field.

NOTE: If you would like to view the programming for the selected attendant station, double-click on **Attendant** to jump to the keyset programming area.

Automatic Fax Detection

4.29 With Automatic Fax Detection, Call Routing Announcement applications can be programmed to automatically route incoming fax calls to a specified extension or to an e-mail address. Call Routing Applications will be able to detect fax tones during the greeting and up to timeout. However, the fax tone detection will be disabled if the caller performs an action that removes them from the Call Routing Announcement (transferring to an extension, transferring to a mailbox, etc.). The two fields that control the Automatic Fax Detection feature are:

- **Fax Delivery Destination**: This field, if programmed, specifies the extension of the fax machine that will receive incoming faxes.
- **Fax Delivery E-Mail Address**: This field, if programmed, specifies the e-mail address of the account that will receive incoming faxes. The fax is converted to a TIFF file and sent to the e-mail address as an attached file. The address can be up to 127 characters. If you enter an invalid character (: ; " \ | ( ) , < > ‘), an error tone occurs. To view a message, use any TIFF file viewer (such as Imaging for Windows).

Default Applications

4.30 *(For STAR Only)* This field determines where calls will go that do not match any of the time-based criteria in tables 1-20. To assign the application, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the current Value. An options box appears. Select **Change Application**. A window appears that asks you to select the type to include.
2. Select the desired application and then select **Next**. A list of applications appears. (You can view them in a list by selecting **List** or view details by selecting **Details**.)
3. Select the desired application and then select **Finish**. The new value appears in the field.

NOTE: If you would like to view the programming for the selected application, double-click on the field to jump to the application's programming area.

Music-On-Hold parameters

4.31 Systems can be programmed to determine the music source a caller hears based on the device for which the caller is waiting. By default, the system determines the music source based on the trunk group on which the call resides.

4.32 The following parameters are programmable for each voice processor application:

- **Audio for Calls Camped onto this Device**: Defines the audio that a caller who is camped onto the device hears. The default is *Board*, and the default extended value is *01:08 Central Processor.*
- **Audio for Calls Holding for this Device**: Defines the audio that a caller who is holding for the device hears. The default is *Board*, and the default extended value is *01:08 Central Processor.*
- **Audio for Calls Ringing this Device**: Defines the audio that a caller who is ringing the device hears. The default is *Ringback*.

4.33 Each of these parameters can be programmed for one of the following:

- Silence
- Tick Tone
- Ringback
- Board

4.34 **To change the value of a music-on-hold parameter:** Use the drop-down list box to make your selection. If you select **Board**, specify the music source in the Extended Value column.

NOTE: If the trunk group's audio field, including Music-On-Hold, is set to Use Next Device's Audio Source (see page 6-234), the system uses the station's programming. If the field is set to any other option, such as Board, the system uses the trunk group's audio source, overriding station programming.

Time Slot Group

4.35 Setting up time slot groups allows you to specify the total number of voice channels available per application for processing calls between the System and the applications. To select the Time Slot Group for the application:

1. Right-click on **Time Slot Group**. An options box appears. Select **Change Time Slot Group**. A window appears that asks you to select the type to include.

2. Select **Time Slot Group** and then select **Next**. A list of time slot groups appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
3. Select the desired group and then select **Finish**. The new time slot group appears in the field.

NOTE: If you would like to view the programming for the selected time slot group, double-click on **Time Slot Group** to jump to the time slot programming area.

Transfer Recall Destination

4.36 This is the station, hunt group, or application that will receive any calls that recall after being transferred by the application. (For the Auto Attendant, this is usually the Automated Attendant Recall application).

4.37 To select the Transfer Recall Destination for the application, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on **Transfer Recall Destination**. An options box appears. Select **Change Transfer Recall Destination**. A window appears that asks you to select the type to include.
2. Select the desired destination type and then **Next**. You will see a list of fax documents, devices, or digit translation nodes that are present in the system.
3. Select the desired application and then select **Finish**. The new destination appears in the Transfer Recall Destination field.

NOTE: If you would like to view the programming for the selected application, double-click on **Transfer Recall Destination** to jump to the application's programming area.

Automatic Speech Recognition (ASR) Setting

4.38 *(EM systems only)* Determines whether the system will use the ASR Enabled setting in the current folder level or in the next folder level. For details about this field, see page 6-348.

Automatic Speech Recognition (ASR) Enabled

4.39 *(EM systems only)* Determines whether or not ASR is enabled for the current folder (and any subfolders or parent folders based on the ASR Setting field). If enabled, the system, application, or mailbox supports voice recognition for accessing that particular feature. If disabled, the system, application, or mailbox will not support voice recognition. For details about this field, see page 6-348.

NOTE: The ASR Enabled field is ignored (that is, displays a red "x") if the ASR Setting is set to Delegated, Ignored, or Ignored and Delegated.

Propagate Original Caller ID on Transfer

4.40 When this flag is enabled, if the endpoint is on a call with an outside trunk that had caller ID information, when a Voice Mail application transfers a call to an endpoint and the endpoint performs a transfer back to the PSTN, the Voice Mail application will forward the caller ID. To forward the caller ID to the PSTN, the eventual trunk must be ISDN and the "Propagate Original Caller ID" flag in its CO trunk group must be set to **Yes**. The default of this flag is set to *Yes* (which means forward caller ID on transfer PSTN calls). The Voice Mail application must be able to perform transfers (for example, Auto Attendant, Auto Attendant Recalls and Call Routing Announcements). For complete information about Caller ID Forwarding, see page 5-77.

Calling Party Name

4.41 The Calling Party Name field allows subscribers to specify the Caller ID [CLID] to use when the voice mail makes an outgoing call. Type the name (up to 15 alphanumeric characters, except for :, ", and ") in the text box.

Calling Party Number

4.42 The Calling Party Number field allows subscribers to specify the Caller ID [CLID] to use when the voice mail makes an outgoing call. Type the number (up to 15 alphanumeric characters, except for :, ", and ") in the text box.

Programming MNR Applications

4.43 A Message Notification/Retrieval (MNR) application is required for message notification, Remote Messaging, and Fax-On-Demand. Double-click on the application you want to program and its programming options appear.

4.44 The MNR application also needs to be programmed for outgoing access in Trunk Group programming (see page 6-414).

4.45 The Attendant, Audio for Calls Camped onto this Device, Audio for Calls Holding for this Device, Audio for Calls Ringing this Device, Time Slot Group, Transfer Recall Destination, ASR Setting, and ASR Enabled fields are programmed exactly as described for the Automated Attendant on page 6-356. The Day and Night Class of Service fields are described below.

Programming RAC Applications

4.46 The Record-A-Call (RAC) feature allows a station user to record an ongoing call and place it in a voice mailbox. When a station user enters the RAC feature code, the system places a call to the station's assigned RAC application using the station's RAC mailbox. When the application answers, the system sets up a conference call.

4.47 Keep the following considerations in mind:

- RAC has no default day/night recording
- If a recording is provided, the user who is using the RAC feature (nor the calling party) never hears any "warning" message indicating that their call may be recorded.

Programming STAR Applications

4.48 Scheduled Time-Based Application Routing (STAR) allows you to have applications with alternate greetings and/or different programming set up for holidays, weekends, and other scheduled events.

4.49 A STAR application is basically a "routing table" for Voice Processing applications. When a direct ring-in call (from a trunk group or call routing table) rings in to a STAR application, it sends the call to another Voice Processing application, according to its programmed schedule. The caller will not be aware of this transfer, but will hear the programmed day or night greeting for the destination application. (The STAR application itself does not play a greeting.)

4.50 STAR can be used with any type of Voice Processing application except Auto Attendant Recall and Record-A-Call. You can even send calls from one STAR application to another, thereby "chaining" the applications together to increase the number of available schedules.

4.51 There can be as many STAR applications as desired, so long as the maximum limit for the number of Voice Processing applications is not exceeded. For more information and a sample STAR setup, see page 4-34 in VOICE PROCESSING FEATURES.

4.52 **Schedules 1-20**: Each schedule can have a description. To enter the description, select the current Value and then enter a description for the schedule, up to 20 characters. To program the remaining information for each schedule, double-click on the schedule to display the following fields:

- **Application**: This is the extension number of the application that will be invoked when the time-based information in the table matches that of the incoming call. To assign the application, use one of the following methods:

  Method A

  a. Select the current Value and enter the new value in the text box.

  b. Press **ENTER**. A screen appears showing you what is associated with the number you entered.

  c. Select **OK**. The new number appears in the field.

  Method B

  a. Right-click on the current Value. An options box appears. Select **Change Application**. A window appears that asks you to select the type to include.

  b. Select the desired application and then select **Next**. A list of applications appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

  c. Select the desired application and then select **Finish**. The new value appears in the field.

  NOTE: If you would like to view the programming for the selected application, double-click on the field to jump to the application's programming area.

- **Specific Date**: If you want the schedule to be active on a single day or for a period of days, enable this flag. To enable this flag, select the check box to place a check in it and change the setting to **Yes**. To disable it, select the check box again to remove the check mark and set it to **No**. Then press **ENTER** or click on another field to save the changes.

- **Start and Stop Date**: If the Specific Date field is set to **Yes**, use these fields to set the date(s) that this schedule will be active (such as a holiday). Enter the date you want the schedule to begin in the Start Date text box and the date you want it to end in the Stop Date text box. After entering information, press **ENTER** or click on another field to save the changes. To have the schedule active on only one day, set the Start and Stop Dates to the same day.

- **Days of the Week**: If the Specific Date field is set to **No**, you can use these fields to determine which days of the week the schedule will be active. To enable a specific day, select its check box to place a mark in it and change the setting to **Yes**. To disable the day, select the check box again to remove the mark and set it to **No**. Then press **ENTER** or click on another field to save the changes.
- **Specific Times**: If you want the schedule to be active for a specific period of time on the selected day(s), enable this flag. To enable this flag, select the check box to place a check in it and change the setting to **Yes**. To disable it, select the check box again to remove the check mark and set it to **No**. Then press **ENTER** or click on another field to save the changes
- **Start and Stop Time**: If the Specific Time field is set to **Yes**, use these fields to set time period that this schedule will be active (such as a after hours). Enter the time you want the schedule to begin in the Start Time text box and the time you want it to end in the Stop Time text box. After entering information, press **ENTER** or click on another field to save the changes.
- **Day/Night Mode**: If the Specific Time field is set to **No**, you can use this field to determine whether the schedule will be active in day and/or night mode. To change this field, select the drop-down list box and scroll to the desired setting. Then press **ENTER** or click on another field to save the change.

B. DIGIT TRANSLATION NODES

4.53 Each digit translation node can be used in one or more Call Routing Announcement application. This allows entire digit translation node trees to be shared or moved without reprogramming.

4.54 There can be up to 200 digit translations in the system. Each digit translation can have its own description, username, digit translation programming, and greetings. The digit translation node is assigned as a destination for a digit translation "Transfer To Node" Action (see page 6-361).

4.55 A digit translation node can only be deleted if it is not associated with a Call Routing Announcement application.

4.56 Digit translation node programming is very similar to that for a Call Routing Announcement application. The programming fields are described in detail below.

4.57 **Description**: Enter a description of up to 20 characters in the Description text box. This description is used in the mailbox directory and should be entered in the form "last name, first name" (with a comma and space separating the names). Do not use Control characters in descriptions.

4.58 Digit Translation Nodes programming includes the following folders:

- Digit Translation (see below)
- Day and Night Greetings (see below)

Digit Translation

4.59 Digit translation allows callers to dial a single digit to select a designated extension number, mailbox, modem, or hunt group pilot number. Digit translation can be programmed for each digit 0-9, #, and *, plus a Timeout that is used when the caller does not enter a digit. Each digit can lead to another "digit translation node" that has its own digit translation actions. This layered Call Routing Announcement digit translation creates a "tree" of programmable digit translation nodes. For a detailed explanation of Call Routing Announcement digit translation nodes and a sample tree, see page 4-21 in VOICE PROCESSING FEATURES.

4.60 To program digit translations for the digit translation node, double-click on **Digit Translations**. A list of the digits and their actions and destinations will appear. To program each digit follow the instructions above.

Day and Night Greetings

4.61 When the digit translation node is used, a custom Audiotex Recording (made using the System Administrator's mailbox, as described in VOICE PROCESSING FEATURES, page 4-58) will play. You can select any Greeting Number for the Day and/or Night Greeting. The Day message is played when the System is in day mode and the night message is played during night mode. The greeting can include pauses and can announce queue position and/or time to wait using special Audiotex selections. The special characters that can be used for programming greetings include the following:

- **Short Pause**: Inserts a 1-second pause.
- **Intermediate Pause**: Inserts a 5-second pause.
- **Long Pause**: Inserts a 10-second pause.
- **Position in Queue**: *(Used for hunt group Call Routing Announcements only.)* Tells the caller how many calls are waiting ahead of the call.
- **Time To Wait**: *(Used for hunt group Call Routing Announcements only.)* Tells the callers how long they can expect to wait, based on the number of calls waiting and the Average Call Length programmed for the hunt group.

4.62 After each message, insert the desired number of pauses to allow the caller to make a response. The length of the pauses determine the "Timeout" for digit translation. Short, intermediate, and long pauses can be used in any combination to create a pause of any desired length. You can combine several recordings and special characters to provide information to the caller.

4.63 To add or change a greeting, double-click on **Day Greeting** or **Night Greeting** to view the current list of greetings.

4.64 To add a greeting:

1. To add above an existing greeting, highlight that greeting and right-click. To add to the end of the list, right-click in any blank area of the screen (if there is no blank space, press **CTRL+N**). An option box appears.
2. Select **Add To Greeting List**. A window appears that asks you to select the type to include.
3. Select **Audiotex Recording** and then select **Next**. The list of Audiotex Recording appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the recording(s) and/or special characters or pauses you want to use. Then select **Add Items**. Your selections appear in the list. Select **Finish** to exit.

4.65 To move a greeting to another location in the list: Drag and drop the greeting to the new position. Or, highlight the greeting to move and press **CTRL** + the up/down arrow to move the greeting up or down in the list.

4.66 To delete one or more greeting: Highlight the greeting(s) and right-click. Then select **Remove Selected Items**. The information is automatically removed from the list.

NOTE: To highlight a series of recordings, hold down **SHIFT** while selecting the first and last recordings you want. To highlight two or more recordings that are not consecutive, hold down **CTRL** while selecting the desired greetings.

C. EXTENSION IDS

4.67 Extension IDs provide the Auto Attendant application with a means for transferring calls to extensions and off-node devices that do not have mailboxes. An extension ID allows the owner to record a name for the directory and establish a password. Extension IDs can be created for stations, hunt groups, modems, and applications.

NOTE: To create a mailbox for an extension that currently has an extension ID, first delete the extension ID, then create the mailbox. Extension IDs cannot be created for off-node device wild-card extensions.

4.68 To create an Extension ID:

1. Right-click in any area of the screen. An option box appears.
2. Select **Create Extension ID**. A window appears that asks you to select the type of device to include.
3. Select the type of device and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the devices for which you want to create IDs. Then select **Add Items**. Your selections appear in the mailbox list. Select **Finish** to exit.

NOTE: To highlight a series of devices, hold down **SHIFT** while selecting the first and last devices you want. To highlight two or more items that are not next to each other, hold down **CTRL** while selecting the desired devices.

4.69 To delete an Extension ID: Highlight the IDs and right-click. Then select **Delete**.

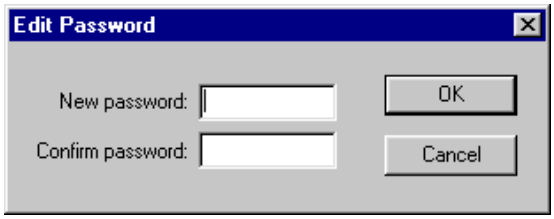
4.70 Extension ID programming includes the following fields:

- Allow Transfer Method Programming
- Auto Attendant Transfer Prompt
- Unlisted Number
- Private Extension
- Password
- Transfer Method

4.71 To program an Extension ID: The description and username fields cannot be changed. They are programmed as part of device programming.

- **Allow Transfer Method Programming**: Determines whether the extension ID user (or the voice mail System Administrator) will be allowed to change the Transfer Method, using the voice mail Personal Options prompts. Select the check box to place a check mark in it and allow user programming. Then press **ENTER** or click on another field to save your change. To disable user programming, select the box again to remove the check mark.

- **Auto Attendant Transfer Prompt**: Determines whether the transfer prompt ("Please hold while your call is being transferred to...") plays after a caller has entered the extension number of the station associated with this extension ID. This applies to calls transferred by Automated Attendant and Call Routing Announcement applications, including transfers to the operator's mailbox or extension ID. To disable the prompt, select the check box to remove the check mark. Then press **ENTER** or click on another field to save your change. To enable the prompt, select the box again.
- **Unlisted Number/Private Extension**: Unlisted numbers are not included in the directory, but can be dialed if the caller knows the extension number. Private numbers can be dialed, but only the name is played in the directory. Select the desired options to place check marks in the check boxes. Then press **ENTER** or click on another field to save your change.
- **Password**: To program a password for the Extension ID right-click on the Password field and select **Edit Password**. A window appears as shown below. Enter the password (up to 12 digits, using digits 0-9) in the New Password box. The digits will not appear on the screen when typed; they will appear as asterisks (***). Then retype the password exactly as before in the Confirm Password box. Then select **OK** to exit and save the password. If the entered passwords match, you will return to the Password field. If not, you must re-enter the new password and verify it again. If you make a mistake while entering the password or want to leave it unchanged, select **Cancel**.



NOTE: To provide system security, ***all*** extension IDs should have a password. To make the passwords difficult to guess, they should not match the mailbox number or consist of one digit repeated several times.

- **Transfer Method**: Determines how transfers will be made to this extension ID. The options are: Announce Only, Screened, and Unannounced. (Defaults to Unannounced.) See page 5-211 for a definition of these terms. Select the drop-down list box and scroll to the desired transfer type. Then press **ENTER** or click on another field to save your change.

4.72 Creating Extension IDs for Phantom Devices: You can create a voice processor extension ID that references a local phantom device or an off-node phantom device (see page 6-196).

NOTE: Older nodes (prior to v9.0) will not recognize off-node phantoms on v9.0 nodes.

4.73 To create an Extension ID for a Phantom device:

1. Open the Voice Processor\Devices\Extension IDs field in DB Studio.
2. Right-click in any area of the right window and select **Create Extension ID**. The Create Extension ID dialog box appears.
3. Select **Phantom Devices** and then select **Next**. The list of devices appears. (You can view them in a list by selecting the **List** button or view details by selecting the **Details** button.)

4. Select the devices for which you want to create IDs. Then select **Add Items**. Your selections appear in the mailbox list. Click **Finish** to exit.

> **NOTE:** To highlight a series of devices, hold down **SHIFT** while selecting the first and last devices you want. To highlight two or more items that are not next to each other, hold down **CTRL** while selecting the desired devices.

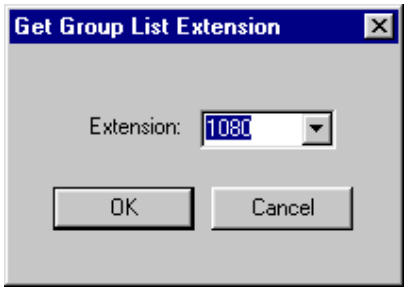
4.74 To allow a device on a node running version 8.2x software or earlier to place a call to a phantom device on a v9.0 node, you must create an off-node keyset on the older node with the phantom device's extension.

D. GROUP LISTS

4.75 Voice Mail group lists can be used by any Subscriber for sending messages to several mailboxes simultaneously. There can be up to 1000 group lists per node and up to 1500 members per group list. The parameters for programming group lists include the following:

- Group List Extension Number
- Description
- Username
- List of Mailboxes

4.76 **To create a Group List:** Right-click anywhere in the right side of the screen. The window, shown on the right, appears. Enter the Group List number and select **OK**.



4.77 **Group List Extension**: To change the group list number, either type the desired number in the text box or use the arrow to scroll to an available number. (If you attempt to enter an invalid number, a number that conflicts with an existing extension, or a number that is already assigned, you will see a warning window that allows you to fix the error. Enter a new number and then select **OK** to continue.) When finished, press **ENTER** or click on another field to save your change.

4.78 **Description and Username**: Enter a description of up to 20 characters in the Description text box. Enter a name of up to 10 characters in the Username text box. This is the name that will appear on phone displays. Do not use slash (/), backslash (\), vertical slash (|) or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

4.79 **Mailboxes included in the list**: To determine the mailboxes that will be included in the group list, double-click on the group list entry. The current list of mailboxes appears in the right screen.

4.80 **To add a mailbox to the group list:**

1. Right-click in any area of the screen. An option box appears.
2. Select **Add To List**. A window appears that asks you to select the type to include.
3. Select **Mailbox** and/or **Off-Node Mailbox** and then select **Next**. The list of mailboxes appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

4. Highlight the mailboxes you want to add. Then select **Add Items**. Your selections appear in the list. Select **Finish** to exit.

4.81 **To remove a mailbox from the list:** Highlight the mailbox(es) and right-click. Then select **Remove Selected Items**. The information is automatically removed from the list.

NOTE: To highlight a series of mailboxes, hold down **SHIFT** while selecting the first and last mailboxes you want. To highlight two or more that are not next consecutive, hold down **CTRL** while selecting the desired mailboxes.

E. MAILBOXES (OFF-NODE AND LOCAL)

4.82 A mailbox is a storage location on the Voice Processing PC hard disk which stores all messages that have been directed to it.

4.83 When you click on **Mailboxes**, what you see next is determined by whether you have previously programmed Voice Processing network nodes. If nodes exist, you will see the nodes listed with the local node. You can click on any of them to program the mailboxes for the selected node, as described below. If there are no Voice Processing networking nodes, you skip directly to the list that shows mailboxes on the local node.

4.84 To create a mailbox for an extension that currently has an extension ID, first delete the extension ID, then create the mailbox.

4.85 You can create associated and non-associated mailboxes on the Local node. However, you cannot create non-associated mailboxes for stations on other System nodes.

Network Mailboxes (Off-Node Mailboxes)

NOTE: Do not create or delete more than 2000 off-node devices at a time. Batch creating more than 2000 off-node devices may cause problems with the system.

4.86 Mailboxes programmed on the local node, but associated with mailboxes on remote Voice Processing nodes are referred to as “network mailboxes.” They are used by the local node to identify and locate the mailboxes located on the other nodes. They are not actual mailboxes; they are just “place holders” that tell the local node where to send messages received for that mailbox number. If the mailbox subscriber logs on to a network mailbox, the options available will be the directory name, the greetings, the password, and possibly the transfer method programming prompt, if enabled. In addition, if a subscriber changes the password, directory name, or greeting selection, these changes will be automatically updated on the corresponding network mailboxes.

4.87 Network mailbox programming is similar to local mailbox programming, except that it is limited to these fields:

- Allow Transfer Method Programming flag
- Auto Attendant Transfer Prompt flag
- Unlisted Number
- Private Extension and Mailbox
- Password
- Transfer Method
- Remote Mailbox Extension

4.88 The only field that is different from local mailbox programming is Remote Mailbox Extension. This field contains the actual mailbox number of the mailbox on the remote node. If there is a universal numbering plan among all the networked voice mail nodes, the Remote Mailbox Number field will be the same as the local mailbox number. If the numbering plan is not universal, the Remote Mailbox Number field can be different than the local mailbox number. The mailbox numbers programmed in this field are not verified because the local Voice Processing Unit does not know about the mailbox numbers on the remote Voice Processing Unit until it actually attempts to deliver a message to the remote Voice Processing Unit. The default value for this field is the local mailbox number. This field can be changed to any number up to five digits. The remote mailbox field is not validated and it should not be programmed a part of a group list or network mailbox on the remote node.

4.89 A network mailbox should have the Unlisted Number and Private Mailbox Number flags enabled. The Unlisted Number flag keeps the network mailbox from appearing in the company directory, and the Private Mailbox Number flag prevents the caller from hearing the mailbox number while using the company directory. These two fields can be programmed only from DB Programming.

Mailbox Programming

4.90 Select the desired node to create and program mailboxes for that node. When you select the node, a list of current mailboxes (if any) is shown. Program and create mailboxes as follows:

4.91 To create an Associated Mailbox:

1. Right-click in the right side of the screen. An option box appears.
2. Select **Create Associated Mailbox**. A window appears that asks you to select the type of device to include.
3. Select the type of device and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the devices for which you want to create mailboxes. Then select **Add Items**. Your selections appear in the mailbox list. Select **Finish** to exit.

> **NOTE:** To highlight a series of devices, hold down **SHIFT** while selecting the first and last devices you want. To highlight two or more that are not next to each other, hold down **CTRL** while selecting the desired devices.

4.92 To create a Non-Associated Mailbox:

1. Right-click in any area of the screen. An option box appears.
2. Select **Create Non-Associated Mailbox**. The following screen appears.

3. Enter the extension number you want to use for the first item in the Starting Extension field. Then indicate the number of items you want to create in the Number of Extensions field. The new mailboxes will be assigned sequentially to the next available numbers. When created, they appear in the programming window. You can program the mailboxes or change the mailbox numbers as described below.

NOTE: If any mailbox extensions conflict with existing extensions, those mailboxes are not created. If this occurs, a dialog box appears, listing the mailboxes that were not created.

— **Extension**: *(Not programmable for associated mailboxes.)* To change an existing mailbox number, select the number and enter the new number (or scroll to an available number using the arrow). Then press **ENTER** or click on another field to save your change.

— **Description and Username**: *(Not programmable for associated mailboxes.)* Enter a description of up to 20 characters in the Description text box. This description is used in the mailbox directory and should be entered in the form “last name, first name” (with a comma and space separating the names). Enter a name of up to 10 characters in the Username text box. This is the name that will appear on phone displays. Do not use slash (/), backslash (\), vertical slash (|) or tilde (~) characters in usernames. Do not use Control characters in descriptions or usernames.

— **Delete Mailbox**: To remove a mailbox from the list highlight the mailbox(es) and right-click. Then select **Delete**.

NOTE: Attempting to delete a mailbox that is in use will cause serious performance issues for the voice processing system. Verify that the mailbox is not in use before deleting it.

— **Clear Mailboxes Messages**: If you have an active direct-connection or remote programming session, you can clear (erase) all the messages in a mailbox. Select the mailbox(es) you want to clear and then right-click. Select **Clear Message(s)** from the Selected Mailboxes option.

— **Copy Mailbox**: To copy a mailbox, right-click on the mailbox number and then select **Copy**. To paste the programming information into another mailbox, right-click on the mailbox where you want the information pasted. Then select **Paste**. You will see a screen like the one shown below that allows you to select the attributes you want to copy.

— **Mailbox Associated Devices**: The Associated Devices and References feature (see page 5-31) allows you to see the associated devices for a station, mailbox, and hunt group. This feature also allows you to query various groups in the database to locate the associated references to the extension.

To search for devices associated with the mailbox, right-click the mailbox and select **Associated Devices and References**. A dialog box, containing associated device(s) and references for the mailbox appears.

4.93 To program a mailbox, double-click on it. (Or, right-click and select **Explore**.) The following options appear:

- Directory Information (see page 6-380)
- Envelope Settings (see page 6-380)
- Unified Messaging (see page 6-381)
- Recording Length (see page 6-385)
- Message Limits (see page 6-385)
- Subscriber Statistics (see page 6-386)
- Primary and Alternate Message Notification (see page 6-386)
- E-mail Reader Profiles (see page 6-389)
- Dial-0 Destination Type and Dial-0 Destination (see page 6-390)
- Remote Messaging (see page 6-390)
- Mailbox Initialized (see page 6-390)
- Receive Only (see page 6-390)
- Allow Transfer Method Programming (see page 6-391)
- Play Recording Instructions (see page 6-391)

- Auto Attendant Transfer Prompt (see page 6-391)
- Deliver Hangup Message (when ANI is available) (see page 6-391)
- Password (see page 6-392)
- Greeting (see page 6-392)
- Transfer Method (see page 6-392)
- Message Notification Station (see page 6-393)
- Time Zone (see page 6-393)
- Automatic Speech Recognition (ASR) Setting (see page 6-393)
- Automatic Speech Recognition (ASR) Enabled (see page 6-393)
- Quota Warning (see page 6-394)
- Quota Grace (see page 6-394)

Directory Information

4.94 The mailbox can have an unlisted number (not included in the mailbox directory but can be dialed if the caller knows the mailbox number) and/or it can be a private extension and mailbox number (extension and mailbox number are not announced in directory, but the number can be dialed). There are no private or unlisted numbers in the default state. To set these flags, double-click on **Directory Information**. The list of flags appears. Then choose the Value for the flag you want to change and click on the check box to place a mark in it and set the flag to **Yes**. (Or click on an existing mark to disable it and change the flag to **No**.) Then press **ENTER** or click on another field to save your change.

Envelope Settings

4.95 E-mail, fax, and voice messages can include a "message envelope" for all message types, including those that are converted to speech. This envelope contains information such as the message duration, source, etc. Announcement options include the following:

- **Announce Message Length**: (*Voice Only*) If enabled, the voice mail message duration is included in the message envelope.
- **Announce Message Subject**: (*E-mail Only*) If enabled, the subject line of the e-mail message is included in the envelope. If the subject line is blank, the envelope ignores this setting and does not announce the subject.
- **Announce Message Pages**: (*Fax Only*) If enabled, the total number of pages that were faxed are included in the envelope.
- **Announce Message Source**: If enabled, the envelope includes the message originator. For e-mail messages, this is the e-mail address or alias of the message sender; for voice mail messages, this is the name and number of the caller (if available); and for faxes, this is the originating fax number.
- **Announce Date and Time**: If enabled, the date and time that the message was received is included in the envelope.

4.96 By default, all available envelope announcement options are enabled for each type of message. To disable an option, double-click **Local Mailbox Envelope Settings (E-Mail, Fax,** or **Voice**) and remove the check mark from the check box for the flag you want to disable.

NOTE: You cannot program envelope options for off-node mailboxes.

Unified Messaging

4.97 This folder is for supporting the features and functionality included with Unified Messaging OSE v2.0. For complete information about Unified Messaging OSE v2.0, refer to the latest version of the *Unified Messaging Open Standards Edition Administrator's Guide* (document part no. 835.3162).

4.98 The first field in the Unified Messaging folder is the Unified Messaging Level. The level that is programmed in this field specifies the Unified Messaging level of integration. This field was previously known as the E-mail Gateway field and it was located in the Fax/E-mail Forwarding folder.

4.99 Options for this field vary depending on the Voice Processor type (see Table 6-44, below):

Table 6-44: *Unified Messaging Levels for BVM, VPU, and EM*

Unified Messaging Level	Voice Processor Type		
	BVM	VPU	EM
Disabled: Delivers a voice message to the user's mailbox just as normal. No e-mail is sent.	✓	✓	✓
Forward Only: Delivers a voice message to the user's e-mail address when the user receives a new voice message in his or her voice mailbox. After the message is sent, the original message is deleted from the mailbox.	✓	✓	✓
Forward and Copy: Delivers a copy of a voice message to the user's e-mail address when the user receives a new voice message in the voice mailbox. The original message can be retrieved from the mailbox.	✓	✓	✓
Unified Messaging: Allows the user to use the Unified Messaging v2.2 functionality.		✓	
Enhanced Forward and Copy: Provides the user with direct access into the voice mailbox from the e-mail client. This level provides full integration with the e-mail client, but does not provide synchronization between the EM server and the e-mail server. With this level, the user has configuration control over the format of the messages.			✓
Basic Integration: Provides the user with direct access into the voice mailbox from the e-mail client. This level offers limited integration with the e-mail client for voice messages because a separate IMAP inbox is required for voice messages. The user has no configuration control over the format of the messages.			✓
Enhanced Integration: Provides full integration for voice messages with the e-mail client, and provides full synchronization between the EM server and the e-mail server. The user has configuration control over the format of the message, and this level supports the Message Waiting Indication (MWI) feature.			✓

4.100 The Unified Messaging level you select determines which additional fields in the Unified Messaging folder are programmable. See page 6-382 for programming instructions and field descriptions.

NOTE: For Unified Messaging OSE v2.0, the values for the fields previously programmed under the Fax/E-mail Forwarding folder remain the same and the fields automatically map to their new location.

4.101 Table 6-45, below, indicates which fields in the Unified Messaging folder must be programmed for each Unified Messaging level.

Table 6-45: *Programmable Fields for the Unified Messaging Folder*

Programmable Fields	Unified Messaging Level					
	Disabled	Forward Only, Forward and Copy	Unified Messaging	Enhanced Forward and Copy	Basic Integration	Enhanced Integration
Unified Messaging Level	✓	✓	✓	✓	✓	✓
Enable SSL for E-mail Server Connection				✓		✓
E-mail Account Folder for Synchronization				✓		✓
E-mail Server				✓		✓
E-mail Account Password				✓		✓
E-mail Account Username				✓		✓
E-mail Address for Voice Messages		✓	✓	✓	✓	✓
Allow User to Configure Settings				✓		✓
Download Format for Mobile Web Page'				✓		✓
E-mail Client Message Format				✓		✓
Synchronize MWI with E-mail Client						✓
E-mail Address for Fax Delivery[1]	✓	✓	✓	✓	✓	✓
Copy Fax to Sender				✓	✓	✓
Fax Delivery Destination[1]	✓	✓	✓	✓	✓	✓
IMAP Synchronization Method[2]				✓		✓
IMAP IDLE Timeout				✓		✓
IMAP Polling Timeout				✓		✓

1. This field is disabled if the voice processor type is BVM, which does not support faxing.

2. The option selected for the IMAP Synchronization Method determines which related fields are programmable:
   • *Event* (default): The IMAP IDLE Timeout field is enabled and the IMAP Polling Timeout field is disabled.
   • *Polling*: The IMAP IDLE Timeout field is disabled and the IMAP Polling Timeout field is enabled.

4.102 To configure the fields in the Unified Messaging folder:

1. Go to Voice Processor – Devices – Mailboxes – <*mailbox*> – **Unified Messaging**.
2. Configure the following fields:

   - **Unified Messaging Level**: Specifies the level of integration for Unified Messaging. This field was previously located in the E-mail Gateway field under the Fax/E-mail Forwarding folder. It is set to *Disabled* by default. Options for this field vary depending on the Voice Processor type:

     – Disabled

- Forward Only
- Forward and Copy
- Unified Messaging
- Enhanced Forward and Copy
- Basic Integration
- Enhanced Integration

See Table 6-44 on page 381 for descriptions of the options above.

- **Enable SSL for E-mail Server Connection**: Indicates whether the integration connection is made using Secure Socket Layer (SSL). It is set to *Yes* by default. This field appears with a red "**X**" unless Unified Messaging Level is set to Enhanced Forward & Copy or Enhanced Integration.
- **E-mail Account Folder for Synchronization**: Indicates the path to the account folder (such as Inbox, Voice Mail, etc.) on the e-mail server to use for message synchronization. The folder name can be up to 127 characters. For example, to synchronize messages to the folder named "Voice Mail" in the user's e-mail client account, type **Voice Mail** in the text box. To synchronize messages to the subfolder named "VM" under the "Inbox" main folder, type `Inbox/VM` in the text box. This field is set to *Inbox* by default. This field appears with a red "**X**" unless Unified Messaging Level is set to Enhanced Forward & Copy or Enhanced Integration.
- **E-mail Server**: Indicates the name of the e-mail server for the account synchronization. The server name can be up to 127 characters. This field is blank by default. This field appears with a red "**X**" unless Unified Messaging Level is set to Enhanced Forward & Copy or Enhanced Integration.
- **E-mail Account Password**: Indicates the password of the account to use for message synchronization. This is the user's network password. The password can be up to 40 characters. This field is blank by default. This field appears with a red "**X**" unless Unified Messaging Level is set to Enhanced Forward & Copy or Enhanced Integration.
- **E-mail Account Username**: Indicates the user name of the account to use for message synchronization. This is the user's network username. The username can be up to 127 characters. This field is blank by default. This field appears with a red "**X**" unless Unified Messaging Level is set to Enhanced Forward & Copy or Enhanced Integration.
- **E-mail Address for Voice Messages**: Specifies the e-mail address to which the voice mail messages will be forwarded. This is also the e-mail address where welcome and error e-mails are sent for the Enhanced Forward and Copy and Enhanced Integration Unified Messaging levels. The content and format of the field depends on the e-mail system being used. The address can be up to 127 characters. If you enter an invalid character (: ; " \ | ( ) , < > '), an error tone occurs. For example, when using Lotus Notes, this field could be set to John Doe/Chandler/Mitel, and when using e-mail, it would be john_doe@mitel.com. This field was originally located in the Fax/E-mail Forwarding folder.
- **Allow User to Configure Settings**: Delegates account configuration control to the individual account users. It is set to *Yes* by default. This field appears with a red "**X**" unless Unified Messaging Level is set to Enhanced Forward & Copy or Enhanced Integration.

- **Download Format for Mobile Web Page**: Specifies which file format to use when downloading voice messages from the mobile device Voice Mail Web page. Options include **.MP3** or **.WAV** file formats. The default value is *MP3*. This field appears with a red "**X**" unless Unified Messaging Level is set to Enhanced Forward & Copy or Enhanced Integration.

  NOTE: Most mobile devices do not support **.WAV** file formats. Check to see which file format is supported by the user's mobile device and program this field accordingly.

- **E-mail Client Message Format**: Indicates whether the message that the user receives contains a URL (a link to the message), a file attachment, or both. It is set to **Attachment** by default. This field appears with a red "**X**" unless Unified Messaging Level is set to Enhanced Forward & Copy or Enhanced Integration.
- **Synchronize MWI with E-mail Client**: Synchronizes message waiting indicator (MWI) support with the e-mail client. If set to "Yes" (default), the message lamp is turned off when the e-mail message is read. This field is applicable to the Enhanced Integration Unified Messaging Level only.
- **E-mail Address for Fax Delivery**: Specifies the e-mail address of the account that will receive incoming faxes. The fax is converted to a TIFF file and sent to the e-mail address as an attached file. The address can be up to 127 characters. If you enter an invalid character (: ; " \ | ( ) , < > ‘), an error tone occurs. This field appears with a red "**X**" if the Voice Processor type is BVM because BVM does not support faxing. This field was originally located in the Fax/E-mail Forwarding folder.
- **Copy Fax to Sender**: Enables the default operation for e-mailing a copy of a fax to a user's account as specified in the E-mail Address for Fax Delivery field (see above). The flag is enabled by default. This field appears with a red "**X**" unless Unified Messaging Level is set to Enhanced Forward & Copy, Basic Integration, or Enhanced Integration.
- **Fax Delivery Destination**: Specifies the extension of the fax machine that receives incoming faxes routed through this mailbox. This field appears with a red "X" if the Voice Processor type is BVM because BVM does not support faxing. This field was originally located in the Fax/E-mail Forwarding folder.

  To program the extension number, use one of the following methods:

  Method A

  1.) Select the current value and enter the new value in the text box.

  2.) Press **ENTER**. A screen displays what is associated with the number you entered.

  3.) Click **OK**. The new number appears in the field.

  Method B

  1.) Right-click on the existing extension number. An option box appears.

  2.) Click **Change Fax Delivery Destination**. A window appears prompting you to select the type of device to include.

  3.) Select the type of device and then click **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)

4.) Highlight the extension number associated with the fax machine then click **Finish**. Your selection appears in Fax Delivery Destination field.

- **IMAP Synchronization Method**: Determines how the IMAP server synchronizes messages between the server and an IMAP client account. By default, it is set to *Event*. This allows message changes on the remote IMAP server to be automatically recognized by the Inter-Tel synchronization client, as they occur. If the IMAP server does not support the IMAP IDLE command, select **Polling**. This will force the Inter-tel synchronization client to query for changes on the account on the remote server. If you do not know if the remote IMAP server supports the IDLE command, default to *Event*. At run time the Inter-Tel synchronization client automatically detects if the IDLE command is supported. If not supported, the client will use the Polling method. Polling can be used as default to reduce load on the EM unit. This field appears with a red "X" unless the Unified Messaging Level is set to Enhanced Forward & Copy or Enhanced Integration.
- **IMAP IDLE Timeout**: Determines the maximum length that the IDLE command will wait for the IMAP server. The field appears with a red "**X**" if the Integration Synchronization Method is set to "Polling." The range is 1–30 minutes; the default value is *20 minutes*. This field appears with a red "**X**" if the IMAP Synchronization Method is set to "Polling" as it only applies for Event or unless Unified Messaging Level is set to Enhanced Forward & Copy or Enhanced Integration.
- **IMAP Polling Timeout**: Indicates how often the synchronization process polls for message updates. Use this when the server does not support the IDLE command. The range is 1–20 minutes; the default value is *10 minutes*. The field appears with a red "**X**" if the IMAP Synchronization Method is set to "Polling" as it only applies for Event or unless Unified Messaging Level is set to Enhanced Forward & Copy or Enhanced Integration.

3. Press **Enter** or click another field to save the change.

Recording Length

4.103 Click on **Recording Length** to display the number of seconds used by the mailbox's directory name recording and the length of the primary and alternate greetings. The information is shown for reference only. It is not programmable.

Message Limits

4.104 The message limits that can be programmed for each mailbox include the following. To change a limit, select the Value field and enter the desired number in the text box. Then press **ENTER** or click on another field to save the change. *If you enter an invalid number,* you will see a warning message.

- **Maximum Mailbox Message Capacity**: The mailbox can be programmed to hold up to 600 minutes of messages or have unlimited message capacity. Default capacity is 30 minutes. The range is 0-600 minutes. Enter 0 for unlimited capacity.
- **Maximum Non-Subscriber Message Length**: Maximum non-subscriber messages can be set to a value between one minute and the Maximum Mailbox Message Capacity setting. Default length is 5 minutes. The range is 1 minute to the Maximum Mailbox Message Capacity value.
- **Maximum Outgoing Message Length**: The length of outgoing messages by this subscriber can be 1-120 minutes. Default length is 5 minutes. The range is 1-120 minutes.

Subscriber Statistics

4.105 You can check the following information by clicking on **Subscriber Statistics**. The statistics reflect the period since the last date that the statistics were cleared. Statistics are cumulative and remain as such until cleared using Report Parameters (see page 6-419). The information is shown for reference only. It is not programmable.

- **Last Logon Date and Time**: Reflects the most recent date and time of the last valid logon by the subscriber. (If the System Administrator makes any change to a subscriber's personal options from the System Administrator's mailbox, no change will be made to this field.)
- **Number of New Messages**: Reflects a count of the number of messages in a subscriber's new message queue. It is the same number that is reported to the subscribers when they log in to the mailboxes.
- **Number of Saved Messages**: Reflects a count of the number of messages stored in the saved message queue for the mailbox. It is the same number that is reported to the subscribers when they log in to the mailboxes.
- **Mailbox Percent Full**: Shows the actual percentage of maximum mailbox message capacity used.
- **Number of Times Mailbox Was 80% Full**: Shows the number of times the mailbox reached 80% of its maximum message capacity.
- **Number of Times Mailbox Was Full**: Displays the number of times a mailbox reached its maximum message capacity.
- **Number of Messages Sent**: Reflects a count of the number of times a subscriber records and sends a message (to one mailbox or a group list of mailboxes), replies to a message sent by another subscriber, or forwards a message with comments.
- **Number of Messages Received**: Shows the number of messages a subscriber has received regardless of where the messages came from (subscriber, non-subscriber, or system).
- **Total Length of New and Saved Messages**: Reflects a combined total of the amount of time represented by the "Number of New Messages" and "Number of Saved Messages" fields.
- **Number of Times 3 Bad Passwords Were Entered**: Increases each time a single call includes three attempts to enter a mailbox and the caller uses an incorrect mailbox/password combination.

Primary and Alternate Message Notification

4.106 If remote messaging is enabled, double-click on either **Primary** or **Alternate Message Notification** to program the remote messaging parameters. You will see the following options:

- **Cascade Levels**: Double-click on **Cascade Levels** to view the list of cascades. (The list is blank if no cascade levels are currently programmed.) To add cascade levels, right-click in any area of the screen and select **Add cascade level(s)**. Then select the starting ID and the number of levels you want to add. (For example, to add 9 levels with IDs 1-9, select 1 as the **Starting ID** and 9 as the **Number of IDs**.) When you click **OK**, the levels are added to the list with default values. You can then change the Notification Number for any cascade by entering the new number in its text box. You can also set the Notification flag by placing a check in the checkbox in this screen or in the cascade settings described below. To delete a cascade level, right-click on the level you want to remove and select **Remove Selected Item**.

NOTE: As of version 7.0, primary and alternate cascade levels are not equipped in pairs. For example, if you add or delete a primary cascade level, a corresponding alternate cascade level is not automatically added or deleted.

When you want to program the cascades settings, double-click on the cascade you want to program to view the following options:

— **Timers and Limits**: Double-click on **Timers and Limits** to program the following. If necessary, enter a new Value and then press **ENTER** or click on another field to save your change.

- *Number Of Call Attempts:* Determines how many times the Voice Processor will attempt to complete a call to this cascade level before moving to the next level. The range is 1-1000 call attempts and the default value is 1.
- *Number Called Busy Timer:* Defines the amount of time the Voice Processor will wait between outgoing call attempts whenever a busy signal is encountered during a remote message notification attempt to a personal number or pager. The range is 0-1440 minutes and the default value is 5.
- *Personal Number No Answer Timer:* Defines the amount of time the Voice Processor will wait between outgoing call attempts whenever a message notification attempt to a personal number is unanswered. The range is 0-1440 minutes and the default value is 30.
- *Pager Notification Retry Timer:* Reflects the amount of time the Voice Processor will wait between outgoing call attempts when the notification number is a pager. The range is 0-1440 minutes and the default value is 20.

— **Enable Notification**: Each cascade can be enabled or disabled individually. To change the setting, click on the check box to place a mark in it and set the flag to **Yes**. (Or click on an existing mark to disable it and change the flag to **No**.) Then press **ENTER** or click on another field to save your change.

— **Call Progress Detection**: This determines if the Voice Processor will perform call progress detection after dialing the notification number and outgoing termination fields on outside calls. If it is enabled, the system will analyze call progress tones after sending the telephone number. If it is disabled, the system will assume the call has been answered, without analyzing tones. This should only be disabled in cases where call progress tones are not used (lighting a message lamp) or when the tones used by the destination cannot be recognized by the system (some pager destinations). To change the setting, click on the check box to remove the mark in it and set the flag to **No**. (Or click on it again to enable it and change the flag to **Yes**.) Then press **ENTER** or click on another field to save your change.

— **Notification Type**: The message notification number can be identified as a personal number (a person will answer the call) or a pager. (This defaults to "personal number.") Select the drop-down list box and scroll to the desired setting. Then press **ENTER** or click on another field to save your change.

— **Notification Destination Type**: This indicates whether the notification call is going to an intercom (IC) or outside (CO) destination. This can also be programmed by the mailbox Subscriber. Select the drop-down list box and scroll to the desired setting. Then press **ENTER** or click on another field to save your change.

- — **Notification Destination**: This is the number (outside number, local extension, or off-node extension) to be notified when the mailbox receives a message. If the Notification Destination Type is IC, right-click on the field and select **Change Notification Destination**. Then program the destination device. If the Notification Destination Type is CO, enter the outside number in the text box. Then press **ENTER** or click on another field to save the change. The outside number can contain up to 24 characters including digits (0-9, #, and *) or P for pauses. (This number should not include pager display numbers; they are programmed below.)
- — **Pager Dial String**: This digit string can contain up to 54 characters. It should include any digits that the paging company requires when the call is answered, the pager LCD number, and the pager termination code, if needed. Valid entries include: any digit 0-9, #, *, P for pause. Also, if you want to have the pager show the number of the mailbox that placed the call, you can use an M in the dial string. To show the number of waiting messages for the mailbox, use an N. For example, if the Pager String is programmed as 9619000*MN# and a pager call is placed by mailbox number 1234 which has 3 waiting messages, the pager display would show 9619000*12343 (the # is used as a call termination digit). Enter the dial string in the text box and then press **ENTER** or click on another field to save the change.
- — **Outgoing Access**: This is used to identify the trunk group that will be used for placing remote notification calls. To change the trunk group, right-click on **Value** and select **Change Outgoing Access**. In the first window, scroll to CO Trunk Groups and then select **Next**. In the next window, highlight the desired trunk group and then select **Finish** to exit.
- — **Outgoing Access Prefix**: This is the dial string that the system uses before an outgoing number, if any. Enter the dial string in the text box and then press **ENTER** or click on another field to save the change.
- — **Outgoing Termination**: This is the dial string that the system uses to terminate an outgoing number, if any. Enter the dial string in the text box and then press **ENTER** or click on another field to save the change.

- **Notification Category**: Message notification can be set to place the notification call for all messages or only when priority messages are received. (This defaults to “all messages.”) Select the drop-down list box and scroll to the desired setting. Then press **ENTER** or click on another field to save your change.
- **Call For Each New Message**: This determines whether the Voice Processor should attempt message notification *every* time a message is received (that meets the notification category set above), or only when a message is received and no other messages are waiting to be picked up. To change the setting, click on the check box to place a mark in it and set the flag to **Yes**. (Or click on an existing mark to disable it and change the flag to **No**.) Then press **ENTER** or click on another field to save your change.
- **Start/Stop Time**: Message notification can be enabled for any time period, up to 24 hours per day. (Default is 8:00am to 5:00pm.) If start and stop times are the same, notification will be enabled 24 hours per day. To set the time, select the current Value. Then select the hours, minutes, or AM/PM field, and use the arrows to scroll to the desired setting. Select another field or press **ENTER** to save your change.
- **Days Of The Week**: Message notification can be set to place notification calls only on certain days. (This defaults to Monday through Friday.) To enable a day, click on the check box to place a mark in it and set the flag to **Yes**. (Or click on an existing mark to disable it and change the flag to **No**.) Then press **ENTER** or click on another field to save your change.

E-mail Reader Profiles

4.107 *(EM Systems Only)* To use the text-to-speech (TTS) functionality to convert e-mail messages to voice mail messages, each mailbox user must have an e-mail reader profile. This profile contains the information that EM requires to access the correct e-mail server and e-mail account. To access e-mail reader profile programming, double-click E-mail Reader Profiles. You can then create profiles as described below.

4.108 To create an e-mail reader profile:

1. Under Voice Processor\Devices\Mailboxes\<*Mailbox*>\E-mail Reader Profiles, right-click in the right pane and select Add to E-mail Reader Profiles List. The Get ID dialog box appears.
2. Enter the starting ID and click **OK**. The ID is added to the list with a blank username.

NOTE: Currently, you can assign only one e-mail reader profile to each mailbox. You cannot, therefore, change the Number of IDs field.

3. Double-click the ID to display the settings for that profile.
4. Complete the following fields:

- **Username**: Enter a username, up to 40 characters, for the profile. This username must match the username used to log on to the associated e-mail account.
- **Password**: Right-click this field and select **Edit Password**. Then enter a password, up to 40 characters, used to access the associated e-mail account. Confirm the password and click **OK**. This password must match the password used to log on to the e-mail account.

NOTE: Some networks are configured so that users are automatically logged onto their e-mail application whenever they log onto the domain. In this case, the Username and Password fields must match the user's domain account username and password.

- **Hostname**: Enter the hostname, up to 40 characters, of the e-mail server PC. This hostname is required for accessing the e-mail system and is case-sensitive.
- **Default Folder**: Enter the name of the mailbox folder that stores the e-mail messages that will be converted to speech. This field supports up to 40 characters. By default, this is *INBOX*.
- **IMAP SSL**: When the this option is enabled (set to Yes), the E-mail Reader uses IMAP SSL (Secure Sockets Layer) to provide an encrypted communication between the EM system and the E-mail Server that the E-mail Reader application is getting e-mails from. By default, it is set to *Yes*.
- **IMAP Port**: When the IMAP SSL option is disabled, the port which E-mail Reader uses to communicate with the E-mail Server is configurable. E-mail Reader will use this port number when connecting to the E-mail Server to retrieve e-mail messages. By default it is set to the standard IMAP port, 143. If IMAP SSL is enabled, then the standard IMAP SSL port of 993 is used and is not configurable. If you enter an invalid character (: ; " \ | ( ) , < > ‘), an error tone occurs.

NOTE: This field has a RED X when IMAP SSL is set to Yes.

- **Account E-mail Address**: Enter the e-mail address for this e-mail account. This field is used to populate the FROM address when originating e-mails. If the receiver of e-mails generated from E-mail Reader replies to the message, this e-mail address is the e-mail address to which the reply will be directed.

NOTE: To edit existing e-mail reader profiles, double-click the profile ID.

Dial-0 Destination Type and Dial-0 Destination

4.109 Each mailbox is assigned an operator destination. When a caller presses 0 while listening to the mailbox's personal greeting or recording instructions, or while recording a non-subscriber message, the operator destination is called. First select the desired extension type by scrolling through the Dial-0 Destination Type drop-down list box to the appropriate type. The options are: Extension, Mailbox, Operator, and None. If Extension or Mailbox is selected, scroll through the Dial-0 Destination drop-down list box to the desired destination. The Dial-0 Destination can be one of the following types:

- **Mailbox**: If the operator is a mailbox, the caller will hear the operator destination's personal greeting after pressing 0.
- **Extension number of a station or a hunt group (including off-node devices)**: If the operator destination is a station's extension number, the caller will hear the Automated Attendant transfer prompt while the call is placed to the extension number ("*Please hold while I transfer your call to...*"), unless it has been disabled at the mailbox. If the station user does not answer the caller will be asked if he wants to leave a message if the operator's extension has an associated mailbox. If the caller chooses not to leave a message, the call returns to the mail menu of the application being used (Call Routing Announcement, Voice Mail, or Auto Attendant.
- **Extension number of an application**: If the operator destination is an application's extension number, the call is transferred to that application's main menu.
- **Operator**: If the operator destination is set to **Operator**, the caller will be transferred to the attendant.
- **None**: If the operator destination is set to **None**, the 0 will be ignored.

Remote Messaging

4.110 To enable remote messaging, select this check box to place a check mark in it. Then press **ENTER** or click on another field to save your change. To disable remote messaging, select it again to remove the check mark.

NOTE: You cannot enable remote messaging unless you have programmed cascade levels (see page 6-386). If you try to enable remote messaging without any cascade levels, an error appears.

Mailbox Initialized

4.111 This field indicates whether the subscriber (mailbox owner) has initialized the mailbox and recorded a name in the company directory. This is shown for reference only and is not programmable.

Receive Only

4.112 The mailbox can be programmed as a standard mailbox that can send and receive messages or it can be a "receive only" mailbox. (In the default state, it is a standard mailbox.) To make the mailbox receive-only select the check box to place a mark in it and change the setting to **Yes**. Then press **ENTER** or click on another field to save your change. To remove the check mark, select the checkbox again.

Allow Transfer Method Programming

4.113 This flag determines whether the user (or the voice mail System Administrator) will be allowed to change the Transfer Method using the voice mail Personal Options prompts. If enabled, it allows user programming. If disabled, the Transfer Method can be changed only in DB Programming. Select the check box to place a check mark in it to allow user programming. Then press **ENTER** or click on another field to save your change. To disable user programming, select the box again to remove the check.

Play Recording Instructions

4.114 If this option is enabled, the system prompt that tells the caller to leave a message after the beep will play after the primary or alternate greeting. If disabled, the beep will occur as soon as the primary or alternate greeting ends. In the default state, this option is enabled. To disable the instructions that play after the primary or alternate greeting, select the check box to remove the check mark. Then press **ENTER** or click on another field to save your change. To enable the option, select the check box again.

Auto Attendant Transfer Prompt

4.115 This option determines whether the transfer prompt ("Please hold while your call is being transferred to...") plays after a caller has entered the extension number of the station associated with this mailbox. This applies to calls transferred by Automated Attendant and Call Routing Announcement applications, including transfers to the operator's mailbox or extension ID. To disable the prompt, select the check box to remove the check. Then press **ENTER** or click on another field to save your change. To enable the prompt, select the box again.

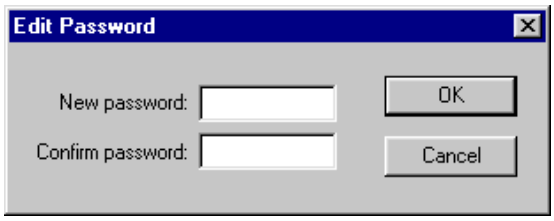
Deliver Hangup Message (when ANI is available)

4.116 This option determines when to leave an "indication message" if the caller hangs up. This indication message is a system-generated message that states there was an incoming call to the voice mailbox but that no voice message was left. The purpose of this indication message is to capture Caller ID [CLIP] information for the call and display it on the phone. If the Deliver Hangup Message (when ANI is available) field is set to *Yes* (default), the system leaves indication messages for calls that are shorter than the Shortest Message Allowed timer (Voice Processing\Timers and Limits\Timers and Limits). If set to *No*, the feature is disabled, and the system does not leave an indication message. To disable the flag, select the check box to remove the check. Then press **ENTER** or click on another field to save your change. To enable the flag, select the box again.

NOTE: In systems that had ANI support and that were running versions prior to 7.0, voice mail messages left by an outside caller that were shorter than the minimum required message length generated a system message (No Message Left) in the associated mailbox.

Password

4.117 A password of up to 12 digits can be programmed for use by the subscriber when retrieving messages. To program a password for the mailbox right-click on **Password** and select **Edit Password**. A window appears as shown below. Enter the password (up to 12 digits, using digits 0-9) in the New Password box. The digits will not appear on the screen when typed; they will appear as asterisks (***). Then retype the password exactly as before in the Confirm Password box. Then select **OK** to exit and save the password. If the entered passwords match, you will return to the Password field. If not, you must re-enter the new password and verify it again. If you make a mistake while entering the password or want to leave it unchanged, select **Cancel**.



NOTE: To provide system security, ***all*** mailboxes should have a password. To make the passwords difficult to guess, they should not match the mailbox number or consist of one digit repeated several times. The default password should be changed the first time the user logs in. This is especially important in the System Administrator's mailbox which allows programming access to the Voice Processor.

Greeting

4.118 The system greeting, the mailbox primary greeting, or the mailbox alternate greeting can be enabled. The greeting selection defaults to the system greeting. To determine the greeting callers will hear when they reach the mailbox, select **Greeting** and scroll the highlight bar to the desired selection (Primary, Alternate, or System Greeting). Then press **ENTER** or click on another field to save your change.

NOTE: Even if re-programmed, the system greeting will continue to play until the greeting selected here has been recorded.

Transfer Method

4.119 *(This is used for associated mailboxes only.)* When a call is received by an automated attendant and the caller enters an extension number, the Transfer Method of the associated mailbox determines how the call will be transferred. Select the drop-down list box and scroll to the desired setting. Then press **ENTER** or click on another field to save your change. The available Transfer Methods are as follows:

- **Announce-Only**: The caller is asked to state his or her name. Then the call is transferred to the associated extension number. When the station user answers the transfer, the system plays the caller's name and completes the transfer.
- **Screened**: The caller is asked to state his or her name. Then the call is transferred to the associated extension number. When the station user answers the transfer, the system plays the caller's name. The station user has the options of replaying the name, sending the call to voice mail (if the extension has a mailbox), transferring the call to another extension, accepting the call, or rejecting the call.
- **Unannounced**: The call is transferred to the associated extension number after the system checks the station to determine its status (busy, available, ringing, etc.).

Message Notification Station

4.120 *(Not programmable for associated mailboxes.)* Each mailbox has a station that will receive message notification whenever the mailbox receives a message. This is usually the same number as the mailbox. However, when a mailbox is shared by several stations, or belongs to a hunt group pilot number, a specific station must be designated to receive the message notification. If non-associated mailboxes are programmed, you must disable the Validate Voice Mailbox Number feature (see System Flags on page 6-258) to allow station users to dial the non-associated mailbox numbers. To determine the station that will receive message notifications, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the existing extension number. An option box appears.
2. Select **Message Notification Station**. A window appears that asks you to select the type of device to include.
3. Select the type of device and then select **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the appropriate station and then select **Finish**. Your selection appears in the Message notification Station field.

Time Zone

4.121 If the mailbox is located in a different time zone than the Voice Processing Unit, you will want to set the Time Zone field to match its location. This will allow the timestamp on the voice mail messages to reflect the correct time for that mailbox's location. To set the time zone, select the drop-down list box and scroll to the desired setting. Then press **ENTER** or click on another field to save your change.

Automatic Speech Recognition (ASR) Setting

4.122 *(EM systems only)* Determines whether the system will use the ASR Enabled setting in the current folder level or in the next folder level. For details about this field, see page 6-348.

Automatic Speech Recognition (ASR) Enabled

4.123 *(EM systems only)* Determines whether or not ASR is enabled for the current folder (and any subfolders or parent folders based on the ASR Setting field). If enabled, the system, application, or mailbox supports voice recognition for accessing that particular feature. If disabled, the system, application, or mailbox will not support voice recognition. For details about this field, see page 6-348.

NOTE: The ASR Enabled field is ignored (that is, displays a red "x") if the ASR Setting is set to Delegated, Ignored, or Ignored and Delegated.

Quota Warning

4.124 Select the threshold that must be met before the system generates a warning for the user. This value is set as a percentage of the Maximum Mailbox Message Capacity (under Message Limits). For example, if this value is set to 80 and the Maximum Mailbox Message Capacity value is set to 30 minutes, a warning message is issued when the number of voice mail messages totals 24 minutes (80% of 30 minutes). This warning prompt is then played each time the user accesses their mailbox. The warning prompt is no longer played once the mailbox storage total falls below the quota warning threshold. The valid range is 0 - 100 percent, and the default is *80*. If set to 0 or 100, no warning message is issued. To set the value, select the drop-down list box and scroll to the desired setting. Then press **ENTER** or click another field to save your change.

Quota Grace

4.125 Select the amount of additional storage that the system allows once the Maximum Mailbox Message Capacity limit (under Message Limits) is met. This value is also set as a percentage of the Maximum Mailbox Message Capacity. To determine the point at which messages are denied, the system adds the quota grace value to the total capacity limit. For example, if this value is set to 80 and the Maximum Mailbox Message Capacity is set to 30 minutes, the system will not store messages once the number of voice mail messages totals 54 minutes (30 + .8 x 30). When this capacity is reached, a voice mail prompt informs the user that their mailbox is currently full. The valid range is -1 - 100 percent, and the default is *0*, which indicates that no grace is allowed. For unlimited messages, set this value to -1. To set the value, select the drop-down list box and scroll to the desired setting. Then press **ENTER** or click another field to save your change.

F. VOICE PROCESSOR NODES

NOTE: The Voice Processor AMIS Networking or Inter-Tel Messaging IP Networking software license is required to use Voice Processor Nodes. See page 6-97.

4.126 See page 4-8 in VOICE PROCESSING FEATURES for a complete description of Voice Processor networking. The Node programming area is used to identify and define information about each node in the Voice Processor network. Before the local Voice Processor can deliver messages to another Voice Processor node in the network, the remote node must have an entry in this table.

4.127 Each node has its own internal message queue (similar to a mailbox's message queues) and will store messages destined for other nodes until they are delivered. There can be up to 100 nodes in the Voice Processor network.

4.128 To create a node: Right-click in the right side of the screen and select **Create Node**. DB Programming will have to close to reset. When you restart the session, the new node will appear in the Nodes folder.

4.129 Each node is programmed with the following fields:

- **Number**: The network node number is a number between PP200 and PP299. The number is assigned automatically when the node is created and cannot be changed.
- **Description**: The Description field is used to provide a meaningful name for the node (such as, "Phoenix" or "Houston"). This field can contain up to 20 characters. The default for this field is "Node xxx". For example, node 1 defaults to "Node 001" and node 10 defaults to "Node 010." Enter the desired description in the text box and then press **ENTER** or click on another field to save the change.

- **System Number**: The System Number field is either a telephone number or an IP address. Enter the appropriate number in the text box and then press **ENTER** or click on another field to save the change. If the Network Type field is set to:
  - — *CO AMIS*, the System Number field is a telephone number used to connect to the remote node. This telephone number should ring directly into the remote Voice Processor.
  - — *TCPIP*, the System Number field should be an IP address.
  - — *IC AMIS*, the System Number field is not programmable.

NOTE: The ESMTP Network Type is reserved for future EM use.

- **Extension**: The Extension field is available only when the Network Type field is IC AMIS. Right-click on the field and select **Change Extension**. Then program the station, CO trunk group, or hunt group extension.
- **Network Type**: The Network Type field identifies how the local node communicated with the remote node. This field can be programmed to None, IC AMIS, CO AMIS, or TCPIP. Select the drop-down list box and scroll to the desired setting. Then press **ENTER** or click on another field to save the change.
  - — **None** indicates that the node will be unavailable for use.
  - — **E-mail** is a proprietary network protocol that uses e-mail as the mode of delivery. When a message is left for a network mailbox configured under an "E-Mail" node, that message is constructed into an e-mail and sent to an e-mail address that another system is monitoring. The protocol allows for greeting and password updates as well as general message delivery. In a way it is similar to VPIM except that VPIM does not support greeting and password updates.
  - — **AMIS** (CO or IC) is an analog networking protocol that uses the public telephone network. AMIS is not supported in Enterprise Messaging.
  - — **TCPIP** is a direct network connection between the local Voice Processor and the remote Voice Processor.
  - — **VPIM** (Voice Profile for Internet Mail) allows Enterprise Messaging v2.1 or later to communicate with other voice processing systems. VPIM networking requires the Inter-Tel Messaging IP Networking premium feature.

NOTE: To receive and send e-mail across the Internet to a remote site, Enterprise Messaging needs access outside its local network. Information needs to come into port 25 on the Inter-Tel Axxess. This may require changes to any firewall between the Inter-Tel Axxess and the Internet.

Keep the following considerations in mind:

- – All of the fields under a VPIM networked node have a red "X" because they do not apply.
- – You can create up to 200 remote VPIM nodes. The number of nodes can be any mix of VPIM or any other node type. For example, if you have 199 TCP/IP nodes, you can have only 1 VPIM node.
- – The value in the System Number/Domain field is the VPIM Home Domain of the node, shown below, and this field is required for the node.

4.130 To program a specific node, double-click on it for access to these programming areas:

- **Non-Inter-Tel Node**: If necessary, you can connect an Inter-Tel Voice Processor to another vendor's voice mail unit using the AMIS protocol. When this flag is programmed to **No**, it means that the remote node is an Inter-Tel Voice Processor. When this flag is programmed to **Yes**, it means that the node is another vendor's Voice Mail. To indicate that the remote node is a non-Inter-Tel product, select the check box to place a check mark in it and change the setting to **Yes**. Then press **ENTER** or click on another field to save your change. To set it back to **No**, select the check box again.
- **Start and Stop Times**: These fields specify the time of day that messages will be delivered to the remote node. If the start time and stop time are the same, it indicates that deliveries are valid the entire day. The default value for this field is 8:00am to 8:00am. To set the time, select the current Value. Then select the hours, minutes, or AM/PM field, and use the arrows to scroll to the desired setting. Select another field or press **ENTER** to save your change.
- **Days of the Week**: These fields specify the days of the week that messages will be delivered to the remote node. Any combination of the days of the week is valid. To enable a specific day of the week, select its check box to place a check mark in it and change the setting to **Yes**. Then press **ENTER** or click on another field to save your change. To set it back to **No**, select the check box again.
- **Timers and Limits**: Double-click on **Timers and Limits** to view the list. To program a timer, click on its Value and enter the new information in the text box. Then press **ENTER** or click on another field to save your change. The following timers and limits can be programmed for each node.
  - — **Latency Time**: This specifies how long the Voice Processor will wait between successful outgoing calls to the remote node if there are pending messages for the remote node. This field is only in effect if the current time on the Voice Processor is within the remote node's delivery time/date settings. The range for this field is 0 to 1440 minutes (24 hours). When this field is programmed to 0, it indicates that the Voice Processor will deliver messages immediately to the remote node. In other words, as soon as the local node receives a message that is destined for the remote node, the Voice Processor will attempt to connect to the remote node and deliver the message (provided that the time is within the remote node's delivery time/date settings). If this field is something other than 0, it specifies the amount of time that a message will remain pending on the local node before the Voice Processor attempts to deliver it to the remote node. For example, if this field is programmed to 60 minutes, the next network call to the remote node will occur 60 minutes after the last successful call to the node. The default value for this field is 30 minutes.
  - — **Priority Latency Time**: This specifies how long the Voice Processor will wait between outgoing calls to the remote node if there are pending priority messages for the node. This field is only in effect if the current time on the Voice Processor is within the remote node's delivery start and stop date/time settings. The range for this field is 0 to 1440 minutes (24 hours). When this field is programmed to 0, it indicates that the Voice Processor will deliver priority messages immediately to the remote node. Any time the Voice Processor delivers pending priority messages, it will also deliver any non-priority messages that are pending for the remote node. Note that this field takes precedence over the Latency Time field. The default value for this field is 5 minutes.

— **Message Threshold**: This specifies the number of messages that must exist in the node's message queue to force the Voice Processor to place a call to the remote node. This field is only in effect if the current time on the Voice Processor is within the remote nodes delivery time/date settings. This field can be used to ensure that the local Voice Processor does not get too backed up with messages destined for the remote node. If this field is programmed to 1, every message the local node receives that is destined for the remote node is delivered as soon as it is received by the local node. If this field is programmed to another number, it means that as soon as the remote node's message queue has that many messages pending, the Voice Processor will attempt to connect to the remote node and deliver the messages. Note that this field takes precedence over both the Latency Time and Priority Latency Time fields. The range for this field is 0-100 messages and the default value for this is 5 messages.

5. E-MAIL GATEWAY

5.1 To read a complete explanation of the E-Mail Gateway and the Unified Messaging OSE features, refer to the latest version of the *Unified Messaging Open Standards Edition Administrator's Guide* (document part no. 835.3162).

5.2 For the E-Mail Gateway to function, its service must be able to log on to the network as a user. Program the E-Mail System field first, as described below.

5.3 To use the TTS functionality to convert e-mail messages to voice mail messages, the E-mail Gateway must be configured and each mailbox user must have an e-mail reader profile. This profile contains the information that EM requires to access the correct e-mail server and e-mail account.

5.4 **E-Mail System**: This system-wide field specifies the type of e-mail system that will be used to transfer messages. The value programmed in this field must correspond to the customer's underlying e-mail system. For example, if the customer's e-mail system is Lotus Notes, this field should be programmed to VIM. This field can be programmed to NONE, MAPI, VIM, or SMTP/POP3. Note that for an EM system, the system only supports SMTP/POP3 E-mails systems. If it is programmed to NONE, the E-Mail Gateway feature is disabled for the entire voice mail system. The default value for this field is NONE. To select the E-Mail system, select the drop-down list box and scroll to the desired value. Then press **ENTER** or click on another field to save the change.

5.5 The other fields that need to be programmed depend on the E-Mail System selected:

- If the E-Mail System field is programmed to MAPI or VIM, only the E-Mail Username and E-Mail Password fields need to be programmed.
- If the E-Mail System is programmed to SMTP/POP3, the E-Mail SMTP Server and E-Mail Address fields must also be programmed, in addition to the username and password fields. The E-Mail Real Name field is optional when using SMTP/POP3.

5.6 **E-Mail Address**: This field specifies the Voice Processing Unit's e-mail address. It is only required if the E-Mail System field is programmed to SMTP/POP3. When the voice mail PC sends an e-mail message, this address will be used as the e-mail address in the "Reply-To" field of the e-mail header. This is the address at which the voice mail PC will receive e-mail messages. This field can contain up to 127 characters and it is empty by default. For example, this field might look like johndoe@inter-tel.com. To enter the address click on the Value and enter the address in the text box. If you enter an invalid character (: ; " \ | ( ) , < > ‘), an error tone occurs. Then press **ENTER** or click on another field to save your change.

NOTE: While this address can contain a "familiar" name, using SMTP, this should be a properly formatted E-mail address to avoid presenting a Reply-To e-mail address without a domain name and potentially being perceived as spam.

5.7 **E-Mail Real Name**: This field specifies the Voice Processing Unit's user name (such as VOICE MAIL). It is only programmable if the E-Mail System field is programmed to SMTP/POP3 and it is optional. When the voice mail PC sends an e-mail message, this name will be included in the FROM field of the e-mail header. This field can contain up to 127 characters, and it is empty by default. To enter the name, click on the Value and enter the address in the text box. Then press **ENTER** or click on another field to save your change.

NOTE: EM takes the domain part of the "Administrator E-Mail Address" and uses that to suffix the "real name" if a properly formatted E-mail is not specified.

5.8 E-Mail Username: This field specifies the username for the voice mail PC's e-mail account. Before the voice mail PC can send or receive e-mail messages, it must log on to the underlying e-mail system. Therefore, the voice mail PC must have an account on the customer's e-mail system, and this field specifies the username for that account. This field can contain up to 127 characters and it is empty by default. To enter the username click on Value and enter the address in the text box. Then press **ENTER** or click on another field to save your change.

NOTE: Currently, EM does not support SMTP authentication. This field is used for MAPI/VIM on the VPU.

5.9 E-Mail SMTP Server: This field specifies the Voice Processing Unit's SMTP mail server. It is only programmable if the E-Mail System field is programmed to SMTP/POP3. The SMTP mail server is the server that the voice mail PC connects to in order to send E-Mail messages. This field can contain up to 127 characters and it is empty by default. To enter the server click on the Value and enter the address in the text box. Then press **ENTER** or click on another field to save your change.

NOTE: The format should be a dotted-decimal IP address or fully qualified SMTP server name to avoid any DNS complications.

5.10 E-Mail POP Server: This field specifies the POP mail server that will be used to receive e-mail messages. It is only programmable if the E-Mail System field is programmed to SMTP/POP3. The POP mail server is the server the voice mail PC connects to in order to **retrieve** e-mail messages through the Internet. This field can contain up to 127 characters and it is empty by default. To enter the server click on the Value and enter the address in the text box. Then press **ENTER** or click on another field to save your change.

5.11 Administrator E-Mail Address: This field specifies the e-mail address of the System Administrator. It will alert the administrator of any problems with the e-mail addresses. Whenever an e-mail message is undeliverable (due to invalid or wrong e-mail address, etc.) it will be forwarded to this address. Or, if a user attempts to reply to a E-Mail Gateway e-mail message, the message will go to this address instead of going to the Voice Processing PC. This field can contain up to 127 characters and it is empty by default. To enter the address click on the Value and enter the address in the text box. Then press **ENTER** or click on another field to save your change.

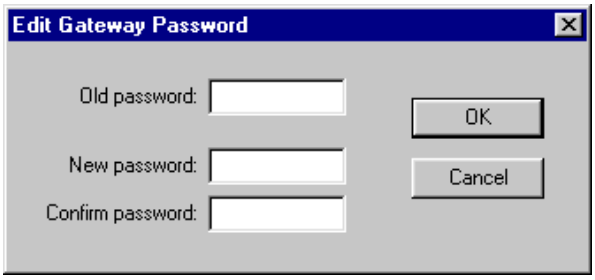
NOTE: Even though this field is not required, Inter-Tel strongly recommends that you provide an e-mail address to forward any e-mail received by the Voice Processing PC. It will alert the Administrator to any problem with the e-mail addresses.

ALSO: If the E-mail System is set to anything other than "NONE," Inter-Tel strongly recommends that you provide an e-mail address to forward any e-mail received by the Voice Processing PC. Having this field blank tends to trigger spam detection.

5.12 **Gateway Password**: This field specifies the E-mail password for the voice mail PC's e-mail account. Before the voice mail PC can send or receive e-mail messages, it must log on to the underlying e-mail system. Therefore, the voice mail PC must have an account on the customer's e-mail system, and this field specifies the password for that account. This field can contain up to 40 characters and it is empty by default.

NOTE: Currently, EM does not support SMTP authentication. This field is used for MAPI/VIM on the VPU.

5.13 To program the password, right-click on the **Password** field and select **Edit Password**. A window appears as shown below. Enter the current password, if one exists. Then enter the new password in the New Password box. The characters will not appear on the screen when typed; they will appear as asterisks (***). Then retype the password exactly as before in the Confirm Password box. Select **OK** to exit and save the password. If the entered passwords match, you will return to the Password field. If not, you must re-enter the new password and verify it again. If you make a mistake while entering the password or want to leave it unchanged, select **Cancel**.



NOTE: To provide system security, the e-mail system must have a password. To make the passwords difficult to guess, they should not consist of predictable patterns, such as one digit repeated several times.

6. FAX-ON-DEMAND

6.1 The Fax-On-Demand feature provides fax services to callers. It is a specially programmed Call Routing Announcement application that uses digit translation to allow callers to select the documents they want to have faxed to them.

6.2 With Fax-On-Demand, callers can use a DTMF telephone to request one or more documents from the company's fax library. When the request is completed, the Voice Processing Unit places a call to the caller's fax machine to deliver the requested documents.

6.3 To use Fax-On-Demand the following fields must be programmed:

- Timers and Limits (see below)
- Fax Documents (see page 6-403)
- Allow International Calls (see page 6-403)
- Outgoing Access - trunk group (see page 6-404)
- Outgoing Access Prefix (see page 6-404)
- Outgoing Access Termination (see page 6-404)
- Start and Stop Time (see page 6-404)
- Days of the Week (see page 6-405)
- Format - Local Fax ID (see page 6-405)
- Format - Company Name (see page 6-405)
- Format - Logo Document (see page 6-405)

A. TIMERS AND LIMITS

6.4 Double-click on Timers and Limits to view the list of current Values. To program a timer or limit, click on its Value and enter the new information in the text box. Then press **ENTER** or click on another field to save your change. The following timers and limits can be programmed for the fax feature:

- **Fax Retry Timer**: When the Voice Processing Unit is unable to complete a fax delivery because the line is busy or there is no answer, it will wait until this timer expires to attempt the delivery again. The range for this field is 1-255 minutes. Default is 10 minutes.
- **Fax Retransmission Timer**: When the Voice Processing Unit is unable to complete a fax delivery because the connection failed (for example, the receiving fax machine had a power failure), it will wait until this timer expires to attempt the delivery again. The range for this field is 1-255 minutes. Default is 1 minute.
- **Maximum Fax Delivery Attempts**: This is the number of times the Voice Processing Unit will attempt to send a fax when the number is busy, there is no answer, or there are transmission errors. If the system encounters unavailable resources (fax ports, documents, outgoing calls, or outgoing trunks) the attempt does not count toward the Maximum Fax Delivery Attempts. The allowed range for this field is 1-15 attempts. It defaults to five attempts.

- **Maximum Fax-On-Demand Ports**: This sets the maximum number of fax ports the system can use for performing Fax-On-Demand (either delivering outgoing faxes or importing fax documents from the system administrator mailbox). By placing a limit on the number of Fax-On-Demand ports, you can reserve fax ports for receiving incoming faxes through mailboxes and Call Routing Announcement applications. For example, if the system has eight fax ports and the Maximum Fax-On-Demand Ports field is set to six, there will always be at least two ports available for faxes received through mailboxes and Call Routing and six ports for outgoing faxes.

NOTE: If the Maximum Fax-On-Demand Ports field is programmed to a number that exceeds the actual fax ports available, the software will automatically adjust the limit.

- **Automatic Header Reduction**: This tells the Voice Processing Unit how much of each document you fax into the system must be removed to erase the sender information at the top of the document. If the fax machine you use to enter the documents does not place sender information at the top of the document, you can set this field to 0. You can reduce the header 0-160 sixteenths of an inch (0-10 inches). This field defaults to 4 sixteenths of an inch (0.25 inches).
- **Fax Tone Wait Timer**: This is the amount of time the Voice Processing Unit will wait for fax tone before sending or receiving a document. If it does not receive fax tone before this timer expires, it will hang up. If the system was sending a fax, it will attempt the call again after the Fax Retry Timer expires. The range for this field is 1-255 seconds. Default is 40 seconds.

NOTE: The Fax Tone Wait Timer is not configurable for Enterprise Messaging. The system uses a value of 40 seconds.

- **Maximum Fax Selections**: This determines the number of faxes a caller can select at a time. When the caller has selected the maximum number of documents, the Voice Processing Unit prompts them for their fax telephone number. After entering the fax delivery information, the caller can then go through the process again to request more documents without hanging up and calling in again. However, each series of requests will generate a separate outgoing call from the Voice Processing Unit, each time the fax delivery information is entered by the caller. The allowed range for this field is 1-20 documents. It defaults to 10.
- **Maximum Fax-On-Demand Library Size**: This determines the amount of Voice Processing Unit hard disk space that will be allotted for storing fax documents. The allowed range for this field is 0-255 megabytes; however, the actual maximum depends on the available disk space on the Voice Processing Unit hard drive. It defaults to 0 megabytes and must be programmed to a higher value before any fax documents can be imported. If the library size is reduced while there are documents stored in the library, the reduction will not affect existing documents (even if they now exceed the new maximum). However, new documents cannot be added unless there is sufficient disk space available.
- **Current Fax-On-Demand Library Size And Percentage Of Maximum Currently Used**: This shows the amount of disk space that is currently being used by the documents in the fax library and the percentage of the Maximum Fax Library Size that has been used. These fields cannot be programmed. They are shown for reference only.

B. FAX DOCUMENTS

6.5 When you double-click on **Fax Documents**, the documents currently existing in the system are shown in the list. You can program the number and description of a document as follows:

- **Document Number**: Enter a number, up to four digits, to identify the fax document in the text box. Then press **ENTER** or click on another field to save your change. This number will be used by programmers to select the destination for digit translations and by callers when selecting fax documents by number.
- **Description**: This is the abbreviated description that is used in programming screens. It can have up to 20 characters. Enter the description in the text box. Then press **ENTER** or click on another field to save your change.

6.6 If you double-click on a document, you will see these additional programming fields:

- **Detailed Description**: This is the description that appears on the fax cover sheet and in any fax programming reports. It can include up to 40 characters. Enter the description in the text box. Then press **ENTER** or click on another field to save your change. If you do not enter a detailed description for the document before exiting, you will receive a warning message.
- **Statistics**: This information cannot be programmed. It is shown as a reference to indicate the following:
  - — **Number of Requests**: How many times the document has been requested, since statistics were last cleared (see page 6-418).
  - — **Last Request**: The last date that the document was requested, since statistics were last cleared. This includes requests that could not be delivered because of transmission errors.
  - — **Last Modification**: The date and time of the last modification to the document being programmed, since statistics were last cleared. Or, if the document has not been modified, it shows the time that it was imported.
  - — **Pages**: The number of pages included in the document.
  - — **Image File Size**: The amount of disk space occupied by this document.

C. ALLOW INTERNATIONAL CALLS

6.7 This field allows you to enable or disable international fax deliveries. If it is disabled, the prompt asks the caller to enter a 10-digit telephone number for their fax destination. If this field is enabled, the prompt gives international dialing instructions and allows international and domestic telephone numbers. The field defaults to *disabled.*

6.8 To allow international calls, select the check box to place a check mark in it to change the setting to **Yes**. Then press **ENTER** or click on another field to save your change. To set it back to **No**, select the check box again.

D. OUTGOING ACCESS

6.9 The next three fields control the outgoing access for calls placed by the Fax-On-Demand application.

6.10 **Outgoing Access**: This is the outgoing trunk access code that the Message Notification/Retrieval application will use to place the outgoing fax delivery call. This field is blank by default. To determine the outgoing access trunk group, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the existing value. An option box appears.
2. Select **Change Outgoing Access**. A window appears that asks you to select the type of device to include.
3. Select **CO Trunk Group** and then select **Next**. The list of trunk groups appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the appropriate trunk group and then select **Finish**. Your selection appears in the Outgoing Access field.

6.11 **Outgoing Access Prefix**: This is the digit dialed by the Voice Processing Unit before the outgoing access code, if needed (e.g., a forced account code). This field can include up to 18 characters. Valid characters include digits 0-9, * and #, and "P" for pause. This field is blank by default. To enter the prefix, click on the Value and enter the address in the text box. Then press **ENTER** or click on another field to save your change.

6.12 **Outgoing Access Termination**: This is the digit dialed by the Voice Processing Unit to terminate the outgoing access code, if needed. This field can include up to 18 characters. Valid characters include digits 0-9, * and #, and "P" for pause. This field defaults to #. To enter the termination digit, click on the Value and enter the address in the text box. Then press **ENTER** or click on another field to save your change.

E. START/STOP TIME

6.13 These fields determine the time period during which faxes will be sent to callers. If a caller requests a fax after the Stop Time, the system will not send the fax until the Start Time. To provide 24-hour fax service, set both fields to the same value. Both fields default to 5:00 PM. To set the time, select the current Value. Then select the hours, minutes, or AM/PM field, and use the arrows to scroll to the desired setting. Select another field or press **ENTER** to save your change.

NOTE: With a U.S. system, the Fax-On-Demand feature uses User Group 1 and the Home Area Code for toll restriction information. Ensure that these are programmed correctly for outgoing fax delivery calls. Also ensure that the Message Notification/Retrieval application has the appropriate trunk access.

F. DAYS OF THE WEEK

6.14 These fields set the days of the week during which faxes will be sent to callers. If a caller requests a fax on a disabled day, the system will wait for the next enabled day to send the fax. At default, all days of the week are enabled. You cannot disable all seven days; at least one must be selected. To enable a specific day of the week, select its check box to place a check mark in it and change the setting to **Yes**. Then press **ENTER** or click on another field to save your change. To set it back to **No**, select the check box again.

G. FAX FORMAT

6.15 The following fields affect the format of the fax documents that are sent by the Fax-On-Demand application.

- **Local Fax ID**: The Local Fax ID appears at the top of each page sent by the Voice Processing Unit. The ID should include the telephone number of the Voice Processing Unit and/or the company name. The ID can be up to 20 characters. Valid characters include upper-case letters, digits 0-9, and the plus sign (+). To enter the Fax ID click on the Value and enter the address in the text box. Then press **ENTER** or click on another field to save your change.
- **Company Name**: The system places Company Name on the "From" line of the fax cover sheet. The company name can include up to 40 characters. Any character is allowed. To enter the company name click on the Value field and enter the address in the text box. Then press **ENTER** or click on another field to save your change.

6.16 **Logo Document**: After you have imported your company logo, you can assign that fax document as the Logo Document for your fax cover sheets. (See page 4-61 for fax importing instructions.) Any fax document in the database can be designated as the logo document. The logo can be up to 5.5 inches tall. If desired, you can choose not to have a logo on your cover sheets by selecting **None**. To indicate which fax document is the logo document, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the existing value. An option box appears.
2. Select **Change Format** and then **Logo Document**. A window appears that asks you to select the type of device to include.
3. Select **Fax Document** (or select **None Device**, if you do not want to use a logo document) and then select **Next**. The list of currently existing documents appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the appropriate document (or None) and then select **Finish**. Your selection appears in the Logo Document field.

7. MAILBOX-RELATED INFORMATION

7.1 When you click on **Mailbox-Related Information**, the following statistics are displayed.

7.2 Statistics continue to accumulate until cleared using the Voice Processor Report Programming window.

7.3 This information is presented for reference only and cannot be programmed.

- **Number of Mailboxes**: This shows the number of mailboxes that have been created in voice mail.
- **Number of Messages Sent**: This is a count of the number of times subscribers have recorded and sent messages (to one mailbox or a group list of mailboxes), replied to a message sent by another subscriber, or forwarded a message with comments.
- **Number of Messages Received**: This shows the number of messages that subscribers have received regardless of where the messages came from (subscriber, non-subscriber, or system).
- **Number of Messages Received From Remote Nodes**: This shows the number of messages that subscribers have received from stations on other nodes.
- **Number of New Messages**: This is the total of the number of messages in all subscribers' new message queues.
- **Number of Saved Messages**: This is the total of the number of messages stored in the saved message queues for all mailboxes.
- **Total Length of New and Saved Messages**: This is a combined total of the amount of time represented by the "Number of New Messages" and "Number of Saved Messages" fields.
- **Average Mailbox Percent Full**: This shows the average percentage of maximum message capacity used by all mailboxes.
- **Number of Times Mailbox Was Full**: This displays the number of times any mailbox reached its maximum message capacity.
- **Number of Times Mailbox Was 80% Full**: This shows the number of times any mailbox reached 80% of its maximum message capacity.
- **Number of Times 3 Bad Passwords Were Entered**: This counter increases each time a single call includes three attempts to enter a mailbox and the caller uses an incorrect mailbox/password combination.
- **Number of Mailboxes Currently Full**: This shows the number of mailboxes that are currently at their maximum capacity.
- **Number of Mailboxes Currently More Than 80% Full**: This shows how many mailboxes are currently over 80% of their maximum message capacity.

8. VOICE PROCESSOR MAINTENANCE

8.1 This section describes the various maintenance types you can perform on the Voice Processor.

A. MAINTENANCE FLAG PROGRAMMING

NOTE: The Maintenance folder and its sub-folders are not supported for Enterprise Messaging. Automated tasks such as backups, shutdowns, and reboots are handled by the Windows Scheduled Tasks and the OAM console for Enterprise Messaging.

8.2 The Maintenance fields allow you to enable automatic scheduling, enable diagnostics, and schedule an automatic reboot and automatic maintenance. Maintenance fields include:

- Auto Reboot (see below)
- Auto Maintenance (see below)
- Enable Diagnostics (see page 6-408)
- Enable Auto Schedule (see page 6-409)
- Auto Schedule Start Time (see page 6-409)

8.3 A program planning sheet that lists the Maintenance fields is included on the system software CD-ROM, Documentation folder.

8.4 You can schedule automatic maintenance and automatic reboots with the Auto Reboot and Auto Maintenance fields. Scheduling these functions ensures that routine maintenance is done on a regular basis and does not conflict with the customer's busiest hours.

NOTE: Automatic maintenance and reboots also require you to set the Enable Auto Schedule field to **Yes**, and set a time for the Auto Schedule Start Time field (see page 6-409).

Auto Reboot

8.5 The Auto Reboot field allows you to schedule which days of the week an automatic shutdown and reboot are performed. Double-click **Auto Reboot** to access the associated days of the week.

Auto Maintenance

8.6 The Auto Maintenance field allows you to schedule the days of the week that automatic maintenance is performed. Double-click **Auto Maintenance** to access the associated days of the week.

8.7 **To enable the days of the week for Auto Reboot and Auto Maintenance**:

1. Select the existing **Value**.
2. Place a check mark in the check box.
3. Press **ENTER** or click another field to save the change.

NOTE: To disable a day of the week, remove the check mark from the check box.

8.8 The type of automatic maintenance performed is up to you. With Windows-based VPU, you can use the Watchdog Configuration Utility (located in the **avdap\utility** directory) to configure automatic backups of the Voice Processor database. You also have the option of creating a batch file to perform this task and/or other custom maintenance tasks. When you create the batch file, name it **system_maintenance.bat** and place it in the **winnt/system32** directory.

8.9 An example of a simple batch file that will copy the entire db\ directory to a directory called backup\avdapdb\ is as follows. (Contact Inter-Tel for more complex examples of batch files.)

```
net stop avdap
del /s /f /q c:\backup\avdapdb
rd /s /q c:\backup\avdapdb
md c:\backup\avdapdb
xcopy /s /e /v c:\avdap\db c:\backup\avdapdb
net start avdap
```

8.10 With Windows-based version, Voice Processing first checks the Auto Reboot and Auto Maintenance fields.

- If the current day is set for Auto Maintenance, the software watchdog will run the Auto Maintenance batch file. When Auto Maintenance is finished, the software watchdog will check if Auto Reboot is enabled for that day. If so, it will shut down Windows-based VPU and reset the PC. If the current day is not set in the auto reboot field, the software watchdog will just restart the Avdap (Voice Processing) service.
- If the current day is not set in the Auto Maintenance field but it is set in the Auto Reboot field, the software watchdog will shut down both the Avdap service and Windows-based VPU, and reboot the PC.

8.11 Note that you can program either Auto Maintenance, Auto Reboot, or both. However, on a day where both fields are programmed, the software watchdog will run Auto Maintenance first and then Auto Reboot.

Enable Diagnostics

8.12 When the Enable Diagnostics option is selected, Voice Processing logs all diagnostics output (including alarms) generated by the Voice Processing PC to a file. Using that file, you can troubleshoot problems dealing with message lamps, delayed messages, remote messaging, etc. By default, this flag is disabled.

8.13 There are five diagnostics log files: log, log.1, log.2, log.3 and log.4. Information is initially written to the log file, when this file becomes full its contents are transferred to the log.1 file. The log.1 file's contents are transferred down to the log.2 file, whose contents are transferred to the log.3 file, whose contents are transferred to the log.4 file. The information contained in log.4 is discarded permanently each time log.3 is transferred.

8.14 Diagnostics may be left on at all times without affecting performance. The number and size of the log files can be changed.

NOTE: Refer to the *Voice Processing Unit Diagnostics Manual* (part number 550.8018—supplied on the System software CD) for a complete list of diagnostic codes.

8.15 To select the Enable Diagnostics option:

1. Select the existing **Value**.
2. Place a check mark in the check box.
3. Press **ENTER** or click another field to save the change.

NOTE: To disable this option, remove the check mark from the check box.

Enable Auto Schedule

8.16 The Enable Auto Schedule field specifies whether the automatic system reboot and/or maintenance is enabled or disabled. By default, this flag is disabled.

8.17 To select the Enable Auto Schedule option:

1. Select the existing **Value**.
2. Place a check mark in the check box.
3. Press **ENTER** or click another field to save the change.

NOTE: To disable this option, remove the check mark from the check box.

Auto Schedule Start Time

8.18 Auto Schedule Start Time: The Auto Schedule Start Time field specifies the time of the day when the automatic system maintenance will be performed (on the days chosen in Auto Reboot or Auto Maintenance). The Auto Schedule Start Time field is programmable only if the Enable Auto Schedule flag has been enabled.

8.19 To program the Auto Schedule Start Time field:

1. Select the existing **Value**.
2. Set the time using the scroll box.
3. Press **ENTER** or click another field to save the change.

8.20 For information on how to set up and perform automatic Voice Processor database backups, see below.

B. VOICE PROCESSOR DATABASE BACKUP WITH WINDOWS-BASED VPU

8.21 This section describes the procedure for performing automatic Voice Processor database backups under Windows-based VPU. These backups can be written to a local drive, a network drive, or to an attached mass-media device. The Voice Processor is not functional for the duration of the backup. When using a network drive, you can configure the Voice Processor to backup to a local drive first to minimize downtime.

8.22 To configure the system to perform automatic backups:

1. Start the Watchdog Configuration Utility.
2. Click on the Maintenance tab.
3. Enable the Maintenance Enabled flag (a check mark appears).
4. Enter the time of day when the backups will be performed in the Maintenance Time field.

5. Select the days of the week to perform the maintenance in the Maintenance Days field.

NOTE: You can perform Steps 2 through 5 through the DB Programming Maintenance menu (Voice Processor\Maintenance).

6. If saving to a local drive:

   a. Clear the Temporary Local Save Directory field.

   b. Set the Voice Processor Database Save Directory field to the location on your local PC where the Voice Processor backup will be restored.

   If saving to an attached mass-media device:

   a. Set the Temporary Local Save Directory field to the location on your hard drive where the Voice Processor backup will be temporarily stored during the backup operation (e.g., **`d:\backup`**). Although this step is optional, it will help to substantially reduce downtime during backups to mass-media drives. The backup files will first be copied to the local hard drive and then to the appropriate destination on the mass-media device. This allows the Voice Processor to start as soon as the backup to the local drive is complete.

   b. Set the Voice Processor Database Save Directory field to the location on the mass-media device where the Voice Processor backup will be stored.

   If saving to a network drive:

   a. Set the Temporary Local Save Directory field to the location on your hard drive where the Voice Processor backup will be temporarily stored during the backup operation (e.g., **`d:\backup`**). Although this step is optional, it will help to substantially reduce downtime during backups to network drives. The backup files will first be copied to the local hard drive and then to the appropriate destination on the network device. This allows the Voice Processor to start as soon as the backup to the local drive is complete.

   b. Set drive B: as the Voice Processor Database Save Directory.

   c. Start the Avdap Configuration Utility.

   d. Select the Others tab.

   e. Specify the network destination path in the Map B: Drive field.

   f. Start DB Programming and connect to the system.

   g. Under Voice Processor, set the Windows Networking Username, Windows Networking Domain, and Windows Networking Password to the values required for the Voice Processing Unit to access the network PC where the network drive is located (see page 6-351 for details about programming these fields).

9. NETWORKING

9.1 The Network programming areas include the following fields. each is described in detail in the following pages.

- AMIS (see below)
- Timers and Limits (see page 6-415)
- Validate Off-Node Mailboxes (see page 6-416)
- Undeliverable Messages Destination Type (see page 6-417)

A. AMIS

NOTE: The Voice Processor AMIS Networking software license is required to use this programming area. See page 6-97 for information.

9.2 AMIS (Audio Messaging Interchange Specification) is a vendor-independent protocol that describes how to network voice mail systems. AMIS has a digital protocol and an analog protocol. The Voice Processor supports AMIS-Analog which uses DTMF control signaling and analog message transmission.

NOTE: AMIS networking is not supported in Enterprise Messaging.

9.3 Call Routing Announcement applications can be programmed to detect and process incoming AMIS calls. For a complete explanation of AMIS networking, see page 4-12 in VOICE PROCESSING FEATURES.

9.4 The following outline can be used as a guide for setting up AMIS networking.

1. Make sure the Voice Processor AMIS Networking software license is enabled.
2. There must be a Message Notification/Retrieval Application programmed with outgoing access. It will be used for placing outgoing AMIS calls.
3. The Voice Processor's IC or CO System Number field should be programmed. This is the field the Voice Processor uses to identify itself when establishing an AMIS session. It allows the remote AMIS system to return calls to the local Voice Processor.
4. At least one Voice Processor Network Node must be programmed for AMIS networking.
5. There must be at least one network mailbox on the local Voice Processor for a mailbox that is located on the network node programmed for AMIS networking. For example, if Voice Processor Node 5 is programmed for AMIS networking, then at least one network mailbox on the local Voice Processor must be created on Node 5.
6. If using a Call Routing Announcement application, the application must have the Accept AMIS Calls flag enabled. If the application does not have this flag enabled, it will not be able to recognize DTMF tone "C" which is transmitted by the originating system to indicate the start of an AMIS call.
7. If using a Call Routing Announcement application, the application cannot have a greeting programmed, and it must have at least a 15 second pause. The 15 second pause allows the application to be used for both Inter-Tel nodes and non-Inter-Tel nodes as well as allowing it to work for calls transmitted over both the private and the public network.

8. The Accept Network Calls flag must be enabled to allow the Call Routing Application, hunt group, CO trunk group, or station to process AMIS calls. If this flag is not enabled, incoming AMIS calls will time-out.
9. The Accept Unknown Network Calls flag must be enabled or there must be a Voice Processor node for each remote node that will be sending calls to the local Voice Processor.

9.5 The AMIS networking programming fields are described individually on the next page.

Timers and Limits

9.6 Double-click on **Timers and Limits** to view the list of current Values. To program a timer or limit, click on its Value and enter the new information in the text box. Then press **ENTER** or click on another field to save your change. The following timers and limits can be programmed for the AMIS networking feature:

- **DTMF Digit Duration Timer**: This field specifies the DTMF digit duration used by the Voice Processor when dialing an AMIS call. (It only applies to the DTMF tones dialed by the local Voice Processor on an AMIS call after the call is answered. It does not apply to the digits dialed in order to establish the AMIS call.) This field can range from 80 to 500 milliseconds. The default value is 80 milliseconds.
- **DTMF Inter-Digit Delay Timer**: When the Voice Processor dials DTMF tones during an AMIS call, this field determines the length of the delay between each digit. This field can range from 80 to 5000 milliseconds (e.g., 5 seconds). The default value is 80 milliseconds.

IC System Number (Link)

9.7 The AMIS IC System Number is the extension number of the associated Call Routing Announcement application. The Voice Processor uses this number to identify itself when it calls other AMIS voice mail systems over the private network, either through the B-channels or ordinary T1 or E1 circuits. It is also the number that the remote AMIS systems will use to call the local Voice Processor. This field need not be programmed before the local Voice Processor can make outgoing AMIS calls. However, the remote AMIS system should be able to return calls to the local Voice Processor.

9.8 To determine the IC System Number, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the existing value. An option box appears.
2. Select **Change IC System Number**. A window appears that asks you to select the type of device to include.
3. Select the device type and then **Next**. The list of devices appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the appropriate application and then select **Finish**. Your selection appears in the IC System Number field.

CO System Number (Link)

9.9 The AMIS CO System Number is the telephone number the Voice Processor uses to identify itself when it calls other AMIS voice mail systems over the public telephone network. This is also the telephone number that the remote AMIS systems will use to call the local Voice Processor. Therefore, this number must ring directly into the Call Routing Announcement that is programmed to accept AMIS calls.

9.10 This number should contain a toll field, area code (U.S. only), and local telephone number. If it does not contain all three, the remote AMIS system may not be able to return calls to the local Voice Processor. An example of the AMIS CO System Number could be 1#602#9619000# [or 011 62#903000# in Europe]. The delimiter, represented by the #, is required to separate the country code, area code, and local telephone number, and must always be present. The delimiter must follow each component of the number.

9.11 This field need not be programmed before the local Voice Processor can make outgoing AMIS calls. However, you should program it so that the remote AMIS system can return calls to the local Voice Processor. To enter the CO System Number, click on its current Value and enter the new value in the text box. Then press **ENTER** or click on another field to save your change.

Outgoing Access

9.12 The next three fields control the outgoing access for calls placed by AMIS.

9.13 **Outgoing Access**: This is the outgoing trunk access code that the Voice Processor will use to place the outgoing AMIS call. This field defaults to ARS (92000). To determine the outgoing access trunk group, use one of the following methods:

Method A

1. Select the current Value and enter the new value in the text box.
2. Press **ENTER**. A screen appears showing you what is associated with the number you entered.
3. Select **OK**. The new number appears in the field.

Method B

1. Right-click on the existing value. An option box appears.
2. Select **Change Outgoing Access**. A window appears that asks you to select the type of device to include.
3. Select **CO Trunk Group** and then select **Next**. The list of trunk groups appears. (You can view them in a list by selecting the List button or view details by selecting the Details button.)
4. Highlight the appropriate trunk group and then select **Finish**. Your selection appears in the Outgoing Access field.

9.14 **Outgoing Access Prefix**: This is the digit dialed by the Voice Processor before the outgoing access code, if needed. This field can include up to 18 characters. Valid characters include digits 0-9, * and #, and P for pause. This field is blank by default. To enter the prefix click on the Value and enter the address in the text box. Then press **ENTER** or click on another field to save your change.

NOTE: For AMIS networking purposes, the first digit of the Outgoing Access Prefix should always be a pause.

9.15 **Outgoing Access Termination**: This is the digit dialed by the Voice Processor to terminate the outgoing access code, if needed. This field can include up to 18 characters. Valid characters include digits 0-9, * and #, and "P" for pause. This field defaults to #. To enter the termination digit click on the Value and enter the address in the text box. Then press **ENTER** or click on another field to save your change.

Network Calls

9.16 **Accept Network Calls**: The Accept Network Calls flag must be enabled to allow the Call Routing Application to process AMIS calls. If it is not, incoming AMIS calls will time-out. To enable the flag, select its check box to place a check mark in it to change the setting to **Yes**. Then press **ENTER** or click on another field to save your change. To set it back to **No**, select the check box again.

9.17 **Accept Unknown Network Calls**: If the Accept Unknown Network Calls flag is enabled, the local Voice Processor can accept AMIS calls from any voice mail system that supports AMIS networking. If this flag is disabled, the local Voice Processor can only accept network calls from Voice Processor nodes that are programmed in its database. To enable the flag, select its check box to place a check mark in it and change the setting to **Yes**. Then press **ENTER** or click on another field to save your change. To set it back to **No**, select the check box again.

B. TIMERS AND LIMITS

9.18 Double-click on **Timers and Limits** to view the list of current Values for general Networking. To program a timer or limit, click on its Value and enter the new information in the text box. Then press **ENTER** or click on another field to save your change. The following timers and limits can be programmed for the Voice Processor networking feature:

- **Failed Connection Retry Interval**: *(EM Systems Only)* Select the number of seconds that the system will wait between attempts to connect to another Voice Processor. When a connection fails, the system waits this number of seconds and then tries the connection again. The valid range is 0 - 32767 seconds, and the default value is *60*.
- **Failed Connection Timeout**: *(EM Systems Only)* Select the number of seconds that the system will wait for the other Voice Processor to respond before the connection attempt is considered a failure. After this timer expires, the system waits until the Failed Connection Retry Interval timer expires before it attempts to connect again. The valid range is 0 - 32767 seconds, and the default value is *60*.
- **Maximum Messages Per Network Call**: Determines the maximum number of messages the Voice Processor will transmit to a remote node during one call. The range is 0-1000 messages. When this field is programmed to 0, the number of messages per network call is unlimited. The default value for this field is 0. Also, AMIS calls are automatically limited to nine messages per call by AMIS protocol. If this field is programmed between 1 and 9 (inclusive), AMIS calls will adhere to this field. However, if this field is programmed to 0 or something greater than 9, AMIS networking calls can still only transmit a maximum of nine messages.
- **Maximum Network Call Attempts**: Determines the maximum number of consecutive, unsuccessful call attempts the Voice Processor will make to a remote node. This field ranges from 1 to 999, and the default value is 30. When call attempts have reached the Maximum Network Call Attempts for a node, all pending messages for the node are returned to the destination programmed in the Undeliverable Network Messages Destination field (see next page). No more calls are attempted for the node until the delivery start and stop date and time are active again (see page 6-396).
- **Maximum Network Calls**: Limits the total number of simultaneous network calls. If the Voice Processor receives a network call when it already has the maximum number of network calls in progress, it will reject the new call. If the Voice Processor needs to deliver messages to a remote node when it already has the maximum number of network calls in progress, the message delivery will be delayed until the number of current network calls drops below the maximum. This field ranges from 1 to 32, and default to 2.

NOTE: This timer is not supported by Enterprise Messaging.

- **Maximum Network Inbound Connections**: *(EM Systems Only)* Select the number of inbound connections that the EM system will support at one time. This limits the number of TCP/IP networking protocols used to network voice processing systems. The valid range is 0 - 32767 connections, and the default value is *1*.
- **Maximum Network Outbound Connections**: *(EM Systems Only)* Select the number of outbound connections that the EM system will support at one time. This limits the number of TCP/IP networking protocols used to network voice processing systems. The valid range is 0 - 32767, and the default value is *1*.

- **Maximum Network Message Length**: Determines the maximum length of a message the Voice Processor will transmit to a remote node. It is applied as the message is being recorded. The field ranges from 1 to 600 minutes. The default message length is 15 minutes. When this field is programmed to 0, the messages length is unlimited.
- **Network Call Failure Threshold**: Determines the maximum number of attempted calls to a node before a diagnostic alarm is printed. The alarm continues to be printed until the Maximum Network Call Attempts for the node has been reached, or a successful call has been completed. This field ranges from 1 to 999, and the default value is 15. The value of this field can equal the value of the Maximum Network Call Attempts field, but it cannot exceed it.
- **Network Call Retry Timer**: Determines the length of time the Voice Processor will wait before retrying a network call when the remote site does not answer or the number is busy. This timer will be ignored if the message threshold has been reached. This field ranges from 1 to 60 minutes and defaults to 5 minutes.
- **Update Message Latency Period Timer**: Determines the length of time the AVDAP will wait between update message delivery attempts. When this field is programmed the update message latency time for an update message will be reset. The notification tasks scan each AVDAP node to determine if the message threshold or message latency has expired. If either is true, the AVDAP will spawn a TCP/IP client task which attempts a connection to a remote AVDAP node. However, if there are only update messages in the node's queue, the AVDAP checks if the update message latency time has expired when it checks the message threshold or message latency. If the update latency time has not expired, that is, it has not been 15 minutes since the last delivery of updates, a TCP/IP client is not spawned. This field ranges from 1 to 1440 minutes and defaults to 15 minutes.

C. VALIDATE OFF-NODE MAILBOXES

9.19 The Validate Off-Node Mailboxes flag indicates whether or not mailbox numbers are checked against DB Programming when a user attempts to deliver a message via a node network mailbox.

9.20 When this flag is enabled, the mailbox number entered must be a valid network mailbox number on the local node. This means that the local node can only send messages to mailboxes on the remote node that have network mailboxes programmed on the local node.

9.21 When this flag is disabled, the mailbox number is not checked. This means the local node can send messages to any mailbox on the remote node, even if it does not have a local direct network mailbox.

9.22 To enable the flag, select its check box to place a check mark in it and change the setting to **Yes**. Then press **ENTER** or click on another field to save your change. To set it back to **No**, select the check box again.

D. UNDELIVERABLE MESSAGES DESTINATION TYPE

9.23 This field is used to determine the destination for all network messages which are undeliverable. It can have a value of Delete, Sender, or System Administrator.

- When the field is set to Delete, all undeliverable messages are deleted.
- When the field is set to Sender, all undeliverable messages are returned to the sender if possible. If the sender is unknown, the messages are returned to the System Administrator's mailbox. If the System Administrator's mailbox does not exist, then the messages are deleted.
- When the field is set to System Administrator, all undeliverable messages are returned to the System Administrator's mailbox, if it exists, otherwise the messages are deleted.

9.24 The default value for this field is Delete. To set the Undeliverable Messages Destination Type, select the drop-down list box and scroll to the desired setting. Then press **ENTER** or click on another field to save the change.

10. REPORT PARAMETERS

10.1 The Voice Processing system includes reporting capabilities to print reports from the Voice Processing PC to a printer or to a file for storage. The reports include Application and Channel Statistics, Directory Listings (by last name or first name or extension), and Group List reporting.

NOTE: In v9.1 or later systems, EVMC now supports Voice Processor Report Parameters. In previous versions, the Report Parameters fields were displayed with a red "x" when using EVMC.

10.2 Report selection options include the following:

- **Application and Channel Statistics**: The following information appears individually for each application and as a summary for all applications.
  - — **Description and extension number of the application**: Shows the programmed name for the application. (Call Routing Announcement applications are all listed together by extension.) The applications are listed in the following order: Message Notification/Retrieval, Voice Mail, Auto Attendant, and Call Routing Announcement. (Auto Attendant Recall applications are reported within the Auto Attendant information.)
  - — **Incoming calls**: Shows the total number of calls received by that extension number. This is shown as a combined total for Call Routing Announcement applications.
  - — **Outgoing calls**: Shows the Message Notification/Retrieval application. These are the remote message notification calls placed by the Voice Mail application.
  - — **Connect minutes**: Shows the total time spent on incoming and outgoing calls (if any) combined. This is shown as a combined total for Call Routing Announcement applications.
  - — **Minutes per call**: Shows a combined total of the average amount of time spent on each call in minutes and seconds for Call Routing Announcement applications.
  - — **Transfers to Operator**: Shows the number of times a caller (within Voice Mail or Auto Attendant) presses the dial pad button 0 for operator access.
  - — **Voice Mail messages left**: Appears in the summary section only. It shows how many voice mail messages were left in all mailboxes combined.
  - — **Channel statistics**: Includes data on the activity of all applications combined. It shows, in 30-minute segments, the total number of minutes and seconds that all of the Voice Processing voice channels were busy simultaneously. The detailed segments begin at 07:00 AM and conclude at 06:00 PM. The "Off Peak Hours" segment shows statistics for the remaining time period (6:00 PM to 7:00 AM). This section ends with a grand total of busy channel occurrences for each of the days being reported.

    NOTE: If the number of voice channels programmed in DB Programming is greater than the number of actual channels available, the statistics reported will be based on the number available and not the programmed number.

- **Directory Listing Reports**: Directory Listings can be sorted by first name, last name, or extension/mailbox number (directory number). The listings show the description for the mailbox or extension ID, the mailbox/extension number, the message notification station (for mailboxes), and mailbox information. The mailbox information tells whether the mailbox is marked Private and/or Unlisted. (An X appears in the Mailbox field to indicate a mailbox that is neither Private nor Unlisted and a blank indicates that it is an extension ID.)
- **Group List Report**: The purpose of the Group List report is to provide the System Administrator with a printed copy of the system's group lists. The report identifies the group list number, the description for the list, and the mailboxes included in the group list.
- **Fax Delivery Reports**: The Fax Delivery Report includes information for up to 200 fax delivery attempts. Each entry contains the date and time of the delivery attempt, the date and time the fax was requested, the delivery status (Successful, Busy, Call Failed, or Transmission Error), the fax telephone number, and the list of requested documents. The Fax Delivery Report displays an asterisk (*) immediately to the left of the delivery status in the Fax Delivery Report for an entry representing a fax delivery that failed and was removed from the delivery queue. Fax deliveries can fail for many reasons, but the most common problem is that the fax number entered was not a fax machine, but was a company's main number or answering service. The System Administrator should review the Fax Delivery Report on a regular basis to check for delivery failures.
- **Fax Document Usage Reports**: This report lists all documents in the fax library by document number. Each entry shows the document number, description, how many times it was delivered to callers, the last request date, and the last revision date and time. (If the document has not been revised, it shows the import date and time.)
- **Directory Sort Order**: *(Available only if Directory Listing Reports are selected.)* The Directory Sort Order drop down list boxes allow you to select First Name, Last Name, or Directory Number sorting order. Select the drop-down list box and scroll the high-light bar to the desired option. Then press **ENTER** or click on another field to save the change.

A. AUTOMATIC REPORT GENERATION

10.3 Automatic Report Generation programming contains controls for establishing the day and time the system should automatically print reports, enabling automatic report printouts, clearing the system's statistics, and a list of the reports to print along with the option to sort the Directory Listing report.

Report Selection

10.4 Double-click on **Report Selection** to view the list of available report types. To enable a report type, select the check box to place a check mark in it. Then press **ENTER** or click on another field to save the change. To disable the report, select it again to remove the check mark.

Statistics To Clear

10.5 You can also choose the statistics that will be cleared after the report is printed. Double-click on **Statistics to Clear** to view the options. To mark a statistic to be cleared, select the check box to place a check mark in it. Then press **ENTER** or click on another field to save the change. To remove the check mark, select it again. The statistics that can be cleared are:

- **Application and channel**: Clears the information that is listed in the Application and-channel Statistics report only.
- **Fax Delivery and Document Usage**: Clears all statistics for the Fax Delivery report and the request count and last request date for the Fax Document Usage report.
- **Mailbox**: Clears all mailbox statistics.

Enable Automatic Reports

10.6 To enable automatic report generation, select the check box to place a check mark in it. Then press **ENTER** or click on another field to save the change. To disable automatic report generation, select it again to remove the check mark.

Print Day

10.7 To select the day for printing automatic reports, select the Print Day drop-down list box and scroll the highlight bar to the desired day. Then press **ENTER** or click on another field to save the change.

Print Time

10.8 To select the time of day for printing automatic reports, select **Print Time** and enter or scroll to the desired time. You can also scroll between AM and PM. When finished press **ENTER** or click on another field to save the change.

B. MANUAL REPORT GENERATION

10.9 Manual Report Generation programming contains a list of the reports to print along with the option to sort the Directory Listing report. It also allows you to clear statistics after printing the reports.

Report Selection

10.10 Double-click on **Report Selection** to view the list of available report types. To enable a report, select the check box to place a check mark in it. Then press **ENTER** or click on another field to save the change. To disable the report, select it again to remove the check mark.

Statistics To Clear

10.11 You can also choose the statistics that will be cleared after the report is printed. Double-click on **Statistics to Clear** to view the options. To mark a statistic to be cleared, select the check box to place a check mark in it. Then press **ENTER** or click on another field to save the change. To remove the check mark, select it again. The statistics that can be cleared are:

- **Application and Channel**: Clears the information that is listed in the Application and-channel Statistics report only.
- **Fax Delivery and Document**: Clears all statistics for the Fax Delivery report and the request count and last request date for the Fax Document Usage report.
- **Mailbox**: Clears all mailbox statistics.

Printing Reports

10.12 To print the selected reports, exit to the Manual Report Generation directory. Then right-click on **Report Selection** and select **Print Selected Reports**.

C. OUTPUT DEVICE

10.13 Select this drop-down list box to scroll to the output device on the Voice Processing PC. For a Voice Processing Unit, your options are FILE or LPT1. For an EVMC system, your option is File only. For an EM system, your options are FILE or EMAIL Then press **ENTER** or click on another field to save the change.

NOTE: If "LTP1" is selected, a printer must be attached to the parallel port on the voice mail server. If "File" is selected, the Output Destination field must also be programmed.

D. OUTPUT DESTINATION

10.14 This is used only if the output device is "FILE." or "E-mail."

10.15 *If the Output Device is File*, enter the name of the desired file in the File text box (for example, C:\avdap\reports.txt). Then press **ENTER** or click on another field to save the change. The system will validate that the specified drive output is defined, but will not validate that the path exists on the Voice Processing PC. Instead, the database manager merely verifies that the file's syntax is correct (e.g., check for valid characters). If only the file name is entered then the system will save the report using the file name under the Avdap directory.

10.16 *If the Output Device is E-mail,* enter the e-mail address (for example, john-doe@abc.com) in the text box. Then press **ENTER** or click another field to save the change.

NOTE: If the e-mail address is invalid or if the destination mailbox is full, the system generates a voice mail message indicating that the report could not be delivered. This message is left in the system administrator's voice mailbox.

11. SMDR BUFFER

NOTE: SMDR buffering is not supported by Enterprise Messaging.

11.1 The Voice Processor will buffer SMDR records from the local node and records that are sent from other nodes in the network. The following information can be programmed for the Voice Processor SMDR Buffering application.

- Clear SMDR Buffer
- SMDR Buffer Size
- SMDR Buffer-Box Bit Rate - COM 2
- Enable SMDR Buffering

11.2 Clear The SMDR Buffer: *(Not available in stand-alone programming sessions.)* During a direct-connect or remote programming session, you can clear the data in the SMDR buffer. To clear the buffer, right-click on **SMDR Buffer** in the Voice Processor directory and select **Clear**. A window appears that asks if you are sure you want to clear the SMDR buffer. Select **Yes** to continue or select **Cancel** to leave the buffer unchanged. If you chose to clear, an information window will appear to tell you that the buffer has been cleared. Select **OK** to continue.

11.3 Programming the SMDR Buffer: You can program the following fields:

- **SMDR Buffer Size**: To determine how much of the Voice Processing PC disk space will be devoted to SMDR buffering, select the current value and enter or scroll to the desired number of megabytes (0-20). Then press **ENTER** or click on another field to save the change.
  - *If you attempt to set the buffer size to a value higher than your available disk space,* you will receive a warning message that says, "The Voice Processing PC did not have enough disk space to set the buffer size to XXX megabytes. The current buffer size is now 0 megabytes. Note: If there was data in the buffer, the data is still intact."; select **OK** to continue and reset the buffer to a valid size.
  - *If you change the size of the buffer,* a window appears that tell you that the buffer will be cleared and asks if you want to continue. Select **Yes** to complete the resizing or select **No** to leave the buffer and buffer size unchanged.
- **SMDR Buffer Box Bit Rate**: To determine the rate of output from the buffer box port (COM2), select this drop-down list box. The available rates are 300, 1200, 2400, 4800, 9600, and 19200. Then press **ENTER** or click on another field to save the change.
- **Enable SMDR Buffering**: *(Buffer size must be greater than 0 for this check box to be selected.)* To enable the SMDR Buffering option, select the check box to place a check mark in it. Then press **ENTER** or click on another field to save the change. To disable SMDR buffering, select the check box again to remove the check mark.

12. VOICE PROCESSOR TIMERS AND LIMITS

12.1 There are programmable timers and limits that control various Voice Processor functions. The timers and their default values, programmable ranges, and purposes are described on the following pages. The default values have been carefully selected to ensure proper operation under most circumstances. Occasionally, one or more of the timers may need to be adjusted.

12.2 The Timers and Limits are divided in the following programming areas:

- Timers and Limits (see below)
- DTMF Detection Information (see page 6-428)
- Number of Voice Channels (see page 6-430)

A. TIMERS AND LIMITS

12.3 Select Timers and Limits to view the list of current Values for the Voice Processor. To program a timer or limit, click on its Value and enter the new information in the text box. Then press **ENTER** or click on another field to save your change. The following system-wide timers and limits can be programmed for the Voice Processor:

Table 6-46: *Voice Processor Timers and Limits*

Timer	Default	Range	Defines
Busy Tone Cycle Detect	2 cycles	1-60 cycles	The minimum number of cycles of tone the system needs to recognize busy, do-not-disturb, or reorder tones. When the Voice Processing PC does not recognize one of these tones, it will assume the call is answered. **NOTE:** This timer is not supported in Enterprise Messaging.
Call-in-Progress Dial Tone	2 seconds	0-5 seconds	The minimum duration of continuous dial tone that the Voice Processing Unit can recognize during an active call (for example, when a caller hangs up while connected to the voice mail application). To disable dial tone detection on active calls, set this timer to 0. **NOTE:** This timer is not supported in Enterprise Messaging.
Call Progress Delay	3000 milliseconds	1-60000 millisec. (0.001-60 sec.)	The maximum length of time the Voice Processing Unit will wait before checking call status when placing a call. It must be set long enough to prevent the Voice Processing Unit from detecting dial tone on the outgoing call. **NOTE:** This timer is not supported in Enterprise Messaging.

Table 6-46: *Voice Processor Timers and Limits (Continued)*

Timer	Default	Range	Defines
Call Progress Detection	4000 hundredths	1-6000 hundredths (0.01-60 sec.)	The maximum length of time the Voice Processing Unit will wait for a call to be answered before aborting the attempted call. If it detects anything other than silence during this time, the system will consider the call answered. **NOTE:** This timer is not supported in Enterprise Messaging.
Deleted Message Hold Duration	1 hour	0-24 hours	The amount of time that the Voice Processing Unit will keep each deleted message. The duration is specified in hours and is based on the time the user deletes the message. For example, if the Deleted Message Hold Duration is set to 6 hours, and a user deletes a message at 9:22 AM, the Voice Processing Unit will keep the message until 3:22 PM. A value of 0 means that the Voice Processing Unit will not hold messages. (That is, as soon as the user deletes the message, it is permanently deleted and there is no way to restore the message).
Dialed Pause Duration	300 hundredths	1-500 hundredths (0.01-5 sec.)	The length of pauses dialed by the Voice Processing Unit as part of feature codes, telephone numbers, and outgoing dialing strings. **NOTE:** This timer is not supported in Enterprise Messaging.
Maximum Deleted Message Space	0 MB	0-20 MB	The maximum amount of space that the Voice Processing Unit allocates to store deleted messages. The deleted message storage size is specified in megabytes (MB) of the hard disk. When a user deletes a message, and the deleted message queue size exceeds this number, the Voice Processing Unit will purge the oldest message to make room for the new one. In this case, the Voice Processing Unit will purge as many messages from storage as necessary to make room for the new one. However, if the hard disk becomes full, the Voice Processing Unit will immediately purge all stored messages. **NOTE:** When Voice Processing is shut down, all deleted messages are permanently deleted and can not be restored.
Maximum Greeting Length	1 minute	1-15 minutes	The maximum time allowed for all mailbox greetings and custom Audiotex recordings.

Table 6-46: *Voice Processor Timers and Limits (Continued)*

Timer	Default	Range	Defines
Call Progress Detection	4000 hundredths	1-6000 hundredths (0.01-60 sec.)	The maximum length of time the Voice Processing Unit will wait for a call to be answered before aborting the attempted call. If it detects anything other than silence during this time, the system will consider the call answered. **NOTE:** This timer is not supported in Enterprise Messaging.
Deleted Message Hold Duration	1 hour	0-24 hours	The amount of time that the Voice Processing Unit will keep each deleted message. The duration is specified in hours and is based on the time the user deletes the message. For example, if the Deleted Message Hold Duration is set to 6 hours, and a user deletes a message at 9:22 AM, the Voice Processing Unit will keep the message until 3:22 PM. A value of 0 means that the Voice Processing Unit will not hold messages. (That is, as soon as the user deletes the message, it is permanently deleted and there is no way to restore the message).
Dialed Pause Duration	300 hundredths	1-500 hundredths (0.01-5 sec.)	The length of pauses dialed by the Voice Processing Unit as part of feature codes, telephone numbers, and outgoing dialing strings. **NOTE:** This timer is not supported in Enterprise Messaging.
Maximum Deleted Message Space	0 MB	0-20 MB	The maximum amount of space that the Voice Processing Unit allocates to store deleted messages. The deleted message storage size is specified in megabytes (MB) of the hard disk. When a user deletes a message, and the deleted message queue size exceeds this number, the Voice Processing Unit will purge the oldest message to make room for the new one. In this case, the Voice Processing Unit will purge as many messages from storage as necessary to make room for the new one. However, if the hard disk becomes full, the Voice Processing Unit will immediately purge all stored messages. **NOTE:** When Voice Processing is shut down, all deleted messages are permanently deleted and can not be restored.
Maximum Greeting Length	1 minute	1-15 minutes	The maximum time allowed for all mailbox greetings and custom Audiotex recordings.

Table 6-46: *Voice Processor Timers and Limits (Continued)*

Timer	Default	Range	Defines
Maximum Number of Deleted Messages	0 messages	0-9999 messages	The maximum number of deleted messages that the Voice Processing Unit will store. If this number is exceeded, and a user deletes a message, the Voice Processing Unit will remove the oldest message in storage to make room for the new one. (Entering a value of 0 means that the number is unlimited.) However, this limit is bound to the availability of hard disk space as specified by the Maximum Deleted Message Space above. That is, the Maximum Deleted Message Space limitation will take precedence over this one.
Maximum Outgoing Calls	2 calls	1-number of ports (max 32)	The number of outgoing message notification and fax delivery calls that can be made at one time by the Message Notification/Retrieval application. Note that the number of calls that can be made are also limited by the availability of intercom voice channels and outgoing trunks.
Minimum Call Progress Silence Duration	15 hundredths	1-21 hundredths (0.01-0.21 sec.)	The minimum duration of a period of silence necessary for the Voice Processing Unit to detect silence. That is, if a period of sound is broken by silence, and the silence is shorter than this timer, the Voice Processing Unit will ignore the silence. This timer applies to outgoing remote notification calls only. **NOTE:** This timer is not supported in Enterprise Messaging.
Minimum Call Progress Signal Duration	18 hundredths	1-18 hundredths (0.01-0.18 sec.)	The minimum duration of a period of sound necessary for the Voice Processing Unit to detect sound. That is, if a period of silence is broken by a burst of sound on the line, and that sound is shorter than this timer, the Voice Processing Unit will ignore the sound. This timer applies to outgoing remote notification calls only. **NOTE:** This timer is not supported in Enterprise Messaging.
Notification No-Answer Detection	4 rings per prompt iteration	1–25 rings 1–5 prompt iterations (automatic)	The Notification No-Answer Detection flag has two purposes: • It sets the number of times remote messaging will ring an IC call or CO call before assuming a no answer condition (from 1 to 25) • It automatically sets the number of iterations remote messaging will cycle through the prompts asking for the correct person (from 1 to 5) DB Programming allows a range of 1 - 25 for this field. The maximum number of rings can be 25, but the maximum prompt cycles is 5 regardless of the field value.

Table 6-46: *Voice Processor Timers and Limits (Continued)*

Timer	Default	Range	Defines
Off-hook Delay	0 seconds	0-5 seconds	Length of time the Voice Processing Unit should wait after coming off hook before playing the first prompt. However, the recommended minimum setting is 1 second to allow a pause before the prompts are played for outside callers, thereby assuring that no text is cut off. It is set to 0 to allow internal callers to hear the prompts immediately. **NOTE:** This timer is not supported in Enterprise Messaging.
Outgoing DTMF Digit Duration	60 milliseconds	30-250 millisec. (0.030-0.250 sec.)	Length of the DTMF tones (and inter-digit pauses) that are sent during remote notification to a pager or personal number. **NOTE:** If this timer is changed, the PC must be reset to update the PC with the new timer value. **ALSO:** This timer is not supported in Enterprise Messaging.
Pause Voice Mail	30 seconds	1-240 seconds	The maximum amount of time that the Voice Processing Unit will pause during the playback of a message or recording. **NOTE:** This timer is not supported in Enterprise Messaging.
Record-A-Call Max Message Length	30 minutes	1-600 minutes	The maximum time allowed for Record-A-Call input to a mailbox. (This overrides the mailbox's maximum message length setting. However, if a Record-A-Call message exceeds the maximum message length, the mailbox will be considered full until that Record-A-Call message is deleted.)
Recording Silence Detection	5 seconds	0-30 seconds	Length of silence required to terminate a recording. If silence in recording exceeds this timer, the recording will automatically end. If lengthy (silent) pauses are included in any recordings, this timer should be adjusted to allow them.
Replay Forward/Rewind Increment	5 seconds	1-60 seconds	The number of seconds a message or recording will be advanced or backed up when a user skips ahead or backward during a replay.
Shortest Message Allowed	3 seconds	1-5 seconds	Length of the shortest message that will be accepted by the Voice Processing Unit. (Messages terminated with # are always allowed, regardless of length.)
Transfer No-Answer Detection	4 rings	1-25 rings	The number of rings during which the Voice Processing Unit will wait for a transferred call to be answered.

B. DTMF DETECTION INFORMATION

NOTE: The DTMF Detection Information field is not supported in Enterprise Messaging.

12.4 In most installations, the DTMF detection information should not need to be changed. However, if the Voice Processor is losing digits or if DTMF detection is causing talk-off, it may be necessary to adjust the DTMF filter parameters.

12.5 If digits are not being detected properly, there are three possible reasons:

- **The "twist" is not within the allowed range.** The Twist parameters specify the maximum acceptable difference between the high frequency energy and the low frequency energy. One frequency can be much stronger than the other, or the frequencies can be off the nominal values. If the frequencies are off, measurements made at the nominal values can show different readings since the peak will be at a different frequency. In some cases, noise or voice on the line can add to or subtract from these frequencies.
- **There is too much noise.** Noisy lines can introduce distortion to the digits. A simple inspection can identify if the lines have too much noise. If this is the case, Twist parameters should be increased and Ratio parameters should be decreased so that digits can be detected even when stronger noise and other signals exist.
- **Another signal is present that is similar to the DTMF digit.** Other signals present on the line, including some types of voice, can be detected as DTMF tones causing "talk-off." Adjusting the DTMF Detect and DTMF Delay parameters may correct this problem. However, changing the DTMF Digit Delay or DTMF Digit Detection timer values is a trade-off between improving DTMF detection and increasing the possibility of talk off. As the values of the timers are reduced, DTMF detection is improved, but the possibility of talk off increases. As the value is increased, the possibility of talk off is reduced, but the possibility of DTMF detection problems is increased.

12.6 There are six DTMF filter parameters that can be adjusted to improve performance when the DTMF detection is not performing as desired. These six filter parameters have one set of values when a recording is being played (Play Mode) and another set during all other functions (Idle Mode). Thus, there are twelve parameters used to control digit detection.

12.7 When digits are lost both in Play and Idle modes, the duration of the digits and the intervals between them may be too close to the DTMF Delay and DTMF Detection values. You should decrease these values to improve the chances that the digit's characteristics will remain within acceptable values during the minimum required duration. After the minimum on and off times are set according to the pattern of the digits being dialed, the next step is to verify the twist. A spectrum analyzer can show the amplitude for the frequencies in the signal, and the values shown can be compared to the twist limits set in the driver. Rhetorex has a utility called "FFT" that will show the basic characteristics of the signal, including frequencies and amplitudes. Increasing the twist parameters for both Play and Idle modes will usually solve the problem.

12.8 When digits are lost only in Play mode, the Twist parameters should be increased and Ratio parameters should be decreased for Play mode only.

12.9 If talk-off is occurring, increasing the minimum duration of the digits should suffice. If this is not possible due to other constraints (e.g., speed dial), Twist and Ratio parameters should be changed to make the filters less tolerant.

NOTE: If the DTMF parameters are changed, the new value will not take effect until the Voice Processor is completely idle.

12.10 Select **DTMF Detection** to view the list of DTMF parameters. To program a parameter, click on its Value and enter the new information in the text box. Then press **ENTER** or click on another field to save your change. The definitions of the parameters are:

- **DTMF Digit Detect On**: Determines the minimum duration of DTMF tones that can be recognized by Voice Processing during Play Mode (for example, when a prompt or message is playing). Valid settings for this parameter are 32, 48, 64, 80, or 96 milliseconds. The default is 48 milliseconds.
- **DTMF Digit Detect Off**: Determines the minimum silence allowed between DTMF tones that can be recognized by Voice Processing during Play Mode. Valid settings for this parameter are 32 or 48 milliseconds. The default is 48 milliseconds.
- **DTMF Digit Delay On**: Determines the minimum duration of DTMF tones that can be recognized by Voice Processing during Idle Mode (for example, when Voice Processing is waiting for input). Valid settings for this parameter are 32, 48, 64, 80, or 96 milliseconds. The default is 32 milliseconds.
- **DTMF Digit Delay Off**: Determines the minimum silence allowed between DTMF tones that can be recognized by Voice Processing during Idle Mode. Valid settings for this parameter are 32 or 48 milliseconds. The default is 48 milliseconds.
- **DTMF Digit Low To High Twist (Play and Idle)**: Reflects the difference between the DTMF digit high frequency and low frequency energy (high minus low). To allow the Voice Processor to detect a digit that consists of frequencies, the high frequency energy cannot be stronger than the low frequency energy by more than the setting of this parameter. In Play Mode the default for this parameter is 9.2dB. In Idle Mode the default is 8.2dB. Valid settings for this parameter are: No Limit, 6.7dB, 8.2dB, 9.2dB, 10.7dB, 12.7dB, 15.2dB, or 18.2dB.
- **DTMF Digit High To Low Twist (Play and Idle)**: Reflects the difference between the DTMF digit low frequency and high frequency energy (low minus high). To allow the Voice Processor to detect a digit that consists of frequencies, the low frequency energy cannot be stronger than the high frequency by more than the setting of this parameter. In Play Mode the default for this parameter is 8.2dB. In Idle Mode the default is 5.2dB. Valid settings for this parameter are: No Limit, 4.2dB, 5.2dB, 6.7dB, 8.2dB, 10.2dB, 13.2dB, or 18.2dB.
- **DTMF Digit In To In Ratio (Play and Idle)**: Indicates a digit's energy minus the next digit's energy. In order for the Voice Processor to detect a digit that consists of frequencies, a digit's energy must be stronger than the next digit's energy by the amount of this parameter setting. The default setting for Play Mode is 2.0dB. The default setting for Idle Mode is 8.0dB. Valid settings for this parameter are 1.0dB, 2.0dB, 3.0dB, 4.0dB, 6.0dB, 8.0dB, 9.0dB or 10.0dB.
- **DTMF Digit In To Out Ratio (Play and Idle)**: Indicates a digit's energy minus the noise energy. In order for the Voice Processor to detect a digit that consists of frequencies, the digit's energy must be stronger than the noise energy by at least the amount of this parameter setting. The default setting in Play Mode is 1.0dB. In Idle Mode, the default is 4.0dB. Valid settings for this parameter are 0.5dB, 1.0dB, 2.0dB, 3.0dB, 3.5dB, 4.0dB, 4.5dB, or 5.0dB.

C. NUMBER OF VOICE CHANNELS

12.11 Voice channels allow communication between the telephone system and the applications. If a voice channel is not available in an application's time slot group, callers to that application will receive busy signals and the call will be sent to the application's attendant.

12.12 At default there are four available voice channels. The actual number of voice channels may be higher or lower depending on the number of Voice Processing Cards used and the port configuration of the Voice Processing Cards.

12.13 The number of available voice channels determines the maximum number of channels that can be assigned to any single Time Slot Group. The combined total of channels assigned to Time Slot Groups may exceed the actual number of voice channels, because it is unlikely that all time slot groups will use their maximum allotment at once.

12.14 To set the number of voice channels:

1. Click on the current Value and enter the new information in the text box.
2. Press **ENTER** or click on another field to save your change.

NOTE: Although DB Programming allows you to configure 64 voice mail ports, Voice Processing using the Windows-based platform currently supports a maximum of only 40 ports. Attempting to configure 64 voice mail ports may cause serious performance issues.

13. UNIFIED MESSAGING

13.1 This folder is only applicable to Enterprise Messaging voice processors, which support Unified Messaging. If the folder is selected for other Voice Processor types, a warning appears and all the fields appear with a red "**X**."

13.2 For complete information about Unified Messaging OSE, refer to the latest version of the *Unified Messaging Open Standards Edition Administrator's Guide* (document part no. 835.3162).

13.3 To program the Unified Messaging fields:

1. Go to Voice Processor – **Unified Messaging**.
2. Configure the following fields:
   - **Attachment Format for Inbound Faxes**: Indicates the format of fax attachments in e-mails sent for Inbound faxes. The available options are TIF or PDF. It is set to *TIF* by default.
   - **Enable Delivery of Incomplete/Failed Inbound Faxes**: Enables e-mail notification for fax receivers regarding failed or incomplete inbound faxes. It is set to *No* by default.
   - **Enable IMAP Login for EM Server**: Enables remote IMAP logins to the EM server. It is set to *No* by default.
   - **EM Server IMAP Connection Timeout**: Determines the maximum duration that the Em server will wait for a remote IMAP connection to be established. The range is 30–1440 minutes; the default is set to *30 minutes*.
3. Press **Enter** or click on another field to save the change.

Troubleshooting

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1. INTRODUCTION

CAUTION

Some of the cards and cables inside the Voice Processing PC may have been spot-glued into place to help prevent them from working loose during shipment. To remove one of these cards or cables from the PC (if necessary), first break the glue seal by gently wiggling the connector with a pair of pliers.

1.1 This section of the manual describes the troubleshooting procedures to follow in the event of a system or station instrument malfunction. System repair is limited to replacing parts (phones, circuit cards, cables, power supply, etc.).

2. TROUBLESHOOTING CHECKLIST

2.1 To save time, perform the troubleshooting procedures in the following order:

1. Check for proper light-emitting diode (LED) indications on the circuit cards (see Figure 7-2 beginning on page 7-5).
2. Check the alarm message output (see page 7-12 to 7-25).

2.2 Isolate the problem and see the appropriate troubleshooting chart (see page 7-26).

3. LIGHT-EMITTING DIODE (LED) INDICATIONS

3.1 LEDs on the front edge of the circuit cards and the ATM Switch indicate specific functions. Figure 7-2 starting on page 7-5 lists the various LEDs and describes their purposes.

A. CIRCUIT CARD LEDS

3.2 If the CPU ACTIVE LED on the CPC or CPU Card is flashing (or if referred to this page to isolate a problem), follow these procedures to check the system voltages:

CAUTION

Always turn OFF the AC POWER before removing or inserting the CPC or CPU Card.

1. Turn OFF the AC power.
2. Remove the CPC or CPU Card and check to make sure that all of the components on the card are properly seated and that no pins are bent. Also, check to make sure that the two parts of the combined card are securely attached, and make sure that the connectors and stand-offs are properly aligned and seated.
3. Wait 10 seconds and re-insert the card.
4. Turn ON the AC power.
5. If the CPU ACTIVE LED is still flashing (or if the problem still persists):
   a. Observe the LEDs on the front of the power supply unit. All of the LEDs should be lit. *If any of the LEDs are blinking or are unlit*, the power supply is defective and the entire unit must be replaced (see page 3-80 or page 3-89 for details).

NOTE: A blinking LED indicates that an overly high voltage has been detected. A lit LED merely indicates that there is some voltage present (as shown in the table at the right); it does not, however, indicate whether the voltage level is acceptable. To determine the exact voltage level present, perform step 5b.

Voltage	LED	Output
+5VDC	OFF	1V
	ON	>1V & <6.5V
	BLINKING	6.5V
-5VDC	OFF	-1V
	ON	<-1V & >-6.6V
	BLINKING	-6.5V
+24VDC	OFF	1V
	ON	>1V & <29V
	BLINKING	29V

b. Measure the following system voltages as outlined on page 3-83 or page 3-92. A digital voltmeter of ±0.5% accuracy is required. Insert the "common" voltmeter probe into the ground point and insert the other probe into the desired voltage test point. (See Figure 3-39 on page 3-79 or Figure 3-43 on page 3-88 for voltage test point locations.)

Table 7-1. *Voltage Test Point*

Voltage	Tolerance
-5VDC	±5%
+5VDC	±5%
+24VDC	±5%
SLPWR (+24VDC)	±5%

6. Perform *one* of the following steps:

- *If the system voltages are not within the specified limits,* replace the defective power supply unit.

CAUTION

DO NOT attempt to adjust the power supply voltages. Damage caused by unauthorized voltage adjustment will not be covered by the warranty.

- *If the system voltages are correct and the CPU ACTIVE LED is* ***still*** *flashing*, try another card.

NOTE: If possible, use the save/restore program to save the customer's database before replacing the CPC or CPU Card (see page 6-65). It can then be restored (loaded) into a new CPC or CPU Card if necessary. Make sure that the database back-up battery on the CPC or CPU Card has a charge of at least 3.3VDC and that the battery back-up jumper strap is in the ON position.

- *If the system voltages are correct and the CPU ACTIVE LED is properly lit, yet the problem still persists*, contact Technical Support for assistance.

Table 7-2. *Card Light-Emitting Diode (LED) Indications*

Location	LED	Description
CPC	CPU Active	Lit green to indicate the card is online.
	DB Error	Solidly lit red to indicate the database is in the default state. If flashing slowly, the backup database is being saved. If flashing rapidly, the local database is being restored.
	Minor Alarm	Lit yellow to indicate a system alarm. If flashing, the software license is invalid (see for details about how this affects the system).
	Modem Active	Solidly lit yellow to indicate that the modem is connected and in sync with the connected device. If flashing, the modem is currently synchronizing with the connected device.
	100Base-T	Lit green to indicate the network connection is operating at 100 Mbps. If not lit, the network connection is operating at 10 Mbps.
	Duplex	Lit green to indicate the network is running in full-duplex mode.
	Collision	Lit yellow to indicate that two or more devices have attempted to send data over the same connection while in half-duplex mode.
	Transmit	Lit green to indicate the IP connection is currently transmitting to the network.
	Battery Alarm	Lit red to indicate the battery is missing or cannot be charged. If unlit, the battery is charged (or charging) and functioning.
	Link (RJ45)	Lit green to indicate a valid network connection.
	XMT (RJ45)	Lit green to indicate that the card is transmitting data to the network. **NOTE:** The faceplate also indicates LINK and SYNC LEDs at the top of the card. These LEDs are reserved for future use.
CPU128 Card or CPU020/PCM Card or CPU020/PCMA Card	CPU Active	Lit green to show that the main processor is operating properly. If flashing at least once every five seconds, see paragraph 3.2 on page 7-3 for troubleshooting procedures. If unlit, check for improper connection (e.g., cabling, card seating, power supply, etc.); if still un lit, try another card. If flashing slower than once every five seconds, remove the battery back-up jumper strap, turn off the system power for thirty seconds, replace the strap, and turn the power back on. If still flashing, see paragraph 3.2 on page 7-3 for troubleshooting procedures.
	Database Error	Lit red when a database error has caused the system to return to default values. When the system is first installed, the LED will remain lit until the clock is initialized through Database (DB) Programming (or other DB Programming is performed). If a CPU020/PCM software upload fails while in progress (see page 3-173), the Database Error (red) and Minor Alarm (yellow) LEDs blink rapidly (ten times per second) and the PC displays the reason for the failure.

Table 7-2. *Card Light-Emitting Diode (LED) Indications (Continued)*

Location	LED	Description
(Continued) CPU128 Card or CPU020/PCM Card or CPU020/PCMA Card	Minor Alarm	Lit yellow when a minor alarm has occurred. The light goes out when the alarm message is cleared as described in FEATURES on page 5-300. If a CPU020/PCM software upload fails while in progress (see page 3-173), the Database Error (red) and Minor Alarm (yellow) LEDs blink rapidly (ten times per second) and the PC displays the reason for the failure.
	External Clock Active	Lit green when the system is receiving clock signals from an external source (e.g., from the public network via a T1 or E1/PRI Card).
	Remote Sync *(PCM Only)*	Lit green to show that the remote fiber-optic connection is functioning properly. If unlit, check for improper connection (e.g., cabling, card seating, etc.); if still unlit, try another card.
	Local Sync *(PCM Only)*	Lit green to show that the local fiber-optic connection is functioning properly. If unlit, check for improper connection (e.g., cabling, card seating, etc.); if still unlit, try another card.
	Battery Off	Lit red when the battery back-up strap on the CPU Card is placed in the OFF position. Will not light if the strap is not attached to any pins. Also, will not light if the strap is in the ON position, but no battery is installed.
CPU020/PCMA Card	ATM Link Bad	Lit red when there is no active connection to the ATM Switch.
	Master and Slave	Lit green to indicate when the chassis is configured as the Master or Slave clock to the system. When the chassis powers up, both LEDs will blip (flash quickly and remain unlit for a second), indicating that the PCMA has not yet connected to the ATM Switch. After the PCMA connects to the switch, the LEDs will flash, indicating that Call Processing has not yet configured the chassis for Master/Slave. After the clocks are configured, both LEDs will stop flashing. These LEDs will then light, flash, or blip based on the configuration or current state of the system. *If the Master and Slave LEDs are both blipping,* the PCMA is not registered with the ATM Switch. *If the Master and Slave LEDs are both flashing,* Call Processing has not yet configured clocking. *If the Master LED is lit but the Slave is unlit,* the Master chassis is providing clock to the system. *If the Master is flashing but the Slave is unlit,* the chassis is free-running. **NOTE:** When a Slave chassis stops receiving timing information from the Master chassis, the Slave will revert to a free-running state. In the free-running state, the Slave chassis functions the same as the Master except the Slave will not make any timing connections to other chassis.

Table 7-2. *Card Light-Emitting Diode (LED) Indications (Continued)*

Location	LED	Description
(Continued) CPU020/ PCMA Card	*(Continued)* Master and Slave	*If the Master is unlit but the Slave is flashing,* the Slave chassis is synchronizing to the Master. *If the Master is unlit but the Slave is lit,* the Slave chassis is synchronized to the Master. **NOTE:** If the Master chassis's reference clock drifts or changes, the Slave chassis will re-synchronize to the new clock. The Slave LED, therefore, may flash during normal operation, even after it has initially synchronized to the Master chassis.
Digital Keyset Card (8-port and 16-port), Analog Keyset Card, or Executone® Keyset Card	Card On-Line	Lit green to show functional readiness (e.g., the card is communicating properly with the rest of the system). If unlit, check for improper connection; if still unlit, try another card or another card slot.
	Circuit Busy	Lit yellow to indicate that at least one of the 8/16 circuits is busy (e.g., connected to a voice channel). Unlit when all the circuits are idle. **NOTE:** As of version 6.0, the Circuit Busy LED also lights when a station residing on a DKSC-16 or DKSC-16+ is listening to background music.
Single-Line Card	Card Active	Lit green to show functional readiness (e.g., the card is communicating properly with the rest of the system). If unlit, check for improper connection; if still unlit, try another card or another card slot.
	Circuit Busy	Lit yellow to indicate that at least one of the eight circuits is busy (e.g., connected to a voice channel). Unlit when all the circuits are idle.

Table 7-2. *Card Light-Emitting Diode (LED) Indications (Continued)*

Location	LED	Description
Internet Protocol Resource Card (IPRC)	Card On-Line	Lit green to show functional readiness (e.g., the card is communicating properly with the rest of the system). If unlit, check for improper connection; if still unlit, try another card or another card slot.
	Circuit Busy	Lit yellow to indicate that at least one of the 16 (32 with a daughter card) VoIP channels is busy. For IPRCs programmed for networking, if this LED is lit, removing the card will drop the call(s) active on the circuit(s). **NOTE:** In software v9.0 and later, the Circuit Busy LED will only light when the VoIP resources are in use. For IP devices, the LED is lit when an endpoint is using VoIP resources. For IP networking calls, the LED is lit when a networking call is in progress. The Circuit Busy LED turns off immediately when no VoIP resources are in use.
	Make Busy	Lit yellow to indicate the current state of the Make-Busy switch. This LED remains lit while the button is pressed but does not change the state of the ports on the card.
	Halted	Lit red to indicate that the card is not functioning due to a software or hardware error. The card will not function properly; return the card for repair.
	Duplex	Lit green to indicate that the network connection is running in full-duplex mode.
	Collision	Lit yellow to indicate that the network connection has experienced a collision while in half-duplex mode.
	100Base-T	Lit green to indicate that the network connection is operating at a speed of 100 Mbps.
	Link (RJ45)	Lit green to indicate a valid network connection.
	XMT (RJ45)	Lit green to indicate that the card is transmitting data to the network.
Trunk Cards (LSC or LGC)	Card On-Line	Lit green to show functional readiness (e.g., the card is communicating properly with the rest of the system). If unlit, check for improper connection; if still unlit, try another card or another card slot.
	Circuit Busy	Lit yellow to indicate that at least one of the four circuits (eight if a daughter card is attached) is busy (e.g., connected to a voice channel). Unlit when all the circuits are idle.

Table 7-2. *Card Light-Emitting Diode (LED) Indications (Continued)*

Location	LED	Description
T1/E1 Cards (T1C, T1/PRI, E1/PRI, or T1/E1/PRI)	Card On-Line	Lit green to show that the primary processor on the card is active and is communicating properly with the rest of the system. If unlit, check for improper connection; if still unlit, return the card for repair.
	Circuit Busy	Lit yellow to indicate that at least one of the T1/PRI or E1/PRI circuits is busy. Unlit when all the resources are idle.
	Red Alarm	Flashes when the T1 or E1/PRI Card has detected some type of red alarm error condition for at least two to three seconds (e.g., loss of the T1/PRI signal), either internally or from outside the system. All calls on the card are dropped and all attendant stations display ALARM #114 CALL TECHNICIAN. Check the error message output (see page 7-25); then attempt to correct the problem. When the card receives a clean signal for 10 to 20 seconds, the Red Alarm LED is unlit and normal call processing resumes. If, while the Red Alarm LED is flashing, the T1 or E1/PRI Card detects a receive signal, it transmits a yellow alarm to outside the system. If the card detects a loss of receive signal, it transmits a blue alarm to outside the system.
	Yellow Alarm	Flashes when the T1 or E1/PRI Card has detected (is receiving) a yellow alarm error condition (e.g., a red alarm error condition from outside the system or a loss of frame synchronization). If the yellow alarm condition occurs for less than two to three seconds, normal call processing continues and no calls are dropped. However, if the yellow alarm condition lasts more than two or three seconds, the card goes into a red alarm condition (see above). If this LED is the only one on the card that is flashing, it may indicate a DB Programming inconsistency. Check the alarm message and field service diagnostic output for a programming conflict; then correct the problem.
	Blue Alarm	Flashes when the T1 or E1/PRI Card has detected (is receiving) a blue alarm error condition (e.g., a stream of continuous ones from outside the system). This “keep alive” signal, also called Alarm Indication Signal (AIS), is sent from the remote site when it is in a yellow or red alarm condition. If the blue alarm condition occurs for less than two to three seconds, normal call processing continues and no calls are dropped. However, if the blue alarm condition lasts more than two or three seconds, the card goes into a red alarm condition (see above).
	Providing Reference Clock	Lit green to show that the card is providing the system with the clock timing necessary to control all communication components in the system (e.g., the entire system is synchronized to the card). Unlit when another card is providing the system reference clock. When this LED is lit on one of the T1 or E1/PRI Cards in the system, the External Clock Active LED on the CPU Card should also be lit.
	Card Halted	Lit red to show that the card has detected an unrecoverable error and has halted the primary processor. The card will not function properly; return the card for repair.

Table 7-2. *Card Light-Emitting Diode (LED) Indications (Continued)*

Location	LED	Description
(Continued) T1/E1 Cards (T1C, T1/PRI, E1/PRI, or T1/E1/ PRI)	Make Busy	Lit yellow to indicate that the card is in the process of "busying out" each circuit. This occurs when someone presses the Make Busy switch. After the circuits are busied out, this LED flashes slowly. If flashing rapidly, the busy out process was canceled because the Busyout Switch Timeout timer expired before the card was removed. If this happens, the LED continues to flash until the Make Busy switch is released or the Make Busy field is disabled (System\Cabinets\<T1 Card>\T1 Diagnostics\T1 Status or System\Cabinets\<E1 Card>\E1 Diagnostics\E1 Status).
Options Card	Card On-Line	Lit green to show functional readiness (e.g., the card is communicating properly with the rest of the system). If unlit, check for improper connection; if still unlit, try another card or another card slot.
	Circuit Busy	Lit yellow to indicate that at least one of the card's resources is busy. Unlit when all the resources are idle.

B. ATM SWITCH LEDS

3.3 Each ATM switch type has LEDs that indicate the status of the Switch and of individual ports.

ForeRunner LE155

3.4 The *ForeRunner* LE155 switch has several LEDs. S1 indicates the software status; S2, the power status; and the port LEDs, the port status. These LEDs will light (or blink) a different color depending on the current status, as described in the following table.

Table 7-3. *ForeRunner LE155 ATM Switch LEDs*

LED	Color	Indication
S1	Green (solid)	The switch software is functioning properly.
	Red (blinking)	The switch software is currently booting. This LED will change to green once the Switch is finished booting. **NOTE:** If this LED continues to blink red, even after a reboot, the SRAM is corrupt. Contact Inter-Tel® for additional assistance.
S2	Green (solid)	The switch has power.
	Unlit	The switch does not have power.
Port (e.g., A1, C3, etc.)	Green (blinking)	The port is receiving packets.
	Red (solid)	The port is not communicating.
	Yellow (blinking)	There is a port alarm.

NOTE: If the expansion slot is populated, each additional port will also have individual LEDs.

ASX-200BX

3.5 The following table describes the LEDs available on the ASX-200BX switch.

Table 7-4. *ASX-200BX ATM Switch LEDs*

LED	Color	Indication
ACT	Amber (solid)	The Ethernet port is transmitting and receiving data.
	Unlit	If unlit, the connection has experienced a collision or there is no activity.
L1	Green (solid)	The Ethernet connection is active.
	Unlit	There is no active Ethernet connection.
PWR	Green (solid)	The Switch Control Processor (SCP) is functioning properly.
	Red (solid)	The SCP is resetting.
	Unlit	There is no power to the SCP.
5 Volts	Green (solid)	The power supply is functioning.
	Red (solid)	There is a problem with the power supply.
RX*n*	Green (solid)	The port is receiving data from the phone system.
	Red (solid)	There is no connection or no data is being received.

Table 7-4. *ASX-200BX ATM Switch LEDs (Continued)*

LED	Color	Indication
TX*n*	Green (solid)	The port is sending data to the phone system.
	Red (solid)	There is no connection.

4. ALARM MESSAGES

4.1 Minor alarm messages can be programmed to appear on all administrator stations' displays or on the primary attendant's display only. (See PROGRAMMING, page 6-259, for details.) Also, when a minor alarm has occurred, the MINOR ALARM LED on the CPC or CPU Card is lit yellow.

4.2 Major alarm messages will appear on all affected display phones, regardless of programming. (See page 7-22 for more information.)

4.3 System messages can be printed to give service personnel and Inter-Tel engineers information about system status during troubleshooting. The programmer may enable any combination of the error message types. (See FEATURES, page 5-313, and PROGRAMMING, page 6-274, for more information.) The available message types are:

- **Information Messages:** These messages give information concerning system operation.
- **Warning Messages:** The warnings in these messages indicate that a condition exists which may affect system performance.
- **Alarm Messages:** Alarm messages indicate that a minor alarm has occurred, but that general system operation was not affected.
- **Severe Messages:** These messages indicate that a severe error has occurred in the system.

4.4 Some of the possible alarm messages, along with their associated troubleshooting procedures, are outlined in the following sections.

NOTE: If a procedure suggests replacing the CPC or CPU Card, first try using the save/restore program to save the customer's database (see page 6-65 in PROGRAMMING). It can then be restored (loaded) into a CPC or CPU Card, if necessary. Make sure that the database back-up battery on the CPC or CPU Card has a charge of at least 3.3VDC and that the battery back-up jumper strap is in the ON position.

4.5 To allow one administrator to monitor multiple nodes, the system provides both system alarms and network-wide alarms:

- **Network-Wide Alarms:** When an event occurs that generates a network-wide alarm, the alarm is broadcast to every node in the system.
- **System Alarms:** System alarms appear only on the node on which the alarm was generated.

4.6 Alarms are classified as follows:

- A000 - A019: Minor System alarms. Can be handled without calling a technician.
- A020 - A039: Minor Voice Processing alarms. Can be handled without calling a technician.
- A100 - A199: Major System alarms. Require assistance from a technician.

- A200 - A224: Major Voice Processing alarms. Require assistance from a technician.
- A225 - A244: Network alarms. These alarms are for either Call Processing or Voice Processing and are handled the same as the local alarm is handled.
- T1 Red LED Alarms (1-6): See page 7-25 for error conditions.

4.7 For a complete listing of all system messages and alarms, see the latest version of the *Message Print Diagnostics Manual* (part number 550.8018).

A. ALARM QUEUE

4.8 Software versions 8.2 and later support the Alarm Queue feature. This feature prioritizes system and network alarms based on severity and allows system administrators to view and handle critical alarms before addressing minor alarms. The administrator can then clear the individual alarm, or clear all the alarms in the queue (up to 30). When clearing alarms individually, the alarms are displayed in order of severity. The Emergency Alarm (A011) is the only priority 1 alarm. Other prioritized alarms have a 2, 3, or 4 priority, based on the severity of the alarm.

4.9 Not all alarms are prioritized. The numbered priority scheme is limited to alarms that can cause a major or minor system reset. Those alarms in the Alarm Queue that have a numbered priority are displayed before the alarms that do not have a numbered priority. (Priority 1 alarms have the highest priority.) Alarms that do not have a numbered priority are prioritized in the queue by date and time. When alarms are generated:

- The highest priority alarm is placed in the front of the queue, regardless of when lower priority alarms are generated. For example, if A114 (priority 3) and A116 (priority 4) are currently in the queue, but A119 (priority 2) is generated, A119 is placed first in the queue.
- Alarms with the same priority level are placed in the queue based on the time the alarm was generated. For example, if A010 (priority 3) is generated at 10:30 AM, and A012 (priority 3) is generated at 10:32 AM, A012 is placed in the queue after A010.
- If the queue contains 30 alarms, the oldest, lowest priority alarm is overwritten with the new alarm. For example, if the queue currently holds 30 alarms, 20 of which are priority 4, and a priority 3 alarm is generated, the oldest priority 4 alarm is overwritten.
- Repetitive alarms such as A125, are placed in the queue only once. If the alarm is regenerated, the alarm that is currently in the queue is overwritten with the new alarm data (if applicable) and time. For example, A125 is overwritten each time it is regenerated, which is every five minutes. This prevents the queue from being filled with duplicate alarms.

4.10 Networked systems containing nodes that are running version 8.1 or earlier require special attention because not all nodes will have an alarm queue. In networks such as this, a broadcast message is sent only when the most recent network alarm is cleared. When the message is sent, that network alarm is cleared on all nodes in the system.

4.11 After the alarm is cleared, any pre-8.2 nodes will not show any other alarms because these nodes do not have a queue to store alarms. 8.2 or later nodes, however, may still have additional alarms in the queue. When this happens, each 8.2 node must manually clear the other network alarms.

4.12 For example, suppose you have a three-node network: node 1 running 8.0; node 2 running 8.2; and node 3 running 8.2. Node 1 then generates two network alarms, one immediately after the other. At this point, node 1 shows one network alarm (the latest) and nodes 2 and 3 show two network alarms. Node 1 then clears the alarm, and a broadcast message is sent. Node 1 now shows no alarms, but nodes 2 and 3 still show the first alarm. Node 2 then clears the first alarm, but a broadcast message is NOT sent. Now nodes 1 and 2 show no alarms, but node 3 still has one in the queue. This alarm will stay in the queue until node 3 clears it or until the alarm is overwritten as previously described.

4.13 The following table shows which alarms have a numbered priority.

Table 7-5. *Alarms with a Numbered Priority*

ALARM	PRIORITY	ALARM	PRIORITY	ALARM	PRIORITY
A010	Low	A032	Low	A117	Low
A011	Critical	A100	Low	A118	Low
A012	Low	A101	Low	A119	High
A013	Low	A102	Low	A120	Med
A014	Low	A103	Low	A121	High
A015	Low	A104	Med	A122	High
A016	Low	A105	Low	A123	High
A017	Low	A106	Low	A124	Med
A018	Low	A107	Low	A125	High
A020	Low	A108	Low	A126	Med
A021	Low	A109	Med	A127	Med
A022	Low	A110	Med	A128	Low
A023	Low	A111	Med	A200	Low
A024	Low	A112	Med	A201	Low
A025	Low	A114	Med	A202	Low
A026	Low	A115	Med	A203	High
A031	Low	A116	Low	A204	High

B. MINOR ALARMS THAT ARE USER CORRECTABLE

4.14 Minor alarm numbers 000-019 are reserved for system problems that can be corrected without calling service personnel. These alarm messages (as they appear in the error message printout) currently include:

- **000 Alarm Automatically Cleared –** The alarm was automatically cleared.
- **001 Alarm Manually Cleared 'Ext. xxxxx'**– An administrator entered the Clear Alarm feature code (9850) and cleared the alarm manually.
- **002 Network Alarm Automatically Cleared** – The network alarm was automatically cleared.
- **003 Network Alarm Manually Cleared 'Ext. xxxxx'**– An administrator entered the Clear Network Alarm feature code (9851) and cleared the alarm manually.

- **010 Device x.y.z `xxxxx' Is Off Hook –** The indicated station is off hook and the Inactivity Alarm timer has expired. (The station's display shows STATION OFF HOOK, and the station's associated button on DSS/BLF Units and other phone speed-dial buttons flutters continuously.) Locate the station and place the handset back in its cradle.

NOTE: Calls being transmitted over the secondary voice path are not affected or interrupted by an off-hook alarm condition. Also, there is a system option to disable this alarm, if desired.

- **011 Emergency Call Made By Device x.y.z `xxxxx' –** A user at the indicated station dialed the Emergency Call feature code, 911 [999 in Europe].
- **012 SMDR Printer Off-Line –** The system has been unable to write to the SMDR printer for at least five minutes. The indicated printer or serial port is not functioning properly. Make sure the printer is turned on and has plenty of paper and ribbon. Check to see that the interface cable and power cord are properly connected.
- **013 Message Printer Off-Line –** The system has been unable to write to the Message Print printer for at least five minutes. The indicated printer or serial port is not functioning properly. Make sure the printer is turned on and has plenty of paper and ribbon. Check to see that the interface cable and power cord are properly connected.
- **014 Real-Time Clock Needs To Be Set –** The system real-time clock on the CPC or CPU Card has not been initialized or has been corrupted. Set the system time and date from an administrator station.
- **015 Hardware Real-Time Clock Inoperative –** During normal operation, the system has detected that the real-time clock circuitry on the CPC or CPU Card is not operating properly. The system is now using a software clock, which may not be as accurate. Check to make sure that all of the components on the card are properly seated and that no pins are bent. If the problem persists, replace the card and return it for repair.
- **016 No Delayed Major Because System Busy –** The system was unable to perform a scheduled delayed major reset because the system was not idle. The system will continue to try to perform the delayed major reset every five minutes. This could be caused if the system is really busy or if DB Programming is left connected. If the "Forced Reset If Not Idle" flag in DB Programming is enabled, the system will perform the delayed major regardless if the system is idle or not.
- **017 System Manager Link Connection Error –** There are many reasons the certificate can fail verification. Common errors are #9 (certificate not yet valid) and #19 (self-signed certificate). Error code 0 typically indicates an incorrect IP address, username, or password. Note that the error "string" comes from a generic SSL library. Because the endpoints can only display 16 characters, for any error strings that are greater than 16 characters, the system will show "CERTIFICATE ERR". The user can go into message print to view the true SSL error string.

4.15 Minor alarm numbers 020-039 are reserved for Voice Processing PC problems that can be corrected without calling service personal.

NOTE: Even though a Voice Processing PC alarm has been registered, the phone system itself may still be functioning properly.

4.16 The alarm messages (as they appear in the error message printout) currently include:

- **020 Voice Processing Unit: Printer Off-Line –** The system has been unable to write to the Voice Processing PC printer for at least one minute. The Voice Processing PC printer or parallel port is not functioning properly. Make sure the printer is turned on and has plenty of paper and ribbon. Check to see that the interface cable and power cord are properly connected.
- **021 Voice Processing Unit: Voice Mail Storage 80% Full –** The portion of the Voice Processing PC hard disk reserved for voice mail is 80% full. Have the users delete all old messages. If the alarm continues to occur, it may be necessary to install a larger hard disk.
- **022 Voice Processing Unit: Voice Mail Storage 100% Full –** The portion of the Voice Processing PC hard disk reserved for voice mail is completely full and cannot accept any new messages until more disk space is made available. Have the users delete all old messages. If the alarm continues to occur, either increase the assigned disk space or install a larger hard disk.

NOTE: In order to send out a broadcast message to all users telling them to delete their old messages, it is necessary to first clear up some disk space by deleting some existing messages.

- **023 Voice Processing Unit: SMDR Storage 80% Full –** The portion of the Voice Processing PC hard disk reserved for SMDR is 80% full. Empty the SMDR buffer using the Inside Track call accounting software. If the alarm continues to occur, empty the buffer more often or increase the size of the buffer.

NOTE: Changing the SMDR buffer size discards all stored SMDR records.

- **024 Voice Processing Unit: SMDR Storage 100% Full –** The portion of the Voice Processing PC hard disk reserved for SMDR is completely full. The oldest SMDR records are discarded as new ones arrive. Empty the SMDR buffer using the Inside Track call accounting software. If the alarm continues to occur, empty the buffer more often or increase the size of the buffer.

NOTE: Changing the SMDR buffer size discards all stored SMDR records.

- **025 Voice Processing Unit: Avdap Message –** This alarm message type is reserved for any alarm message text that the AVDAP wants to send.
- **026 Voice Processing Unit: E-mail Gateway Is Disabled –** The Voice Processing Unit E-mail Gateway is disabled because the number of mailboxes programmed for E-Mail Gateway has exceeded the maximum number permitted by the license. Disable E-mail Gateway for some mailboxes so that the total number of E-mail Gateway enabled mailboxes would not exceed the permitted license number.
- **031 VM Hacker Alert –** A voice mail subscriber entered an invalid password three times in a row.
- **032 Received Insufficient Bandwidth –** The IPRC [b.p.d] detected that the audio-receive timer has expired without receiving the first audio packet or the average-in-time-frame percentage has fallen below the threshold specified in the Voice Audio Session Configure Command.

C. MAJOR ALARMS REQUIRING ATTENTION FROM SERVICE PERSONNEL

4.17 Major alarm numbers 100-199 are reserved for system problems that require attention from service personnel. These alarm messages (as they appear in the error message printout) currently include:

- **100 Device x.y.z `xxxxx' Reset Due To Excessive Data Errors –** The indicated phone has detected an unusually large number of data errors from the corresponding keyset card and has been reset (dropping any active call). If the resets continue, the system will "lock out" the phone. The data errors may be caused by a defective phone (perform keyset self-test), defective wiring, defective station cable, or a defective keyset card.
- **101 Device x.y.z `xxxxx' Reset Due To Excessive Hardware Failures –** The indicated phone has been reset (dropping any active call) because the corresponding keyset card has detected an unusually large number of hardware errors. If the resets continue, the system will "lock out" the phone. The hardware errors may be caused by a defective phone (perform keyset self-test), defective wiring, defective station cable, or a defective keyset card.
- **102 Logical Port nn Tx Interrupt Is Not Operational –** The indicated serial port (nn) has a transmit interrupt problem. Replace the component that has the defective serial port (CPC, CPU128 Card, CPU020/PCM Card, OPC, or PCDPM) and return it for repair.
- **103 Logical Port nn Clock Is Not Operational –** The indicated serial port (nn) has a clock problem. Replace the component that has the defective serial port (CPC, CPU128 Card, CPU020/PCM Card, OPC, or PCDPM) and return it for repair.
- **104 CPU: Database Corruption –** The system database stored in battery-backed memory on the CPU Card appears to be corrupt. Although it may be corrupt, try saving the database onto backup disks (but not on top of a previously known-good database). Turn off the system AC power and turn it back on. The system should default the database and the red DATABASE ERROR LED on the CPU Card should be lit. Initialize the system clock through DB Programming (or do some other DB Programming) to turn off the LED. Then turn the system power off and back on again. If the database does not default the second time, the CPU CPU020/PCM Card may be okay. Try restoring the database. If the restore is not successful, restore an older version of the database. If the database defaults the second time, replace the defective CPU Card and return it for repair. Then try restoring the database in the new CPU Card. If the restore is not successful, restore an older version of the database.
- **105 CPU: Write-Protect Is Not Operational –** The circuitry that prevents the software or hardware from accidentally writing to the database is no longer functioning properly. Replace the CPC or CPU Card and return it for repair.
- **106 CPU: Real-Time Clock Is Not Operational –** During power-up, the system has detected that the real-time clock circuitry on the CPC or CPU Card is not functioning properly. The system is now using a software clock, which may not be as accurate. Check to make sure that all of the components on the card are properly seated and that no pins are bent. If this does not solve the problem, replace the card and return it for repair.
- **107 CPU: Watchdog Is Not Operational –** The watchdog circuitry on the CPC or CPU Card is not functioning properly. Replace the CPC or CPU Card and return it for repair.
- **108 CPU: Interval Timer Is Intermittent –** The interval timer on the CPC or CPU Card is not functioning properly. Replace the CPC or CPU Card and return it for repair.

- **109 CPU: Watchdog Expired During Minor Reset –** When the system attempted a minor reset, it did not recover and refresh the watchdog timer, so the hardware watchdog assumed the system was down. Freeze the system history immediately. Then use DB Programming to save the database and obtain the CP Error Information. Contact Inter-Tel Field service with this information.
- **110 CPU: Database Feature Bundling Error –** The system has detected an inconsistency between the PAL on the CPU Card (e.g., the hardware information about the number of feature bundling units available) and the information stored in the database. When the problem is detected, the system automatically defaults the database to the basic feature configuration. (This will prevent a computer "hacker" from stealing feature units.) This alarm also occurs when attempting to restore a database that has more features enabled than the PAL allows. In DB Programming, enable the appropriate number of features. If the problem persists, try another PAL. If the problem still persists, replace the CPU Card and return it for repair.
- **111 CPU: Battery Backed Memory Too Low –** The system has detected a battery-backed memory problem. Either the database is so large that the battery-backed memory cannot store further information (such as message requests, reminder messages, queue requests, hardware configuration changes, etc.) or there is a system software problem. Cancel all message requests, reminder messages, and queue requests. If this does not solve the problem, try saving the faulty database (for possible future evaluation) and contact Technical Support for assistance.
- **112 CPU: Running Low On System Dynamic Memory (Heap) –** The system has detected a dynamic memory problem. Either there is so much call handling activity that the dynamic memory on the CPC or CPU Card is running low or there is a software problem. The system uses "dynamic" memory for performing call handling functions, storing temporary information, etc. Each device in the system (CO trunk, phone, single-line set, Voice Processing PC port, etc.) requires some dynamic memory. In addition, each call that is currently active requires some dynamic memory.

  When the first memory-low threshold is met (currently set to less than 160k free), the system starts forcing multiple CO ring-ins to ring only at the primary attendant since ringing one station will consume less dynamic memory than ringing several stations. The alarm and a delayed major reset are activated at this time.

  If the condition persists and the second low-memory threshold is met (currently set to less than 60k free), the system starts to prevent new calls from being made. CO ring-ins are ignored, and station users cannot make calls (they see an ALL CIRCUITS BUSY display). The alarm and a delayed major reset are activated at this time.

  If the condition persists and no dynamic memory is available, the system immediately performs a major reset (drops all active calls) to free up the dynamic memory.

  If the customer's configuration has multiple trunks ringing into multiple stations and/or has multiple all-ring hunt groups with large numbers of stations in them, it can consume a lot of dynamic memory each time a call rings in because the call "fans out" to make one call to each device in the list. Try reducing the number of stations in each multiple ring-in group. If the problem still persists, contact Technical Support for further assistance.

NOTE: A CP Server is required to support off-node members of an all-ring hunt group.

- **114 T1 Card [dd] In Red Alarm –** The indicated T1 Card has detected some type of red alarm error condition for at least two to three seconds (e.g., loss of the T1 signal), either internally or from outside the system. All calls on the card are dropped. Check the alarm message and field service diagnostic output; then attempt to correct the problem.
- **115 Alarm History Frozen Due To System Exception –** The system has frozen the call processing history automatically because a system exception has occurred. Save the database and the CP Error Information and contact Inter-Tel Technical Support for further assistance.
- **119 Link To Secondary Cabinet Lost –** The system has detected a loss of communication with the PCM-F Card in the slave (third) chassis in a tri-chassis or quad-chassis configuration. Check to make sure the fiber-optic cables between the first (master) and third (slave) chassis are properly connected.
- **120 Static Memory Fault –** The system was unable to establish a region of static, write-protected memory. This may happen because the system could not open or create the static database file on disk, or the system found or created the database file on disk, but could not load it into program memory.
- **121 System History Frozen Due to a Freeze-On String –** The system froze the System History automatically because a "freeze-on" string was output by MSG Print. Use DB Programming to save the database and obtain the CP Error Information. These operations appear on the "Database Save and Restore" screen in the "Service" area. Contact Inter-Tel Field service with this information.
- **122 Hardware Key Not Found –** The system is equipped with a CP Server, and it does not have the proper security key attached to its parallel port or the proper PAL is not in socket U13 on the CPU Card. If the key is attached, make sure the parallel port is configured and working (LPT1 using IRQ7). If the parallel port is not functioning correctly, replace it or add a parallel port card (use LPT1 and IRQ7). If the key is defective, replace it.
- **123 PCMA is Offline** – One of the remote equipment chassis is offline in an ATM system. Make sure the indicated PCMA has the correct ATM PAL and that the cables are connected. If the PCMA Card is defective, replace it.

NOTE: The v7.0 ATM PAL is not compatible with v6.0 software. Make sure the ATM PAL in U13 is the correct part number (827.9683 for 7.0 and 827.9413 for 6.0). Version 8.0, however, supports any ATM PAL.

- **124 System Running on Backup Key: <xx> Hours Remaining** – The connection to the master redundancy CP Server has been lost, and the backup CP Server will continue to run for <xx> hours (up to 100 hours). Either the cable is removed or compromised, or the PC lost power or is failing. All devices in the equipment chassis and attaching chassis will continue to function until the time expires, at which point the audio is terminated. If the master PC had a major failure, remove the key on the master and place it on the backup PC to reduce the risk of losing audio resources.

NOTE: No alarm 124 will be posted if you manually stop the services on the master, or if you manually shut down Windows.

- **125 License Error - Reset in XXX Minutes** – The software license on the CPC or CP Server is invalid. An invalid software license is detected if the correct software license for the system version is not loaded or if the hardware does not match the software license. When this occurs, the *License Error - Reset in XXX Minutes* message appears on Administrator stations. Network alarm 244 is also generated, indicating *Call Technician*. When Alarm 125 is generated, the reset timer counts down from 240 minutes (four hours). The alarm is then generated every five minutes. If the software license is not validated before this timer expires, the system resets, dropping all calls. The alarm is generated again until the license issue is resolved. If these alarms are generated, you must log into DB Programming and upload the correct software license (see page 6-45). If you have uploaded the correct software license and the alarm is still generated, the license may be corrupt. Try downloading the software license from the Web site (the *edGe*) again and then upload the license. If this does not work, transfer the software license to a new CPC or CP Server security key (see page 7-103) and notify Inter-Tel immediately. For readers located in Europe, contact the Inter-Tel Europe office for licensing-related issues.
- **126 System History Frozen Due to Interval Freeze** – Automatic Diagnostics Delivery froze the System History automatically because of an interval freeze. (This alarm will not appear on the administrator's display, but it will print out in Message Print.)
- **127 Unlicensed IP Device Failed to Come Online** – The CPC or CPS received an on-line request from an IP device but there was not an IP device license available, therefore, the IP device cannot come on-line.
- **128 Audio Diagnostics Feature Activated** – Someone has activated the Audio Diagnostic Feature from an endpoint (extension).
- **136 Automatic Diagnostics Delivery Transmission Failure—**A transmission failure occurred while attempting to do an Automatic Diagnostics Delivery. This may be caused by networking problems, ADD Server failure, or bad configuration. The user should make sure that Automatic Diagnostics Delivery is programmed correctly, and the ADD Server is accessible. Contact Tech Support if this persists.

4.18 Major alarm numbers 200-224 are reserved for Voice Processing PC problems that require attention from service personnel.

NOTE: Even though a Voice Processing PC alarm has been registered, the phone system itself may still be functioning properly.

4.19 The alarm messages (as they appear in the error message printout) currently include:

- **200 Voice Processing Unit: Hard Drive Failure –** The system has detected a possible Voice Processing PC hard disk failure. If the problem persists, save the current database (if possible) and replace the hard disk.
- **201 Voice Processing Unit: Channel(s) Failure –** The system has detected a failure of one or more of the voice channels between the phone system and the Voice Processing PC. If the problem persists, check the cabling between the two and reboot the PC. If the problem still persists, check for a defective Audio Interface Card, Voice Processing Card, CPU020/PCM Card, CPC, or Options Card.

- **202 Voice Processing Unit: Database Corruption –** The Voice Processing applications database stored on the PC's hard disk appears to be corrupt. Reboot the PC. If the problem persists, try saving both the system and Voice Processing databases (for possible future evaluation), then restore older valid versions or reprogram manually. (It is possible to restore just the system database and then reprogram the Voice Processing database, if desired.) If the problem still persists, try replacing the hard disk. If this does not solve the problem, contact Technical Support for further assistance.
- **203 Voice Processing Unit: Communications Link Down –** The system has detected a loss of communications between the phone system and the Voice Processing PC. This alarm is generated whenever the PC crashes or is rebooted, the RS232 cable is disconnected, or a communications channel is rendered inoperable (e.g., due to a bad serial port or RS232 cable). First verify that the problem still exists by placing a call to a valid voice mail extension number.

  NOTE: If any message other than "xxxxx IS UNPLUGGED" appears, the communications link is now okay. If not, check the RS232 cable and make sure it is properly connected, and verify that the Voice Processing PC communications port is still enabled and set to 9600 bps and software handshaking. If the problem persists, reboot the Voice Processing PC. If the communications link is still down, reboot the phone system. (This will drop all calls). If rebooting both systems does not clear the problem, replace the RS232 cable or replace the component that has the defective serial port (CPU/MEM Card, CPU020/PCM Card, CPC, or Disk Controller Combination Card) and return it for repair.

- **204 Voice Processing Unit: Version Mismatch –** The version of software on the Voice Processing PC is not compatible with the version of software on the phone system. Upgrade one or both to the appropriate version.

4.20 If any other alarm messages are encountered, note the circumstances under which the message(s) occurred and contact Technical Support.

D. MAJOR ALARMS THAT REQUIRE IMMEDIATE ATTENTION

4.21 If a system-wide failure is detected, a MAJOR ALARM message appears on all display phones. If the major alarm message appears only on a group of phones, their associated Keyset Card may have failed. If the major alarm message appears on a single phone, the phone or its cabling may be defective.

4.22 Because major alarms indicate that all or part of the system is inoperative, they require immediate attention from service personnel. When a system-wide major alarm occurs, do the following:

1. Attempt to reset the system by using the reset switch on the CPU Card or by using the programming PC (see PROGRAMMING, page 6-271). If the system does not recover from the alarm, continue to the next step.

   NOTE: The CPC does not have a reset switch.

2. Turn OFF the AC POWER for at least ten seconds and then turn it ON again.
3. If the system still does not recover from the alarm, check the system voltages as outlined on page 7-4. Using the troubleshooting charts beginning on page 7-26, try to isolate the defective part(s). If it is determined that the CPC or CPU Card (or any other part) is faulty, return it for repair and include any indicated error messages in the problem description.

   NOTE: When returning a faulty card, indicate all applicable error messages on the material return authorization (MRA) tag. For more information on returning equipment, see page 7-108.

E. NETWORK ALARMS

4.23 Network alarms (225-244) are used internally to distinguish a local alarm from a network alarm. These network alarms indicate remote node alarms. Network alarms indicate the name of the node on which the alarm occurred. The node name is obtained from the username. For example, when a remote node receives an emergency call alarm the main administrator will see NET ALARM #11 x<EXT> EMERGENCY. The second line of the display will state the node where the alarm originated. Alarms 225-244 are not displayed on the administrator's endpoint.

4.24 Each node has two flags (see page 6-261) which determine whether the node broadcasts or receives network-wide alarms.

- The **Send Network Alarms** flag determines whether a node will broadcast alarms that occur on that node to the rest of the network.
- The **Receive Network Alarms** flag determines whether the node will receive and display alarms sent by other nodes in the network.

4.25 To differentiate between network-wide and local alarms, network-wide alarms appear on administrator phones preceded by "NET ALARM" and local system alarms are preceded with "SYS ALARM." Network-wide alarms will override system alarms on an administrator's phone display. An administrator can clear a network-wide alarm on the local node only or on every node in the network.

- **Clear Network Alarm (9851):** This feature code clears network-wide alarms on every node in the network, but will not affect system alarms. The Clear Network Alarm feature code may be entered on any node in the network, but the Send Network Alarms flag must be set for the administrator to clear alarms on other nodes in the network.
- **Clear System Alarm (9850):** This feature code clears all local and network-wide system alarm displays on your node.

4.26 Network alarms correspond to local alarms, so alarms 225 - 244 are not displayed on the administrator's endpoint. The following network alarm numbers are shown for reference only:

- **225 Emergency Call Made by Device –** A user at the indicated station and node dialed the Emergency Call feature code, 911 [999 in Europe].
- **226 Voice Mail Storage 80% Full –** The hard disk on the Voice Processing Unit system is filled to 80% of capacity.
- **227 Voice Mail Storage 100% Full –** The hard disk on the Voice Processing Unit system is filled to capacity.
- **228 SMDR Storage 80% Full –** The portion of disk space on the Voice Processing Unit hard disk reserved for SMDR storage is filled to 80% of capacity.
- **229 SMDR Storage 100% Full –** The portion of disk space on the Voice Processing Unit hard disk reserved for SMDR storage is filled to capacity.
- **230 AVDAP Generic –** This alarm message type is reserved for any alarm message text that the Voice Processor wants to send. Refer to the *Voice Processing Diagnostics Manual* (part number 550.8019) under alarm number 25.
- **231 Voice Mail Reset –** The Voice Processing E-mail Gateway is not functioning.
- **232 Hard Drive Failure –** The hard drive on the Voice Processing Unit has failed.
- **233 Channel(s) Failure –** This alarm indicates that one of the Voice Processing Unit voice boards is faulty or that a voice board software driver problem has occurred.
- **234 Database Corruption –** One of the Voice Processing Unit database files has been overwritten or otherwise corrupted (e.g. there is an error on the hard disk which has resulted in invalid data being placed into the database files). A system reset may occur due to the invalid database.
- **235 Communications Link Down –** The Voice Processing Unit was rebooted, the Voice Processing Unit RS-232 cable was disconnected, or the communication channel was otherwise rendered inoperable (e.g., the System serial port is faulty, the Voice Processing serial port is faulty, the RS-232 cable is faulty, etc.).
- **236 Version Mismatch –** The version of Voice Processing Unit connected to the system is incompatible or out-of-date.
- **237 No Response from Other Fiber Controller –** The fiber-optic link between the primary and secondary cabinet is not functioning properly. This could be due to either the fiber-optic cable being removed or compromised, or due to the secondary cabinet losing power. All devices in the secondary cabinet will be non-functional (i.e., off-line). The system will drop all calls that were in progress between cabinets when the link was lost.
- **238 T1 Card <card> In Red Alarm –** Refer to M8000 and M8001 in the latest version of the *Message Print Diagnostics Manual* (part number 550.8018).
- **239 CP History Error –** The system froze the system history automatically because a software exception occurred.

- **240 Battery Backed Memory Too Low –** A user either requested a reminder message, left a message for another station, or queued onto another device. However, the system was too low on Battery Backed memory, so it could not store the information. It is also possible that a technician tried to add more stations, trunks, or boards to a system when Battery Backed memory was already low.
- **241 Running Low on System Dynamic Memory (Heap) –** Too much call activity (see below) or a software error.
- **242 System History Of Node <frozen_node> Has Been Automatically Frozen Due To A Software Exception –** The system history was automatically frozen due to a software exception. There are places within the code of Call Processing Software that request automatic freezes when certain exceptional conditions arise.
- **243 System History Of Node <frozen_node> Has Been Automatically Frozen Due To A Freeze-On String –** The system froze the System History automatically because a "freeze-on" string was output by MSG Print.
- **244 Invalid Software License Loaded –** The software license on the CPC or CP Server is invalid. An invalid software license is detected if the correct software license for the system version is not loaded or if the hardware does not match the software license.

F. T1 ALARMS

4.27 When a T1 Card's Red Alarm LED is flashing, check the error message output to see if any of the following error conditions have been detected:

- **Alarm 1, Receiver Loss of Signal from Network:** The card is not properly receiving the T1 signal from the public network. Check for improper connection. If the problem persists, contact the T1 provider.
- **Alarm 2, Transmitter Loss of Signal from T1 Card to Network:** The card is not properly transmitting the T1 signal to the public network. Check for improper connection. If the problem persists, replace the card and return it for repair.
- **Alarm 3, Receiving Yellow Alarm:** The card is receiving a yellow alarm error condition from outside the system. Check for improper connection or wiring. If the problem persists, contact the T1 provider.
- **Alarm 4, Receiving Blue Alarm:** The card is receiving a blue alarm error condition (also called Alarm Indication Signal [AIS]) from outside the system. Check for improper connection or wiring. If the problem persists, contact the T1 provider.
- **Alarm 5, T1 Signal Out-of-Frame:** There is a problem with the master/slave synchronization or the framing scheme. Check DB Programming. and check for improper connection or wiring. If the problem persists, contact the T1 provider.
- **Alarm 6, Severely Errored Seconds:** Indicates that one or more severe errors have been detected for at least one second. Check for improper connection or wiring. If the problem persists, contact the T1 provider.
- **Alarm 7, Transmitting Blue Alarm:** If error condition (1) above is detected (loss of receive signal), the card sends out a blue alarm signal.
- **Alarm 8, Remote Loopback State:** Currently reserved for engineering diagnostic purposes only.

5. TROUBLESHOOTING CHARTS

5.1 The simplified troubleshooting charts located on the following pages list symptoms, possible causes, and corrective actions for problems. Look up the problem in the appropriate chart and perform the corrective actions in the order given. The troubleshooting procedures for correcting equipment failures have been divided into the following categories:

- System (see page 7-27)
- CO Trunks (see page 7-32)
- CPC Modem (see page 7-37)
- Features (see page 7-37)
- Phones (see page 7-42)
- Single-Line Sets (see page 7-47)
- Phantom Devices (see page 7-50)
- DSS/BLF Units (see page 7-52)
- Voice Processing PC and Applications (see page 7-53)
- IPRC—Version 9.0.0 (see page 7-59)
- IP Devices (see page 7-61)
- Network (see page 7-63)
- IP Networking (see page 7-65)

NOTE: Throughout the troubleshooting section of the manual, there are numerous references to replacing the defective part and returning it for repair. However, before returning any part, proper troubleshooting procedures should be used to verify that the part is actually defective. For example, if a Digital Keyset Card (DKSC) appears to be defective, swap it with a "known good" DKSC presently installed in the system. If the problem follows the suspect card, it can be considered defective. For more information on returning defective equipment, see page 7-108.

5.2 For v1.x firmware device IPRC troubleshooting information, see the *IP Devices Installation Manual* (version 1.3 or later).

A. SYSTEM ISSUES

Table 7-6. *System Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
Repeated occurrence of all calls in progress dropping	AC line is not isolated and dedicated	Have isolated, dedicated line installed. See page 3-10 in INSTALLATION for details.
	Defective power supply	See INSTALLATION, page 3-83 or 3-92, to perform the power supply electrical test. Replace the power supply if faulty.
	Equipment chassis located near a strong magnetic field or other potential source of interference (copy machines, power transformer, etc.)	Relocate the equipment chassis a minimum of 20 feet (6 meters) from any equipment that is a potential source of interference.
	IC-CO/CO-CO Disconnect timer(s) need(s) adjustment	See *dialed digits* field in SMDR, page 5-311 in FEATURES. Set timer(s) to a higher value. See PROGRAMMING, page 6-329.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
	NOTE: The central office must provide a minimum of 18mA loop current	
All phones in the system are inoperative; no LED indication when a trunk or call button is pressed; +24V measurement is not present	Open or loose connection in the cable between the power supply and the chassis backplane, or a defective cable	Turn off the AC power. Check to see that the backplane-to-power supply interface cable is properly connected. Repair or replace the power supply, the cable, and/or the backplane if the connection is faulty.
	Defective power supply or connector	Use a voltmeter to check the +24VDC voltage on the chassis backplane or power supply. If the volt age is not +24VDC ±5%, replace the power sup ply and/or the cable.
	Defective keyset card	Remove all keyset cards from the chassis. Replace the cards one at a time and check the system voltages on the chassis backplane or power supply (see page 7-3), until the defective card is isolated. Replace the faulty card.
	Defective chassis backplane	Check the system voltages on the chassis backplane (see page 7-3). Replace the chassis assembly or the chassis backplane if necessary.
Database restore aborts before finishing	PC's power saver feature shuts down the PC and severs the communications link	Disable the PC's power saver feature.
DISA inoperative	User error	See FEATURES, page 5-118, for correct procedures.
	Telephone not compatible	User must dial in from a DTMF telephone.
	Programming error	Ensure that the trunk group is identified correctly as a day or night DISA trunk group. See PROGRAMMING, page 6-228.
	Defective trunk card	Replace the associated card.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.

Table 7-6. *System Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
(Continued) DISA inoperative	**NOTE:** Due to the natural characteristics of the CO trunk, the volume level of DTMF tones transmitted over the trunk may be substantially reduced before reaching the Inter-Tel system. This natural degradation in tone volume may adversely affect the reliability of the DISA feature. Other factors which can affect DISA performance are CO trunk noise and the quality and strength of the DTMF tones generated by the off-premises phone itself	
Unable to interface with computer call-up device (banking machine, answering machine, auto dialer, etc.)	Equipment being called is defective	Ensure that the called equipment is functioning correctly.
	DTMF digit duration/pause specifications of called equipment is incompatible with Inter-Tel system	Check with the equipment manufacturer for DTMF digit duration/pause specifications. Adjust DTMF Digit Duration/Pause timer. (See page 6-428 in PROGRAMMING.) Default value is 6/100 second.
	CO trunk is designated for dial-pulse signaling	CO trunk must be designated as DTMF. See page 6-244 in PROGRAMMING.
	Defective trunk card	Replace the associated trunk card.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
RFI/EMI present over conversations	AC power source or grounding incorrect	Verify that the AC circuit is isolated and dedicated and check for proper grounding (see pages 3-10 and 3-81 in INSTALLATION).
	Grounding point is source of RFI/EMI	While the system is running on AC power, temporarily remove the grounding wire to see if it is the source of the RFI/EMI. See page 3-81 in INSTALLATION for proper ground requirements.
	AC power source is causing RFI/EMI	If an external battery back-up power source is installed, switch system operation to battery back-up power by unplugging the power source's AC power cord (with grounding wire connected to chassis). If RFI/EMI stops, the AC power source is the cause. Install an RFI/EMI filter or equivalent on the AC outlet.
	Trunk from central office is picking up interference	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to check for interference. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telco.

Table 7-6. *System Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
(Continued) RFI/EMI present over conversations	**NOTE:** For further RFI/EMI troubleshooting assistance *while on site*, certified technicians should contact Technical Support with the following information: 1. Modulation (AM, FM, or other) and frequency of the interfering station (in Hz) 2. Broadcast power and distance between equipment chassis and broadcast antenna 3. Who hears RFI: – Outside call - inside party only? – Outside call - outside party only? – Outside call - both parties? – Intercom call - one or both parties? 4. Type of instrument(s) on which RFI is heard (e.g., Basic Digital Phone, Standard Display Phone, etc.)	
Faulty DID numbers displayed at attendant stations	User error (after dialing the correct 3- or 4-digit "base" number, the user entered in correct or incomplete "start" digits)	Only valid DID "start" digits can be properly processed by the system.
	Programming error	Ensure that all valid DID numbers have been programmed to ring in to the appropriate locations. (See page 6-244 in PROGRAMMING.)
	Defective DID trunk from central office (the system is receiving incomplete or mutilated DID "start" digits)	At the DKSC block, remove the bridging clips for the trunk. On the system side of the block, use a test set to verify the DID trunk connection and the system's ability to accept and process valid digits. Also, move the trunk to a known good DKSC circuit. If the problem follows the trunk, contact the telephone company.
	Defective SLA, DKSC, or T1C	Replace the associated SLA, DKSC, or T1C.
Out of memory error while loading DB Programming software on the programming PC	Memory conflict with the PC's autoexec.bat and/or config.sys files	Rename autoexec.bat file to autoexec.sav (or something similar) and rename config.sys file to config.sav (or something similar). Then create a basic config.sys file with: files=51 device=c:osimem.sys buffers=30 Reboot the PC without an autoexec.bat file and then re-install the software.
DID numbers routed before all digits are dialed	Non-programmable system timer expiring before CO completes digit transmission	The phone system allows seven seconds for DID input. After seven seconds, the system routes the call based on information received to that point. The Central Office must complete DID digit transmission within seven seconds.
The RX*n* LED on the ASX-200BX ATM switch is lit red	The connection to the chassis or CPS is down	Verify that the cables are connected properly. If using a fiber connection, switch the RX/TX connectors. If necessary, test the connection with a known, good cable. Replace the defective cable, if necessary.

Table 7-6. *System Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
The 5 Volts LED on the ASX-200BX ATM switch is lit red for one of the power supplies	The power supply is not plugged in, not turned on, or defective	Make sure the power supply is plugged into a properly grounded socket. If it is, make sure the power switch is turned on. If the LED is still red, see the Ericsson® (Marconi) documentation for information on replacing the power supply.
When the system is routing a normal amount of traffic through the switch, the Slave LED on slave PCMA cards is blinking. This LED does not blink if the system is idle	The slave PCMA cards are having difficulty synchronizing their local clocks to the master clock source	Configure the interfaces on the switch so that the constant bit rate (CBR) uses roundrobin instead of guaranteed bits. See page 7-69 for details.
DB Programming is not recognizing a PCMA card (displays "Unknown") after upgrading the system to include a new PCMA group. The Master and Slave LEDs on the new PCMA card are also blinking.	The backup CP Server does not have an updated static memory file that reflects the new card. This backup server is then trying to take over Call Processing.	Stop the Watchdog Service on the backup CP Server. Then start DB Programming. After the card is recognized, you can restart the Watchdog Service.
A delayed major reset was scheduled after programming changes were made.	The delayed major reset may have been scheduled for reasons other than changes to the database (e.g., minor reset, a heap corruption message or heap nearing full message).	Contact Tech Support for instructions on how to dump the delayed major reset information.
	Programming changes may have caused an exception (minor reset) or a heap corruption, which caused the system to schedule a delayed major reset.	
A CPS site has just upgraded to 9.0 or later. Upon the first system freeze, the system has a problem compressing the freeze directory.	The "Freeze" directory may have over 2 GB of data in it. The compression utility on the CPS cannot zip more than 2 GB of data.	Remove/backup the old "Freeze" directory before installing 9.0 or later software.
The new dump options in Diagnostics Monitor do not exist.	The user is using an older version of Diagnostics Monitor.	Use Diagnostics Monitor version 3.2.1 or later.
Executing the diagnostics dump resource command returns "The server did not recognize the last command."	The user has connected to a system running pre-v9.0 Call Processing software and does not support the new diagnostics feature.	Make sure the system is running v9.0 Call Processing software or greater.
The traceroute command always displays "no reply."	The routers or destination host on the network are programmed to ignore ICMP Echo requests or to not send ICMP Time Exceeded messages.	Contact the network administrator to enable these ICMP messages.

Table 7-6. *System Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
With the diagnostics mode enabled, the administrator cannot access the dump from his or her endpoint. The administrator sees “Invalid Extension Number.”	The administrator has not enabled the diagnostics mode.	Make sure the administrator has enabled the diagnostics mode (9900 in the U.S. and 9100 in the U.K.).
	The user has not entered the correct digits.	Verify the correct diagnostics code is used (9933 for the U.S., 9133 for U.K.). Also, verify that the correct digits are used for the dump.
	The node does not support the new diagnostics command.	Verify that the system is running v9.0 Call Processing software or greater.
The administrator station is not displaying Alarm 128.	The Audio Diagnostics Suppression System Flag is enabled.	Disable the Audio Diagnostics Suppression flag (under System\Flags).
Alarm 128 displays on an administrator’s endpoint.	A user completed the Audio Diagnostics feature.	Collect the Freeze which contains Message Print entry and other diagnostics data to submit to Technical Support for analysis.
Message Print output indicates that a SIP phone extension is invalid or incomplete.	The SIP phone is calling its own number or a number that is programmed but not networked.	Verify the cause but do nothing to attempt to fix the condition. The cause of the printout is the SIP server performing its normal cycle of events.

B. CO TRUNK ISSUES

Table 7-7. *CO Trunk Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
CO trunk inoperative throughout the system	Defective CO trunk from central office	At the associated CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.
	Trunk is dedicated to a secondary carrier requiring an access code	Verify the type of CO trunk. Instruct users to dial access code if required.
	Defective cabling or mis-wired amphenol connector on the trunk card	Using a test set, ensure presence and correct location of the CO trunk at the associated CO block.
	Programming error	Ensure that the trunk is assigned to the correct trunk group and stations have been given access to it. See page 6-244 in PROGRAMMING.
	Defective trunk card	Replace the associated trunk card.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Cannot obtain CO dial tone	If reorder tone is heard, programming error	Ensure that phone has outgoing access. See page 6-198 in PROGRAMMING. Also, check to see if the phone is programmed for ARS only. If so, check ARS programming, see page 6-291 in PROGRAMMING. Ensure trunk is assigned to the correct trunk group. See page 6-244 in PROGRAMMING.
	If progress tone is heard, user error	System is programmed to expect a forced account code. See PROGRAMMING, page 6-301.
	Selected trunk is designated as incoming-only	Cannot dial out on an incoming-only trunk.
	Defective trunk card	Replace the associated trunk card.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Low volume on all CO trunks; cannot break CO dial tone	Defective trunk card	Measure the system voltages. If not within tolerance, remove all trunk cards and check voltages again. If the voltages return to normal, replace the cards one at a time until the defective card is isolated and replaced.
	Defective CO trunk from central office	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.

Table 7-7. *CO Trunk Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
(Continued) Low volume on all CO trunks; cannot break CO dial tone	Open or loose connection in the cable between the power supply and the backplane	If, with all trunk cards removed, the system volt ages are still not within tolerance, turn off the system and check the cable and connector with an ohmmeter. Replace or repair the faulty cable.
	Defective power supply	With all trunk cards removed, measure the system voltages. If not within tolerance, replace the power supply.
	Hybrid balance mismatch	Select the balance network – normal or short – that best matches the trunk's impedance. See PROGRAMMING, page 6-244.
	NOTE: See also phone problems on page 7-43.	
Cannot break CO dial tone	CO circuit is programmed for wrong signaling type	Ensure that CO trunk and CO circuit use same type signaling (DTMF or dial pulse). See PROGRAMMING, page 6-244.
	Defective CO trunk from central office	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.
	Timer error	If trunk is DTMF, the DTMF Digit Duration/ Pause timer setting may not be compatible with the trunk. See PROGRAMMING, page 6-329.
	Defective trunk card	Replace the associated trunk card.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
	NOTE: See also phone problems on page 7-43.	
Cannot place an outgoing call; CO dial tone is present; intercom works	Programming error	Check station class of service (COS). See PROGRAMMING, page 6-198. Check that equal access and absorbed digit programming for the trunk group are correct. See PROGRAMMING, page 6-244. If using ARS, check ARS programming on page 6-291 in PROGRAMMING.
	Defective CO trunk from central office	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.
	Defective station instrument	Replace the station instrument and/or perform the keyset self-test as outlined in INSTALLATION, page 3-182.
	Selected trunk is designated as incoming-only	Cannot dial out on an incoming-only trunk.
	Defective trunk card	Replace the associated trunk card.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.

Table 7-7. *CO Trunk Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
Other station conversations can be heard on the CO trunk (crosstalk)	Defective CO trunk(s)	Isolate the trunk(s) with crosstalk by removing the bridging clips from the CO block. On the telco side of the block, attach a test set to each trunk and check for crosstalk. If present, contact the telephone company.
	Defective cabling or mis-wired amphenol connector on the trunk card	Using a test set, ensure presence and correct location of the CO trunk at the associated CO block.
	Defective station card	Replace the associated station card.
	Defective trunk card	Replace the associated trunk card.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Outside calls dropped during conversation or upon answering CO trunk	User error (trunk button being pressed after initial connection is made)	Instruct users not to press the trunk button while on a call. Or, if necessary, set the CO Reseize timer to a higher value. Default value is 3 seconds. See page 6-329 in PROGRAMMING. Or, program the station to disallow CO reseize. See PROGRAMMING, page 6-258.
	Defective CO trunk from central office	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.
	Insufficient loop current supplied by central office	Central office must supply 18mA minimum loop current.
	IC-CO Disconnect timer value is set too short	Ensure timer value is long enough to ignore normal interruptions in CO loop current. Default value is 0.6 seconds. See page 6-329 in PROGRAMMING.
	Defective station instrument	Replace the station instrument and/or perform the keyset self-test as outlined in INSTALLATION, page 3-182.
	Defective trunk or station card	Replace the associated card.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
CO trunk cannot be re seized	CO Reseize feature is disabled	Check station programming as described on page 6-198 in PROGRAMMING. If the CO Reseize option has been disabled, the user cannot reseize a trunk until it has been disconnected by hanging up or pressing another trunk button.
	CO Reseize timer is set too high	Set the timer to a lower timer value. Default value is 3 seconds. See page 6-329 in PROGRAMMING.
	User error	If the station is programmed with the validate account codes option and the user enters an invalid Account Code For All Calls Following, calls will not be allowed at the station until the code is removed or reprogrammed. See page 5-200 in FEATURES.

Table 7-7. *CO Trunk Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
Noise on CO trunk at all stations	Defective CO trunk	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.
	Defective trunk card	Replace the associated trunk card.
	Defective power supply	See INSTALLATION, page 3-83 or 3-92, to perform the power supply electrical test. Replace the power supply if faulty.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
T1 trunks have frequent “Controlled Slip” and “Out-Of-Frame” errors	Line Build-Out (LBO) setting incorrect	Use T1 Diagnostics to determine the current LBO status. Then use T1 Programming to change the setting. See page 6-126 in PROGRAMMING.
	Defective T1 trunk	Check for improper connection. If the problem persists, contact the T1 provider.
	Defective T1C	Replace the associated T1 card.
CO trunk remains seized after a call has been ended	Characteristic of some ESS central offices	Central office must provide disconnect signal. Or, install a trunk release module available from most supply houses.
	IC-CO or CO-CO Disconnect timer value set too long	Central office disconnect signal was not detected by IC-CO or CO-CO Disconnect timer. Default value of the IC-CO timer is 0.6 seconds; the CO-CO timer is 0.35 seconds. See page 6-329 in PROGRAMMING.
	Defective CO trunk	At the CO block, remove the bridging clips for the trunk. On the telco side of the block, use a test set to verify the CO trunk connection. Also, move the CO trunk to a known good CO circuit. If the problem follows the trunk, contact the telephone company.
	Defective station instrument	Replace the station instrument and/or perform the keyset self-test as outlined in INSTALLATION, page 3-182.
	Defective trunk card	Replace the associated trunk card.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Ground start trunks not functioning properly	System not properly grounded	Verify that the system AC circuit is isolated and dedicated (see page 3-10 in INSTALLATION) and check for proper grounding (see page 3-81 in INSTALLATION).
	Tip and ring are reversed	At the CO block, check to make sure that tip and ring are not reversed. See INSTALLATION, page 3-24. Unlike loop start trunks, ground start trunks are polarity sensitive.
	Programming error	Ensure that the trunks are identified correctly as ground start trunks. See PROGRAMMING, page 6-244.

Table 7-7. *CO Trunk Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
(Continued) Ground start trunks not functioning properly	60 Hz AC power is getting onto the trunks	Use a voltmeter to check the AC voltage between tip and earth ground while the trunks are connected to an LGC. If more than 5V (RMS) are measured, there is AC power contamination (e.g., due to faulty wiring insulation). Locate and re move the source of AC power contamination.

C. CALLER ID FORWARDING ISSUES

Table 7-8. *Caller ID Forwarding Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
The customer reports incoming trunk caller ID does not forward to an outbound PSTN call.	The original incoming trunk call did not have caller ID.	Enable service with the CO provider to ensure incoming trunk calls have caller ID.
	The system administrator has not enabled the “Propagate Original Caller ID for Unanswered Calls” flag for the correct CO trunk group.	Enable the “Propagate Original Caller ID for Unanswered Calls” flag for the correct CO trunk group.
	The outbound call is sent out a non-ISDN PSTN line.	For Caller ID Forwarding, the customer must use an ISDN line to forward caller ID.
	The original trunk is answered by the system (either through a Voice Mail application, human operator, or Unified Communicator).	Currently, the system does not support propagating caller ID for incoming trunk calls that are answered and then transferred back out to the PSTN.
	If the answer is none of the above, the CO may not support the ISDN message for propagating caller ID.	Call the CO provider to determine why the ISDN caller ID information is not forwarded. If the CO cannot resolve this issue, contact Inter-Tel Technical Support and provide the following information: 1) the ISDN switch type where outbound calls was made; and 2) freeze of this call scenario.
The customer reports station caller ID does not forward to external PBX.	The system administrator has not enabled the “Send Station Caller ID to Attached PBX” flag for the correct CO trunk group.	Enable the “Send Station Caller ID to Attached PBX” flag for the correct CO trunk group.
	The outbound call is sent out a non-ISDN PSTN line.	For Caller ID Forwarding, the customer must use an ISDN line to forward caller ID.
	An unsupported SIP gateway is used to connect the system to an external PBX.	Verify that the SIP gateway has been officially tested. If the SIP gateway is officially supported, call Inter-Tel Technical Support and provide the following information: 1) the ISDN switch type where outbound calls was made, 2) type of gateway used to connect to external PBX, 3) ISDN switch type settings on SIP gateway, and 4) freeze of this call scenario.

Table 7-8. *Caller ID Forwarding Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
The customer reports station caller ID name (and only name) is not propagating to an external PBX. The customer is using a SIP gateway to communicate to an external PBX.	Two specific conditions cause this to happen: 1) the outbound ISDN line is configured as a DMS-100; and 2) the originating extension is calling out ARS and ARS is routing the call to a remote node.	There is a scenario where station Caller ID Forwarding is used over a networked ARS configuration, and the station user name will not be passed in the ISDN calling party name field. If the outgoing ISDN line uses the “display IE” method to transfer the calling party name, the calling party name will not use the station's user name. Instead of the user name, the system uses the calling party name as the name.

D. MUSIC-ON-HOLD ISSUES

Table 7-9. *Music-On-Hold Issues*

Symptom	Possible Cause	Corrective Action
No Music-On-Hold (external music source connected).	The external music source is turned off or inoperative.	Check the external music source for proper operation.
	There is a programming error.	Make sure the following fields are programmed correctly: • Music-On-Hold settings for CO Trunk Groups • Music-On-Hold Profiles for CRTs
	There is a defective Music-On-Hold port.	Replace the defective Call Processing Card (CPC) or the Options Card (OPC).

E. CPC MODEM ISSUES

Table 7-10. *CPC Modem Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
Users cannot connect to the modem	The Force Minimum Bit Rate flag is enabled, and the connection speed is below the specified minimum.	Disable the Force Minimum Bit Rate flag.
	Poor line quality or other external factors are affecting the connection rate.	Set the Minimum Bit Rate field to 9600 or lower.
An incoming CO call is disconnected when the modem is disabled.	The primary attendant station is not set.	Make sure the primary attendant station is set in the Station-Related Information field so that the call is sent to the attendant when the modem is disabled.

F. FEATURE ISSUES

Table 7-11. *Feature Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
Feature does not work properly	User error	See FEATURES for procedures. Also, ensure that the feature is available on the software package installed.

Table 7-11. *Feature Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
	Programming error	Check feature code programming. See PROGRAMMING, page 6-159. Also, check user-programmable feature key programming. See FEATURES, page 5-132, and PROGRAMMING, page 6-308.
	Defective station instrument	Replace the station instrument and/or perform the keyset self-test as described in INSTALLATION, page 3-182.
	Defective station card	Replace the associated station card.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Cannot transfer CO or intercom calls to other stations	User error (e.g., wrong feature code)	See FEATURES, page 5-209, for procedures.
	Called station is in do-not-disturb	A station in do-not-disturb cannot receive transferred calls.
	Call transferred to an illegal intercom number	Use only valid intercom numbers.
(Continued) Cannot transfer CO or intercom calls to other stations	Defective station instrument	Replace the station instrument and/or perform the keyset self-test as described in INSTALLATION, page 3-182.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Cannot transfer calls to outside numbers	User error (e.g., wrong feature code)	See FEATURES, page 5-209, for procedures.
	Programming error	Check trunk access and toll restriction.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Cannot transfer incoming CO calls or place them on hold	User error	See FEATURES, page 5-209, for procedures.
	User error (trunk button being pressed after initial connection is made)	Set the CO Reseize timer to a higher value. Default value is 3 seconds. See page 6-329 in PROGRAMMING. Or, program the station to disallow CO Reseize. See PROGRAMMING, page 6-258.
	Programming error	Check user-programmable feature button programming. See FEATURES, page 5-156, and PROGRAMMING, page 6-308.
Cannot initiate a conference	User error	See FEATURES, page 5-213, for correct procedures.
	System capacity exceeded	See the maximum system capacities on page 1-5 in OVERVIEW.
	Defective station instrument	Replace the station instrument and/or perform the keyset self-test as described in INSTALLATION, page 3-182.
	Defective station card	Replace the associated station card.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.

Table 7-11. *Feature Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
Last number redial feature inoperative	User error	See FEATURES, page 5-253, for procedures. Phone may be programmed for last number saved.
	System speed-dial number identified as non-display	A system speed-dial number identified as non-display cannot be redialed.
	Defective station instrument	Replace the station instrument and/or perform the keyset self-test as described in INSTALLATION, page 3-182.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Cannot initiate a page	User error	See FEATURES, page 5-256, for correct procedures.
	All stations in the paging zone are busy, or a voice channel is not available	Reorder tone is heard. Wait several seconds and then attempt to place the page again. Paging requires a voice channel.
	All stations in the paging zone are in do-not-disturb	Reorder tone is heard if all stations in the zone are in do-not-disturb and if external paging for the zone is disabled.
	No stations are programmed to receive pages	Reorder tone is heard. Check paging assignment. See page 6-194 in PROGRAMMING.
	Defective station instrument	Replace the station instrument and/or perform the keyset self-test as described in INSTALLATION, page 3-182.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
House phone is not working properly or is inoperative	User error	Incoming calls take precedence over outgoing calls. See FEATURES, page 5-251.
	Programming error (data base)	Ensure that the station is designated as a house phone. See page 6-198 in PROGRAMMING. Also, make sure the station has been assigned the proper COS and trunk access.
	Programming error (data base or speed-dial)	Ensure that the correct numbers are in appropriate day number and night number (speed-dial) locations. See page 5-236 in FEATURES.
Cannot initiate a call for ward	User error	See FEATURES, page 5-229, for correct procedures.
	User attempting illegal for ward	Stations are not allowed to set call forward if it forms an unconditional loop, the receiving station is in do-not-disturb, or an invalid intercom number is dialed. COS and outgoing access are checked when a call is forwarded to an outside telephone number. Also, ARS cannot be used to forward to an outside number.
	Defective station instrument	Replace the station instrument and/or perform the keyset self-test as described in INSTALLATION, page 3-182.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.

Table 7-11. *Feature Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
Calls will not forward	User error	See FEATURES, page 5-229, for procedures.
	Illegal forward	Conditional forwards (e.g., if busy, if unanswered) may form an undetected loop. If a call forward request forms a conditional loop, the call returns to the first station.
	Defective station card	Replace the associated station card.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Station not receiving hunt group calls	User error	Hunt group calls may have been halted using the Hunt Group Remove feature code (see page 5-36 in FEATURES). Or, the station may be in do-not-disturb.
	Programming error	Check hunt group programming. See page 6-160 in PROGRAMMING.
Station is not receiving pages	User error	Pages may have been halted using the Page Re move feature code (see page 5-257 in FEATURES). Or, the station may be in do-not-disturb.
	Programming error	Check page zone programming for the station. See PROGRAMMING, page 6-209.
Station cannot be placed in do-not-disturb	User error	See page 5-257 in FEATURES for correct procedures.
	Programming error	Station is programmed to disallow do-not-disturb. See PROGRAMMING, page 6-198.
	Defective phone	Perform the keyset self-test as described in INSTALLATION, page 3-182, and replace the phone if faulty.
The supervisor cannot silent monitor a call.	The supervisor is not in the member's hunt group.	Program the appropriate station as the hunt group supervisor in DB Studio.
	The agent is on a peer-to-peer (P2P) call.	Program the agent in a network group that does not allow P2P calls.
	The agent is on a conference call.	The agent must terminate the conference call.
	The agent is receiving agent help.	Wait until the agent help hangs up the call.
	Another supervisor is currently joining the hunt group member's call.	Wait until the other supervisor hangs up the call.
The supervisor's record-a-call does not record an agent helper, or the hunt group member's record-a-call does not record the supervisor after the supervisor joined the call.	Known problem.	None. This problem is due to a hardware limitation in the current Axxess system. This problem may be corrected in future versions of Axxess.

Table 7-11. *Feature Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
A supervisor's silent monitor terminates.	The hunt group member has requested agent help.	None. The supervisor has terminated the call.
	The hunt group member puts the call on hold.	None. The supervisor has terminated the call.
	Another supervisor was silent monitoring and joined the call.	None. Only one supervisor can join a call per call.
	The call was transferred.	None. A silent monitor on a transfer announcement call will terminate after the system completes the transfer.
The supervisor cannot steal a call from an agent and sees VERSION MISMATCH STEAL REJECTED in the display.	The agent is talking to a party on a pre-v9.0 node. A supervisor cannot steal a call away when the other party is on a pre-v9.0 node.	Upgrade all nodes in the network to version 9.0 or later.
The user cannot access feature code 320.	The Audio Diagnostics station flag is not enabled.	Enable the Audio Diagnostics station flag (under System\Devices and Feature Codes\Stations\<Node>\<station>\Flags).
The user does not see the Record-A-Call menu after activating the Audio Diagnostics feature.	The Record-A-Call application does not exist.	Create a Record-A-Call application if one does not exist.
	The Record-A-Call application is not active.	Make sure the Record-A-Call feature is enabled in DB Studio.
	A supervisor is silent monitoring a call.	None. The Record-A-Call feature cannot be used on certain calls (such as Agent Help, Station Monitor, Conferences, Paging, etc.).

G. PHONE ISSUES

Table 7-12. *Phone Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
Phone inoperative; LED indication present while any button with an LED is held down; reorder tone is heard when button is pressed	Programming error (circuit identified as dual single-line sets (SLA); no reorder tone is heard)	Identify the circuit for phone use, not dual single-line sets (SLA). See page 6-118 in PROGRAMMING.
	System lockout caused by excessive data errors (displays SYSTEM LOCKOUT)	Remove and replace the line cord to reset the phone.
	Defective cabling or connections	Ensure that all cables are correctly connected to the modular jack as shown in Figure 3-1 on page 3-19 in INSTALLATION. Check for loose or open connections in the station cabling and the line cord.
	Defective phone	Perform the keyset self-test as described in INSTALLATION, page 3-182, and replace the phone if faulty.
	Defective DKSC, EKSC, or AKSC	Replace the associated DKSC, EKSC, or AKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Phone inoperative; no LED indication when any button is pressed; no audio is present	Open or defective thermistor on the associated DKSC, EKSC, or AKSC	If a short causes a thermistor to open, it automatically closes when the short is removed. Replace DKSC, EKSC, or AKSC if the thermistor is faulty.
	Defective phone	Perform the keyset self-test as described in INSTALLATION, page 3-182, and replace the phone if faulty.
	Defective cabling or connections	Ensure that 24VDC is present at the modular jack and polarity is correct. Check for loose or open connections in the station cabling and the line cord. See Figure 3-1 on page 3-19 in INSTALLATION.
	Defective DKSC, EKSC, or AKSC	Replace the associated DKSC, EKSC, or AKSC.
Erratic phone operation (lamp status incorrect)	Station cable exposed to interference	Ensure proper station cable runs. See INSTALLATION, page 3-16.
	Station loop limits exceeded	Perform the station loop resistance test as out lined on page 3-69 in INSTALLATION.
	Defective cabling or connections	Ensure that 24VDC is present at the modular jack and polarity is correct. Check for loose or open connections in the station cabling and the line cord. See Figure 3-1 on page 3-19 in INSTALLATION.
	Defective phone	Perform the keyset self-test as described in INSTALLATION, page 3-182, and replace the phone if faulty.
	Programming error	Ensure that the proper keymap has been assigned to the phone. See PROGRAMMING, page 6-306.

Table 7-12. *Phone Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
(Continued) Erratic phone operation (lamp status incorrect)	Defective DKSC, EKSC, or AKSC	Replace the associated DKSC, EKSC, or AKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
A group of 8 or 16 phones inoperative; no LED indication when a trunk button is held down; all affected phones are located on one DKSC, EKSC, or AKSC	Defective station amphenol connector or cable attached to the associated DKSC, EKSC, or AKSC	Remove the cable from the card. Using an ohm meter, verify the pinout of the cable. (See the information on page 3-44 to 3-59 in INSTALLATION.) Ensure that 24VDC is present at the modular jack and polarity is correct. Check for loose or open connections in the station cabling and the line cord.
	Defective cabling or DKSC, EKSC, or AKSC	Replace the associated cabling or card.
	Defective receptacle on the chassis backplane	Replace the chassis backplane.
Phone squeals on outside calls or when receiving a handsfree intercom call from a single-line station (feedback)	Speaker volume is too loud	Reduce feedback by lowering speaker volume using phone volume controls.
	Poor acoustics	Poor acoustics can cause poor quality on handsfree calls. Try placing a private call.
	NOTE: A two- to four-wire converter is used during communication between phones and single-line stations. Reflection is a normal characteristic of these converters. Feedback on intercom calls is eliminated when the single-line station user places a private intercom call by pressing the pound [hash] (#) button before dialing the intercom number or by entering the ring intercom always feature code – 377 (provided the phone user does not press the SPKR button to respond).	
Cannot break CO dial tone	Defective phone	Perform the keyset self-test as described in INSTALLATION, page 3-182, and replace the phone if faulty.
	Defective trunk card	Replace defective trunk card.
	Defective DKSC, EKSC, or AKSC	Replace the associated DKSC, EKSC, or AKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
	NOTE: See also CO trunk problems on page 7-33.	
Cannot obtain intercom dial tone; no tone heard; CO trunk works	Defective phone	Perform the keyset self-test as described in INSTALLATION, page 3-182, and replace the phone if faulty.
	Defective DKSC, EKSC, or AKSC	Replace the associated DKSC, EKSC, or AKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Cannot obtain intercom dial tone; no tone heard; CO trunk works	Defective phone	Perform the keyset self-test as described in INSTALLATION, page 3-182, and replace the phone if faulty.

Table 7-12. *Phone Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
	Defective DKSC, EKSC, or AKSC	Replace the associated DKSC, EKSC, or AKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Cannot break intercom dial tone	Defective phone	Perform the keyset self-test as described in INSTALLATION, page 3-182, and replace the phone if faulty.
	Defective DKSC, EKSC, or AKSC	Replace the associated DKSC, EKSC, or AKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Cannot place intercom call; IC dial tone is present, but reorder tone is heard when call is attempted	User error	Invalid intercom number or improper dialing procedure. See FEATURES, page 5-174, for procedures.
	Defective phone	Perform the keyset self-test as described in INSTALLATION, page 3-182, and replace the phone if faulty.
	Defective DKSC, EKSC, or AKSC	Replace the associated DKSC, EKSC, or AKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Cannot receive off-hook voice announce calls	Programming error	Ensure that the system option for off-hook voice announce is enabled (PROGRAMMING, page 6-258), that the called phone is programmed to receive OHVA calls, and that the calling phone is programmed to transmit OHVA calls (PROGRAMMING, page 6-209).
	User error	The called station is a single-line set or a phone that is programmed not to receive OHVA calls. Or, the called phone user may have blocked the OHVA call. See FEATURES, page 5-190, for details.
	Secondary voice path busy	The phone will not receive off-hook voice announce calls when the secondary voice path or speakerphone are in use. See FEATURES, page 5-190.
	Defective PCDPM Module	Check installation and connections. See INSTALLATION, page 3-191. Replace if defective.
	Defective phone	Perform the keyset self-test as described in INSTALLATION, page 3-182, and replace the phone if faulty.
	Defective DKSC, EKSC, or AKSC	Replace the associated DKSC, EKSC, or AKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.

Table 7-12. *Phone Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
Cannot place off-hook voice announce calls	Programming error	Ensure that the system option for off-hook voice announce is enabled (PROGRAMMING, page 6-258), that the called phone is programmed to receive OHVA calls, and that the calling phone is programmed to transmit OHVA calls (PROGRAMMING, page 6-209).
	User error	The called station is a single-line set or a phone that is programmed not to receive OHVA calls. Or, the called phone user may have blocked the OHVA call. See page 5-190 in FEATURES for more information.
	Secondary voice path busy	The called phone cannot receive off-hook voice announce calls when its secondary voice path or speakerphone are in use. See FEATURES, page 5-190.
	Defective phone	Perform the keyset self-test as described in INSTALLATION, page 3-182, and replace the phone if faulty.
	Defective DKSC, EKSC, or AKSC	Replace the associated DKSC, EKSC, or AKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Data device connected to phone not operating properly	User error	See manufacturer's operating instructions.
	Secondary voice path busy	The phone will not transmit data calls when the secondary voice path or speakerphone are in use. See page 5-269 in FEATURES.
	Problem with data device	Disconnect data device and check operation according to the manufacturer's instructions.
	PCDPM, MDPM, or AC transformer not installed properly or defective	Check installation and connections. See INSTALLATION, pages 3-191 and 3-204. Replace if defective.
	Data Port Module not installed properly or defective	Check Data Port Module installation and jumper strap settings. See INSTALLATION, page 3-204. Replace if defective.
	Defective phone	Try another phone.
	Defective DKSC, EKSC, or AKSC	Replace the associated DKSC, EKSC, or AKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Data noise in any off-hook condition	Defective cabling or connections (e.g., line cord, amphenol connector, bridging clip, etc.)	Check for loose or open connections, or crossed wires.
	Wrong type of line cord for 8-line AIM phones installed two to a circuit.	Replace the flat line cord with a twisted-pair line cord.
	Defective phone	Try another phone.

Table 7-12. *Phone Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
(Continued) Data noise in any off-hook condition	Defective DKSC, EKSC, or AKSC	Replace the associated DKSC, EKSC, or AKSC.
Optional headset inoperative	User error	Ensure the enable headset feature code (315) was entered. Check feature code programming to see if code was changed. See page 6-159 in PROGRAMMING.
	Incorrect or defective head set	Ensure the headset contains the appropriate microphone (dynamic or electret). Replace headset if necessary.
	Defective phone	Try another phone.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Delayed audio connection on incoming calls when using a headset with an Attendant Console PC	Headset in battery saving mode	Some headsets have a battery saving feature that turns the headset off after a certain period of inactivity. To prevent these headsets from missing portions of incoming calls, the station flag can be programmed to generate a “Send Alert Burst To Headset” that activates the headset before connecting to an incoming call. See PROGRAMMING, page 6-209. Or, if necessary, disable the feature and/or replace the batteries with a transformer.
	User error	Ensure the enable headset feature code (315) was entered. Check feature code programming to see if code was changed. See page 6-159 in PROGRAMMING.
	Incorrect or defective head set	Ensure the headset contains a dynamic microphone, or a carbon microphone and an external AC power source. Replace headset if necessary.
	Defective phone	Try another phone.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Optional AgentSet inoperative	Not in headset mode	Ensure the enable headset feature code (317) was entered. Check feature code programming to see if code was changed. At the phone, unplug the headset or handset (to take the AgentSet off hook) and enter the Headset On/Off feature code (317) to enable headset mode. Then plug in the headset or handset. See page 6-159 in PROGRAMMING.
	Incompatible headset	Use only an Inter-Tel headset
LRA device connected to analog phone not operating properly	Problem with LRA device	Disconnect LRA device and check operation ac cording to the manufacturer's instructions.
	Data Port Module not installed properly or defective	Check Data Port Module installation and jumper strap settings. See INSTALLATION, page 3-204. Replace if defective.
	Defective analog phone	Try another phone.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.

H. SINGLE-LINE SET ISSUES

Table 7-13. *Single-Line Set Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
Single-line set completely inoperative	Defective set	Replace the single-line set.
	Defective cabling	Check amphenol connector and station cabling. See INSTALLATION, pages 3-44 and 3-53 to 3-59.
	Defective SLC, SLA, DKSC, or T1C	Replace the associated SLC, SLA, DKSC, or T1C.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Single-line set inoperative; talk battery present	Defective set	Replace the single-line set.
	Defective SLC, SLA, or DKSC	Replace the associated SLC, SLA, or DKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Single-line set inoperative; calls ring in; talk battery is present	Defective set	Replace the single-line set.
	Defective SLC, SLA, or DKSC	Replace the associated SLC, SLA, or DKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
A group of single-line sets is inoperative; all are on the same SLC, SLA, DKSC, or T1C	Loose amphenol connector or cable	Ensure connector or cable is securely attached to the SLC, SLA, DKSC, or T1C.
	Defective cabling or SLC, SLA, DKSC, or T1C	Replace the associated cabling or card.
Single-line set will not ring for CO or intercom calls; talk battery is present; calls can be placed	Single-line set ring jumper strap not set for correct ringing	Set the single-line set's ring jumper strap for AC ringing. See INSTALLATION, page 3-211.
	Defective set	Replace the set.
	Defective SLC, SLA, or DKSC	Replace the associated SLC, SLA, or DKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Ring trip is not provided to a group of single-line sets; outgoing calls are not affected	Proper voltage absent from SLA or DKSC	Defective connection or incorrect installation of SLA or DKSC. Using a voltmeter, measure the voltage at the DKSC block. See installation, pages 3-53 to 3-59.
	Defective cabling	Using a voltmeter, measure the input at the am phenol connector. See Figure 3-7 on page 3-59 in INSTALLATION.
	Defective SLC, SLA, or DKSC	Replace the associated SLC, SLA, or DKSC.

Table 7-13. *Single-Line Set Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
A group of AC ringer-equipped single-line sets will not ring for CO or intercom calls; talk battery is present; calls can be placed; all stations are on the same SLA or DKSC	Defective power supply	See INSTALLATION, page 3-83 or 3-92, to perform the power supply electrical test. Replace the power supply if faulty.
	Defective SLC, SLA, or DKSC	Replace the associated SLC, SLA, or DKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
No AC ringer-equipped single-line set will ring for CO or intercom calls; talk battery present; calls can be placed	Defective power supply	See INSTALLATION, page 3-83 or 3-92, to perform the power supply electrical test. Replace the power supply if faulty.
	Defective SLC, SLA, or DKSC	Replace the associated SLC, SLA, or DKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Cannot obtain intercom dial tone; no sound is heard	Either a DTMF decoder, voice channel, or tone generator is not available	Single-line station user will hear busy tones when any of the necessary resources are not available. User may camp on.
	Defective set	Replace the single-line set.
	Defective SLC, SLA, or DKSC	Replace the associated SLC, SLA, or DKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Cannot place intercom call; dial tone present, but reorder tone heard	User error	Try the call again. User may have dialed an in valid number.
	Defective set	Replace the single-line set.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
Cannot break CO dial tone	Defective set	Replace the single-line set.
	Defective SLC, SLA, or DKSC	Replace the associated SLC, SLA, or DKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
	NOTE: See also CO trunk problems on page 7-32.	
Cannot place off-hook voice announce calls	Programming error	Ensure that the system option for off-hook voice announce is enabled (PROGRAMMING, page 6-258), that the station is programmed to transmit OHVA calls (PROGRAMMING, page 6-209). Also, Ensure the Ring Intercom Always feature is disabled.
	User error	Called station is a single-line set or a phone that is not equipped with a secondary voice path.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.

Table 7-13. *Single-Line Set Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
Single-line sets not receiving audio message waiting indications	Programming error	Audible Message indication for SL Sets flag must be enabled in the database. See page 6-258 in PROGRAMMING.
Single-line set does not receive caller ID information.	Single-line station flags may be programmed incorrectly in DB Studio.	Set the Caller ID-IC and Extended Ring Cadence flags to **Yes**.
	The DKSC 16+ with caller ID enabled firmware v5.5 or later or OPC with caller ID transmitters is not equipped.	If using a DKSC 16+ for DSP resources, verify that it has been updated to firmware v5.5 or later. If using the OPC for DSP resources, verify that the DSP resources have been properly configured as caller ID transmitters.
	The single-line device may not support the shortened number of an extension.	Replace the device with a single-line device that supports the shortened number of an extension number and the split ring of an IC call.
	The single-line device may stop collecting data at the second IC ring burst.	

I. PHANTOM DEVICE ISSUES

Table 7-14. *Phantom Device Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
When adding more phantoms, the system slows down.	Adding too many phantoms in the current database configuration. Phantoms, though they do not use physical hardware, still use similar processor resources as a physical endpoint.	Remove some phantoms from the system.
Axxess v9.x or earlier nodes cannot call an Axxess v11.0 phantom.	The off-node device for the phantom does not exist on the earlier node.	On the earlier node, create an off-node device for the v11.0 phantom extension. This will allow the older node to recognize the phantom extension and call it.
Endpoint displays INVALID EXTENSION NUMBER and swap endpoint fails.	An administrator accessed the swap endpoint operation for a phantom device.	As designed, a phantom device cannot swap endpoints because it does not have a physical hardware address.
When an administrator is scrolling through a list of endpoints, the endpoint displays CIRCUIT NONE.	An administrator is accessing administrator features for a phantom device.	As designed, there is no board:port:device number for a phantom device because it does not have a hardware address.
A phantom cannot forward to the public network.	A phantom does not have access to forward to the public network.	Enable the phantom's Manual Forward to Public Network flag.
	The phantom is not in the CO trunk group's outgoing access list.	Assign the phantom's outgoing extension to the corresponding outside access CO trunk group.
A phantom cannot forward off-node.	The phantom is not in the node trunk group's outgoing access list or node IP connection group.	Assign the phantom's outgoing extension to the corresponding outside access node trunk group or IP connection group.
	The local node does not recognize the off-node device.	Create the off-node device on the local node.
A phantom hunt group member cannot forward to the public network.	Like other forwarded endpoints, hunt group members cannot forward to the public network.	Phantom hunt group members cannot forward to the public network.

Table 7-14. *Phantom Device Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
The user cannot add more phantoms.	The default phantom value is too low for the current system.	If additional phantoms are necessary, Inter-Tel recommends increasing the phantoms value in On-Line Monitor (OLM) mode so the combined TDM/IP devices and the phantoms value does not exceed the system's license capacity. There are three ways to tell how many devices you have in your system: 1) Dump the System Device Information to message print through an administrator's endpoint. 2) Use Diagnostics Monitor/System Monitor (v3.2.1 or later) to dump the System Device Information. 3) Use DB Studio to view the System Device Information (View\System Device Information). If the system needs more phantoms that the system's licensable capacity, contact Inter-Tel Technical Support. The system may run slower by increasing the default phantom value. **NOTE:** Do not use OLM mode unless you are instructed to do so by Inter-Tel support personnel.

J. DSS/BLF UNIT ISSUES

Table 7-15. *DSS/BLF Unit Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
DSS/BLF Unit inoperative; no LED indication present while button is pressed	Defective cabling or improper installation	Check for loose or open connections. See INSTALLATION, page 3-205.
	AC transformer unplugged or defective	Check the transformer. Replace if defective.
	Defective PCDPM	Check installation and connections. See INSTALLATION, page 3-191. Replace if defective.
	Defective DSS/BLF Unit	Replace the DSS/BLF Unit.
DSS/BLF Unit LED indications incorrect	Programming error	Check DSS/BLF button assignments in PROGRAMMING, page 6-306.
	Defective DSS/BLF Unit	Replace the DSS/BLF Unit.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
	Cable exposed to interference	See INSTALLATION, page 3-16, for correct cabling procedures.
DSS/BLF Unit LED indications incorrect	Programming error	Check DSS/BLF button assignments in PROGRAMMING, page 6-306.
	Defective DSS/BLF Unit	Replace the DSS/BLF Unit.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.
	Cable exposed to interference	See INSTALLATION, page 3-16, for correct cabling procedures.
Calls are transferred to the wrong station	User error	See FEATURES, page 5-153, for procedures.
	Programming error	Check DSS/BLF button assignments. See PROGRAMMING, page 6-306.
Cannot place immediate off-hook voice announce calls (using the DSS/BLF button)	Programming error	Ensure that the system option for off-hook voice announce is enabled, that the immediate off-hook voice announce option is enabled, that the DSS/BLF station is programmed to transmit, and that the called phone is programmed to receive. See page 6-209 in PROGRAMMING.
	User error	The called station is a single-line set or a phone that is programmed not to receive OHVA calls. Or, the called phone user may have blocked the OHVA call. See page 5-190 in FEATURES for more information.
	Defective phone	Perform the keyset self-test as described in INSTALLATION, page 3-182, and replace the phone if faulty.
	Defective DSS/BLF Unit	Perform the DSS/BLF Unit self-test as described in INSTALLATION, page 3-206, and replace the DSS/BLF Unit if faulty.
	Defective DKSC	Replace the associated DKSC.
	Defective CPC or CPU Card	See page 7-3 for LED and system voltage information. Replace the card if faulty.

K. VOICE PROCESSING ISSUES

NOTE: If a Voice Processing applications problem cannot be resolved, Technical Support may instruct the installer on how to use the diagnostics software. Also, for additional troubleshooting procedures and important diagnostic information, refer to the new voice processing system diagnostics manual, which will be published at a later date.

Table 7-16. *Voice Processing Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
Phone system unable to communicate over voice channels with Voice Processing PC	Programming error	Make sure the appropriate time slot group (no. of Voice Processing voice channels) is assigned to each Voice Processing application. Voice channels are used for processing calls between the phone system and the Voice Processing PC. See PROGRAMMING, page 6-354.
	Loose or defective cabling between the phone system and the Voice Processing PC	Check the cabling connecting the system to the PC. The CPC or CPU connection controls ports 1-8; the OPC connection controls ports 9-40.
	Incorrect strap settings on the AIC or one or more of the VPCs	Ensure that the AIC and all VPCs have their jumper straps set in the proper positions. See page 3-213 in INSTALLATION.
	Defective AIC or VPC	Replace the defective card.
System programming PC unable to communicate with Voice Processing PC (the monitor may display a message about being unable to communicate with or losing its connection with the Voice Processing PC)	Programming error	Voice Processing programming cannot be accessed until a Voice Processing communications port has been established.
	Loose or defective RS232 cabling between the phone system and the Voice Processing PC	Check the RS232 cabling connecting the system to COM1 on the PC.
	Defective CPC, CPU, MEM, or Disk Controller Combination Card	Replace the defective card.
Monitor and keyboard unable to communicate with Voice Processing PC	Loose or defective cabling between the Voice Processing PC and the monitor or keyboard	Check the cabling connecting the PC to the monitor or keyboard.
	Monitor is wrong type for the Video Card installed in the PC	Make sure the monitor is the proper type (generally VGA) for the Video Card installed in the PC.
	Defective monitor or keyboard	Replace the defective monitor or keyboard.
	Defective Video Card or PC Motherboard	Replace the defective card.
Automated attendant not responding properly	User error	The caller must use a DTMF phone to enter digits. See VOICE PROCESSING FEATURES, page 4-16.
	DTMF tones not being interpreted correctly (see NOTE below)	If numbers appear to be misdialed frequently, due to trunk noise or other problems, use the digit translation feature as described on page 6-371 in PROGRAMMING.

Table 7-16. *Voice Processing Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
(Continued) Automated attendant not responding properly	Programming error	Automated attendant station(s) must be designated in the database. See PROGRAMMING, page 6-303.
	Defective AIC, VPC, or hard disk	Replace the defective PC component.
	NOTE: Due to the natural characteristics of the CO trunk, the volume level of DTMF tones transmitted over the trunk may be substantially reduced before reaching the Inter-Tel phone system. This natural degradation in tone volume may adversely affect the reliability of the automated attendant feature. Other factors which can affect automated attendant performance are CO trunk noise, the quality of the recording device, and the quality and strength of the DTMF tones generated by the calling phone itself.	
Automated attendant not receiving incoming calls; callers routed to primary attendant instead	Programming error	Ensure that the desired trunks (in day and/or night mode) are programmed to ring in directly to the proper automated attendant extension number. See PROGRAMMING, page 6-244.
	Loose or defective cabling between the phone system and Voice Processing PC	Check the cabling connecting the system to the PC.
	Defective AIC or VPC	Replace the defective card.
Automated attendant indicates that an extension number is invalid even though it exists on the system	The extension does not have an assigned mailbox or an extension ID number	Extension ID numbers provide the automated attendant application a means for transferring calls to extensions which do not have mailboxes. See PROGRAMMING, page 6-373.
Automated attendant transfers calls directly to voice mail without trying the extension number first	Programming error	Make sure the mailbox is associated with a corresponding extension number. See PROGRAMMING, page 6-376.
Unable to access desired Voice Processing application (intercom callers hear reorder tones and see a CALL CANNOT BE COMPLETED display; outside callers are rerouted to the primary attendant)	The number of call attempts exceed the ports available through the VPCs	If the problem persists, it may be necessary to purchase and install additional VPCs.
	Loss of RS232 connection between the phone system and the Voice Processing PC	Check the RS232 cabling connecting the system to COM1 on the PC.
	Voice Processing has slowed down and call processing has timed out	If the problem continues to occur, even during periods of low activity, reboot the Voice Processing PC.
Unable to access desired Voice Processing application (intercom callers see a XXXXX IS BUSY display and camp on until a voice channel is available; outside callers hear ringing until a voice channel is available)	All programmed voice channels are currently busy	Make sure the appropriate time slot group (number of Voice Processing voice channels) is assigned to each Voice Processing application. Voice channels are used for processing calls between the phone system and the Voice Processing PC. See PROGRAMMING, page 6-354.

Table 7-16. *Voice Processing Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
Unable to access desired Voice Processing application (intercom callers hear reorder tones and see a XXXXX IS UNAVAILABLE display; outside callers are rerouted to the primary attendant)	A database save or restore is in progress, the PC is being reset, or the PC is starting up for the first time	Try again within five or ten minutes.
Unable to access desired Voice Processing application (intercom callers hear reorder tones and see a XXXXX IS UNPLUGGED display; outside callers are rerouted to the primary attendant)	The Voice Processing PC power switch is off, the cabling between the system and the PC is disconnected or defective, or the Voice Processing PC is inoperative	Check to make sure the Voice Processing PC power switch is on, and check the cabling connecting the system to the PC.
Voice Processing applications are slow (e.g., voice prompts are slow and display messages are delayed)	The Voice Processing PC's high-speed (turbo) processor mode is turned off	For optimal performance, the Voice Processing PC is configured with the high-speed (turbo) mode *always* enabled. Do *not* disable the turbo mode setting.
Voice Processing PC is programmed with 12-32 voice channels, but no more than eight calls at a time are accepted	The OPC is not programmed to have a Voice Processing Unit connection	Configure the OPC to have a Voice Processing Unit connection. See PROGRAMMING, page 6-117.
	The number of programmed voice channels is less than the available hardware	Make sure the appropriate time slot group (number of Voice Processing voice channels) is assigned to each Voice Processing application. See PROGRAMMING, page 6-354.
	Loose/defective cabling between the Voice Processing Unit AIC and the chassis OPC	Check the AIC-to-OPC cabling.
	AIC in slave chassis	In tri-/quad-chassis installations, make sure the OPC is installed in the master dual-chassis.
When power is turned on, the Voice Processing PC does not boot up properly	Diskette in floppy drive	Ensure that the PC is not trying to boot off a diskette in the floppy drive.
	The two-pin AIC-to-reset switch cable is not connected properly	Be sure that pin 1 marked on each end of the cable matches pin 1 on the AIC and PC Mother board. (If connected incorrectly, the PC will be reset continuously or will not boot up at all.)
	The DB-25 modular adapter (supplied with the Voice Processing PC) is still connected to the COM2 serial port on the PC	Remove the DB-25 modular adapter when not being used
	The Voice Processing PC's high-speed (turbo) processor mode is turned off	The Voice Processing PC boots up much faster in turbo mode. Before shipping, the PC is configured with the high-speed (turbo) mode *always* enabled. Do *not* disable the turbo mode setting.

Table 7-16. *Voice Processing Troubleshooting Chart (Continued)*

<table>
<tr><th>Symptom</th><th>Possible Cause</th><th>Corrective Action</th></tr>
<tr><td rowspan="6">When power is turned on, the Voice Processing PC beeps once (for two seconds) and does not boot up properly. (See note below for a list of possible AIC driver errors.)</td><td>Incorrect DIP switch or jumper strap settings on the Audio Interface Card (AIC)</td><td>Ensure that the AIC DIP switches and jumper straps are set in the proper positions. See step 4 on page 3-214 in INSTALLATION.</td></tr>
<tr><td>Incorrect strap settings on one or more of the VPCs</td><td>Ensure that all VPCs have their jumper straps set in the proper positions. See step 5 on page 3-215 in INSTALLATION.</td></tr>
<tr><td>Software problem</td><td>The PC should be used for Voice Processing applications only. If necessary, delete any unneeded peripheral software and re-install the Voice Processing applications software as outlined in INSTALLATION on page 3-226.</td></tr>
<tr><td>Incorrect CMOS settings</td><td>Refer to the diagnostic materials generally included in a plastic sleeve attached to the inside of the chassis cover.</td></tr>
<tr><td>Defective AIC, VPC, or PC</td><td>Replace the defective card(s) or the PC.</td></tr>
<tr><td colspan="2">NOTE: If the corrective actions listed above do not solve the problem, connect a monitor and keyboard to see which AIC driver error is being generated, as listed below, and then contact Technical Support.

<table>
<tr><th>Error No.</th><th>Meaning</th></tr>
<tr><td>AIC0000</td><td>No AIC address specified.</td></tr>
<tr><td>AIC0001</td><td>Invalid AIC address specified. Valid addresses fall in the range from 200 through 3F8.</td></tr>
<tr><td>AIC0002</td><td>The driver did not find an AIC at address <card address>.</td></tr>
<tr><td>AIC0003</td><td>The driver was unable to initialize AIC. This problem may indicate an I/O address conflict or a problem with the AIC.</td></tr>
</table></td></tr>
<tr><td>Both internal and remote users have no message notification after receiving voice messages in their mailboxes</td><td>The Message Notification/ Retrieval application has not been set up</td><td>The system must be programmed with the Message Notification/Retrieval application in order to allow voice mail message notification and quick mailbox access. See PROGRAMMING, page 6-369.</td></tr>
<tr><td>Audio volume levels from Voice Processing applications (such as voice mail and automated attendant) are too low or too high</td><td>Programming error</td><td>To adjust the volume level for all of the voice channels used by the Voice Processing PC, see PROGRAMMING, page 6-346.</td></tr>
<tr><td rowspan="2">No audio from Voice Processing applications (such as voice mail and automated attendant)</td><td>Loose or defective cabling between the AIC and the chassis CPC, CPU, or OPC</td><td>Check the AIC-to-CPU/OPC cabling.</td></tr>
<tr><td>Incorrect strap settings on one or more of the Voice Processing Cards (VPCs)</td><td>Ensure that all VPCs have their jumper straps set in the proper positions. See step 5 on page 3-215 in INSTALLATION.</td></tr>
</table>

Table 7-16. *Voice Processing Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
(Continued) No audio from Voice Processing applications (such as voice mail and automated attendant)	On Windows-based Voice Processing Unit, Watchdog Service not running	On the AIC card, the same component that controls the voice paths is started only by the watchdog; if you do not start the watchdog, you will not get audio or hardware-triggered reboots. The watchdog is supposed to start automatically when the system comes up. You can check this by opening the Services Control Panel (Start/Settings/Control Panel, then choose the Services icon). The window that comes up lists alphabetically all of the services currently registered with that system. Voice Mail (Avdap Service) uses “Manual” startup because the Watchdog starts it. And the Watchdog Service should be “Automatic.” If necessary, change the settings. If the watchdog is not currently started, start it by highlighting it, then selecting Start.
	Invalid or missing prompt files	When the Voice Processing Unit starts up, if the prompt or menu files (english.pmt, english.ini, & menuinit.dat) are not valid (missing, defective, or the wrong version), the message “Faking an active language” will appear in the log file. You must reinstall the prompt and menu files as described on page 3-226.
	OPC not programmed to support voice mail	The Options Card must be configured to have a Voice Processing Unit connection. See page 6-117.
	One or more ports are blocked	This problem blocks one or more ports from handling a call. If a caller is connected to a blocked port, there will be no audio; you will hear a short period of silence, then the call is disconnected. The message “Allocating busy port” appears in the log. You must shut down the Voice Processing Unit, wait until several minutes have passed with no activity showing in the log display on the screen (or in avdapmon on an system), and reboot the system. Select Start/Shut Down/Restart to reboot the computer.
	Defective AIC, CPC, CPU, or OPC	Replace the defective card.
Fax-On-Demand feature not working properly	Programming error.	User group 1 and the home area code tables must be programmed before using the Fax-On-Demand feature. A fax library must also be established. See page 4-29 in VOICE PROCESSING FEATURES for more information. Check to make sure that a fax library has been established. See page 4-29 in FEATURES for more information.
	Incorrect switch or jumper settings on the Fax Card.	Ensure that the switches and jumpers are set in the proper positions.
	Defective Fax Card	Replace the Fax Card.

Table 7-16. *Voice Processing Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
Unable to record any voice mail greetings or messages and/or unable to hear any voice prompts	Incorrect strap settings on one or more of the Voice Processing Cards (VPCs)	Ensure that all VPCs have their jumper straps set in the proper positions. See step 5 on page 3-215 in INSTALLATION.
	Loose/defective cabling between the Voice Processing Unit AIC and the chassis CPC, CPU, or OPC	Check the AIC-to-CPU/OPC cabling.
	Loose/defective cabling between the AIC and the VPCs	Check the AIC-To-VPC cabling.
	Defective VPC	Replace the VPC.
Japanese language voice prompts unavailable	Japanese voice prompts not installed	Install the Japanese voice prompts as outlined on page 3-226 in INSTALLATION.
	User and/or programming error	See page 5-160 in SYSTEM FEATURES for proper operation.

L. IPRC ISSUES

Table 7-17. *IPRC Troubleshooting Chart*

SYMPTOM	PROBABLE CAUSE	SOLUTION
User cannot make IP networking calls.	The system allocates a VoIP resource for each IP networking call (above what is needed for making the simple call). If resources are unavailable, the call will not be completed and will camp-on at the node.	Get more VoIP resources.
		Dedicate VoIP resources for networking usage.
User cannot make IP networking calls.	The system allocates a VoIP resource for each IP networking call (above what is needed for making the simple call). If resources are unavailable, the call will not be completed and will camp-on at the node.	Get more VoIP resources.
		Dedicate VoIP resources for networking usage.
IP endpoints will not come online.	The system pre-allocates one VoIP resource for each IP device. If the system has insufficient VoIP resources, the IP device will not be functional.	Get more VoIP resources.
		Dedicate VoIP resources for endpoint usage.
	The IPRC's ethernet cable is unplugged.	Verify that the ethernet cable is plugged in.
IPRC doesn't come online.	By design, the system will prevent IPRC firmware version older than 8.1 to come online.	Upgrade the IPRC to the latest version.
IPRC networking or 32-Devices card can't be found in the system.	By design, the system automatically converts IPRC networking or 32-Devices card to be the IP Resource Card. The IPRC networking and 32-Devices cards are replaced by the new IPRC configuration.	Find the card in the same slot as the previous IP Networking or 32-Device IPRC.
Pals do not work.	Version 9.0 or later system doesn't support pals.	Check that the appropriate license is loaded to allow for the desired number of VoIP resources.
A call has no audio on a node with multiple IPRCs.	An IPRC's Ethernet Cable is unplugged.	Check that the IPRC's Ethernet cable is plugged in.
Networking data connections between nodes do not load balance between multiple IPRCs.	An IPRC's ethernet cable is unplugged.	Check that the IPRC's ethernet cable is plugged in.
A node cannot connect to a node with multiple IPRCs.	An IPRC's ethernet cable is unplugged on the node with multiple IPRCs.	Check that the IPRC's ethernet cable is plugged in.

Table 7-17. *IPRC Troubleshooting Chart (Continued)*

SYMPTOM	PROBABLE CAUSE	SOLUTION
IPRC does not appear in IPRC diagnostics page even though it is plugged into the cabinet.	CP agent is not connected to System Manager.	Make sure CP agent is connected to System Manager.
	CP agent is running a pre-9.0 version of Axxess.	Make sure CP agent is running Axxess 9.0 or later.
	IPRC is not correctly plugged into the cabinet.	Make sure IPRC is correctly plugged into cabinet.

M. IP DEVICES ISSUES

Table 7-18. *IP Devices Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
IP Phone displays VOIP RESOURCE IS UNAVAILABLE.	The number of programmed IP devices exceeds the number of available voice channels, and all programmed voice channels are in use.	Make sure the voice channels are set to IP Endpoint (see page 6-146). If necessary, consider adding another IPRC or a daughter card.
A call was established between two IP phones, but there is no audio.	The IP phones are programmed for P2P audio, but they do not support the same vocoder.	Remove the IP devices from the Network Group. Consider adding them to a Network Group that contains IP devices that support the vocoder.
	The IP phones are programmed for P2P audio, but they do not support the same number of audio frames per IP packet.	Remove the IP devices from the Network Group. Consider adding them to a Network Group that contains IP devices that support the same number of audio frames per IP packet.
	There is a firewall or NAT between the two devices.	Place the devices into different Network Groups or disable the **Use Peer-To-Peer Audio** flag for the group.
A call was established, but the device cannot send DTMF digits.	The IP phones are programmed for P2P audio, but they do not support the same DTMF encoding setting.	Remove the IP devices from the Network Group. Consider adding them to a Network Group that contains IP devices that support the same DTMF encoding setting.
The Upload Utility is not connecting to the IP devices.	The IP devices have version 1.x firmware, and the IPRC has v8.1 firmware. Because the IP devices cannot connect to the card, they are constantly retrying the connection and blocking the Upload Utility.	In general, you should upgrade the IP devices before you upgrade the IPRC (see page 3-156 for details). If, however, you have already upgraded the IPRC, you must manually place the devices in the download state before you upload the firmware.
The network is experiencing audio and/or connection problems.	There is a port conflict or the firewall, NAT, or router is blocking the port.	Try changing the ports associated with IP call control and/or audio. Make sure none of them conflict with ports that other protocols use (e.g., SIP uses 5060).
The Upload Utility is not working.	You have not entered a password for the connection.	Make sure the Password field is complete before you click **Start**. See page 3-178 for details.
Accessing the Record-A-Call or Agent Help feature while using an IP SLA results in the call ringing and then disconnecting as soon as the agent helper or voice mail application answers.	The IP SLA is currently on a peer-to-peer (P2P) call with another IP device. Record-A-Call and Agent Help features are not available on P2P calls.	Remove the IP SLA from a Network Group that supports P2P.

Table 7-18. *IP Devices Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
Incoming and outgoing calls do not reach the destination.	Incorrect Database or SIP gateway programming. SIP gateway is not on the network. CPC or CPS is not on the network.	Verify the SIP gateway is plugged into the network and the correct IP address is programmed in.
		Verify the SIP gateway is configured using the AudioCodes MP-114 Web interface.
		Ping the SIP gateway address and if successful, hook up a network monitor and determine if the SIP packets are being sent or received properly.
There is a One-Way Audio or No Audio problem.	Incorrect Database or SIP gateway programming, There may be Session Description Protocol (SDP) problems.	Verify that the network groups are programmed correctly for both the SIP trunks and any IP endpoints involved in the call.
		Identify if the audio should be peer-to-peer or not.
		If the audio is one-way, hook up a network monitor tool and verify if the SDP sent contains the destination's correct IP address and port number, and the vocoder type supported.

N. NETWORK ISSUES

Table 7-19. *Network Node Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
Cannot access a station on another node. Display shows DESTINATION UNREACHABLE or OUTGOING ACCESS DENIED.	The station or application making the call does not have outgoing access for the Node Trunk Group that it needs	Program a Node Trunk Group, if one does not exist. Program outgoing access for the station or application. See PROGRAMMING, page 6-228. **NOTE:** You can use an extension list to make this task easier. See page 6-158 for information about extension lists.
When calling a station on another node, using the Automated Attendant, calls go directly to voice mail without trying the extension number first	The off-node device is not associated with a mailbox of the same number	Make sure the mailbox is associated with a corresponding extension number. See PROGRAMMING, page 6-376. If so, make sure the off-node device exists on the node attached to the Voice Processing Unit using the Export/Import feature. See PROGRAMMING, page 6-83. Then delete the off-node device for the station that was associated with the mailbox and create a new off-node device with a temporary extension. Change the extension number of the temporary off-node device to the extension number associated with the mailbox.
When using a PRI span on another node, calls do not go out properly (e.g., callers hear ringing but there is no answer)	The PRI span does not have the “CO Provides Tones” flag enabled	Enable the CO Provides Tones flag for the span. See PROGRAMMING, page 6-126.

O. 99 NODES SUPPORT ISSUES

Table 7-20. *99 Nodes Support Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
Increased Node Capacity: A system is continuously experiencing software exceptions and resetting. The system resides in a network of 63+ nodes. The node is running older software (pre-v10.0).	The node that is resetting is running an invalid software version. After the network is operating at a 63+ node network, nodes cannot downgrade below v10.0.	Upgrade the node to v10.0 or later.
Attendant Console Causes Slow-Downs: The system slows down when an Attendant Console application connects to a system.	This is a “brute force” OAI command that requests all the off-node device information from the system. The slow down is directly related to the number of off-node devices in the system.	Leave the Attendant Console running continuously rather than shutting off the application during non-work hours. This will reduce the number of refreshes the console performs to get status of the devices in the network.
“Brute Force” Network Broadcasts: The system uses the last IP resource. Per design in Axxess 7.0 IP networking, the system sends a broadcast message to all nodes in the network to inform them that it cannot handle IP networking calls. As an IP resource frees up, the system sends a broadcast message to all nodes in the network again to inform them that it can handle IP networking calls. If the system oscillates between this last IP resource, it can cause a tremendous amount of IP traffic on the entire network.	There are not enough IP networking resources.	This condition of oscillating on the last IP resource is not good. If the system is in this state, install an additional IPRC to increase the number of IP resource.

P. IP NETWORKING ISSUES

CAUTION

Security is the responsibility of the customer's network administrator. Inter-Tel is not responsible for network problems due to security violations involving the IP address exposure of the IPRC. This includes, but is not limited to, toll fraud and interrupted network service. Place the IPRC in a private IP network such as point-to-point or frame relay; however, the IPRCs do not work behind firewalls and NATs. It is recommend that you place v1.x firmware device IPRCs in DMZ areas or behind firewalls with appropriate port access. Firewalls are still recommended where technically feasible to prevent unauthorized access from areas such as the public Internet, wireless IP, or other media.

Table 7-21. *IP Networking Troubleshooting Chart*

Symptom	Possible Cause	Corrective Action
Cannot establish an IP connection	The Ethernet cable is not plugged into the IPRC and/or data network (i.e., hub).	Plug the Ethernet cable into the IPRC and/or data network.
	The IPRC has the incorrect boot code.	Upload the correct version of the IPRC code (iprc_ipnc.hex). See page 3-178 for details.
	The IPRC is not seated properly in the chassis.	Make sure the IPRC is properly seated.
	NATs/firewalls are present.	Remove the NATs/firewalls. The IPRC cannot work behind firewalls or use NATs. See page 2-94 for details.
	The IP connection extension for the node is not unique.	Make sure the IP connection extensions (P6XXY) are unique. See page A-13 for a networking example.
	There is no Node IP Connection Group programmed for the node.	Program a "P8XXX" Node IP Connection Group for the networked nodes. See page 6-191 for details.
	The IPRC does not have a valid networking PAL.	Make sure the IPRC has the correct networking PAL. See page 8-7 for a list of the available PALs.
	The IP address and subnet mask information for each IPRC is not valid.	Make sure each IPRC has the correct IP address and subnet mask programmed (see page 6-174).
Cannot make calls across IPRCs	The IPRC does not have a valid PAL.	Make sure the IPRC has the correct networking PAL. See page 8-7 for a list of the available PALs.
	No VoIP channels are available.	Make sure the IPRC has the correct networking PAL. See page 8-7 for a list of the available PALs.
	The remote IP address is programmed incorrectly.	Make sure the off-node IP connections have the correct Remote IP Address programmed (see page 6-177). This address must match the IP address programmed locally on that node.
	The node cannot find the IP connection.	Make sure the correct Node IP Connection Group is added to the appropriate node (see page 6-191).

Table 7-21. *IP Networking Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
No audio on network calls	Two NATs or firewalls are present in the path.	Remove the NATs/firewalls. The IPRC cannot work behind firewalls or use NATs. See page 2-94 for details.
	The audio stream receive ports are not programmed correctly.	Make sure the audio stream ports fields match for the affected nodes. See page 6-177 for details.
	The IPRC is faulty.	Replace the faulty IPRC.
Phone display shows INSUFF BANDWIDTH FOR VOICE	The Ethernet cable came unplugged from the IPRC and/or data network (i.e., hub) during a call.	Make sure the Ethernet cable is plugged into the IPRC and data network.
	The data network does not meet the minimum specifications required for IP networking.	Run the Network Qualifier to assess your data network. Then verify that your data network meets the minimum specifications (see page 2-94 for more information).
	The data network is experiencing problems (e.g., excessive packet loss and traffic).	Contact your network administrator.
One-way audio on network calls	A NAT or firewall is present in the path.	Remove the NAT/firewall. The IPRC cannot work behind firewalls or use NATs. See page 2-94 for details.
	The audio stream receive ports are not programmed correctly.	Make sure the audio stream ports fields match for the affected nodes. See page 6-177 for details.
Poor call quality on network calls	The data network is experiencing problems (e.g., excessive packet loss and traffic).	Contact your network administrator.
	The data network does not meet the minimum specifications required for IP networking.	Run the Network Qualifier to assess your data network. Then verify that your data network meets the minimum specifications (see page 2-94 for more information).
	The G.711 vocoder is using too much bandwidth.	Switch to a G.729 vocoder, which uses less bandwidth.
	The G.729 vocoder is compressing audio data, resulting in poorer quality.	Switch to a G.711 vocoder, which does not compress audio data, resulting in higher call quality.
DTMF tones are not detected	DTMF tone length is too short.	Set the DTMF Digit Duration/Pause field to 100ms or higher (see page 6-331).
	The wrong DTMF vocoder is programmed.	Make sure the DTMF Encoding field is set to either G.711 or RFC2833 (see page 6-186).
The network is experiencing audio and/or connection problems	There is a port conflict or the firewall, NAT, or router is blocking the port.	Try changing the ports associated with IP call control and/or audio. Make sure none of them conflict with ports that other protocols use (e.g., SIP uses 5060).
Cannot fax across the network	Currently the IPRC does not support faxing over the IP network.	

Table 7-21. *IP Networking Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
Echo is heard (half-duplex)	Noise levels are too high.	Talk at a lower volume while on a call.
	Callers are talking at the same time (doubletalk).	When simultaneous and similar audio on both sides of the call is detected, the card may try to cancel this perceived “echo,” causing audio and echo problems. Try to avoid doubletalk.
	The data network does not meet the minimum specifications required for IP networking.	Run the Network Qualifier to assess your data network. Then verify that your data network meets the minimum specifications (see page 2-94 for more information).
	The backplane configuration settings (echo settings in particular) are not optimized.	Change the backplane configuration settings. See page 6-173 for more information.
Cannot call devices on other nodes after using the IPRC Networking Wizard.	Wrong Off-Node IP Connection IP Address.	Check IP Connection IP Addresses on the local and the remote nodes.
	Missing IP Connection Group or Off-Node IP Connection.	Verify that there's a Node IP Connection Group in the target node's Node Trunk/IP Connection Groups folder. Make sure there's an Off-Node IP Connection in that NIPCG, and verify the IP Address.
	Missing IP Networking Port License(s) or IP Networking software License.	Check the License and premium features for Networking.
	Wrong version of IPRC Firmware installed.	Check your internet connection, and Upgrade IPRC Firmware to version 8.1.2 or higher.
	Internet connection is down.	Check your Internet connection.
Cannot call devices on other nodes after using the T1/E1 Networking Wizard.	No B-Channels.	Make sure the PRI Card is online, and enough B-Channels are equipped.
	No Reference Clock.	Check reference clock list on this and other nodes.
	Bad License, no premium features.	Make sure license and premium features are valid for networking.
	Old firmware.	Make sure firmware is up-to-date.
Cannot import/export to/from new target node after using the Networking Wizard.	Networking is not programmed correctly.	Verify that there's a Node Trunk/IP Connection Group to the remote node.
		Verify that the Node Trunk/IP Connection Group includes the Node Trunk Group or Off-Node IP Connection to the remote node Depending on the type of programming.
		Refer to the Help file for each programming area.

Table 7-21. *IP Networking Troubleshooting Chart (Continued)*

Symptom	Possible Cause	Corrective Action
No audio on a call between two IP devices (in the same Network Group) configured to use peer-to-peer audio.	The Network Group is set for peer-to-peer audio, but there is a firewall or NAT in between endpoints.	Verify that there is no firewall or NAT between the two endpoints in the Network Group.
	A Network Group is programmed to use peer-to-peer audio on one node, but not on another node.	Verify that the Use Peer-to-Peer Audio flag is set to **Yes** on all nodes.
	A Network Group is not configured to use peer-to-peer audio.	Verify that the Use Peer-to-Peer Audio flag is set to **Yes** on all nodes.
A node cannot export or import a Network Group.	One of the nodes involved is not running v9.0 or later software.	Upgrade the node to v9.0 or later software.

6. CONSTANT BIT RATE (CBR) FOR THE ATM SWITCH

6.1 If using an ASX-200BX ATM switch and the Slave LED on a slave PCMA card is blinking irregularly, you may have to set the constant bit rate (CBR) to *roundrobin* instead of *guaranteed*. This ensures the bit rate does not change and that the master/slave clocks can remain in sync.

6.2 To configure the CBR:

1. Establish a terminal session on the serial port and access the root menu (see steps 1-5 on page 3-168).
2. At the root menu, type `interfaces atmif schedule` and press **ENTER**.
3. For each interface (module) installed, type the following command:

   `modify -atmif 1XY -servcat CBR -schedmode roundrobin`

   Where *1XY* indicates the interface number (1a1, 1a2, 1a3, etc.).
4. A message appears that informs you that the change requires a reset. Type `Y` and press **ENTER** to acknowledge the message. Then type `Y` and press **ENTER** to reset the module.
5. Repeat steps 3 and 4 for each module.

6.3 The following is an example of what the output should resemble.

```
<switch name>:interfaces atmif schedule-> modify -atmif 1a1 -servcat CBR -
schedmode roundrobin

WARNING: The new connection scheduling mode will be effective after a network
 module reset
Do you wish to continue? (y or n): y

WARNING: Changes have been made that require the following netmod(s) to be
 reset: 1A. Would you like to reset them now? (y or n): y
Waiting for netmod(s) to come back up...
The netmod(s) have been reset. All connections associated with them will take
some time to come up.

<switch name>:interfaces atmif schedule-> modify -atmif 1a2 -servcat CBR -
schedmode roundrobin

WARNING: The new connection scheduling mode will be effective after a network
 module reset
Do you wish to continue? (y or n): y

WARNING: Changes have been made that require the following netmod(s) to be
 reset: 1A. Would you like to reset them now? (y or n): y
Waiting for netmod(s) to come back up...
The netmod(s) have been reset. All connections associated with them will take
some time to come up.

<switch name>:interfaces atmif schedule->
```

6.4 After you have programmed the CBR for all of the modules, type `show` and press **ENTER** at the `interfaces atmif schedule->` prompt. The following is an example of what should be displayed.

```
<switch name>:interfaces atmif schedule-> show
                                S c h e d u l i n g
AtmIf        Service Category  Mode          RR Priority   Override
1A1             CBR           roundrobin    prio1         off
1A1            rtVBR          guaranteed    prio2         off
1A1           nrtVBR          guaranteed    prio2         off
1A1             ABR           guaranteed    prio3         off
1A1             UBR           roundrobin    prio3         on
1A2             CBR           roundrobin    prio1         off
1A2            rtVBR          guaranteed    prio2         off
1A2           nrtVBR          guaranteed    prio2         off
1A2             ABR           guaranteed    prio3         off
1A2             UBR           roundrobin    prio3         on
1A3             CBR           roundrobin    prio1         off
1A3            rtVBR          guaranteed    prio2         off
1A3           nrtVBR          guaranteed    prio2         off
1A3             ABR           guaranteed    prio3         off
1A3             UBR           roundrobin    prio3         on
1A4             CBR           roundrobin    prio1         off
1A4            rtVBR          guaranteed    prio2         off
1A4           nrtVBR          guaranteed    prio2         off
1A4             ABR           guaranteed    prio3         off
1A4             UBR           roundrobin    prio3         on
1C1             CBR           roundrobin    prio1         off
1C1            rtVBR          guaranteed    prio2         off
1C1           nrtVBR          guaranteed    prio2         off
1C1             ABR           guaranteed    prio3         off
1C1             UBR           roundrobin    prio3         on
1C2             CBR           roundrobin    prio1         off
1C2            rtVBR          guaranteed    prio2         off
1C2           nrtVBR          guaranteed    prio2         off
1C2             ABR           guaranteed    prio3         off
1C2             UBR           roundrobin    prio3         on
1C3             CBR           roundrobin    prio1         off
1C3            rtVBR          guaranteed    prio2         off
1C3           nrtVBR          guaranteed    prio2         off
1C3             ABR           guaranteed    prio3         off
1C3             UBR           roundrobin    prio3         on
1C4             CBR           roundrobin    prio1         off
1C4            rtVBR          guaranteed    prio2         off
1C4           nrtVBR          guaranteed    prio2         off
1C4             ABR           guaranteed    prio3         off
1C4             UBR           roundrobin    prio3         on
```

7. IP RESOURCE CARD DIAGNOSTICS

7.1 The IP Resource Card (IPRC) has many diagnostic tools used for troubleshooting purposes. These tools are separate from Call Processing and display only information local to the IPRC. There are also limited programming displays to control the card.

7.2 To access diagnostic information, you can use a serial (RS232 or Telnet) or Web/HTML interface. The serial interface contains text displays that are common to all IP cards as well as information that is specific to IPRCs. The IPRC-specific displays show the current status of a call, how Call Processing has mapped the crosspoint connections, and how Call Processing has mapped the sessions and channels. The Web/HTML interface uses a graphical user interface (GUI) to display information that is common to all IP cards.

NOTE: You can also access Message Print and Exception Log information for the IPRC when using a serial interface.

7.3 To access the serial displays, use an RS232 interface or the Telnet protocol (if the option is enabled in DB Programming; see page 6-154). Using an RS232 interface for an extended period of time, however, may interfere with Message Print, which also uses RS232. While in the diagnostic screens, Message Print will send messages to a queue until the RS232 connection becomes available. If the queue limit is exceeded, the IPRC may discard Message Print messages.

7.4 To access the Web/HTML displays, the Web Server option must be enabled in DB Programming (see page 6-154). You can then use a Web browser to display the desired information. Although the Web interface is easier to use, the serial interface offers access to a larger variety of diagnostic screens.

7.5 The serial and Web diagnostic tools are detailed throughout this section.

NOTICE
Although the Web and serial interfaces provide access to several programming options for the IPRC, use only DB Programming to program the IPRC. Any settings you define in the Web or serial interface are automatically overridden by DB Programming.

A. SECURITY

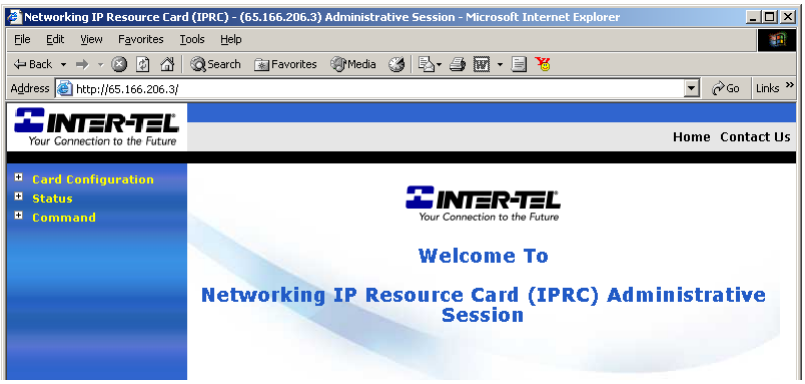
7.6 The IPRC requires a password when you log on to a Telnet session or enter a programming display on an RS232 session. For the Web/HTML interface, the IPRC requires a username and password. By default, “IPRC” is the username and “itpassw” is the password. Because the password allows access to the IPRC, Inter-Tel recommends changing the defaults. To change the password, see page 6-154 in PROGRAMMING. You cannot change the password using the serial or Web/HTML interface.

7.7 In addition to requiring a password, you can disable the Telnet and/or Web/HTML interfaces. If disabled, the only way to access the IPRC is through an RS232 interface. Although the Telnet and Web/HTML options can be enabled/disabled through any interface, the changes do not take effect until Call Processing sends the settings to the IPRC, which is usually after a reset or a database change.

B. WEB/HTML INTERFACE

7.8 To use the Web/HTML Interface:

1. Make sure the IPRC is connected and online.
2. Point a Web browser to the IP address of the IPRC. This IP address is set in DB Programming (see page 6-172).



3. When prompted, enter your username and password. See page 6-154 for information about the username and password.

NOTE: The username and password are case-sensitive.

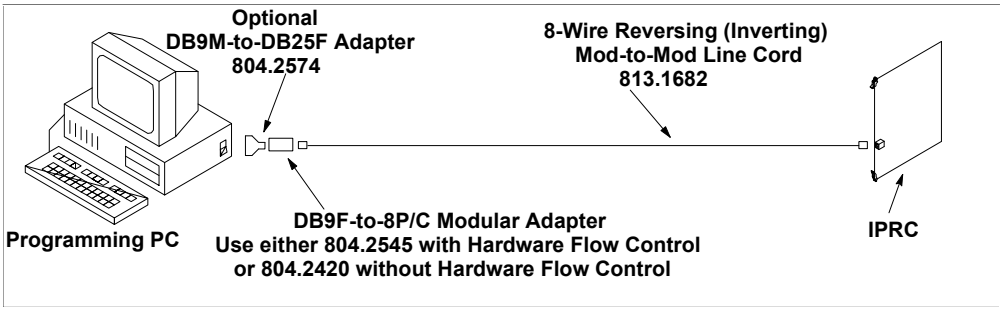
4. Expand the Status menu by clicking the plus sign (+) next to **Status**. A list of options appears.
5. Select the desired display (see page 7-75).

C. SERIAL INTERFACE

7.9 To use a serial interface via an RS232 connection:

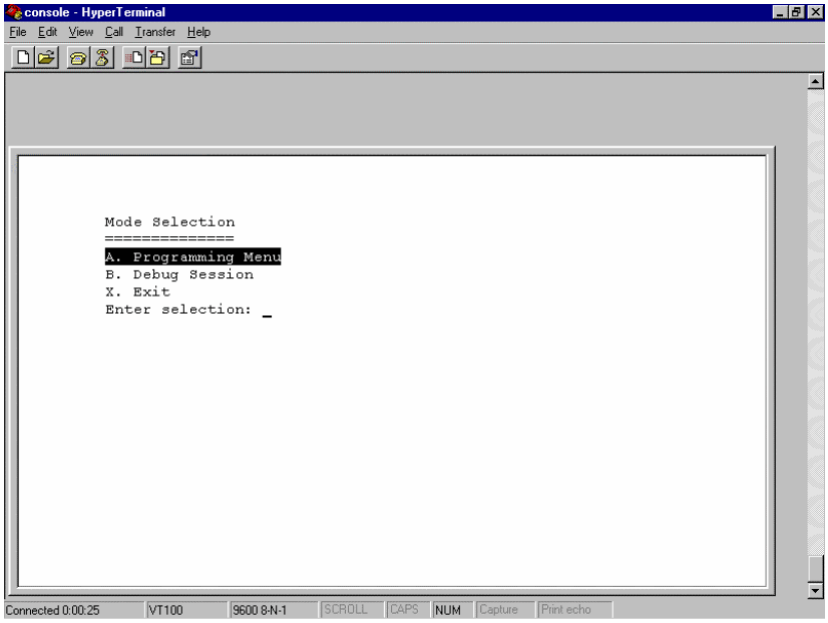
1. Make sure the IPRC is connected and online.
2. Connect a programming PC to the IPRC serial port jack as follows (see the diagram on the following page). The necessary connectors can be purchased separately or as part of the Universal RS232 Kit (part no. 828.1282). This RS232 connection is the same as the RS232 connection to the CPC or CPU cards used for DB Programming.
   a. Attach one end of an 8-wire reversing mod-to-mod line cord to the serial port jack on the IPRC.
   b. Attach the other end of the line cord to a DB9F-to-8P/8C modular adapter with the proper flow control. (This application does not require flow control unless the bps rate exceeds 9600.)
   c. If the PC does not have an available DB9M COM port but does have an available DB25M COM port, attach the DB9M-to-DB25F converter to the DB9F end of the modular adapter.

d. Connect the cable to the appropriate COM port on the programming PC.



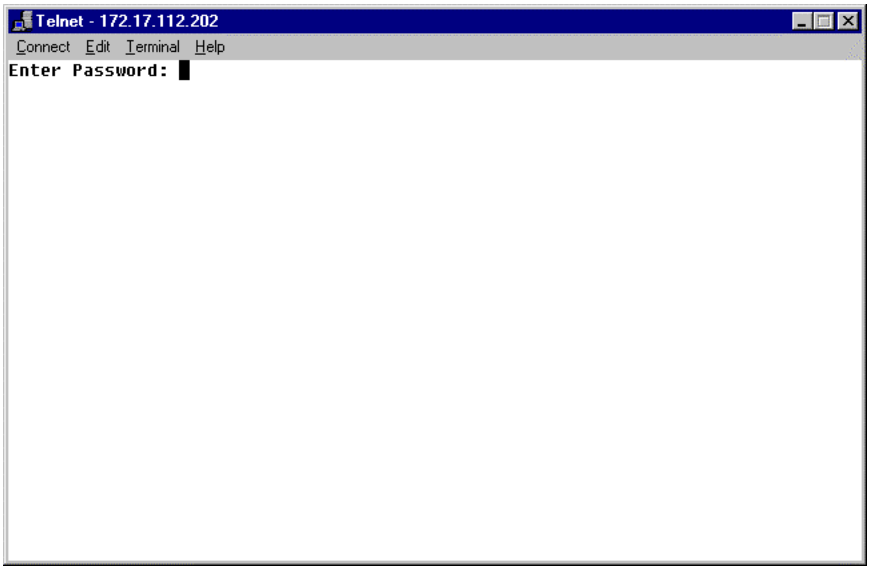
3. Using a terminal program (such as HyperTerminal) on the PC, press **ENTER** several times to establish communication. The same screen used for a Telnet session (described on the next page) should be displayed.

4. Follow the instructions for the Telnet session (beginning with step 4) to view the diagnostic displays.



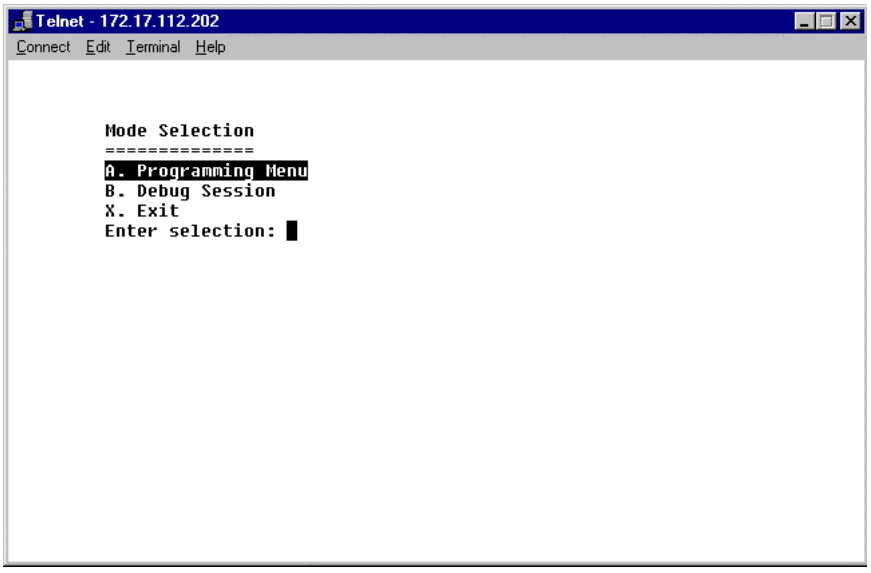
7.10 To use a serial interface via a Telnet connection:

1. Make sure the IPRC is connected and online.
2. Establish a Telnet connection to the IPRC's IP address using a Telnet program (such as the Microsoft Windows Telnet program) on a PC configured to run the IPRC's subnet. A Telnet page similar to the one shown below appears.

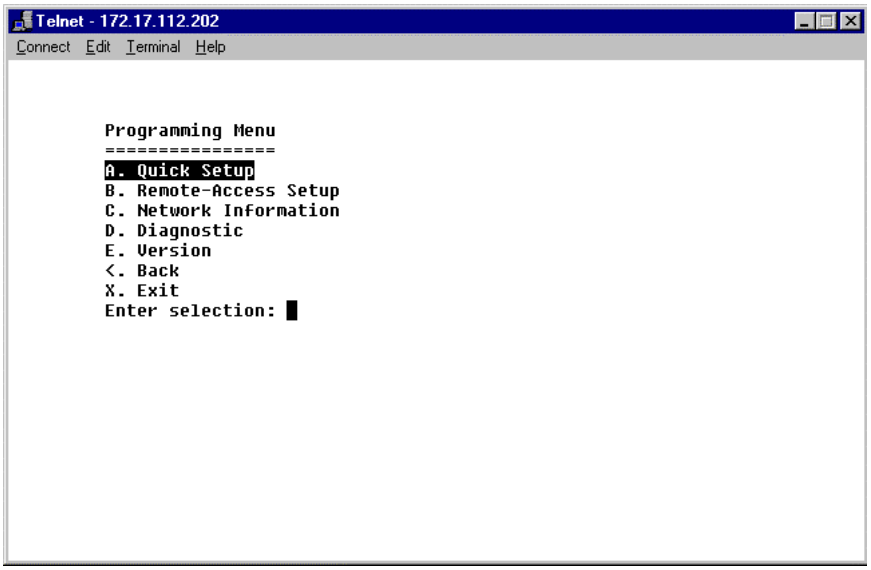


3. At the prompt, enter the IPRC's password and press **ENTER**. The following page appears.

NOTE: The password is case-sensitive.



4. Enter A (not case-sensitive) at the Enter selection prompt. The Programming Menu appears, as shown below.



5. Enter D (not case-sensitive) at the Enter selection prompt to access the diagnostic information.

6. Enter the letter associated with the desired display. See the following section for a list of available screens.

D. IPRC DIAGNOSTIC DISPLAYS

7.11 The following table describes the available displays and the information associated with each screen. The display name for both the serial and Web interface is also indicated where applicable.

NOTE: "N/A" indicates that the display is not available when using a Web interface.

Table 7-22. *IPRC Diagnostics Displays*

Serial Display Name	Web Display Name	Contains
Network Information	Network	Information about the status of the network, the card's Ethernet address, subnet mask, gateway IP address, and WINS information.
Version Information	Firmware Version	Information about the version of the boot code and the application code.
NICS Statistics	N/A	Information about the Ethernet driver. This display may point out problems with the Ethernet driver or assist in solving bandwidth- and Ethernet-related problems.
Thread Registry	N/A	Information about the threads in the system, showing the name, priorities, and state of each thread.
Timer Alarm Statistics	N/A	Information about the timer alarms that may be running on the IPRC. Currently, the IPRC uses timer alarms for LED flashing and setting the audio-receive timers.

Table 7-22. *IPRC Diagnostics Displays (Continued)*

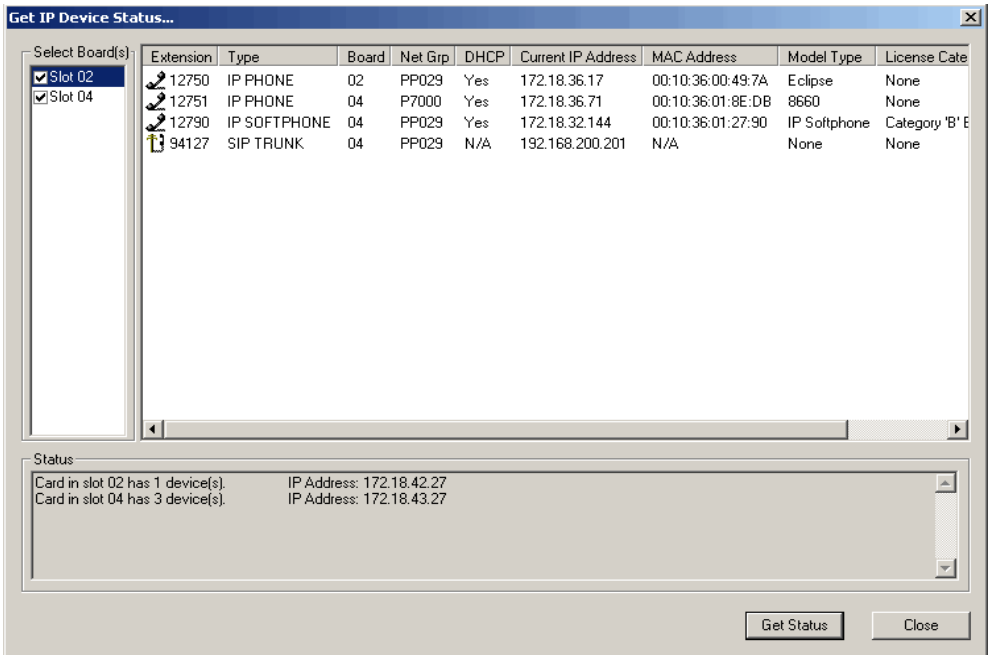
Serial Display Name	Web Display Name	Contains
On-Line Monitor	N/A	Options for entering commands for a variety of functions. To see a list of the available commands, enter "?" at the prompt.
Connection Status	Session Information*	Information about all of the sessions and channels. While in this screen, you can switch between sessions and channels or scroll through the multi-page display. For each session or channel, the following appears: • mapped channel or session information • any crosspoint connections, including Call Processing's highway/timeslot pairs and the card's local highway/timeslot pairs • description sent down by Call Processing for that particular session • destination IP address • statuses of the RTP streams for that channel
In-Use Audio Resources	N/A	Information about which sessions and channels are in use. Although this display contains the same information as Connections Status, the In-Use Audio Resources display offers a quick way to examine the use of these resources without scrolling. Also, a tester or developer can use this display while attempting to place all calls through a specific range of DSPs (e.g., for testing the daughterboard).
Receive-Audio Statistics	Receive Session Information*	Information about the receive-audio statistics for any session or channel. While in this screen, you can switch between session and channel modes and scroll through each mode. For any session/channel, information about the following appears: • percentage of in-time audio frames • resynchronization count • playback adjustment control • jitter statistics • early and late frame statistics • buffer lead-time statistics • Call Processing thresholds **NOTE:** While in the Receive Session Information Web display, you can click on a circuit to access an in-time packet graph of the last thirty seconds of a session's call. This allows you to determine the quality of the call based on the percentage of in-time frames.

Table 7-22. *IPRC Diagnostics Displays (Continued)*

Serial Display Name	Web Display Name	Contains
Send-Audio Statistics	N/A	Information about the send-audio statistics. Unlike the Receive-Audio Statistics display, Send-Audio Statistics contains global information about the send buffers (i.e., the information is not on a per-channel/session basis).
DSP Events	N/A	Information about DSP events for any session or channel. While in this screen, you can switch between session and channel modes and scroll through each mode.
Echo Canceller	N/A	Information about the echo canceller for any session or channel. While in this screen, you can switch between session and channel modes and scroll through each mode.
Crosspoint Connections	N/A	Information about the crosspoint connections. This display constantly queries the crosspoint device and displays any connections. Because this display offers information only about the crosspoint device, the highways and timeslots may differ from the numbers that Call Processing sends to the IPRC. This is because the IPRC translates Call Processing's numbers before accessing the crosspoint device. For untranslated numbers, see the Connection Status display.
Audio Session Send Configuration	N/A	Information related to the send-RTP side for any session or channel. While in this screen, you can switch between session and channel modes and scroll through each mode. For any session/channel, information about the following appears: • current activation state • destination address • SSRC number, which identifies the synchronization source • number of frames per packet • RTP profile in use • current buffered-frame count • current sequence number • current 10 ms timestamp • current 125 µs timestamp • current encoding type • frame length • payload types for all of the encoding types that the IPRC supports

* These Web interface displays offer only a subset of the information included in the serial display.

7.12 IP Device Status: You can view the connection status of Inter-Tel IP devices on the local node at any time using the **IP Device Status** option under the Operations menu. When selected, the Get IP Device Status screen similar to the one below appears.



NOTE: There must be at least one IPRC installed and equipped to use the IP Status feature. If you select this option on a system with no IPRCs, an error message appears.

7.13 To view the status of the IP devices that terminate at a particular IPRC, select the slot number of the IPRC. The right frame displays the following information for each IP device:

- **Extension:** Indicates the extension number assigned to the IP device.
- **Type:** Displays the device type associated with the extension. Possible options include IP PHONE and IP SLA.
- **Board:** Indicates the board number of the device.
- **Port:** Displays the port number of the device.
- **DHCP:** Indicates if the device is obtaining its IP address from a DHCP server.
- **Current IP Address:** Displays the IP address currently assigned to the device.
- **MAC Address:** Displays the MAC address assigned to the device.
- **Model Type:** Describes the model of endpoint being displayed. The following list shows the possible entries for the model type and license categories. Click **Get Status** to retrieve IP Device status.
  - — Offline (no device is installed)
  - — IP Phone +
  - — IP Single-Line Adapter
  - — SIP
  - — SpectraLink
  - — Models 8864, 8665, 8668

— 8600

— 8620

— 8622

— 8660

— 8690

— IP Softphone

— Model 8602

— Third Party

— Third Party phone

— Future Phone (reserved for future phone model)

- **License Category:** When an endpoint comes online, it searches for a valid license. The license category column displays which category of license the endpoint has consumed. The available categories are listed in the table below.

Table 7-23. *IP Device Status Dialog Box License Category Column*

License Category	Description
Unlicensed	The endpoint was unable to retrieve a valid license and is currently not licensed.
Category A	The endpoint is using a license from Category A.
Category B	The endpoint is using a license from Category B.

- **Status:** Indicates if the device is offline or online. If there is no license available for the device, this displays "No LICENSE."

7.14 This screen also includes a Status area. This area displays information about the devices and the boards, including how many devices terminate at a particular board. It also displays the IP address of the IP Connection used for the card.

NOTE: If this screen indicates that a device is offline, there may not be enough VoIP resources for that device. Make sure all of the available voice channels are programmed for Voice over IP. If there are no more available voice channels, consider adding a daughter card or another IPRC.

7.15 As of v9.1, the IP Device Status menu option has changed. The "Need License" column has been removed. The "Model Type" and "License Category" columns were added. Network Group Diagnostics

NOTE: The Network Group diagnostics feature works on Inter-Tel proprietary IP devices for the local node only. It does not work across nodes.

7.16 Versions 8.1 and later support a Network Group Diagnostics feature that allows an administrator to verify that the Network Groups on the local node are programmed properly. When the Network Group Diagnostics feature code is entered, the phone system initiates pings from each Inter-Tel IP device on the local node and determines if the other devices respond to the ping. If a device does not respond to the ping or if a firewall is detected, the system issues a Message Print message.

NOTE: Enable a port for Message Print before you run this diagnostics feature.

ALSO: Because this feature affects system performance, run the diagnostics program when the system is idle (i.e., after normal business hours).

7.17 To run the Network Group Diagnostics:

1. At an administrator station, enter the Diagnostics On/Off feature code, 9900 [9100 in Europe] by default, followed by the Network Group Diagnostics feature code (9963 by default). The display shows NET GRP CHECK (YES = 1 NO = 2).
2. When prompted, press 1 or **ACCEPT** to start the diagnostics feature. The display shows NET GROUP DIAG IN PROGRESS.

NOTE: In the U.K., the default Diagnostics On/Off feature code is 9100, and the Network Group Diagnostics feature code is 9163.

7.18 When the diagnostics is complete, the phone displays one of the following messages:

- NET GROUP CHECK COMPLETED: Indicates that all IP devices within the Network Groups are capable of communicating via P2P audio.
- NET GROUP CHECK ERRORS FOUND: Indicates that either some of the IP devices are offline or there are NATs/firewalls located between the devices. Check Message Print to determine which errors occurred.

8. CALL PROCESSING CARD (CPC)

8.1 This section provides additional troubleshooting information specific to the CPC. These instructions specifically address the following problems:

- RS-232 connections
- Local area network (LAN) connections
- DB Programming logons
- Default the CPC
- * button
- Web Interface for CPC Diagnostics

8.2 Instructions are also included to restore the flash database default settings.

A. TROUBLESHOOTING THE RS-232 PORTS

8.3 This section describes how to troubleshoot the CPC if your PC cannot communicate with the CPC using RS-232. Because each CPC has two RS-232 ports, perform these procedures on each RS-232 port.

8.4 To troubleshoot the RS-232 ports:

1. Make sure the CPC is powered OFF.
2. Launch a PC terminal emulation program, such as Hyperterm, Procomm, or Teraterm, and configure the COM port settings as follows:
   - **Bit Rate:** 9600
   - **Data Bits:** 8
   - **Stop Bit:** 1
   - **Parity:** None
   - **Flow Control:** None
3. Connect a serial cable from the COM port to the CPC RS-232 port that has Message Print enabled.
4. Power ON the CPC and watch the terminal emulation screen.
5. *If you do not see any boot messages,* connect the PC to a known good serial port on a different CPC, an Internet Protocol Resource Card (IPRC), or a T1 card and look for online responses. To verify transmit and receive messages, enter `/olm` when the card has completely booted. One of the following should happen:
   — *If the card responds with a command prompt or menu,* your PC serial port is working properly. Turn OFF the power to the CPC and reconnect the RS-232 cable to the CPC. Power ON the CPC and verify that no boot messages are present. If there are still no boot messages, there may be a problem with the CPC port. Return the card to Inter-Tel.
   — *If the card does not respond or if you do not see any boot messages,* check the settings for your PC serial port. If they are correct, your PC serial port is not functioning properly.

If you see the boot messages from the CPC, the serial port is working correctly. THE SERIAL PORT HARDWARE IS VERIFIED AS WORKING, AND THE CARD SHOULD NOT BE RETURNED. The CPC boot code controls this RS-232 port until the "Decompressing card-specific code to RAM..." message appears. The boot code uses the default RS-232 port settings listed in step 2. After the message appears, the CPC database takes control of the RS-232 port using the settings programmed in the Session Manager. Continue to step 6.

NOTE: After the CPC boot code has finished setting up the port, the database takes control of the RS-232 port. While the database is setting up the port, the port appears to stop functioning properly. This is normal behavior and does not indicate any hardware issues.

6. *If you see some non-readable symbols after the "Decompressing" message,* Call Processing has changed the bit rate on the CPC or the port may be set for OAI. (See below for an illustration of the output.) If this occurs, set the bit rate of the terminal connection to a higher value and reboot the CPC. Continue to try different values until you can log on to DB Programming using the CPC RS-232 port. In general, the greater the difference between the CPC and PC baud rates, the less likely it is that you will see any output in the terminal program. If the port is not set for OAI, you should still be able to log on to DB Programming. If you cannot log on, continue to step 7.

```
Decompressing card-specific code to RAM ... successful!
(decompressing 2789072 bytes into 8885956 bytes)
ž l<9Ù$ƒ!'"!ÜÃ'C8q;-ê!!Èò'8xq{-"!!Ñ³ÈRàÃ'88qL{-
êa!ÈPE<Q9õpj²!ÈR~$
@Âh"4úÖF^"-M J¸Ô-€šPŽ!!!!Ö!9@¼òÒ@ Ä°?Ä!!a!!ü÷?
```

If nothing appears after the "Decompressing" message, Message Print may be disabled or redirected to another port. If you have verified that Message Print is enabled for this port, continue to step 7.

If you see nonsensical Message Print information and the CPC resets several times, the flash database is corrupt. See page 7-86.

If you see Message Print information specific to the CPC, the port is functioning. THE SERIAL PORT HARDWARE IS VERIFIED AS WORKING, AND THE CARD SHOULD NOT BE RETURNED.

7. Log on to DB Programming using another connection method and reset the RS-232 port to its default values. These default values are as follows:

   - **Bit Rate:** 9600
   - **Hardware Handshaking:** No
   - **Software Handshaking:** Yes
   - **Auto Baud:** Yes

   NOTE: While in the DB Programming session, make sure to enable the **Message Print Output Active** flag for PP100 (System\Maintenance\Message Print).

8. Try to access the port again. *If this fails,* restore the CPC database default settings as described in page 7-85.

B. TROUBLESHOOTING THE IP PORT

8.5 This section describes how to troubleshoot the CPC if your PC cannot communicate with the CPC using an Internet Protocol (IP) connection.

8.6 To troubleshoot the IP port:

1. Make sure the CPC is powered OFF.
2. Connect the PC LAN port to the CPC IP port using a crossed cable or two straight cables through a hub. Remove all other IP devices from the hub to ensure there are no conflicts.
3. Power ON the CPC and verify that the LNK light-emitting diode (LED) on the CPC is lit. If the LNK LED is not lit, connect the cable to a known good CPC or IPRC and verify the LNK LED is lit.
4. *If the LNK LED is lit on the known good CPC or IPRC,* there may be a problem with the IP port on the CPC you are troubleshooting.
5. *If the LNK LED is not lit on the known good CPC or IPRC,* there is a cabling problem.
6. Connect the PC LAN port to a known good CPC or IPRC. If a connection is made, compare the IP settings between the PC and the CPC you are troubleshooting (see page 7-85 for more information). If the IP address for the CPC is unknown, verify that the PC is compatible with the default address.
7. Try to connect the PC to the default CPC address (192.168.200.201). If the CPC address has been changed, try the IP address that was assigned to the card. If you do not know the IP address, continue to step 6.
8. Connect a PC serial port to the CPC RS-232 port that has Message Print enabled. Boot the CPC and wait for the CPC IP address to be displayed in Message Print, as shown below.

```
-01:023- 04:04 01-08 M2002 INF HW CP [8.6.1]: On-Line From CPC IP Port
-01:024- 04:04 01-08 *** CPC HW Address: [8.6.1] IP Address:
192.168.200.201
+Hostname: 'TEST_CPC'
```

9. Check the Address Resolution Protocol (ARP) table listing and attempt to ping the CPC IP address. See page 7-85 for information on finding the CPC IP address.
10. Power OFF the CPC, press and hold the star (*) button, and power ON the CPC. After a few seconds release the * button. After the DHCP and BOOTP attempts pass, the CPC enters download mode, and the MAC and IP address are displayed. Verify that the PC is compatible with this address. After the CPC has reset, try to log on to the port again.
11. If you cannot connect to the CPC using the correct IP address:
    a. Use a terminal emulation program, such as Hyperterm, to access the Message Print-enabled serial port.
    b. Type `/olm` when the CPC has finished booting. A menu appears.

c. Select Programming Menu, then **Quick Setup**. Enter the values as shown in the illustration below. For the IP address and subnet mask, enter the values required for your network.

```
Quick Network Setup

Warning! A non-default Call Processing database will overwrite these
settings!

Address-assignment protocol selection:

Would you like to use BOOTP (Y/N)? n
Would you like to use DHCP (Y/N)? n

No address-assignment protocol selected; please enter static
information.

IP Address:  10.16.240.32 ? 192.168.200.201 (or your values)
Subnet Mask:  255.255.200.0 ? 255.255.255.0
Default IP Gateway Address:  192.168.20.21 ? (leave alone)
Hostname:  TEST_CPC - Change (Y/N) ? n (leave alone)
IP Address:  192.168.200.201
Subnet Mask:  255.255.255.0
Default IP Gateway Address:  192.168.20.21
Hostname:  TEST_CPC

Would you like to continue with these values (Y/N)? y
To activate all changes, a RESET must occur.
Would you like to reset the card (Y/N)? y

Resetting ...
```

NOTE: Some of the fields should be left alone, as indicated in the illustration. These settings are temporary and must be duplicated in DB Programming.

12. Attempt to log on again. If you cannot log in using these network settings, repeat step 9 using the following default settings:
    - **IP Address**: 192.168.200.201
    - **Subnet Mask**: 255.255.255.0
    - **Default IP Gateway Address**: 192.168.0.1
13. Attempt to log on again.

8.7 If you still cannot log on, restore the CPC to the default settings as described in page 7-86.

IP Addressing

8.8 Most small, local static networks are in the class C range. These IP addresses are in the range 192.0.1.0 to 223.255.254.0. If your network is in this range, the first three octets of each IP address must be the same for each device. For example, all members of a given static subnet must have an IP address of 192.168.200.x, where *x* is a number 1 to 254. The subnet mask must be 255.255.255.0 so that each device on the subnet can see all of the other devices. If you are skilled at segmenting a subnet, then the masking can be changed; otherwise, leave it at the default.

8.9 Every device on the subnet should have the same subnet mask. In other words, if the subnet mask is at default (i.e., 255.255.255.0), the IP address designated as *xxx.xxx.xxx.yyy* will see any other IP device with identical *x* values in the IP address. These devices, however, will not see any other devices that do not have matching *x* values. For example, if the subnet mask is default, a device with the IP address of 192.168.200.201 will not communicate with IP address 192.168.1.200, but it will communicate with 192.168.200.200. For this reason, Inter-Tel strongly recommends that you consult a networking professional prior to changing the IP address or subnet mask of any device.

Finding the CPC IP Address

8.10 A server dynamically assigns the IP address to the IP port on the CPC if DHCP or BOOTP is enabled in DB Programming. You can view Message Print at boot up to determine this IP address. If Message Print is not active, you can check the IP address in the ARP table.

8.11 To check the ARP table:

1. Open a command prompt console on your PC.
2. Type `arp -a` and press **ENTER**. The ARP table appears, listing the MAC addresses and their corresponding IP addresses.
3. *If the CPC has a known IP address,* it should be listed in the table.

   If the CPC is not in the table, try to "ping" the IP address that you believe is assigned to the CPC.

NOTE: For more information about CPCs, see page 2-32.

Defaulting the CPC IP Settings

8.12 The values in the System\Devices and Feature Codes\IP Ports\PPXXX folder are **NOT** changed when you use Session Manager to default the database. To default these settings manually, set the following values in DB Programming (System\Devices and Feature Codes\IP Ports\<IP Port>):

- **Current IP Address:** 192.168.200.201
- **Current Subnet Mask:** 255.255.255.255.0
- **Current Gateway:** 192.168.0.1
- **Current WINS Server:** 0.0.0.0
- **Static IP Address:** 192.168.200.201
- **Static Subnet Mask:** 255.255.255.0
- **Static Gateway:** 192.168.0.1
- **Static WINS Server:** 0.0.0.0

- **Hostname:** None
- **IP Address Assignment: BOOTPC:** Yes
- **IP Address Assignment: DHCP:** Yes
- **Telnet Server:** Yes
- **Web Server:** Yes
- **Listening Port:** 4000
- **Web Listening Port:** 80

C. LOG ON TO DB PROGRAMMING

8.13 With any CPC, there must be at least one reliable method to log on to the CPC. The available connection methods (from fastest to slowest) are as follows:

- IP port
- RS-232 port
- Modem

AXXESS Version 8.002 Known Issues

8.14 There is a known issue with loading DB Programming v8.002 on a PC that has not had 8.000 previously loaded. If you install v8.002 without having installed v8.000, an incorrect `.dll` is installed. This `.dll` prevents you from logging on to the database. If this occurs, uninstall v8.002, install v8.000, and then re-install v8.002.

Modem Connection

8.15 If connecting via the CPC modem, be sure to set the **Minimum Bit Rate** field (under System\Devices and Feature Codes\Modems\<Modem>) to a low value (even less than 9600 bps may be necessary). The modem can always negotiate faster rates, but the CPC cannot connect at a rate slower than this setting. Do **NOT** enable the **Force Minimum Bit Rate** field unless the connection is being made to a fixed-rate device.

8.16 By default, the modem port is set to the following values:

- **Modem Enabled:** Yes
- **Minimum Bit Rate:** 9600
- **Force Minimum Bit Rate:** No

D. DEFAULT THE CPC

NOTE: To default the CPC, the card must have boot code 2.1.0 or later.

8.17 This section details how to return the CPC to a "from factory" state. It is to be used as a last resort when you are unable to access the CPC using the procedures described in the previous sections. Following this procedure will erase or return default settings to the following areas:

- Back-up database
- Static database
- License

- IP port
- Exception log information
- History

8.18 Prior to performing this procedure, make sure you have:

- All necessary history and exception log information.
- A copy of the license to reload on the CPC.
- A backup copy of the CPC database.

8.19 **To default the CPC:**

1. Turn ON the power to the CPC and wait for the LEDs to begin cycling. This process takes 10-15 seconds.
2. Press the * button exactly four times. All LEDs will light when the button is pressed.
3. Wait for the LED cycle process to complete.
4. *If the button was pressed exactly four times*, all LEDs light for 1-2 seconds.
5. *If the button was not pressed four times*, the CPC boots as normal. Repeat steps 1-3.
6. Press the button exactly once more while the LEDs are all lit. After a few seconds, the CPC will boot as normal.

   If the button was pressed exactly once, the CPC is defaulted and restored to the "factory" settings. The card then resets.

   If the button was not pressed exactly once, the CPC boots as normal. Repeat steps 1-4.

CAUTION

It is extremely important that you wait until the CPC resets itself at the end of the process. Resetting the CPC while it is erasing/debugging can have catastrophic effects on the CPC. You can use a terminal emulation program to view this process. See page 7-81 for details.

8.20 If the process was successful, the LEDs begin rapidly flashing while the CPC erases or restores default settings. The CPC will remain in this state for approximately 60 seconds. After the procedure is successful, the CPU Active and DB Error LEDs should be solidly lit. Alarm 125 is then generated to indicate that the CPC does not have a valid license. When this occurs, upload the license.

E. * BUTTON

8.21 The * button allows you to:

- Place the CPC into the download state.
- Check the status of Call Processing and perform a minor reset.
- Bypass the backup database automatic load.
- Factory default the CPC.

8.22 Each of these is described in the following sections.

Download State

8.23 When the CPC is in the download state, you can use the Board Utility to download software. To place the card in the download state, press and hold the * button while powering on the system. The CPU Active and DB LEDs alternately light to indicate the card is in the download state.

8.24 The message print will show the following information during this time: "The CPC will time out after 66 seconds if no download is started." An example of the Download State is shown below.

8.25 The following example is from Message Print when the * button is held during power up and then released. In the example below no file was downloaded so the system timed out after 66 seconds where the system will reset.

```
--------------
ColdFire CPU Card (CPU-CF) Boot Code, Version 2.1.12 CPC
Build date: Thu Dec 22 2005 13:34:45

CS 0 flash 0 is type AM29LV320D.
CS 0 flash 1 is type AM29LV320D.
CS 1 flash 0 is type AM29LV320D.
CS 1 flash 1 is type AM29LV320D.
Total flash memory available is 16777216 bytes.

Parameter blocks OK!

Production CPC detected.

Checking the CPLD contents ...
CPLD is current (CPLD checksum: 0x0017BC54).

CPLD-generated TA (DTACK) not detected, using 7 wait states for SRAM.
SRAM detected: width 16 bits, size 4194304 bytes.

Waiting for FEC ... initialized!
Starting BOOTP attempts: 1 ... 2 ... 3 ... failed!
Starting DHCP attempts: 1 ... successful!
```

```
(received reply from 192.168.1.1)



Button pressed on power-up.
Begin download or cycle power to reset. Awaiting download.
Awaiting initiation of code download at 9600 baud.
EA = 00:10:36:00:A2:DB
IP = 192.168.1.119
Idle time (seconds): 66



Timed out waiting for download.


Resetting ...
--------------
```

Call Processing Status

8.26 With the * button, you can also send an “are you alive” message to Call Processing. If Call Processing does not acknowledge this message within 15 seconds, a minor reset is performed.

8.27 To check the Call Processing status, press the * button while the system is active. The CPC Active LED flashes, and a message is sent to Call Processing. If Call Processing receives and acknowledges the message, the CPC Active LED returns to its normal state (solidly lit). If, however, Call Processing does not receive or acknowledge the message, the system performs a minor reset.

8.28 In Message Print the following message will appear in the logs to confirm that the * button was pressed and acknowledged. If the Button Acknowledged message does not appear within 15 seconds then the system will reset.

Press and hold the * button until the Acknowledged message displays.

```
-01:551- 09:54 03-03 M2040 INF HW CP CPC '*' Button Pressed
-01:552- 09:54 03-03 M2041 INF HW CP CPC '*' Button Acknowledged
```

Backup Database Bypass

8.29 The * button on the CPC can be used to prevent the CPC from restoring the backup database once the system has been defaulted through DB Programming.

8.30 After the system database has been defaulted through DB Programming, the CPC will reset and will power up with a default database. At this point, any additional resets of the CPC will cause it to restore the database backup unless the * button is pressed.

8.31 This Backup Database Bypass is only useful after the system has been defaulted and for reasons when the CPC must be reset or powered cycled additional times after defaulting. Pressing this * button on power up without defaulting first will not cause the system to restore a default database. It only bypasses restoring the backup database if the system has been defaulted using DB Programming and the CPC has a backup database saved.

8.32 If the Backup Database is corrupted, use the following procedure to resolve the situation.

1. Save the current database if allowed. If not, locate the last known good database to restore.
2. Run DB test on the database and have it fix the errors.
3. Default the CPC through DB Programming.
4. After the CPC resets it will load a default database this time only. (If the CPC is reset or loses power, it will restore the backup database unless the * button is pressed)
5. Immediately connect to DB Programming and run the Backup Database Save thus saving the defaulted database to backup memory. (This will replace the backup database with a default database).

NOTE: If for some reason the system lost power or reset before implementing items 4 or 5, press the * button as indicated below to bypass the backup database restore.

8.33 **The * button procedure for the Backup Database Bypass:** To bypass the backup database restore feature, press the * button after power up while the top four LEDs are flashing the up/down sequence. This will prevent the backup database from being restored on additional resets after the system database has been defaulted.

NOTE: Bypassing the backup database does not erase the backup database from flash memory. If bypassing the backup database is necessary, use the procedure listed above for defaulting the backup database or use the factory default procedure. If necessary, contact Inter-Tel Technical Support.

8.34 See page 6-69 for more information about the flash database. Message Print does not provide any indications that the * button is pressed for this procedure.

Factory Default Procedure

8.35 The * button on the CPC can be used to factory default the CPC using the procedure below. The following procedure will remove all databases and license files from the CPC. Use caution when using this procedure because there you cannot recover any files on the CPC once this process has been implemented.

8.36 This procedure will return the CPC to a “factory default” state as if the CPC were new. The CPC must have boot code 2.0.4 or higher to complete this procedure. It is to be used when programming is not accessible or when it is necessary to restore the CPC to default state. Performing the following procedure will erase or default the following:

- Back-Up database
- Static Database
- License
- LAN Port Settings
- Exception-log information
- History

8.37 Prior to performing the procedure, review the following items:

- Retrieve the system logs if the past history is relevant to default the CPC.
- Have the necessary license files and software to reload on the CPC.
- A database save of a known good database.
- Confirm or convert the database if necessary (64, 128 or 256).

8.38 To factory default the CPC:

1. Turn on power to CPC and wait for LEDs to begin cycling (this process will last 10-15 seconds).
2. Press the * button exactly four times (all LEDs will light when the button is pressed).
3. Wait for the LED cycle process to complete.
4. If the button was pressed exactly four times all LEDs will light for one to two seconds. if the button was not pressed four times, CPC will boot as normal.
5. Quickly press the * button one more time while the LEDs are all lit (after two seconds, the CPC will boot as normal).
6. If the button press process was successful, the LEDs will begin flashing rapidly while the CPC erases or defaults itself.
7. The CPC will remain in this state for approximately 60 seconds. It is extremely important that you wait until the CPC resets itself at the end of the process. Resetting the CPC while it is erasing/debugging can have catastrophic effects on the CPC. This process is visible through a terminal program connected to RS-232-1.

8.39 Message print provides the following information when using this procedure.

- - - - - - - - - -

Power is cycled on the CPC and the * button has bee pressed 4 times.

```
ColdFire CPU Card (CPU-CF) Boot Code, Version 2.1.12 CPC
Build date: Thu Feb 02 2006 10:27:13



CS 0 flash 0 is type AM29LV320D.
CS 0 flash 1 is type AM29LV320D.
CS 1 flash 0 is type AM29LV320D.
CS 1 flash 1 is type AM29LV320D.
Total flash memory available is 16777216 bytes.

Parameter blocks OK!

Production CPC detected.

Checking the CPLD contents ...
CPLD is current (CPLD checksum: 0x0017BC54).
```

```
CPLD-generated TA (DTACK) not detected, using 7 wait states for SRAM.
SRAM detected: width 16 bits, size 4194304 bytes.

Testing RAM ... successful!

Decompressing card-specific code to RAM ... successful!
(decompressing 2800668 bytes into 8773820 bytes)

Default-CPC procedure requested, please press the * button again to
confirm …
------------
```

8.40 All the LEDs light solid for 3 seconds. If the * button is not pressed again at this time it will not default the board.

8.41 If the * button is not pressed the following message appears: "`No confirmation received, skipping default-CPC procedure ...`"

8.42 If the * button is pressed, the CPU will continue with the default procedure, and the following message appears: "`Initiating default-CPC procedure; do not power down the system!`"

```
----------
Resetting OS database ...

Programming the Ethernet address into the flash device ...
Erasing parameter block 2 ...
Programming parameter block 2 ...
Erasing parameter block 1 ...
Programming parameter block 1 ...
Ethernet address successfully programmed.
OS database reset.

Deleting SRAM (4194304 bytes) ... successful!
Erasing file-system sector ... successful!
Default-CPC procedure completed!
Resetting ...
n
```

F. WEB INTERFACE FOR CPC DIAGNOSTICS

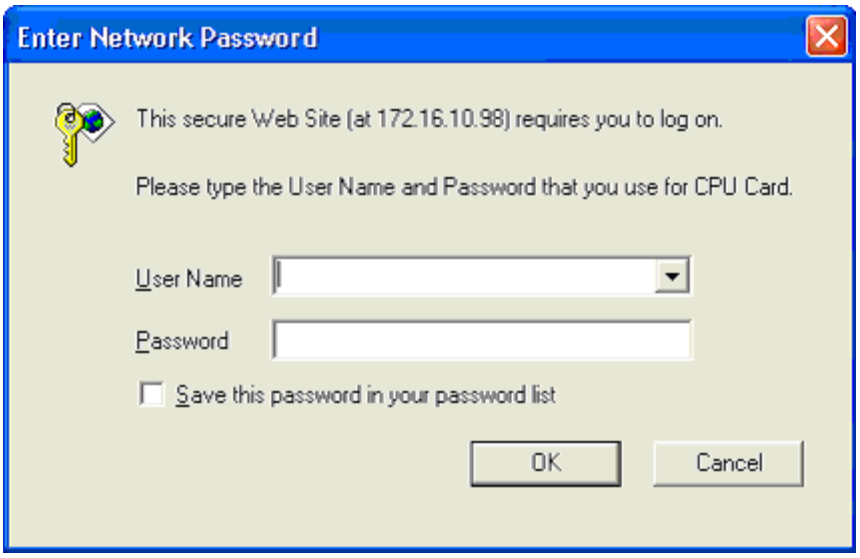
8.43 Because the CPC has an IP port, you can access the CPC using a Web browser. Although you can program the IP port settings from the Web, Inter-Tel recommends that you use the Web interface for diagnostic purposes only.

NOTE: Changes made in the Web interface do not take effect until the system is reset.

Accessing the CPC

8.44 To access the CPC via the Web:

1. Enter the IP address of the CPC in the Address line of your browser. You are prompted for a username and password.

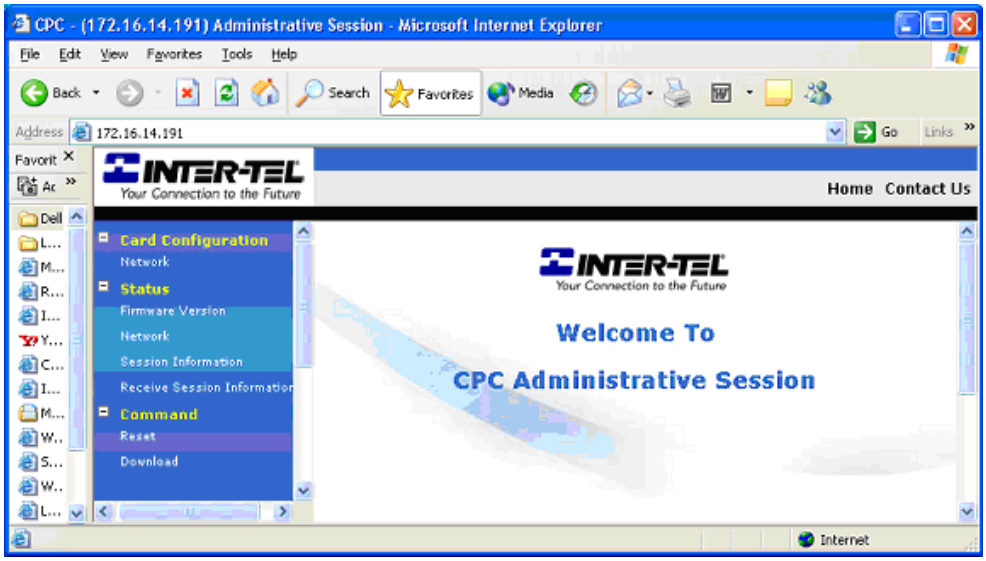


2. Enter `CPU` as the username and `itpassw` as the password and click **OK**.

NOTE: If desired, enable **Save this password in your password list** to automatically populate this screen the next time you access the CPC.

Navigating the Web Site

8.45 After you have successfully accessed the CPC, the Welcome page appears, as shown below.



8.46 The left side of the Web page lists the main menu items, each of which expands to display different options. The available main menu items are:

- **Card Configuration:** Includes the Network page, which allows you to program the following IP port parameters:
  - — Static IP Address
  - — Static Subnet Mask
  - — Static Gateway Address
  - — WINS Server
  - — Hostname
  - — IP Address Assignment (Enable or Disable BOOTP or DHCP)
  - — Web and Telnet Server (Enable or Disable)

  NOTE: For details about these fields, see page 6-178.

- **Status:** Includes the following options:
  - — *Firmware Version:* Displays the boot code and application code version uploaded onto the CPC.
  - — *Network:* Displays the current network settings for the CPC.
  - — *Session Information and Receive Session Information:* Not available.
- **Command:** Allows you to reset the card or place it into the download state. If you reset the card from the Web site, your connection is eventually terminated.

9. SOFTWARE LICENSE MANAGEMENT

9.1 This section provides information about obtaining a software license. It also includes additional software license information such as how to transfer a license and how to troubleshoot license errors.

A. OBTAINING A SOFTWARE LICENSE

9.2 In general, the site should already have a software license available. If necessary, however, you can "build" a license to meet the customer's needs, as described in this section.

NOTE: The software license is tracked by the serial number of the CPC or CP Server software security key and by the software site license (SSL). The serial number appears on the outside of the card or security key while the site license is shipped with the CPC or CP Server site license kit. The serial number is required to build a license, and the SSL number is required to download a license.

9.3 To order a license:

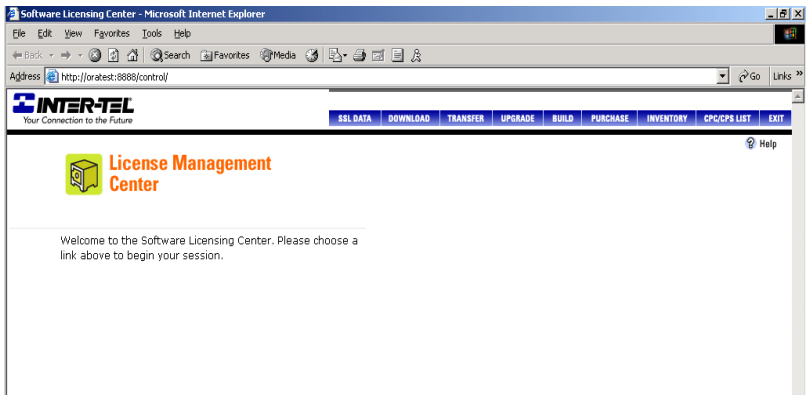
1. Determine the CPU type (e.g., CPC64, CPC128, etc.).
2. Select the desired features. See page 8-12 for a list of part numbers.

   NOTE: If upgrading from a CPC - Single (64) to a 128 (or higher) system, you must purchase the premium feature upgrades for the license. For example, if the CPC64 is currently licensed for Advanced CO Interfaces, you must purchase part no. 840.0062 to continue using this premium feature on the 128 (or higher) system. This does not apply if upgrading from a 128 system.

3. Purchase the features through Inter-Tel (via the *edGe*) or your Inter-Tel sales representative. For readers located in Europe, contact the Inter-Tel Europe office for licensing-related issues.

   NOTE: If you do not have Web access or if you need assistance, contact the Inter-Tel Software Maintenance Center.

4. Access the License Management Center through the Inter-Tel Web site (the *edGe*).

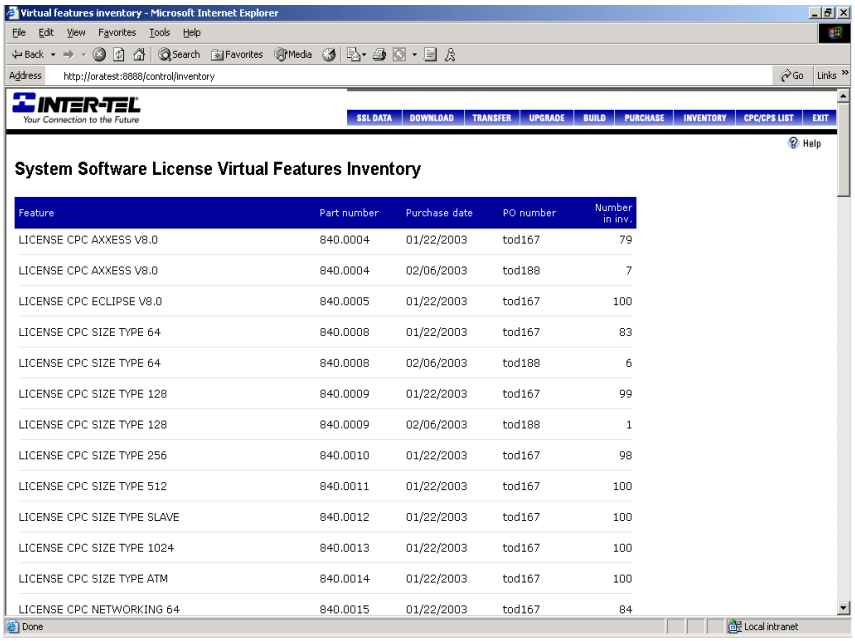


While in the License Management Center, you can:

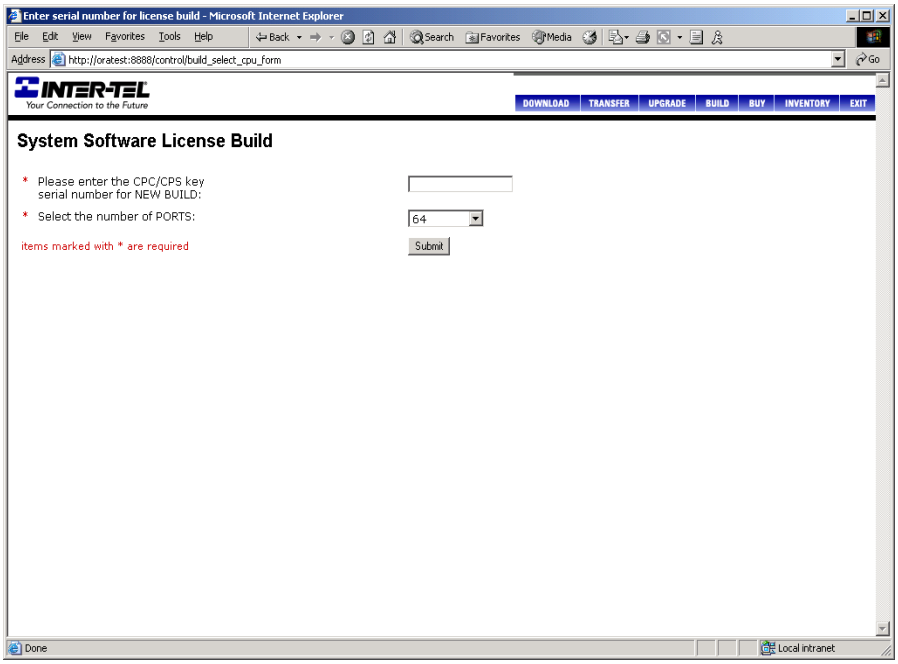
- Enter SSL data.
- Check your virtual warehouse inventory.
- Purchase more features.

- Build licenses.
- Upgrade licenses.
- Download licenses.
- Transfer licenses (see page 7-103).

5. Click **Inventory** to view the features you have purchased. Your "Virtual Warehouse" inventory appears.

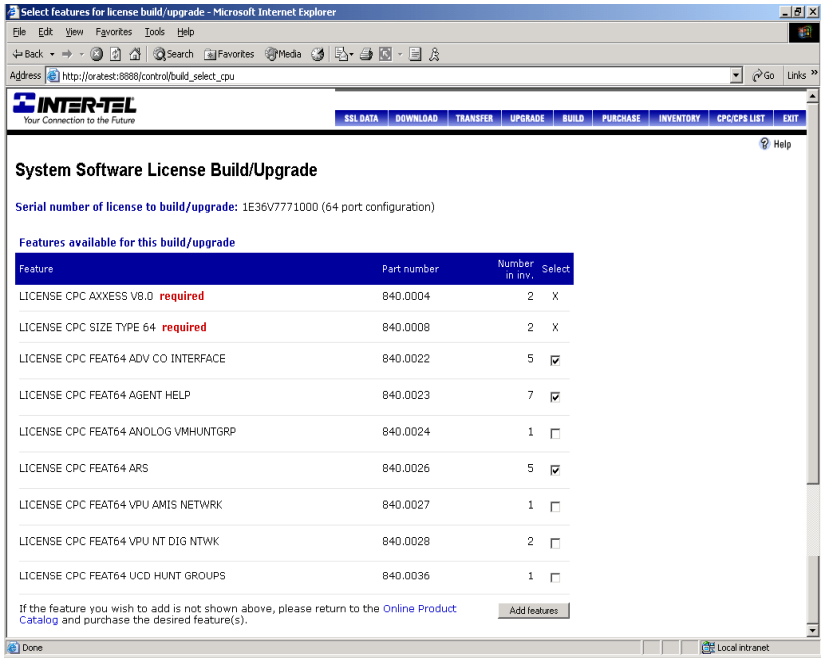


6. Click **Build** to build your license. The following screen appears.

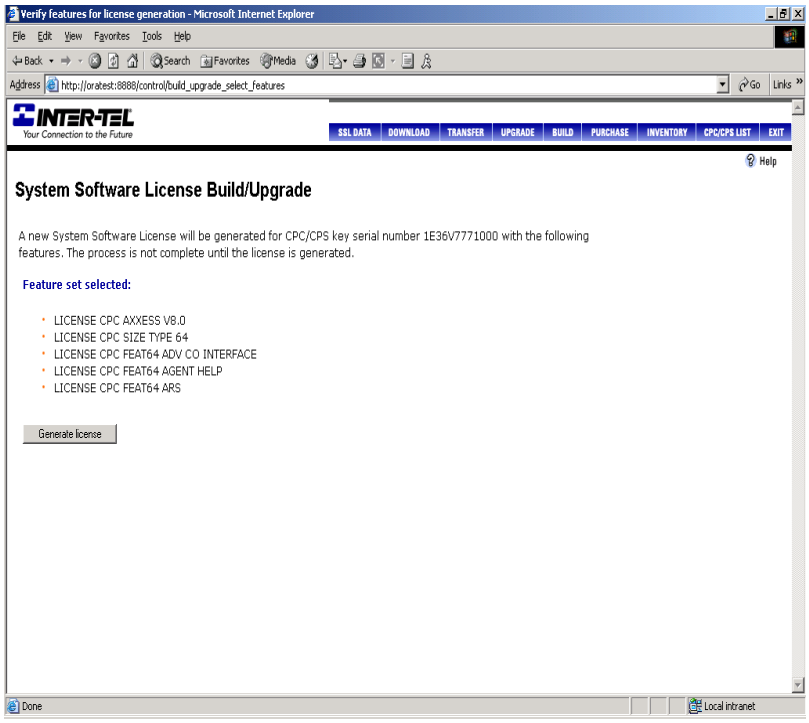


7. Enter the serial number of the CPC or CP Server software security key and select the number of ports (devices) you want the license to support.

8. Click **Submit**. If the inventory contains the license features required for the CPU type selected, the screen below appears. If, however, the inventory does not contain the required features, an error page appears. This page identifies the missing feature.

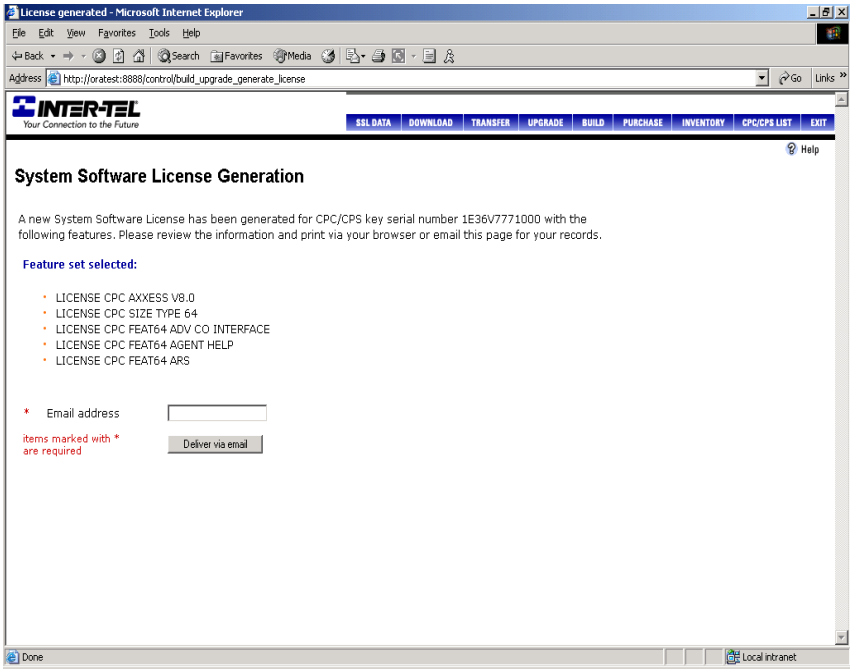


9. Select the features you want supported on this license, then click **Add features**. A screen summarizing the license features appears, and the number of features available in inventory is adjusted accordingly.

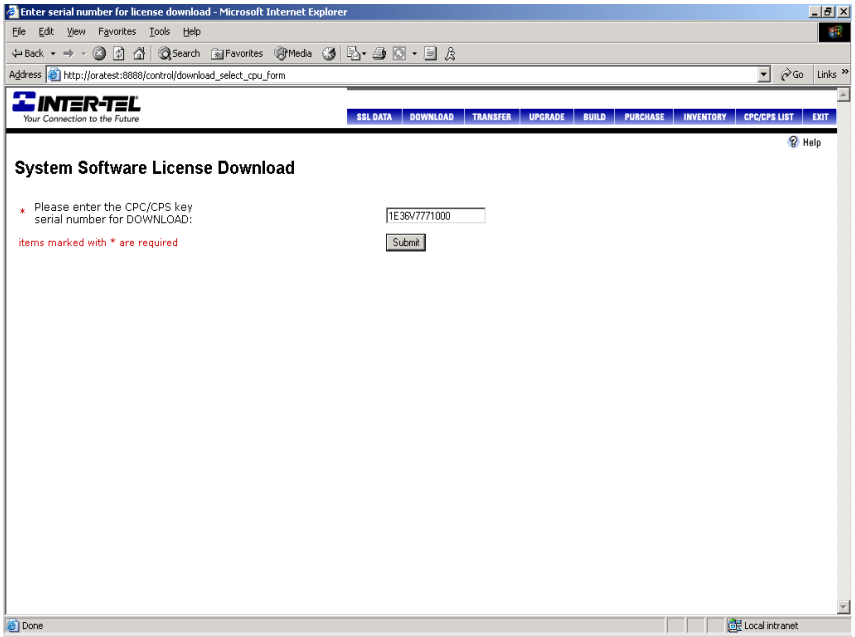


NOTE: Because you use a Web interface to build the software license, more than one person could build a license for the same serial and SSL numbers at the same time. If this occurs, the available feature units are allocated to the license that is built first.

10. Verify that the information in this screen is correct, then click **Generate license**. The license is generated, and a verification screen appears. While in this screen, you can e-mail the confirmation to any address.

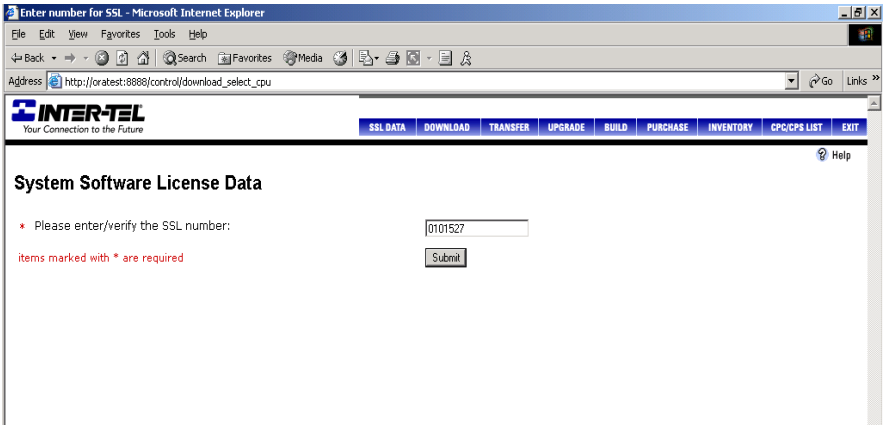


11. Click the **Download** tab. You are prompted for the serial number.

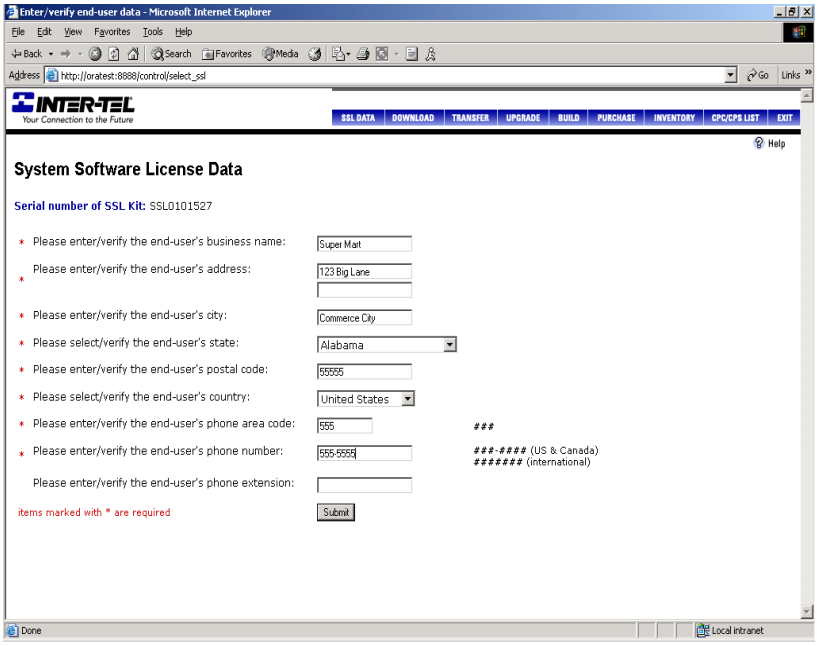


12. Enter the serial number of the CPC or CP Server software security key. Then click **Submit**. You are then asked to enter the Software Site License (SSL) number that is included with the CPC or CP Server.

NOTE: The SSL number is required for downloading a software license. If desired, you can click **SSL Data** to enter the SSL number and customer information at any time. You can also fax the SSL Data to the Software Management Center (480-961-8080) or send the information in an e-mail to smc@inter-tel.com.

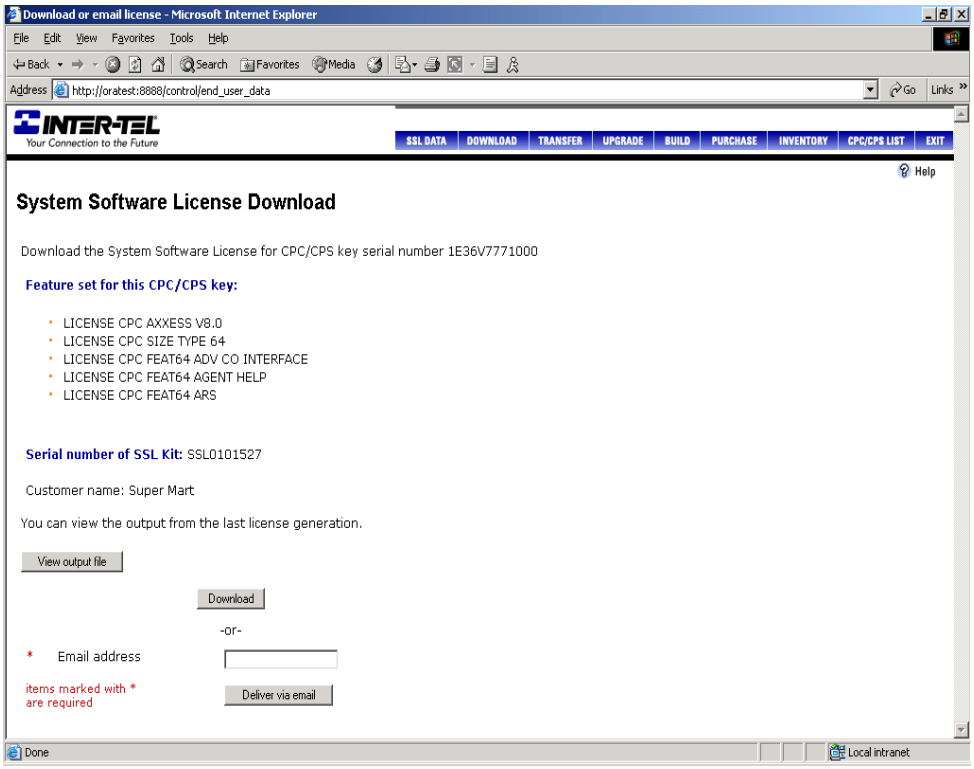


13. Click **Submit**. The System Software License Data prompts you to enter data about the customer site.



14. Enter the appropriate information. If you have already entered this data using the **SSL Data** option in the button bar, the fields are automatically populated.

15. Click **Submit**. A screen, similar to the one below, displays a summary of the software license.



16. *EITHER,* Click **Download** to download the license directly to your PC. If you select this option, a standard Windows Save screen appears, allowing you to browse for the file destination. Set the Save as type field to "All Files." The license is saved as an **`.isl`** file.

> **NOTE:** The Save as type field is set to "Text Document" by default. If you do not change the setting to "All Files," the file will be downloaded as a .txt file.

If you select this option, a standard Windows Save screen appears, allowing you to browse for the file destination. The license is saved as an **`.isl`** file.

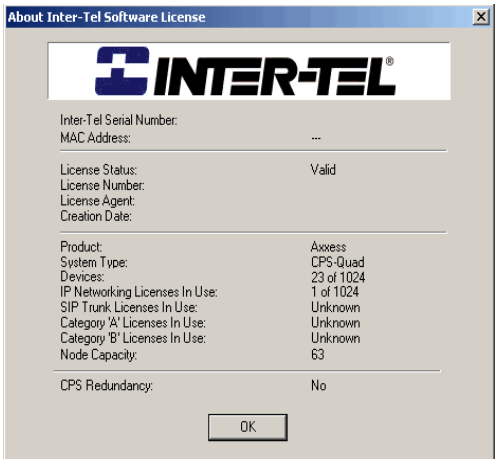
OR, Click **Deliver via email** to e-mail the license to an address. If you select this option, you are prompted for an e-mail address. The license is attached to the e-mail message as an **`.isl`** file. You must then save the file to a location that is accessible by the DB Programming PC.

> **NOTE:** The **View output file** button allows you to view the license information in a text-based file.

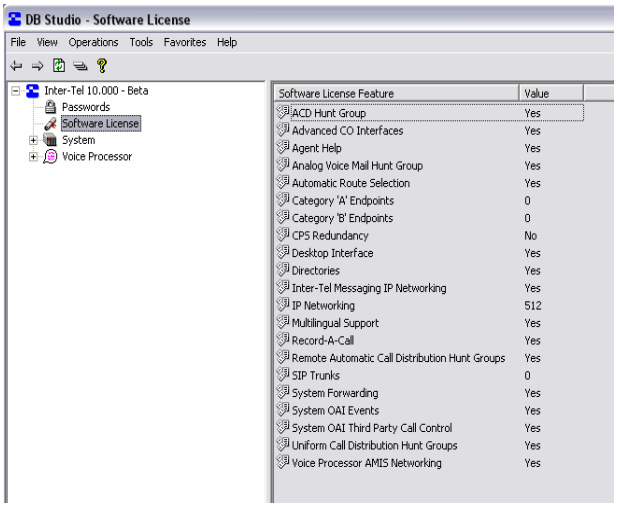
B. VIEWING THE CURRENT LICENSE

9.4 From the menu bar, select Help - **About Inter-Tel Software License** to see the new license features. Because there is now a Software License folder, much of the information previously located on the About Inter-Tel Software License dialog box no longer displays in the About Inter-Tel Software License menu option. The dialog box now contains main license and system information along with the current status of dynamic usage information. Also, the current node capacity appears. See the following examples. For information about the Software License folder, see page 6-97.

Help - About Inter-Tel Software License



Software License Folder



9.5 The About Inter-Tel Software License screen displays the software license information as described in the following table.

Table 7-24. *Software License Information*

Feature	Indicates
Inter-Tel Serial Number	The serial number (printed on the CPC or CP Server key) assigned to the software license.
MAC Address	The MAC address of the CPC. (The CP Server license does include a MAC address.)
License Status	Whether the license is valid or invalid.
License Number	The unique license number assigned by Inter-Tel.
License Agent	The person who created the license.
Creation Date (and Time)	The date and time the software license was created. This appears in the *MMM DD, YYYY (HH:MM:SS)* format.
Product	The product type (Axxess).
System Type	The system capacity (CPC-Single, CPC-Dual, CPU256, CPU512, CPU1024, or ATM CPU).
Devices	The number of devices currently programmed and the maximum number of physical devices supported. This is in the format *XX of YY* where *XX* is the number of devices currently programmed and *YY* is the maximum number of devices supported.

Table 7-24. *Software License Information (Continued)*

Feature	Indicates
IP Networking Licenses in Use	The total number of licenses available for IP networking and the number of IP networking that are currently consuming a license. This field is in the format XX of YY where XX indicates the number of endpoints currently programmed and YY indicates the total number of licenses available.
SIP Trunk Licenses in Use	The total number of licenses available for SIP trunks that are currently consuming a license. This field is in the format XX of YY where XX indicates the number of endpoints currently programmed and YY indicates the total number of licenses available.
Category A Licenses in Use	The total number of licenses available for Category A license. This field is in the format XX of YY where XX indicates the number of endpoints currently programmed and YY indicates the total number of licenses available.
Category B Licenses in Use	The total number of licenses available for Category B license. This field is in the format XX of YY where XX indicates the number of endpoints currently programmed and YY indicates the total number of licenses available.
Node Capacity	Indicates whether the system supports 63 or 99 nodes.
CPS Redundancy	Whether the CP Server redundancy is supported (Yes or No).

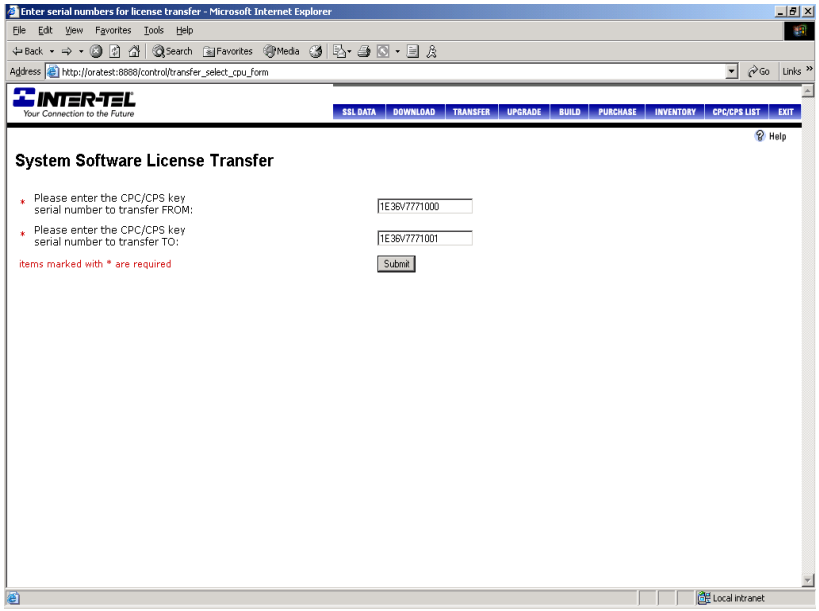
9.6 Click **OK** when finished to exit this screen.

C. TRANSFERRING LICENSES

9.7 If there are problems with the CPC or CP Server security key (see page 6-46 for a list of error messages), you may have to transfer a license.

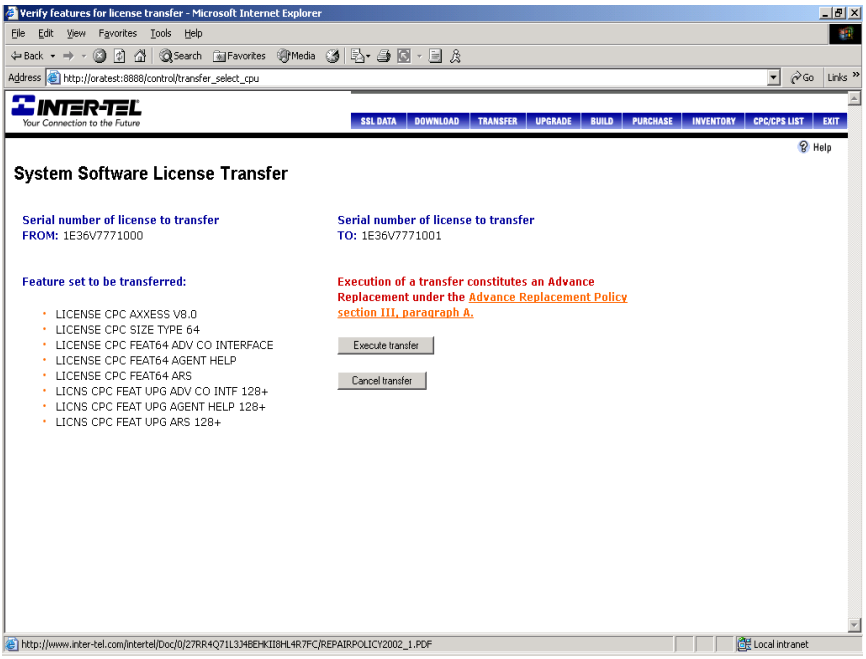
9.8 To transfer a license:

1. Access the License Management Center on the Inter-Tel Web site (the *edGe*).
2. Click **Transfer**. The following screen appears.



3. Enter the serial number of the CPC or CP Server security key currently associated with the license.
4. Enter the serial number of the new CPC or CP Server security key to which the license will be transferred.

5. Click **Submit**. A screen that provides a summary of the license appears.



6. Click **Execute transfer** to transfer the license or click **Cancel transfer** to cancel the transfer.

NOTE: The cost of the feature set is incurred when you transfer a license. However, a credit is applied when the defective CPC or CP Server is returned.

7. *EITHER,* Click **Download** to download the license directly to your PC. If you select this option, a standard Windows Save screen appears, allowing you to browse for the file destination. Set the Save as type field to "All Files." The license is saved as an **`.isl`** file.

NOTE: The Save as type field is set to "Text Document" by default. If you do not change the setting to "All Files," the file will be downloaded as a .txt file.

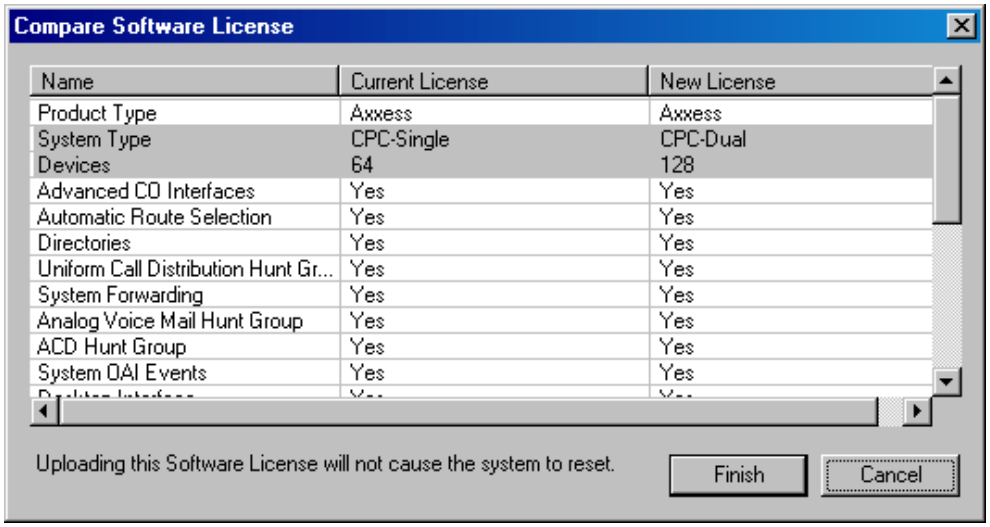
OR, Click **Deliver via email** to e-mail the license to an address. If you select this option, you are prompted for an e-mail address. The license is attached to the e-mail message as an **`.isl`** file.

9.9 If necessary, contact your local sales dealer for information on returns and/or upgrades.

D. CHANGING THE LICENSE

9.10 If necessary, you can upload a new license that provides a different set of features. If the new license, however, supports new capacities, you must make sure that you have the appropriate hardware installed.

9.11 To change a license, obtain a new license (see page 7-95) and then upload the license (see page 6-45). The system compares the new license to the existing license and then displays a screen similar to the one shown below.



9.12 In this screen, the areas of the license that are changing are highlighted. For example, in the above screen, the System Type is changing from CPC-Single to CPC-Dual, and the number of devices supported is increasing to 128. In addition, the status line indicates whether the system requires a reset once the license is uploaded.

9.13 After you have verified that you want to upload the new license, click **Finish**. The new software license is loaded, and the old software license is stored on the system. The licenses currently uploaded to the system are displayed in the Software License folder (see page 6-97). For more information about system and software licenses, see page 5-11.

NOTE: The system may require a reset if the CPU card type has changed.

10. AUTOMATIC DIAGNOSTICS DELIVERY

10.1 Automatic Diagnostics Delivery (ADD) is a new troubleshooting feature available for the CPC and CP Server. If programmed, the system automatically sends information to Inter-Tel whenever the system freezes. Because the information is sent directly to Inter-Tel, Technical Support is able to retrieve system freezes without involving a technician, providing quicker response times.

10.2 The CPC and CP Server use port 443 for communicating with the Inter-Tel ADD server. If this port is blocked when attempting to send freezes, an error message appears in Message Print. ADD is available only in On-Line Monitor (OLM) mode. To enable ADD, you must contact Inter-Tel Technical Support.

11. SYSTEM BACKUP AND RECOVERY

11.1 This section describes system backup and recovery.

A. RECOVERY BACKUP

11.2 Hard drives are now partitioned with a C drive partition (for the operating system and applications) and an E drive partition (for recovery backup files). The D drive is the CD/DVD drive. Windows is set up at the factory to automatically back up (save) important application data to the E drive partition using the built-in Windows backup/scheduling programs.

NOTE: This backup is not to be used as a replacement for the database or voice mail save procedures. See pages 6-65 and 6-71 for details on performing database and voice mail saves, respectively.

11.3 If the product originally shipped with an E:\Backup directory, the unit is configured to back up critical database files to E:\Backup every Sunday at 2:00 A.M. using the Local System Account (NT Authority). The batch file that performs this operation can be found in ITAdministrator\My Documents. This backup is solely intended to be used in the event of a system re-image from the supplied recovery CD/DVD. It is strongly recommended that an off-unit backup be performed before applying this new image. However, if a backup is not available, use the files on E:\Backup to restore the customer's data. Deleting the Windows Scheduled Task or the batch file itself will prevent this backup from occurring.

NOTE: There is an "ITAdministrator" account that is used by Technical Support only. To remain eligible for remote support and assistance, do not modify this account.

11.4 For details about the recovery backup procedure, refer to the *Recovery Instructions for PC-Based Products* (document part number 835.2759) that is included in the recovery CD/DVD. For details about the recovery backup procedure on an Enterprise® Messaging system, refer to the *Enterprise Messaging Installation Manual* (document part number 780.8006).

B. GHOST RECOVERY CD/DVD

11.5 The ghost recovery CD/DVD that is shipped with the C/E partitioned products only re-images the C partition, not the E partition. This way, you can re-image the C partition to bring the unit back to the original factory installation, but still have access to the current application data on the E drive. You will, however, need to reload any Inter-Tel applications and service packs that are not included in the original factory installation. If the unit did not originally contain an E:\Backup directory, the entire drive will be re-imaged. To recover the system, follow the instructions in the *Recovery Instructions for PC-Based Products* (document part number 835.2759) that is included in the recovery CD/DVD-ROM.

NOTICE

In some circumstances beyond Inter-Tel's control, the recovery process may re-label the drive partitions different from Inter-Tel's factory settings. This is usually caused by third party partitioning programs, changes in cabling, or the addition of non-factory devices. After recovering the unit, verify that the system contains the C:\ partition, the CD/DVD is labeled D:\, and the Recovery Backup partition is labeled E:\. These settings can be viewed by accessing the Disk Administrator within Windows. If these settings are not as described, use Disk Administrator to rename the partitions to their original factory settings. Failure to employ these settings will prevent the recovery backup script from executing.

12. CUSTOMER SUPPORT

12.1 Keep the following information in mind for troubleshooting purposes.

A. TECHNICAL SUPPORT

12.2 If problems persist when installing or servicing Inter-Tel equipment: While on site and with the proper troubleshooting tools available, certified technicians may contact Inter-Tel's Technical Support Department at 1-888-777-EASY (3279) in the U.S., or (41162) 903000 in Europe.

B. EMERGENCY ASSISTANCE

12.3 After office hours and on weekends, call 1-888-777-EASY (3279) and leave your message with the voice mail service. A Technical Support Product Specialist will return your call as soon as possible, usually within an hour. As a backup (e.g., the main number does not answer or you do not receive a call back within an hour), you can call 480-961-0277. Please remember that this is an emergency number for *critical system problems only.* Sales questions, equipment orders, etc., can only be handled during normal business hours.

C. DIAGNOSTICS

12.4 For advanced troubleshooting purposes, the Voice Processing applications software includes special diagnostics. If a Voice Processing applications problem cannot be resolved, Technical Support may instruct the technician to connect a monitor and keyboard and perform certain diagnostic procedures.

12.5 For example, if the system is not performing properly, log files on the PC's hard drive can be copied to a floppy disk and sent back to Inter-Tel for evaluation. (Some examples of unusual system behavior include: the PC continually resets at specific time intervals, the link between the PC and the telephone system keeps dropping, and greetings and recordings are scrambled and/or show up in the wrong location.)

12.6 To retrieve the system log files:

1. Insert a blank, properly formatted floppy disk into the floppy drive.
2. Press and hold **SHIFT**, then press the vertical bar (|) button on the keyboard. (The floppy disk drive LED lights as the files are copied to the floppy disk.)
3. When the floppy drive LED goes out, remove the floppy disk. (The log files can be examined using a text editor.)

NOTE: The amount of time it takes to copy the log files is dependent on the amount of information in the files. It could take from a few seconds to a few minutes. Copying the log files to a floppy disk does slow down the system; therefore, it is best to perform this procedure when the system is idle.

D. END OF SALE/END OF LIFE POLICIES

12.7 The End of Sale and End of Life policies were created to help maintain focus on the development and delivery of new products and technologies while halting the sale and support of products that have reached the end of their useful lives. Most of these products have been replaced by a new product or enhanced solution. The end-of-sale date marks the beginning of a period when a product is no longer sold and support is limited. End-of-life marks the date a product is no longer supported. Refer to the *edGe* Product Central for a complete list of products that are officially listed as "end-of-sale" or "end-of-life." For readers located in Europe, contact the Inter-Tel Europe office for product information.

13. DEFECTIVE UNIT RETURN POLICY

13.1 For complete information on returning equipment, refer to the current Inter-Tel *Repair and Return Policy* (document part no. 835.1065). This document includes specific information on the following subjects: warranty, procedures to follow when returning equipment, equipment damaged in shipment, insurance, repair policy, and advance replacement policy.

NOTE: Some of the cards and cables inside the Voice Processing PC may have been spot-glued into place to help prevent them from working loose during shipment. To remove one of these cards or cables from the PC (if necessary), first break the glue seal by gently wiggling the connector with a pair of pliers.

Replacement Parts

CONTENTS PAGE

1. Introduction 8-2
2. Ordering Procedure 8-2
3. Replacement Parts and Software License Part Numbers. 8-2
4. Recommended Spare Parts 8-2

1. INTRODUCTION

1.1 This section provides the information necessary to order and stock replacement parts for the Inter-Tel® Axxess® System.

NOTE: See a version 7.0 or earlier *Eclipse™ Installation and Maintenance Manual* for Eclipse part numbers.

2. ORDERING PROCEDURE

2.1 When ordering equipment for the Inter-Tel phone system, provide the following information to the order processing clerk:

- Company name
- Purchase order number
- Required date of shipment
- Part number(s) of equipment ordered
- Quantity required

3. REPLACEMENT PARTS AND SOFTWARE LICENSE PART NUMBERS

3.1 Figure 8-1 lists authorized parts available for replacement on the Inter-Tel phone system. Figure 8-2 lists the part numbers required when ordering a software license for use on an 8.0 or later system.

4. RECOMMENDED SPARE PARTS

4.1 Inter-Tel, Inc. is dedicated to providing the best possible customer service. To eliminate interruption of service to our customers' business operations, Inter-Tel, Inc. recommends that Inter-Tel dealers/branches maintain an inventory of spare parts.

4.2 Critical components of your installed based should be kept in inventory to minimize the outage for our customers. Because each of the customer installations is different, identifying a standard list of spare parts is unreasonable. Inter-Tel's recommendation is that you evaluate your installed customer base and identify the critical components that in the event of failure would result in a significant loss of business operations. These components should be considered your spare parts inventory.

4.3 The spare parts inventory should be maintained and, in some cases, increased when new customers are sold equipment. Keep in mind that hardware, firmware, and software revisions change and could affect the revision level of your inventory.

Figure 8-1. *Replacement Parts*

DESCRIPTION	PART NUMBER (U.S./EUROPE)
Original Equipment Chassis and Power Supply Parts	
Backplane Termination Resistor	831.5570
Chassis Expansion Cable 5" (12.5 cm)	813.1576
Chassis Expansion Cable 44" (112 cm)	813.1636
New-Style Equipment Chassis and Power Supply Parts	
Chassis Assembly (with backplane only)	550.1300
Chassis Backplane	550.1002
Power Supply Unit – 4 Amp	550.0121 (PS2747)
Power Supply Unit – 6 Amp	550.0104
Power Supply Unit – 9 Amp	550.0131 (PS2735)
Chassis Expansion Cable 5" (same as original)	813.1576 (C99)
Chassis Expansion Cable 44" (same as original)	813.1636 (C131)
ATM Switches	
ForeRunner LE 155 12-Port UTP5 Switch	550.5251 (IT128)
ForeRunner LE 155 12-Port OC-3 Switch	550.5252 (IT132)
ForeRunner LE 155 12-Port UTP5 w/4-Port OC-3 Switch	550.5253 (IT130)
ForeRunner LE 155 12-Port UTP5 w/4-Port OC-3 (2 SMF and 2 MMF) Switch	550.5270 (IT139)
ASX-200BX 8-port UTP5 Switch	550.5296
ASX-200BX 8-Port MMF Switch	550.5297
ASX-200BX 4-Port UTP5 and 4-Port MMF Switch	550.5298
ASX-200BX 4-Port UTP5, 4-Port MMF, and 4-Port SMF Switch	550.5299

Figure 8-1. *Replacement Parts (Continued)*

DESCRIPTION	PART NUMBER (U.S./EUROPE)
CPU Cards	
With Extended Memory Installed, with EAC Connector:	
CPU020/PCM-D (CPU 256) with Extended Memory	550.9037
CPU020/PCM-F (CPU 512 Slave) with Extended Memory	550.9036
CPU020/PCM-F (CPU 512 Master) with Extended Memory	550.9035
CPU020 with Extended Memory and EAC Connector	550.2023
CPC (64, 128, or 256 Devices)	550.2030
CPU128 with page port	550.2015
CPU128 without page port	550.2010
CPU128 with Memory Daughter Card	550.2017
CPU 64	550.2014
Without Extended Memory, with EAC Connector:	
CPC (64, 128, or 256 Devices)	(PB2030)
CPU020/PCM-D/EAC (CPU 256)	(PB2599)
CPU020/PCM-F/EAC (CPU 512)	(PB2600)
Cards used with CP Server:	
CPU020/PCM-D/EAC (CPU 256)	550.9026 (PB2599)
CPU020/PCM-F/EAC (CPU 512 Master)	550.9025 (PB2600)
CPU020/PCM-F (CPU 512 Slave) with Extended Memory	550.9036 (PB2592)
CPU020/PCMA-F (ATM MMF)	550.9041 (PB6000)
CPU020/PCMA-FS (ATM SMF)	550.9044 (PB6002)
CPU020/PCMA-C (ATM CAT5)	550.9042 (PB6001)
CPUs without EAC Connector or Extended Memory:	
CPU020-D/Q Card (requires D or Q software kit)	550.2003
CPU020-X Card (requires X software kit)	550.2005
CPU020-Q/PCM-F (CPU 512 Master)	550.9015 (PB2591)
CPU020-X/PCM-F (CPU 512 Slave)	550.9016 (PB2592)
CPU020-D/PCM-D (CPU 256)	550.9017 (PB2593)
CPU128	(PB2590)
CPU Card Add-Ons:	
CPU020-X/PCM Upgrade Kit	828.1456
CPU020-D/PCM Upgrade Kit	828.1457
CPU020-Q/PCM Upgrade Kit	828.1458
Ethernet Adapter Card (EAC)	550.3033
PCM-D Card	550.2007
PCM-F Card	550.2006
PCM-D Card (A-Law/Mu-Law)	550.2027
PCM-F Card (A-Law/Mu-Law)	550.2026

Figure 8-1. *Replacement Parts (Continued)*

DESCRIPTION	PART NUMBER (U.S./EUROPE)
CP Server	
Windows NT-Based PC Desktop Unit	550.5199
Windows NT-Based PC Rackmount Unit	550.5187
Windows 2000-Based PC Desktop UnitNon-ATM	550.5260 (IT135)
Windows 2000-Based PC Rackmount Unit Non-ATM	550.9089 (IT136)
Windows 2000-Based PC Desktop Unit CAT5	550.5256 (IT134)
Windows 2000-Based PC Rackmount Unit CAT5	550.5258 (IT137)
Windows 2000-Based PC Desktop Unit MMF	550.5257 (IT135)
Windows 2000-Based PC Rackmount Unit MMF	550.5259 (IT138)
Network Interface Card (NIC)	550.5304
ATM Adapter Card with OC-3 Connector	550.5255 (IT131)
ATM Adapter Card with UTP5 Connector	550.5254 (IT129)
56K Internal Modem	550.5193
14-inch SVGA Monitor	440.5080
Non-ATM NT CPU Security Key (v5.2/5.3)	827.9360
Non-ATM CP Server Security Key (v7.0)	827.9641
Non-ATM CP Server Redundancy Kit	828.1550
ATM CP Server Redundancy Kit	828.1551
Base Security Key (v8.0)	827.9775
Redundancy Backup Security Key (v8.0)	827.9776
10Base-T CAT5 Cross-Connect Cable (14 ft)	813.1724
Multi-Mode Fiber SC/SC Cable (14 ft 5 m)	813.1772 (C158)
Multi-Mode Fiber ST/SC Cable (14 ft/5 m)	813.1773
UTP5 Cable (14 ft)	813.1743
Windows NT-Based CPU Memory Map PAL	827.8967
Axxess CPU020 Operating System Software	827.9092
Axxess Installation Software	827.9806
WinZip Software	827.8811
PC Anywhere Host Software	827.8797
Diag Mon Software	827.9826
Upgrade Kit CPS Windows 2000 Professional to Windows 2003 Server for Non-ATM	828.1754
Upgrade Kit CPS Windows 2000 Professional to Windows 2003 Server for ATM	828.1755

Figure 8-1. *Replacement Parts (Continued)*

DESCRIPTION	PART NUMBER (U.S./EUROPE)
Other Circuit Cards	
Options Card	550.2600 (PB2547)
Digital Keyset Card – 8-Port	550.2200 (PB2545)
Digital Keyset Card – 16-Port	550.2250 (PB2563)
Digital Keyset Card – 16-Port +	550.2255 (PB2605-A)
Analog Keyset Card	550.2208 (PB2579)
Embedded Voice Mail Card (EVMC) with OS/2 Operating System	550.5030 (PB2602)
Embedded Voice Mail Card (EVMC) with Linux Operating System	550.5040
EVMC with OS/2 Operating System, spare hard drive	550.5363
Executone Keyset Card	560.2212
Single-Line Adapter (includes AC transformer)	550.2500 (PB2554-A)
Single-Line Card – 8-Port	550.2101 (PB2552)
Single-Line Card – 16-Port	550.2116 (PB2586)
Single-Line Card – 16-Port +	550.2126
SLC-16 Power Supply (for 1 card)	550.0114 (PS2743)
SLC-16 Power Supply (for 4 cards)	550.0116 (PS2744)
Lamp Option Card – 8-Port	550.2102 (PB2553)
Lamp Options Card – 16-Port	550.2104 (PB2587)
Loop/Ground Start Card	550.2309
Loop/Ground Start Daughter Card	550.2310
Digital Loop/Ground Start Card	550.2320
Digital Loop/Ground Start Daughter Card	550.2321
Loop Start Card	550.2300 (PB2546)
Loop Start Daughter Card	550.2301 (PB2555)
Internet Protocol Resource Daughter Card	550.2266
IPRC-DE (Daughter Enhanced)	550.2277
IPRC16-DE	550.2276
Internet Protocol Resource Card	550.2265
T1 Card	550.2730
E1 Card (ISDN)	(PB2580)
T1/E1/PRI Card	550.2740 (PB2606)
Primary Rate Interface Card	550.2733 (PB2581)
Network Clock Card	550.2732
BRS Card (Basic Rate S/T Interface Card)	550.2220
BRU Card (Basic Rate U Interface Card)	550.2230
Firmware	
DKSC-16+ v5.5 (Kit Part No. 828.1560)	827.9976
SLC 16+ v4.0	827.9964
IPRC Station Firmware v1.5.2	841.0176
IPRC Networking Firmware v7.0.5	841.0033
IPRC Networking Firmware v8.2.0	841.0162
IPRC Firmware v9.0.0	841.0148
EKSC Firmware v7.0 (Kit Part No. 828.1670)	841.0054
CPS Boot Code v2.1.4	841.0077
Mu-Law OPC Firmware Upgrade Kit (for the U.S.)	828.1561
A-Law OPC Firmware Upgrade Kit	828.1671 (IT158)
EAC 512 OS v8.1.5	841.0161
ATM OS v8.1.10	841.0425

Figure 8-1. *Replacement Parts (Continued)*

DESCRIPTION	PART NUMBER (U.S./EUROPE)
CT Gateway/Transceiver	
Switch Transceiver Software	827.9000
Applications Transceiver Software	827.9001
CT Gateway Software	827.9002
System Software	
Version 11.0 System Software CD-ROM (contains all software)	841.0935
Axxess Software Upgrade v10.x to v11.0	840.0829
EVMC PALs	
64-Mailbox PAL	(DI176)
200-Mailbox PAL	(DI177)
CP Server Security Keys (ATM Systems)	

NOTE: See the *Software License Part Numbers* section for v8.0 information. Version 8.0 does not use feature unit or networking PALs.

25-Feature Units	827.9509
50-Feature Units	827.9510
75-Feature Units	827.9511
100-Feature Units	827.9512
125-Feature Units	827.9513
150-Feature Units	827.9514
Additional 25 Feature Units for ATM CP Server	827.9649
Digital Station Instruments	
Axxess Basic Digital Phone (with red/green LEDs)	520.4300 (KT534)
Eclipse™ Basic Digital Phone	560.4100 (N/A)
Associate Display Phone	560.4200 (N/A)
Professional Display Phone	560.4300 (N/A)
Standard Display Phone (with red/green LEDs)	550.4400 (KT535)
Executive Display Phone (with red/green LEDs)	550.4500 (KT536)
Model 8500 Phone (with red/green LEDs)	550.8500 (KT540)
Model 8520 Phone (with red/green LEDs)	550.8520 (KT539)
Model 8560 Phone (with red/green LEDs)	550.8560 (KT538)
60-Key 12-Row DSS/BLF Unit (includes AC transformer & PCDPM cable)	560.4000
60-Key 6-Row Digital DSS/BLF Unit (includes AC transformer & PCDPM cable)	550.4200 (KT532-A)
50-Key Digital DSS/BLF Unit (with AC transformer & PCDPM cable)	550.8450 (KT542)
16-Key Mini-DSS	550.8416 (KT543)

Figure 8-1. *Replacement Parts (Continued)*

DESCRIPTION	PART NUMBER (U.S./EUROPE)
Analog Station Instruments	
IMX 24-Line Display Phone – charcoal, electret	660.7600
IMX 24-Line AIM Phone (display only) – charcoal, electret	660.7000
IMX 12-Line Standard Phone – charcoal, electret	660.7900
IMX 8-Line Phone (non-display only) – charcoal, electret	660.7500
IMX 8-Line AIM Phone (display only) – charcoal, electret	660.7400
IMX 12-Line Standard Phone – gray, electret	661.7900
Inter-Tel/DVK 24-Line Display Phone	662.3400
Inter-Tel/DVK 12-Line Display Phone	662.3900

NOTE: See the *IP Devices Installation Manual* (part number 835.2195) for the list of hard and soft IP endpoint part numbers.

IP and SIP Endpoint Instruments	
Model 8600 Multi-Protocol Endpoint with No Display	550.8600
Model 8601 SIP Softphone for Pocket PC	827.9901
Model 8602 IP Softphone Application	
Requires Category B IP Endpoint License	840.0477
Model 8622 – 2-Line Display Multi-Protocol Endpoint (PoE Class 0)	550.8622
Model 8622 – 2-Line Display Multi-Protocol Endpoint (PoE Class 2)	550.8622P
Model 8660 – 6-Line Display IP Endpoint	550.8660
Model 8662 – 6-Line Display Multi-Protocol Endpoint (PoE Class 0)	550.8662E
Model 8662 – 6-Line Display Multi-Protocol Endpoint (PoE Class 2)	550.8662P
Model 8664 – 2-Line Display 802.11b Wireless IP Endpoint	550.8664
Model 8665 – 2-Line Display 802.11b Wireless IP Endpoint	550.8665
Model 8668 – 2-Line Display 802.11b Wireless IP Endpoint	550.8668
Model 8690 Multi-Protocol, Multimedia Touchscreen Endpoint	550.8690
Internet Protocol Single Line Adapter (IP SLA)	770.3000

Figure 8-1. *Replacement Parts (Continued)*

DESCRIPTION	PART NUMBER (U.S./EUROPE)
Voice Processing	
IBM-486 (or compatible) Personal Computer (PC) Assembly	550.5124
Power Supply Replacement Kit NT Version	828.1478
PC Motherboard (80486 without memory)	550.5126
8 MB SIMM821.1052 Floppy Disk Drive (3½-inch double-sided/high-density)	550.5111
Hard Disk Drive	550.5118
Disk Controller Card (with serial & parallel ports)	691.3007
Floppy Disk-To-Controller Interface Cable	813.1623
Hard Disk-To-Controller Interface Cable	813.1626
NT PC Upgrade Unit	550.5220
NT 4-Port PC	550.5221 (IT101)
NT 8-Port PC	550.5222 (IT102)
NT 12-Port PC	550.5223 (IT103)
NT 16-Port PC	550.5224 (IT104)
NT 24-Port PC	550.5225 (IT105)
NT 32-Port PC	550.5226 (IT106)
Audio Interface Card (AIC)	550.2800 (PB2565)
Voice Processing Card (VPC) – 4 Port	550.3006 (PB2572)
Voice Processing Card (VPC) – 8 Port	550.5121 (PB2573)
Voice Processing Card (VPC) – 4 Port Mini-card	550.5209
Voice Processing Card (VPC) – 8 Port Mini-card	550.5210
Voice Processing Card (VPC) – 16 Port	550.5120 (PB2574)
Video Card – VGA	440.5069
Optional Fax Card	550.5122 (PB2571)
AIC-To-Reset Switch Interface Cable	813.1588
AIC-To-VPC Interface Cable	813.1586
AIC-To-CPU/OPC Interface Cable	813.1587 (C122)
Optional SMDR Retrieval Cable Kit (includes items below)	828.1282
Line Cord (standard, two-pair, reversing)	813.1592
DB9 Modular Adapter (female, non-reversing, four-wire) – MOD-TAP part no. 08-988-610-00	804.2420 (DMS67)
DB25 Modular Adapter (female, reversing, four-wire) – MOD-TAP part no. 06-986-G99-33	804.2425
Optional Diagnostics Kit	550.5117
14” SVGA Monitor	440.5080
Keyboard	440.5081
Optional PC Wall-Mount Shelf (fits most PC models)	823.1270

NOTE: Depending on the specific model of PC being used, some of the parts and part numbers listed above could vary somewhat.

Figure 8-1. *Replacement Parts (Continued)*

DESCRIPTION	PART NUMBER (U.S./EUROPE)
Documentation	
Axxess Installation & Maintenance Manual	550.8000
Administrator's Guide	550.8001
Model 8690 Administrator's Guide	550.8120
Professional Display Phone User Guide & Quick Reference	560.8002
Executive Display Phone User & Quick Reference Guide	550.8108
Axxess Basic Digital & Standard Display Phone User & Quick Ref. Guide	550.8109
Associate/Basic Phone User Guide & Quick Reference	560.8003
Model 8500 Phone User & Quick Reference Guide	550.8114
Model 8520 Phone User & Quick Reference Guide	550.8112
Model 8560 Phone User & Quick Reference Guide	550.8113
Model 8664/8665/8668 Phone User & Quick Reference Guide	550.8026
Model 8600 Phone User & Quick Reference Guide (IP Mode)	550.8115
Model 8620 Phone User & Quick Reference Guide (IP Mode)	550.8119
Model 8662 Phone User & Quick Reference Guide (IP Mode)	550.8117
Model 8690 User Guide (ITP Mode)	550.8116
Model 8690 User Guide (SIP Mode)	550.8025
Model 8000 Series Phone User & Quick Reference Guide (SIP Mode)	550.8024
Analog Phone User & Quick Reference Guide	550.8110
Executone Keyset User Guide & Quick Reference	560.8004
Single-Line Set User & Quick Reference Guide	550.8107
Unified Messaging Installation Manual	835.2296
Unified Messaging User & Quick Reference Guide	550.8111
CT Gateway Installation Manual	835.2161
Axxess IP Devices Installation Manual	835.2195
Eclipse IP Devices Installation Manual	835.2410
NetLink SVP Server and Model 8664/8665/8668 Installation Manual	935.4521
Axxess Call Processing Server Upgrade Kit Instructions	835.3211

Figure 8-1. *Replacement Parts (Continued)*

DESCRIPTION	PART NUMBER (U.S./EUROPE)
Miscellaneous Equipment	
Rechargeable Battery (CPC Only)	815.1026 (DB5)
Fiber-Optic Cable (8.5-foot/2.5-meter)	813.1628 (C118)
Digital Signal Processor (DSP) Chip—for the U.S.	834.2566
DSP Chip—for Europe	834.2728 (DI241)
DSP PAL Chip – for an OPC with one DSP installed	827.8023
DSP PAL Chip – for an OPC with two DSPs installed	827.8024
DSP PAL Chip – for an OPC with three DSPs installed	827.8025
DSP PAL Chip – for an OPC with four DSPs installed	827.1020
Executive Display, Standard Display, Model 8520, or Model 8560 Phone Modem Data Port Module (MDPM) – includes PCDPM-To-MDPM cable & AC transformer	550.3015 (PB2549-A)
Executive or Standard Display Phone PC Data Port Module (PCDPM) – does not require an external AC transformer	550.3018 (PB2585)
Model 8520 or Model 8560 Phone PC Data Port Module (PCDPM) – does not require an external AC transformer	550.8400 (PB2612)
PCDPM-To-Keyset Interface Cable	813.1567 (C108)
PCDPM-To-MDPM Interface Cable	813.1566 (C107)
PCDPM-To-RS232C Interface Cable	813.1631 (C106)
PCDPM-To-DSS Interface Cable	813.1595 (C109)
DSS-To-DSS Interface Cable	813.1519 (C105)
Plug-Top Transformer (for MDPMs and DSS/BLF Units)	(PS2732)
AC Transformer – 24VAC, 450mA (for PCDPMs, MDPMs, and DSS/BLF Units)	806.1045
AC Transformer – 24VAC, 1.25A (for SLAs	806.1046 (PS2731)
T1/PRI Firmware Upgrade Kit	828.1521
Modem Kit (includes modem and custom cable)	550.3026 (IT46-A)
Digital Attendant Playback Device	828.1315
Database Back-Up Battery (3.6V, ½AA lithium	815.1011 (DB4)
IMX 24/12-Line Phone Data Port Module Kit	828.1094
IMX 24/12-Line Phone LCD Kit.	828.1072
IMX 24/12/8-Line Phone Blank Keycap Kit (100)	812.1041
Inter-Tel/DVK Phone Data Port Module Kit.	828.1094
Inter-Tel/DVK 24-Line Phone LCD Kit (Large).	828.1166
Inter-Tel/DVK 12/8-Line Phone LCD Kit (Small)	828.1165
Zip Drive Backup Kit	828.1460
Universal RS232 Kit	828.1282

Figure 8-2. *Software License Part Numbers*

DESCRIPTION	PART NUMBER (U.S./EUROPE)
Axxess (v10.0) CD-ROM	841.0580
Axxess CPC Software License	840.0680
Category A Endpoints—8 Endpoints	840.0132
—16 Endpoints	840.0133
—32 Endpoints	840.0134
Category B Endpoints—IP Endpoint (Model 8602)	840.0633
IP Networking Ports—4 Ports	840.0115
—8 Ports	840.0116
—16 Ports	840.0117
—32 Ports	840.0118
Axxess License Enables 99 Nodes	840.0632
CPC - Single (64)	840.0008 (SW0008)
CPC - Dual (128)	840.0009 (SW0009)
CPC - Dual (256)	840.0010 (SW0010)
512CPS Master	840.0011 (SW0011)
1024CPS	840.0013 (SW0013)
ATMCPS 2 PCMA Groups	840.0081 (SW0081)
ATMCPS 3 PCMA Groups	840.0082 (SW0082)
ATMCPS 4 PCMA Groups	840.0083 (SW0083)
CP Server Redundancy Enabled	840.0020 (SW0020)
CPS Redundancy (Non-ATM)	(SW1550)
CPS Redundancy (ATM)	(SW1551)
System Manager (v1.1)	840.0262 (SW0262)
System Manager CP Snap-In (v9.0)	840.0263 (SW0263)
SIP Server (1 Trunk)	840.0206 (SW0206)
SIP Server (1 Trunk, Orderable)	840.0213 (SW0213)
SIP Gateway (1 Trunk)	840.0266 (SW0266)
SIP Gateway (1 Trunk, Orderable)	840.0267 (SW0267)
Multi-Protocol Endpoints	
8 Endpoints	840.0132
16 Endpoints	840.0133
32 Endpoints	840.0134
IP Networking Ports	
4 Ports	840.0115
8 Ports	840.0116
16 Ports	840.0117
32 Ports	840.0118
Networking Feature	
CPC - Single (64)	840.0015 (SW0015)
CPC - Dual (128)	840.0016 (SW0016)
CPC - Dual (256)	840.0017 (SW0017)
512CPS	840.0018 (SW0018)
1024CPS or ATM	840.0019 (SW0019)

Figure 8-2. *Software License Part Numbers (Continued)*

DESCRIPTION	PART NUMBER (U.S./EUROPE)
CPC64 Premium Features	
Advanced CO Interfaces	840.0022 (SW0022)
Agent Help	840.0023 (SW0023)
Voice Processor Analog Hunt Groups	840.0024 (SW0024)
ACD Hunt Groups	840.0025 (SW0025)
ARS	840.0026 (SW0026)
Voice Processor AMIS Networking	840.0027 (SW0027)
Inter-Tel Messaging IP Networking	840.0028 (SW0028)
Desktop Interface	840.0029 (SW0029)
Directories	840.0030 (SW0030)
Record-A-Call	840.0031 (SW0031)
Remote ACD Hunt Group	840.0032 (SW0032)
System Forwarding	840.0033 (SW0033)
System OAI Events	840.0034 (SW0034)
System OAI Third Party Call Control	840.0035 (SW0035)
UCD Hunt Groups	840.0036 (SW0036)
Multilingual	840.0037 (SW0037)
CPC - Dual (128), CPC - Dual (256), 512CPS, and ATM CPS Premium Features	
Advanced CO Interfaces	840.0038 (SW0038)
Agent Help	840.0039 (SW0039)
Voice Processor Analog Hunt Groups	840.0040 (SW0040)
ACD Hunt Groups	840.0041 (SW0041)
ARS	840.0042 (SW0042)
Voice Processor AMIS Networking	840.0043 (SW0043)
Inter-Tel Messaging IP Networking	840.0044 (SW0044)
Desktop Interface	840.0045 (SW0045
Directories	840.0046 (SW0046)
Record-A-Call	840.0047 (SW0047)
Remote ACD Hunt Group	840.0048 (SW0048)
System Forwarding	840.0049 (SW0049)
System OAI Events	840.0050 (SW0050)
System OAI Third Party Call Control	840.0051 (SW0051)
UCD Hunt Groups	840.0052 (SW0052)
Multilingual	840.0053 (SW0053)
CPU Type Upgrades	
64 to 128	840.0054
128 to 256	840.0055
256 to 512CPS	840.0056
512 to 1024CPS	840.0057
CPU Network Upgrades	
64 to 128	840.0058
128 to 256	840.0059
256 to 512CPS	840.0060
512 to 1024CPS	840.0061
CPU Product Upgrades	
Axxess (v8.x to v10.0)	840.0683
Axxess (v9.x to v10.0)	840.0682

Figure 8-2. *Software License Part Numbers (Continued)*

DESCRIPTION	PART NUMBER (U.S./EUROPE)
System Manager Upgrades	
System Manager (v1.0 to 1.1)	840.0265
System Manager CP Snap-In (v8.x to 9.0)	840.0264
CPU64 Feature Upgrades to 128 (or Higher)	
Advanced CO Interfaces	840.0062
Agent Help	840.0063
Voice Processor Analog Hunt Groups	840.0064
ACD Hunt Groups	840.0065
ARS	840.0066
Voice Processor AMIS Networking	840.0067
Inter-Tel Messaging IP Networking	840.0068
Desktop Interface	840.0069
Directories	840.0070
Record-A-Call	840.0071
Remote ACD Hunt Group	840.0072
System Forwarding	840.0073
System OAI Events	840.0074
System OAI Third Party Call Control	840.0075
UCD Hunt Groups	840.0076
Multilingual	840.0077

NOTE: You must purchase each feature upgrade if upgrading to a 128 (or higher) system. CPU64 features do not work on 128 (or higher) systems.

Appendix A — Networking

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1. INTRODUCTION

1.1 With system software version 7.0 and later you can connect two or more Inter-Tel® systems to form a network that provides an interface between the systems. To the user, the network appears as though it is one integrated system. With few exceptions, the user can perform all of the functions across the network that he can within a single phone system.

1.2 In this manual all references to a "network" mean two or more Inter-Tel systems connected by a network span. Each system in a network is called a "node."

1.3 There can be up to 99 nodes per network with a combined total of up to 40,000 devices on the nodes. The maximum capacities are:

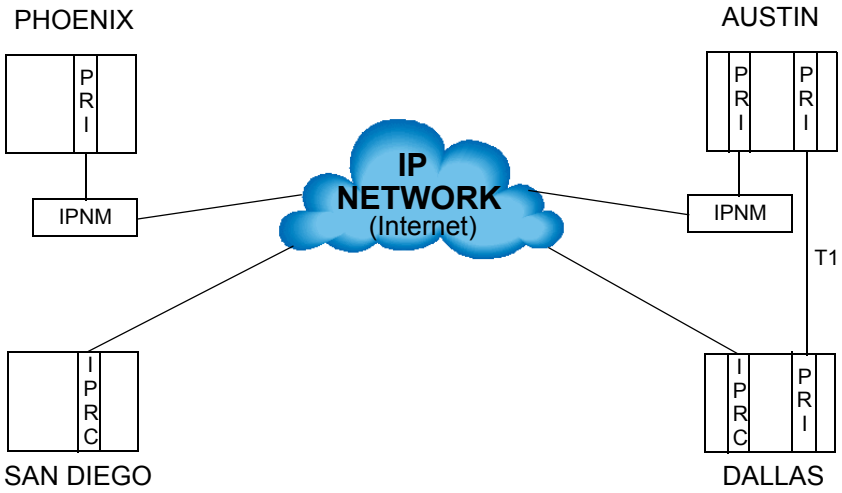
- 1792 devices on each node
- 25,000 entries in the database for off-node devices on each node
- 32 system serial ports on each node

1.4 The system supports several private networking configurations. Between any two endpoints, there can be any combination of IPRCs, T1/PRI or T1/E1/PRI cards, and IP Networking Modules (3200Ns).

- **T1 Connections:** Are used for connecting system nodes in a network. Each network span requires two T1/PRI or T1/E1/PRI cards, one for each end of the span. See page A-7 for more details.
- **IP Networking Module (3200N):** Is basically an extension of ISDN designs. It utilizes one channel of a primary rate interface (PRI) to transfer data in the form of high-level data link control (HDLC) packets containing all of the information necessary to process calls on the remaining channels. The Networking Module networking is derived from the Q.SIG standard (an international signaling standard for PBX-based private networks) and modified to accommodate the proprietary messages required by Telephone System networking. For more details, refer to *the Inter-Tel IP Networking Module Manual* (part no. 730.8001).
- **IP Networking using an IP Resource Card (IPRC):** System Versions 7.0 and later allow an IPRC to be programmed in the Database (DB) Programming to provide private networking using an IP network. See page A-9 for more details.

1.5 The figure on the next page illustrates an example of private networking using T1/PRI Cards, IPRCs, and IP Networking Modules.

Figure A-1. *A Private Network Combining IPRCs and T1/PRIs Modules*



1.6 This figure shows the three types of networking: IP Resource Card (IPRC), IP Networking Module (IPNM), and T1 connections (PRI). Each type can communicate directly with another node that uses the same type. For example:

- Phoenix and Austin can speak directly to each other via the IP Networking Module.
- Austin and Dallas can speak directly to each other via the T1 connection (PRI).
- San Diego and Dallas can speak directly to each other via the IP Resource Card (IPRC).

1.7 Nodes with different networking types can also communicate with each other, but not directly. To communicate with a dissimilar node, the call must be routed through a node with the same networking type. For example, if Phoenix needs to call Dallas, the call must first go through Austin. For Austin to speak with San Diego, the call must first go through Dallas. With proper planning and design of call routing patterns, an IP private network can be optimized such that no more than one hop is required between dissimilar nodes.

NOTE: ATM nodes can also be included in the private network.

2. PLANNING THE NETWORK

2.1 The most important step in setting up a network is planning. Without careful planning, the network will be difficult to program and maintain.

A. GATHER INFORMATION

2.2 As you read through this section, fill out the planning sheet on page A-33 as you answer questions such as these:

- How many systems (nodes) are in the network? How will they be numbered?
- Where are they located?
- What additional hardware will you need?
- *(For T1 connections)* What additional services need to be ordered? Does the local telephone company offer point-to-point clear channel connections to the areas for the desired nodes? Does the customer have existing cable/fiber/microwave, etc., connections that can be used? In many cases, the network connection can simply be a customer-provided cable. The maximum length of the customer-provided network cable should not exceed 6000 ft (1800 m). If more than 6000 ft (1800 m) of cable is required, some type of amplification, such as a repeater, is required.
- Does the customer already have a data network? It is very possible the customer already has an existing data network. Is it possible to use bandwidth from the existing data network for voice? Does the customer want to combine voice and data on the network? What type of drop and insert devices will be included in the network plan?

NOTE: For T1 connections, the System requires T1 channel 24 in any network span.

- How will the network be configured? How will the trunks connect to the network? Will all the trunks connect to one node or will the trunks be distributed?
- *(For IP connections using IP Networking Module)* Will an IP or frame relay device, such as the IP Networking Module, be used for routing calls to other nodes? This can substantially impact the spans between each node. Each span between nodes can carry a maximum of 23 [30 for Europe] simultaneous calls.
- Call patterns for each node should be determined to configure the appropriate number of channels necessary for the network connections. Remember to include incoming/outgoing intercom, incoming/outgoing central office, automated attendant, operator, voice mail, hunt group announcement and overflow messages, and Record-A-Call traffic in your calculations.

NOTE: *(For IP connections using the IP Networking Module)* No special programming is required when using IP or frame relay devices, such as the IP Networking Module, in place of T1 connections. Each remote node will still have programmed node trunk groups and off-node devices.

When planning the system, note that each node trunk group can have up to 23 [30 for Europe] trunks, thus limiting the number of incoming and outgoing calls. Make sure you have enough T1/PRI or T1/E1/PRI cards connected to the IP or frame-relay device(s) to support the customer's call load.

- What is the numbering plan for the extensions, Voice Processing applications, feature codes, hunt groups, node trunk groups, etc? Remember to leave room for expansion in your numbering plan.
- Does each node have a local trunk group for emergency calls?
- Which system in the network will have the Voice Processing Unit?
- How will calls ring in to the Voice Processing Unit? Will they be routed to other nodes?
- Will call routing tables and/or ARS route calls to other nodes?
- Which station(s) will be the network-wide primary attendant(s)? Which stations are the node attendants?
- Will SMDR include all network calls? Where will the output port(s) be located?
- Where will the message print port(s) be located?
- Where will fax machines, modems, and external paging ports be located?

B. DESIGN THE NETWORK

2.3 For optimum network operation, follow these guidelines:

- **Carefully manage the traffic on a central node.** The central node is susceptible to unintentional bottlenecks. Good rules of thumb for preventing bottlenecks on the central node are:
  - — If the node contains a Voice Processing Unit voice mail system with more than 16 ports and more than 128 Executive Display, Professional Display, or Model 8560 Phones, redistribute the load by moving some of the Executive Display, Professional Display, or Model 8560 Phones to another node, creating a new node, if necessary. Or, use the CP Server on the node.
  - — If networked node with a Voice Processing Unit with Automated Attendant and/or Call Routing Applications is expected to handle more than 1000 calls in an hour, use a separate Voice Processing Unit to provide the announcements. Or, use the CP Server on the node.
  - — If a node with a centralized voice mail supports more than four nodes or a total of 1500 off-node devices, redistribute the load by reducing the number of stations on the central node to 200 (adding another node, if necessary). Or, use the CP Server on the node.
  - — A node used as an outgoing call center should not have more than 100 agents. Redistribute the load among several nodes, if possible, or place the call center on its own node. Or, use the CP Server on the node.
  - — A node should not have more than five ISDN Primary Rate trunks for private networking or to the central office. Redistribute the load by moving ISDN trunks to another node, adding a node, if necessary. Or, use the CP Server on the node.
  - — For very busy systems or large networks, equip the satellite nodes with their own local and long distance trunks where possible.
  - — If the central node is used for all long distance calls, make sure it is equipped with enough trunks to handle the peak traffic.
- **Limit the maximum number of display phones to 355 per node.** Redistribute the load; adding another node if necessary. Or, use the CP Server on the node or upgrade to an ATM system.

- **Eliminate data errors whenever possible.** Currently, the system can experience slow-downs during error dumps. The most common preventable errors are:
  - — ***T1 errors:*** Among other things, ensure any Primary Rate Daughter Cards are Revision F or above or have ECN 5669 applied (EPROM 827.8894 installed in socket U4).
  - — ***Fiber errors:*** Ensure all PCM-F cards are revision M or above or have ECN 5574 applied.
  - — ***Glare:*** Ensure the Inter-Tel system and central office are programmed to minimize glare by initiating outgoing calls on the opposite end of a trunk groups from the starting point for incoming calls. Even when configuring in this manner, glare is possible when the system gets very busy, but the likelihood of a glare condition is greatly reduced.
- **Turn off diagnostics except when working with Engineering or Technical Support to troubleshoot a problem.** Diagnostics require computer resources and they can adversely affect the speed of heavily loaded nodes.
- **Ensure enough T1 spans are installed between nodes to handle the traffic.** In most cases, a single T1 span between nodes will suffice. However, if lots of calls are anticipated between two nodes, add another T1 span to handle the traffic.
- **Avoid doing DB Programming during peak business hours.**
- **Avoid using System OAI on a heavily loaded node.** Redistribute the load by moving traffic not associated with the System OAI application to another node (adding a node, if necessary). Or, use the CP Server on the node.
- **When using Paging:**
  - — Avoid using a lighted DSS/BLF Button for the main page zone (e.g., 9600 or 9601).
  - — Limit page zones to 60 or fewer stations for customers who need to make more than 30 pages per hour. Use external paging whenever possible. Or, use the CP Server on the node.
- **Avoid overloading all-ring hunt groups.** Avoid all-ring hunt groups with ten or more stations that are expected to handle more than 500 calls in an hour. Use linear or distributed ringing instead of all-ring. Or, use the CP Server on the node.

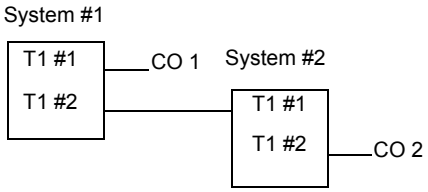
NOTE: A CP Server is required to support off-node members of an all-ring hunt group.

- **Avoid large hunt groups.** Avoid creating node with hunt groups containing more than 100 members. Distribute the hunt groups over several nodes, if possible. Or, use the CP Server on the node.
- **When using an ATM Switch:**
  - — Make sure the chassis within the node do not overlap time zones. The ATM Switch cannot synchronize across time zones.
  - — Make sure the chassis within the node do not overlap area codes unless the area codes are considered local. If there is an overlap with local area codes, enable ten-digit dialing for that node.

3. T1 NETWORKING

3.1 A T1 connection is used for connecting two system nodes in a network. Each network span requires two T1 or T1/E1/PRI cards, one for each end of the span. The cards require the most recent firmware to support networking (use upgrade kit part number 828.1506). Point-to-point clear channel T1 spans are used to connect the nodes. For each network span, order a clear channel, point-to-point T1 span from the service provider (not PRI service, even though you will be using T1/PRI or T1/E1/PRI cards). The following figure illustrates a sample T1 connection.

Figure A-2. *Sample T1 Connection*



3.2 In a T1 network that involves multiple CO and T1 card connections, the line buildout is determined by each connection. In the example shown above, the two T1 cards connected to the CO receive reference clock from the public network and are slave clocks. The two that are connecting the Inter-Tel systems do not draw reference from the public network. T1 #2 on system #1 is the master clock and T1 #1 on system #2 is a slave clock.

A. HARDWARE INSTALLATION

3.3 To install a network, you will need the following:

- **IP Networking Software License:** Each node in the network must have a software license that supports the networking premium feature.

> **NOTE:** Nodes that are running version 7.0 or earlier require a specific networking PAL. See Replacement Parts for a complete list of software PALs.

- **T1 Service:** For each network span, order a clear channel, point-to-point T1 span from the service provider (not PRI service, even though you will be using T1/PRI cards).
- **T1/PRI or T1/E1/PRI Cards:** Each network span requires two cards, one for each end of the span. The cards require the most recent firmware (use upgrade kit part number 828.1459).
- **CPU Cards:** CPU cards should be upgraded using the proper upgrade kit. The kit part numbers are:

Table A-1. *CPU Card Kit Part Numbers*

CPU Card	Kit Part #
CPU020/PCM-X	828.1456
CPU020/PCM-D	828.1457
CPU020/PCM-Q	828.1458

> **NOTE:** You do not need to upgrade the CPC.

APPENDICES

3.4 Installation recommendations include the following:

- Each node ***must*** have at least one central office trunk. Emergency calls should be programmed to use the local trunk.
- If the Voice Processing Unit is shared by multiple nodes, all of the nodes should have access to the node where the Voice Processing Unit is located, with a minimum number of hops through other nodes. This will have significant impact on the network layout.

3.5 The network does not allow one node to use another node's resources (DSP resources, serial ports, etc.) with the exception of external page ports, SMDR output ports, and message print output ports.

B. T1 PROGRAMMING

NOTE: System software v9.0 and later provides a networking wizard to simplify the required programming for T1/E1 PRI networking. See page page 6-87 for more information about the networking wizard.

3.6 T1 programming, described on page 6-126, has a field called "ISDN Switch Type." In addition to the ISDN switch types that the system supports (AT&T 5ESS Custom, AT&T 4ESS Custom, DMS-100, National ISDN 2, Private Networking, and IP Private Networking). To program a T1/PRI or T1/E1/PRI card for networking, use the ISDN Switch Type list box to select Private Network.

3.7 When a T1 card is programmed for networking (Private Network switch type), the system automatically creates a node trunk group. (All B-Channels residing on the same card are in the same node trunk group.)

3.8 The system allows up to eight node trunk groups per node (one for each T1/PRI or T1/E1/PRI card).

4. IP NETWORKING USING THE IPRC

4.1 As of v7.0, an Internet Protocol Resource Card (IPRC) can be programmed to provide private networking using an IP network.

NOTE: When using the networking feature in systems running software version 7.0 and later, to avoid one-way audio, do not use two IPRCs on separate networks.

A. HARDWARE INSTALLATION

4.2 To install a network, you will need the following:

- **IP Networking Software License:** Each node in the network must have a software license that supports the networking premium feature.

  NOTE: Nodes that are running version 7.0 or earlier require a specific networking PAL. See Replacement Parts for a complete list of software PALs.

- **IPRC and IPRC-D Cards:** The IPRC provides up to 16 ports, and the IPRC-D provides up to an additional 16 ports (32 ports total). For v1.x firmware IPRCs, the number of available ports depends on the PAL installed, as listed in the following table.

Table A-2. *IPRC/IPRC-D PALs*

PAL (P/N)	IPRC/IPRC-D	Available Ports
4 port (827.9595)	IPRC	4
8 port (827.9596)	IPRC	8
16 port (827.9597)	IPRC	16
24 port (827.9598)	IPRC	16
	IPRC-D	24
32 port (827.9599)	IPRC	16
	IPRC-D	32

For v8.x and later firmware IPRCs, the number of available ports depends on the software license, as listed in the following table.

Table A-3. *IPRC Software Licenses*

License (P/N)	IPRC/IPRC-D	Available Ports
4 port (840.0115)	IPRC	4
8 port (840.0116)	IPRC	8
16 port (840.0117)	IPRC	16
32 port (840.0118)	IPRC	16
	IPRC-D	32

An IPRC requires 0.34 amps and an IPRC-D requires an additional 0.14 amps (0.48 amps total). See page 3-155 for details on the IPRC-D.

- **CPU Cards:** CPU cards should be upgraded using the proper upgrade kit. The kit part numbers are listed in the following table.

Table A-4. *CPU Card Kit Part Numbers*

CPU Card	Kit Part #
CPU020/PCM-X	828.1456
CPU020/PCM-D	828.1457
CPU020/PCM-Q	828.1458

NOTE: You do not need to upgrade the CPC.

- **Firmware Configuration:** You must configure all IPRC settings, other than those specified by the PAL, using the DB Programming interface. An IPRC does include web-based and telnet configuration interfaces. However, the system does not propagate changes made using these interfaces to call processing, and call processing may quickly override such changes. Therefore, you should only use these interfaces for troubleshooting purposes when call processing is not communicating with the IPRC. When you attempt to save any configuration changes using these interfaces, the IPRC warns you that those changes will be overwritten the next time the card communicates with call processing. The firmware operation is as follows:
  - **Companding Type:** An IPRC transmits audio data using the system-wide companding type.
  - **DTMF:** When DTMF tone is detected, the system uses the DTMF encoding settings dictated by DB Programming (see page 6-154 for more information about the DTMF encoding settings).
  - **Minor and Major Resets:** When an IPRC encounters a software fault (minor reset), the system attempts to reconnect all conversations and data connections. After a major reset, the system cannot re-establish existing connections.

4.3 Installation recommendations include the following:

- Each node **must** have at least one central office trunk. Emergency calls should be programmed to use the local trunk.
- If the Voice Processing Unit is shared by multiple nodes, all of the nodes should have access to the node where the Voice Processing Unit is located, with a minimum number of hops through other nodes. This will have significant impact on the network layout.

4.4 The network does not allow one node to use another node's resources (DSP resources, serial ports, etc.) with the exception of external page ports, SMDR output ports, and message print output ports.

B. IPRC PROGRAMMING

NOTE: Prior to v9.0, the 8.x firmware IPRC could be used for private networking **or** for IP devices, as determined by the configuration chosen in DB Studio. The v9.0 and later firmware IPRC supports **both** private networking and IP devices on the same card.

ALSO: System software v9.0 and later provides a networking wizard to simplify the required programming for private networking. See page 6-87 for more information about the networking wizard.

4.5 The IPRC relies on a unique numbering plan to make connections across the network. This means that you must assign each IPRC in the network a unique extension that the other IPRCs must be able to recognize. For this reason, Inter-Tel strongly suggests that you use the P6XXY numbering convention, where *XX* is the node number and *Y* is the IPRC number. For example, if the IPRC is the second IPRC on node 1, the extension number would be P6012. It is also recommended that you complete the IP Networking Planning Sheet (available at the end of this appendix) before you begin programming the IPRC. If you do not follow a numbering convention and identify the extensions before you program the IPRCs, it will be difficult to successfully set up your network. (See page A-13 for a programming example.)

4.6 If using an IPRC for private IP networking, follow the guidelines below. See the reference pages for details.

NOTE: IP Networking Planning sheets are included at the end of this section for your convenience.

1. Program the equipment chassis slot where the IPRC is installed as an **IPRC** (see page 6-145).
2. Choose a "P6XXY" extension where *XX* is the node number where the IPRC resides and *Y* is the IPRC number.

   NOTE: The extension you assign to each IPRC must reflect the node where the IPRC resides. If it does not, it will be difficult to establish the IP connections between IPRCs on a multi-node system. This limits the number of IPRCs configured for private networking within a private network to 1000.

3. **Program the Voice Resources:** Program one or more voice resources as a *Networking* VoIP resource (see page 6-146).

   NOTE: While in local mode, DB Programming allows you to program up to 32 voice resources, regardless of the number of resources actually installed or the PAL configuration. The system ignores programming for voice resources rendered invalid by the PAL configuration.

4. **Program the IP Connection:** The IPRC screen contains a link to the IPRC's IP connection. Program the IP connection as described on page 6-172. Whenever the system creates an IP connection, the system broadcasts the IP information to the other nodes as a database update, and the new IP connection becomes an off-node device on the other nodes (provided the remote node does not block the database change such as when a DB Programming session is active on the remote node).

NOTE: This does not automatically configure the IPRC for use by the network. You must manually program networking parameters locally as well as on each remote node.

5. **Program the IP Connection for the remote node:** You can also create IP connections to represent IPRCs on other nodes within the network. To see these off-node IP connection fields, you must have a node created in System\Devices and Feature Codes\Nodes.

NOTE: Each node in an IP network must contain an off-node IP connection for all other IP nodes in the network. For example, in a three-node network, node 1 must have an off-node IP connection for nodes 2 and 3; node 2 must have off-node IP connections for nodes 1 and 3; and node 3 must have off-node IP connections for nodes 1 and 2. Nodes that do not have the correct off-node IP connections programmed may not be able to communicate with each other via IP.

6. Create an off-node IP connection in the IP Connections field.
7. Choose a "P6XXY" extension for the off-node IP connection. Again, you must assign a unique extension to each IP connection within the network.

NOTE: When creating off-node connections and other IP-related extension numbers, Inter-Tel recommends using a numbering plan that associates the extension to the device and the node on which it resides. For example, the first IPRC on node 2 would be P6021 (P6 followed by the node number, then the IPRC number). The second IPRC would be P6022, etc.

8. Program a remote IP address, remote audio receive port, and remote listening port in the IP Connection screen (see page 6-177).
9. **Program the Node IP Connection Group:** The off-node IP connection screen also contains a link to the Node IP Connection Group field to which the IP connection belongs. The system automatically creates a node IP connection group for each remote node you program. A node IP connection group corresponds to an IP network connection between the local node and a remote node. Program the Node IP Connection Group field as described on page 6-176.

NOTE: Changes to the Node IP Connection Group parameters do not take effect until the next call. Changes do not affect existing calls.

C. IP NETWORKING PROGRAMMING EXAMPLE

4.7 This section provides an example of how to set up a network using the IPRC based on the information provided in the following tables.

Table A-5. *Programming for Node 1*

Description	Extension	IP Address
Local IP Connection	P6010	192.168.0.1
Remote IP Connection (N2)	P6020	192.168.0.2
Remote IP Connection (N3)	P6030	192.168.0.3

Table A-6. *Programming for Node 2*

Description	Extension	IP Address
Local IP Connection	P6020	192.168.0.2
Remote IP Connection (N1)	P6010	192.168.0.1
Remote IP Connection (N3)	P6030	192.168.0.3

Table A-7. *Programming for Node 3*

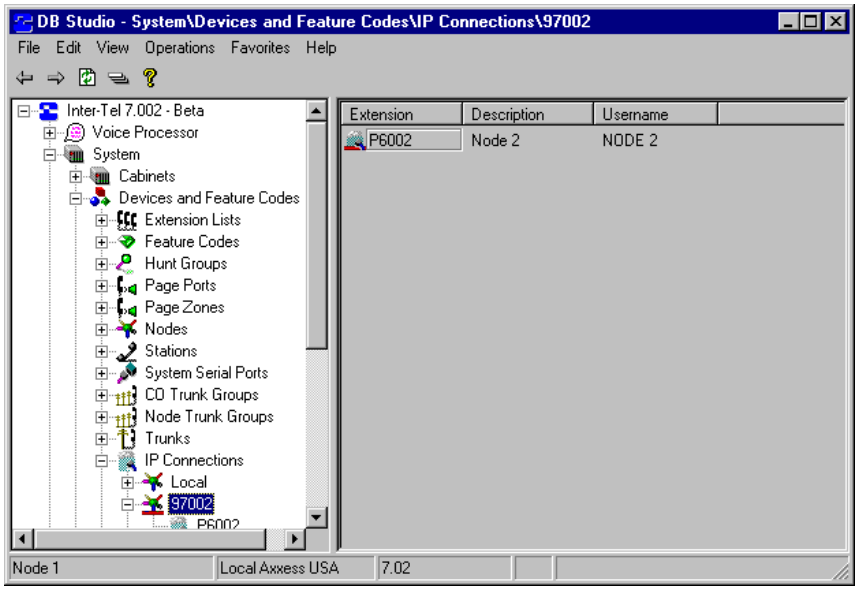
Description	Extension	IP Address
Local IP Connection	P6030	192.168.0.3
Remote IP Connection (N2)	P6020	192.168.0.2
Remote IP Connection (N1)	P6010	192.168.0.1

4.8 To program the database:

NOTE: System software v9.0 and later provides a networking wizard to simplify the required programming for private networking. See page 6-87 for more information about the networking wizard.

1. Under System\Cabinets, add and equip an IPRC (see page 6-145).
2. When prompted, enter the P6XXY extension (i.e., P6010) for the local node.
3. Under System\Devices and Feature Codes\Nodes, create all of the remote nodes (see page 6-190). Program the Description and Username for each node.
4. Under System\Devices and Feature Codes\IP Connections\Local\P6XXY, program the Static IP Address (i.e., 192.168.0.1), Static Subnet Mask, and Static Gateway Address fields (see page 6-172).

5. Under System\Devices and Feature Codes\IP Connections\97XXX, create an off-node IP connection for all connections available on that remote node. Program the Description and Username for each. Make sure the extension numbers match the unique values assigned at each node. For example, if creating an off-node IP connection for Node 2, the extension number must be P6020.



6. Double-click on the remote IP extension and program the Remote IP Address for the remote node (i.e., 192.168.0.2).

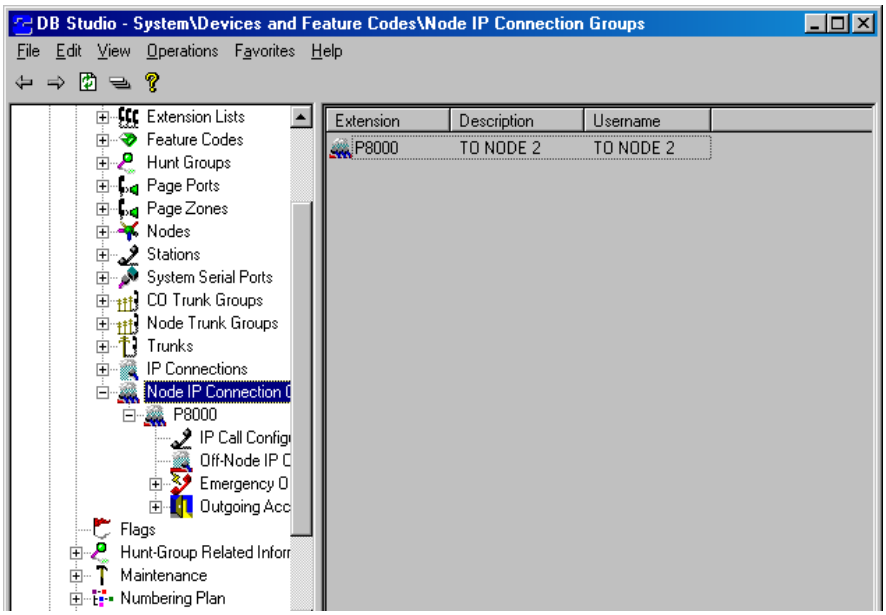
7. Verify that the remote IP connection is located under the Node IP Connection Group folder. To do this:

   a. Double-click on the remote node's extension under System\Devices and Feature Codes\IP Connections. There should be a P8XXX extension listed under the Node IP Connection Groups field.

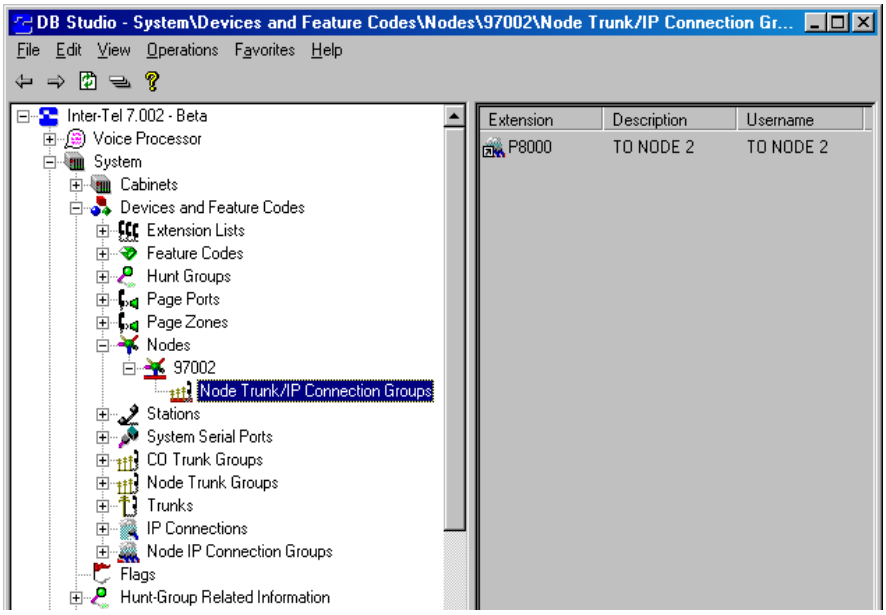
   b. Double-click on the P8XXX extension to access the folder for that node IP connection group.

   c. Double-click on the Off-Node IP Connections field to display the remote nodes programmed for that IP connection group.

8. Under System\Devices and Feature Codes\Node IP Connection Groups, enter a description and username for the each P8XXX extension. Use a description/username that identifies where the off-node extensions reside. For example, if P8000 includes the off-node IP connection extension for connecting to Node 2, use TO NODE 2 as the description/username for P8000.



9. Under System\Devices and Feature Codes\Nodes\97XXX\Node Trunk/IP Connection, add the Node IP Connection Group extension (P8XXX) to the appropriate node (i.e., the remote node created in step 1).



10. Repeat steps 1-9 for all nodes that will be networked via the IPRC.

5. EXTENSION NUMBERS

5.1 Each node has its own extension and feature code programming. However, the network should have a universal numbering plan so that extension numbers on the various nodes do not overlap and do not conflict with feature codes. That is, when planning the extension numbers for each of the nodes in the network, set aside a block of extension numbers (for stations, hunt groups, Voice Processing applications, etc.) for each node.

A. PROGRAMMING OFF-NODE EXTENSION NUMBERS

5.2 You can give users access to station, hunt group, Voice Processing application, and page port extensions on other nodes in two ways:

- You can manually program off-node devices (extensions from other nodes) into the database of the node you are programming. To give a node access to an off-node device, program the extension number, username, description, device type (station, hunt group, Voice Processing application, or page port), and node identification number. The network uses the username and descriptions for directory features only. If you define an extension conflict, DB Programming does not allow you to exit the screen until you correct the conflict.
- You can import and export new extensions as you create them. When you add a new extension, DB Programming automatically makes the new extension visible to the other nodes in the network. Or you can manually export local devices to other nodes (or import devices from other nodes) using the Export/Import option in DB Programming. The network broadcasts the new extension number (along with the username, description, device type, and node identification number) to the other nodes in the network and places it in their off-node device lists. If there is an extension conflict on another node, the node with the conflict ignores the new extension in favor of the previous extension number. Also note that if the network is unable to route the broadcast message to a node, that node will not be able to receive and re-broadcast the new information and it must be entered manually. (If Message Print is enabled, error messages will indicate any unsuccessful broadcasts.)

5.3 Each node can have only 8000 off-node device entries in its database. After that limit has been reached, new devices received through network broadcasts cannot be added to the node's database. You can make room in the database for new extensions by setting up wildcard extensions as described below.

5.4 If you change a device's extension, username, or description, the network automatically broadcasts that extension change to the entire network and updates any existing off-node devices on the other nodes. If the network is unable to export or import an extension because there is an active programming session on that node, the node is unable to communicate with its Voice Processing port, the node is down, or the links to the node are down, the new extension will not be added to or changed on that node. You must manually add or change the new extension in the node's database. You may want to check each node to verify that their off-node device lists are programmed properly to allow access between the nodes.

5.5 Nodes are assigned extension numbers that users can dial to reach stations which are not programmed as off-node devices on their node. Also, if the same extension number exists on two nodes, the only way a user can call the extension on the other node is to dial a node number before dialing the extension number. For example, if extension 1200 exists on both Node 1 and Node 2, users on Node 1 must first dial the node number for Node 2 to before dialing extension 1200 on Node 2. (Note that each node trunk group also has an extension, but the number cannot be dialed by users.)

B. WILDCARD CHARACTER X IN OFF-NODE EXTENSIONS

5.6 You can use the wildcard character X in extension numbers to represent "any digit." This allows you to include a range of extensions as one off-node device entry. Wildcard extensions can only be used for off-node devices.

5.7 Wildcard extensions are made up of digits (1-9), followed by wildcard digits X. Valid wildcard extensions include 1XXX (range of 1000 to 1999), 14XX (range of 1400 to 1499), 7X (range of 70 to 79). For example, if 14XX is a wildcard extension, users can then dial a 14XX number, such as 1433, and be connected to the off-node device.

5.8 There are some limitations to using wildcard extensions:

- There will be no entry for wildcard extensions in the IC Directory.
- DSS buttons cannot be programmed for the individual devices the wildcard extension is representing. For example, if the off-node device 4:14XX exists, the user will now be restricted from programming extensions 1400 through 1499 as a DSS/BLF button. As a result of this action, the system will display an error message INVALID EXTENSION NUMBER and then re-prompt for another button.
- Voice Processing cannot create extension IDs or associated mailboxes for stations included in wildcard ranges. Each associated station must have its own off-node device entry.
- Wildcard extensions cannot be used for Call Routing destinations. For example (using the 4:14XX case above), the user will now be restricted from using unconditional forward feature code (355) and setting the destination to extensions 1400 through 1499. As a result of this action, the system will display an error message FORWARD TO INVALID DEST and re-prompt for another destination.

6. NODE PROGRAMMING

6.1 Each node is programmed individually and contains its own database. Only one DB Programming session (including administrator DB Programming) may be active on any single node at a time. However, several nodes can have DB Programming sessions active throughout the system. ***Note that simultaneous sessions on different nodes can cause conflicting database changes, but they also allow the flexibility of easier system administration.***

6.2 Each node is given a unique identification number (1-99). The programmer is responsible for insuring that each node's identification number is unique and that all of the nodes agree with respect to their node identification numbers.

6.3 A **node** contains a list of node trunk groups used to access other nodes. For each node in the network, you must define the routes to every other node. For example, in a network with four nodes, you would define three routes for each node (one to each of the other three nodes).

6.4 Each node contains:

- An optional username
- An optional description
- The identification number of the remote node
- A list of the node trunk groups used to establish calls and communications to the node

> **NOTE:** If desired, you can assign the same node trunk group to multiple nodes. This is a common setup when using IP or frame relay devices, such as the IP Networking Module. However, note that each node trunk group can have up to 23 trunks, thus limiting the number of incoming and outgoing calls. Make sure you have enough T1/PRI or T1/E1/PRI cards to support the customer's call load.

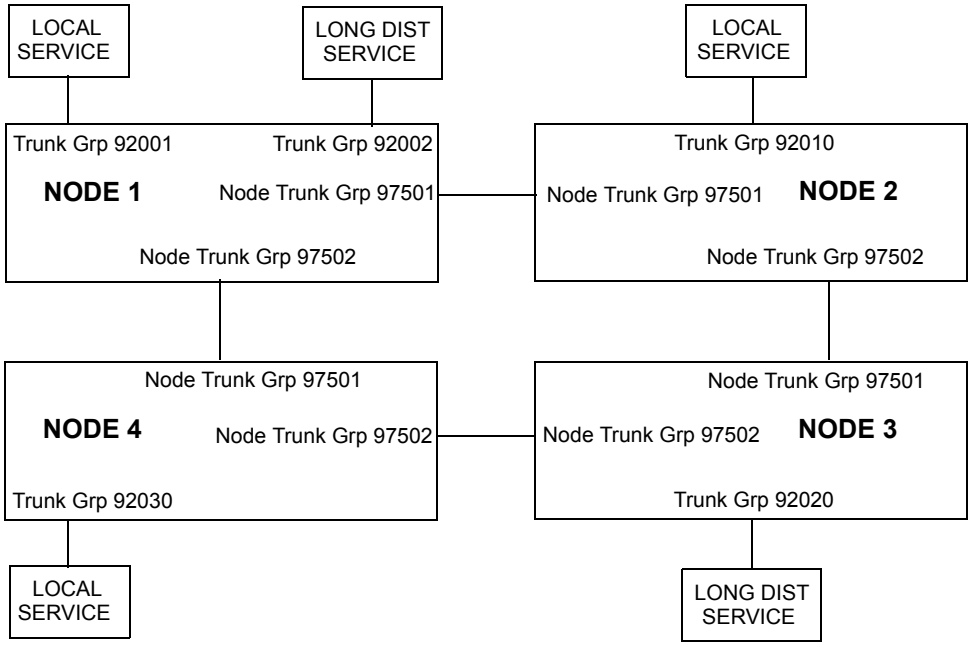
- A search algorithm that determines whether the node trunk groups are accessed in linear or distributed order:
  - — **Linear:** The node first attempts to route through the first node trunk group listed in the route. If the node is unable to route through that node trunk group, it attempts to route through the second node trunk group listed in the route. The node continues to attempt to route through subsequent node trunk groups listed in the route until it successfully routes or exhausts all node trunk groups listed in the route.
  - — **Distributed:** The node shifts the first node trunk group it attempts to use. For example, if the node routed the previous call or communication through the first node trunk group listed in the route, the node routes the second call or communication through the second node trunk group listed in the route.

6.5 If an off-node device is not programmed on the local node for a station on a remote node, users can still call that device by first dialing the node number for the station's node and then the station's extension number. For example, in the sample shown on the next page, a caller on Node 1, calling extension 1100 on Node 2, could dial Node 97002 and then 1100 to reach the station on Node 2.

A. SAMPLE NODE PROGRAMMING

6.6 The programmer must define the routes to every node in the network to give the system the information necessary to route calls.

6.7 Each node trunk group (and its B-Channels) in a route group must connect the node to another node in the network. However, that node does not need to be the destination node. For example, consider the following network:



6.8 Using the map above, the programmer can determine the necessary information about each route to every node in the network, as shown in the chart on the next page.

Table A-8. *Programming for Node 1*

Node Route Grp #	Node #	Description	Node Trunk Grps	Search Type
NRG 02 97002	2	To Node 2	97501, 97502	Linear
NRG 03 97003	3	To Node 3	97501, 97502	Distributed
NRG 04 97004	4	To Node 4	97502, 97501	Linear

Table A-9. *Programming for Node 2*

Node Route Grp #	Node #	Description	Node Trunk Grps	Search Type
NRG 01 97001	1	To Node 1	97501, 97502	Linear
NRG 03 97003	3	To Node 3	97502, 97501	Linear
NRG 04 97004	4	To Node 4	97501, 97502	Distributed

Table A-10. *Programming for Node 3*

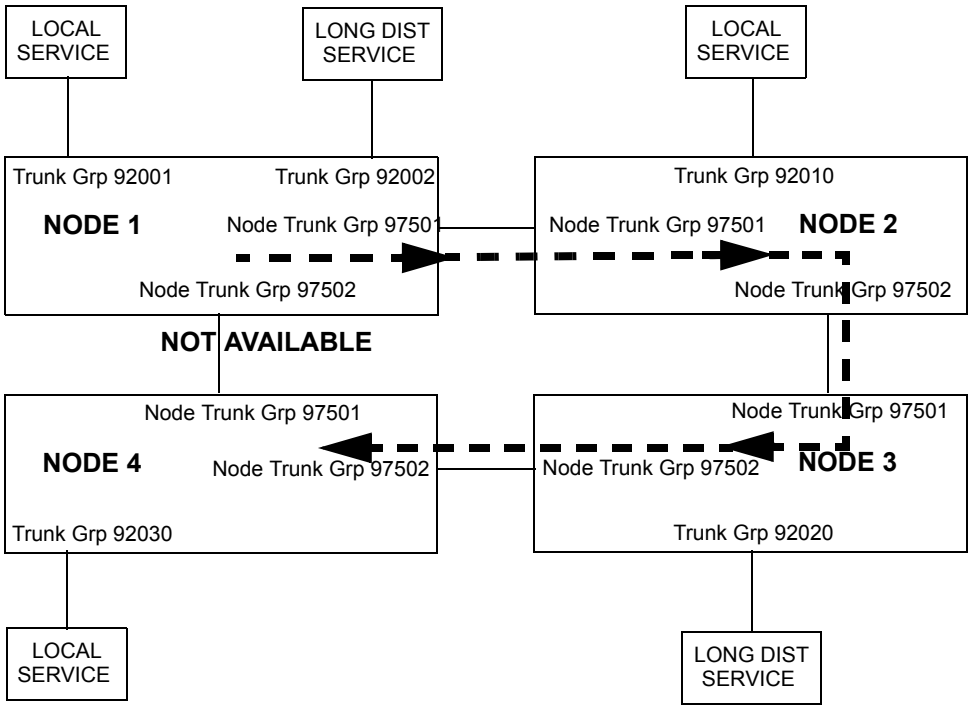
Node Route Grp #	Node #	Description	Node Trunk Grps	Search Type
NRG 01 97001	1	To Node 1	97501, 97502	Distributed
NRG 02 97002	2	To Node 2	97501, 97502	Linear
NRG 04 97004	4	To Node 4	97502, 97501	Linear

Table A-11. *Programming for Node 4*

Node Route Grp #	Node #	Description	Node Trunk Grps	Search Type
NRG 01 97001	1	To Node 1	97501, 97502	Linear
NRG 02 97002	2	To Node 2	97501, 97502	Distributed
NRG 03 97003	3	To Node 3	97502, 97501	Linear

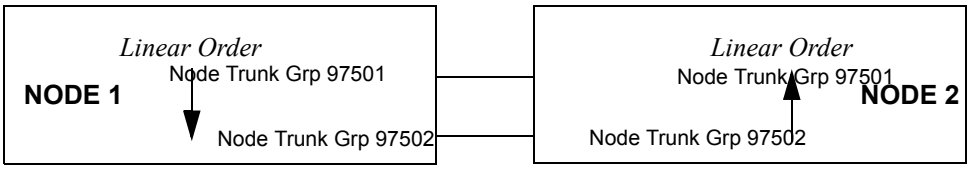
6.9 To prevent circular or excessive routes, the system limits the number of "hops." A call or communication can take up to 10 hops to other nodes. (However, 2-3 hops maximum is recommended.) Using the sample shown on the previous page, here is an example of a call that makes three hops through intermediate nodes to reach its destination (the path is also shown in the diagram below):

1. A call is placed from Node 1 to Node 4.
2. Node 1 first attempts to route through Node Trunk Group 97502, but that route is not available. So Node 1 routes the call using Node Trunk Group 97501 and uses Nodes 2 and 3 as intermediate nodes.
3. Because the network has already routed the call through Node 1, Node 2 will not use Node Trunk Group 97501 and routes the call through its Node Trunk Group 97502.
4. Node 3 routes the call through its Node Trunk Group 97502 to Node 4. (It will not use 97501 because the call came from Node 2.)



6.10 Determining in what order to list the Node Trunk Groups in a Node and when to use Linear or Distributed search type depends on the system configuration and traffic.

6.11 For example, **Linear** search type is the preferred method for avoiding glare between two nodes that have two or more connections. Place the Node Trunk Groups in opposite order in the Nodes on each side of the connection and use Linear search type, as shown below.



6.12 Programming For Node 1:

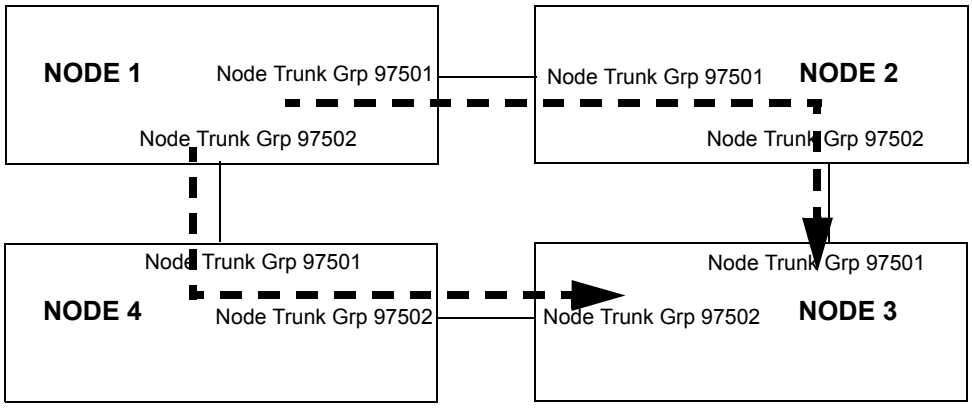
Node Route Grp #	Node #	Description	Node Trunk Grps	Search Type
NRG 02 97002	2	To Node 2	97501, 97502	Linear

6.13 Programming For Node 2:

Node Route Grp #	Node #	Description	Node Trunk Grps	Search Type
NRG 01 97001	1	To Node 1	97502, 97501	Linear

6.14 In the above example Node 1 uses Linear searches to send calls first through Node Trunk Group 97501 and then through Node Trunk group 97502. Node 2 reverses the order and begins searching with Node Trunk Group 97502. With calls beginning at opposite ends of the trunk lists, it is less likely that both nodes will try to seize the same trunk and cause glare.

6.15 Distributed search type is best for balancing traffic when there are multiple-hop paths between nodes. In the example shown below, the path to Node 3 from Node 1 has two hops using either Node Trunk Group. Therefore, the route from Node 1 to Node 3 should be Distributed to balance the call traffic.



6.16 Programming For Node 1:

Node Route Grp #	Node #	Description	Node Trunk Grps	Search Type
NRG 03 97003	3	To Node 3	97501, 97502	Distributed

B. NODE PROGRAMMING PROCEDURES

6.17 Programming a node involves several steps. You should program the network in the following order:

1. Local Node Number
2. Nodes
3. T1/PRI or T1/E1/PRI Cards
4. Node Trunk Groups
5. Node Trunk Lists
6. Off-Node Devices

Local Node Number

6.18 To program a node's local node number and description:

1. Make sure the software license supports the required premium features, including Advanced CO Interfaces and Automatic Route Selection.

   NOTE: System versions prior to 7.0 require the appropriate number of premium feature units.

2. In System\Devices and Feature Codes\Nodes, program the local node number and the local node description. This causes the system to reset, and you will have to re-establish the DB Programming connection.

Nodes

6.19 To program each of the nodes:

1. In System\Devices and Feature Codes\Nodes, create the Node.
2. Set the remote node number to the destination node.
3. Name the node (for example To Node X).
4. Repeat for all nodes.

T1/PRI or T1/E1/PRI Cards

6.20 To program the T1/PRI or T1/E1/PRI cards:

1. In Cabinets, program the card slots and then program B-channels and extension numbers for the circuits.
2. Select the card to program. Then set the ISDN Switch Type to Private Network. This will automatically place the trunks on this card in a node trunk group.
3. Set the Remote Node Number to the node where the T1 span will be connected. (This span can also be used to route calls to other nodes, through the remote node, if the remote node is properly programmed.)

NOTE: If using an IP or frame relay device, such as the IP Networking Module, the Remote Node Number can be set to any valid node number. The device will use this field for identification purposes only.

Node Trunk Groups

6.21 To program the node trunk groups:

1. In System\Devices and Feature Codes\Node Trunk Groups, select the node trunk group.
2. Name the node trunk group, set outgoing access assignments.
3. Repeat for the remaining node trunk groups.

Node Trunk Lists

6.22 To add the node trunk groups to the node's list:

1. In System\Devices and Feature Codes\Nodes, select the remote node.
2. Select Node Trunk Groups and add the node trunk groups.
3. Program the Search Algorithm.
4. Repeat for each node.

Off-Node Devices

6.23 There are two methods for programming off-node devices, as described on page A-16. The automatic broadcast method works only for newly created stations. Any extensions that existed before the nodes began communicating must be exported (or imported) to other nodes using the Export/Import button in DB Programming or programmed as "off-node devices" manually at each of the other nodes.

7. NETWORK OPERATION OF SYSTEM FEATURES

7.1 When the phone system is programmed for use in a network the following features are affected:

- **Administrator Feature Codes:** Administrator stations are explained in detail on pages 26 and 273 in FEATURES and the feature codes are listed on page 5-21. See page 6-209 in PROGRAMMING for information about creating administrator stations. These administrator feature codes apply in network settings:

  — **Clear Network Alarm:** Administrators who receive network-wide alarms can clear them. The default feature code for the clear network alarm feature is 9851. (See page A-32 for more information on network alarms and page 5-301 for procedures.)

  — **Enable Network Night or Day Mode:** Administrators can place any or all remote nodes into night mode or day mode. The default feature code for Enable Network Night is 9861. The default feature code for Enable Network Day is 9862.

  NOTE: The Night Ring On/Off feature code (9860) affects only the node on which the administrator resides. (See page 5-273 for procedures.)

  — **Set Network Date And Time:** Administrators can synchronize the time of day, date, and day of week across the network. The default feature code for the synchronize network time of day feature is 9810. Note that administrators can still set the time of day, date, and day of week on their local node. (See page 5-275 for procedures.)

  — **Synchronize Network Time:** Administrators can synchronize the minutes past the hour across the network without changing the hour. This is useful when the nodes are in different time zones. The default feature code for the synchronize network minutes past the hour is 9811.

  NOTE: If a node's time is off by more than 30 minutes, synchronizing the minutes may cause the hour to change. Also, network time is automatically synchronized every day at 12:30 AM (00:30), using the time setting on the node with the lowest number. (See page 5-276 for procedures.)

  — **Network Freeze/Unfreeze Zone:** Administrators can perform a history freeze or unfreeze on any freeze zone. The default feature code for Network Freeze is 9939 [9139 in Europe] and Network Unfreeze is 9989 [9189]. (See page 5-303 for procedures.)

- **Administrator DB Programming:** The administrator can change extensions and usernames of only those users residing on the same node as the administrator. The network prohibits the administrator from programming an extension conflict. The network automatically broadcasts changes to extensions and/or usernames made by an administrator to the other nodes in the network. (See page 5-276 in FEATURES for a detailed explanation of DB Programming using an administrator's phone.)
- **Attendant Stations:** An attendant does not need to reside on the same node as the stations it serves. (See page 5-27 for a detailed description of attendant stations.)
- **Node Attendants:** There can be a primary attendant for each node. (See page 5-27 for a detailed description of attendant stations.)

- **Network Primary Attendant:** There can be a network primary attendant programmed on each node. (There is usually one Network Primary Attendant shared by all of the nodes.) When the network needs to direct a call to an attendant, it attempts to direct the call to the network primary attendant first. If the network primary attendant is unavailable because the network is unable to route the call, it directs the call to the attendant on the node where it originated. (See page 5-27 for a detailed description of attendant stations.)
- **Extension Numbers:** When an extension number is changed or deleted, the network broadcasts the extension change to the other nodes in the network, and the network updates all off-node device lists appropriately. If the network is unable to export or import an extension because there is an active programming session on that node, the node is unable to communicate with its Voice Processing port, the node is down, or the links to the node are down, the new extension will not be added to or changed on that node. You must manually change or delete the extension from the node's database.
- **Extension Lists:** Extension lists are limited to the stations or trunks on a single node. (See page 5-30 for an explanation of extension lists.)
- **Hunt Groups:** UCD hunt group members can be off-node devices. ACD hunt group members can also span nodes. Supervisors, however, cannot span nodes. All versions support supervisors, recall destinations, announcement stations, and overflow stations on other nodes. (See page 5-32 for a description of hunt groups.)
- **Station Monitor:** Because supervisors must reside on the same node as the hunt group, station monitoring can not occur across nodes. (See page 5-40 for more information about hunt group supervisors and station monitoring.)
- **Trunk Groups:** See page 5-87 in FEATURES and page 6-237 in PROGRAMMING for trunk group information.
  - **Trunk Lists:** All trunks in a trunk group must reside on the same node as the trunk group.
  - **Outgoing Access:** Because the network does not allow users to directly access trunks on other nodes, each trunk group's Outgoing Access must contain only stations on the local node. Users must use ARS to access trunks on other nodes. After an ARS call leaves the originating node, the network ignores outgoing access restrictions.
  - **Ring-In Assignment:** A trunk can ring in to a station or Voice Processing application on another node.
  - **Allowed-Answer Assignment:** Stations cannot have allowed-answer assignment for trunk groups on other nodes.
- **Caller ID [CLIP], DNIS, and ANI:** The network retains trunk and outside party information and passes that information along when a call is routed from node to node. (See page 5-72 for a complete description of this feature.)
- **Call Routing Tables:** Call routing tables can be used on a network. (See page 5-89 in FEATURES and page 6-237 in PROGRAMMING for more information.)
  - You cannot route a call from one Call Routing Table to a Call Routing Table on another node. When a Call Routing Table sends a call to another node, it cannot be routed to any additional nodes.
  - Trunks that ring in to call routing tables can only ring in to tables that are located on their local node. They cannot ring in to a call routing table on another node.

— The network attempts to optimize routes so that calls do not route through a node more than one time (looping). However, the network does not have the ability to dynamically reroute. That is, if a node trunk group that was previously unavailable becomes available after the network has already chosen another route, the network does not reroute using that node trunk group.

— The network will try to optimize routes when calls have been transferred or reverse transferred across several nodes (possibly forming loops).

- **Automatic Route Selection (ARS):** Each node has its own ARS programming. All trunk groups and nodes in a facility group must reside on the same node as the facility group. You cannot place a node ***trunk*** group into a facility group. (See also Outgoing Access on this page.) Using ARS is the only way a user can access trunks on other nodes. If the system routes an outgoing call to another node using an intermediary node, the intermediary node's ARS cannot use the nodes in its facility group to move the outgoing call to another node. This eliminates the system from getting into an infinite loop searching for a node to route the outgoing call. (See page 5-99 in FEATURES for a complete explanation of ARS.)
- **ARS Toll Restriction:** When determining toll restriction for an ARS outgoing call, the network only checks the station toll restriction for the node on which the station resides. The system does not check the trunk class of service for ARS calls.
- **Emergency Calls:** Emergency calls, by default, use the first local trunk group and will not be sent using node trunk groups on other nodes. However, when ARS is used to place an emergency call, Route Group 1 is used even if it contains nodes. This means that the network ***can*** access a trunk on a node other than the user's node if the user accesses ARS and dials the emergency number. ***Inter-Tel highly recommends that local trunks be installed and used for emergency number trunk access and that nodes not be used in Route Group 1.*** When a user places an emergency call, every administrator in the network receives an emergency alarm. (See page 5-112 for a feature description.)
- **Direct Inward System Access (DISA):** DISA callers can select off-node stations, hunt groups, and Voice Processing applications.
- **Keymaps:** Each node contains its own keymap programming.
- **DSS/BLF Keys:** You can program a DSS button for an extension that resides on another node. The buttons will show lamp status for stations on the other nodes.
- **Secondary Extension Keys:** The primary station and the secondary station must reside on the same node. (See page 5-139 for a description of secondary extension buttons.)
- **Music-On-Hold and Background Music:** Each node has its own music source. If a caller on Node 1 is holding for a user on Node 2, the caller hears the music on node 2. Note that if a caller is holding for a user on Node 1 and the call moves to Node 2 (due to a transfer, forward, or recall), the user will stop hearing the music on Node 1 and will begin hearing the music on Node 2. (See page 5-159 for a detailed description of this feature.)

NOTE: As of 7.0, the system can be programmed to determine the music source a caller hears based on the device for which the caller is waiting. See page 5-159 for details about how this affects music-on-hold in a networked system.

- **Handsfree Calls:** The network allows handsfree intercom calls when calling from one node to another.

- **Camp On:** A user can camp on to busy resources on another node. When the system has several routing options for a call, it tries to use each route, in order, until it finds a free route. If it is unable to find a free route, the system tries once again to use the first route. If the first route is still busy, the system camps on to the first route. (See page 5-177 for details.)
- **Queue:** A user cannot queue onto busy resources on another node. (See page 5-177 for details.)
- **Inter-Station Messages:** The network allows users to leave messages for, and respond to messages from, users on a different node. A station's message center or alternate message source does not need to be on the same node as the station. It can be a station, hunt group, Voice Processing application, or any off-node device.
- **Off-Hook Voice Announce (OHVA):** The network allows intercom callers to establish OHVA calls to phones on other nodes. (See page 5-190 for more information about this feature.)
- **Account Codes:** The network validates account codes against the account code table on the same node as the caller. The account code follows the call as it moves from node to node and appears on every SMDR record associated with the call. (See page 5-198 for an explanation of account codes.) However, if an optional account code is entered during the call, that account code will be used in the SMDR report.
- **System Hold:** When a user puts a trunk on system hold, only that user and users on the same node as the trunk can access the call. (See page 5-203 for a description of the system hold feature.)
- **Transfer Recall Destination:** A station's transfer recall destination can be an off-node device. (See page 5-210 for an explanation of transfer recall destinations.)
- **Call Screening:** In a network setting, the Voice Processing can provide call screening for a destination extension on another node. However, the node where the Voice Processing is connected must have an off-node device programmed for the destination extension and access to the remote node. (See page 5-211 for a complete description of this feature.)
- **Conference Calls:** A user can build a conference with party members (stations and/or trunks) on other nodes. The conference circuits used to build the conference will be allocated from the originating party's node. Any party, not just the originating party, can add a party member to the conference. (See page 5-213 for more information about conference calls.)
  - — If the party who adds the new member is on the originating party's node, the party will be added to the conference circuit already allocated. This conference is still restricted to the four party limit.
  - — If the party who adds the new member is not on the originating party's node, a new conference circuit is allocated from the second node. Because this allows more than four parties in a conference, users should be warned that voice levels can be degraded if more than four parties are on a conference.
- **Reverse Transfer:** A user can reverse transfer a call from a station or a hunt group on another node. (See page 5-212 for details.)
- **Record-A-Call:** The Record-A-Call destination does not have to be on the same node as the station, but the Voice Processing Unit must have a mailbox for that station. (See page 5-218 for a complete explanation of this feature.)
- **Agent Help:** The Agent Help Extension does not need to reside on the same node as the user requesting Agent Help. (See page 5-220 for an explanation of this feature.)

- **System Forwarding:** System forward points can reside on other nodes. (See page 5-223 for a description of system forwarding.)
- **Call Forwarding:** The network allows the user to program a manual forward destination to stations on another node. (See page 5-229 for more information about call forwarding.)
- **System Speed Dial:** Each node in a network has its own system speed dial numbers. System speed dial numbers can be used only on the node on where they are programmed. (See page 5-233 for more information about speed dialing.)
- **Directories:** The network displays intercom directory names and extension numbers for all devices the user can access directly. Devices that are represented by wildcard extensions will ***not*** show up in the intercom directory. (See page 5-241 for a complete explanation of the directories.)
- **House Phone:** The house phone can be programmed to dial the extension number of an off-node device. (See page 5-251 for a description of the house phone feature.)
- **Paging:** All phones and trunks within a page zone must reside on the same node as the page zone. However, a page zone can contain external page ports on other nodes. (See page 5-256 for more information about paging.)
- **Do-Not-Disturb:** Do-not-disturb works as follows in a network setting. (Do-not-disturb is described in detail on page 5-257.)
  - — Each node has its own list of do-not-disturb messages that can be used only on that node.
  - — An intercom caller will see another user's do-not-disturb display when calling across nodes.
  - — The network allows DND override across nodes.
- **Remote Feature Programming:** Remote Feature Programming is used for setting a station's do-not-disturb and call forwarding modes from another station or through DISA. Remote programming can be performed on one node for any phone that is programmed as an off-node device on that node. You must go through DISA for remote programming of any phones that are not programmed as off-node devices. (See page 5-262 for a complete explanation of remote programming.)
- **Reminder Messages:** Each node has its own list of reminder messages that can be used only on that node. (Reminder messages are described in detail on page 5-267.)
- **Data Ports:** A station's data port extension can reside on another node. (See page 5-269 for an explanation of data ports.)
- **Day and Night Modes:** The network determines the day/night mode status of a call based on the day/night mode status of the node where the trunk resides. (See page 5-273 for more information.)
- **Call Cost:** The call cost shown on the display and SMDR output use the factors and rates for the node on which the trunk resides. In other words, if a station on Node A dials what is considered a long-distance number on node A, but the call is routed to Node B where the number is considered local, the phone will use the local call cost rate from Node B. The call rate used for calls between nodes will be based on the Network call rate. (See page 5-305 for a description of call cost and for programming information.)

- **SMDR:** Each node has its own SMDR programming, SMDR output port, and SMDR output port backup. (See page 5-308 for complete details about SMDR and page 6-276 for programming information.) You can enable or disable network call records on each node. When network calls are enabled, the following applies:
  - The system generates SMDR records for outgoing calls on the node where each trunk used resides. For example, if a caller on Node A places a call using a trunk group on Node B, the SMDR report for Node A will show the outgoing call to Node B and the SMDR reports on Node B will show the incoming call from Node A and the outgoing call on the trunk group.
  - The call record type for network SMDR records is NET.
  - If a node's SMDR output port is a node, the network sends SMDR records to the specified node.
  - You cannot select a node as the SMDR output port backup.
  - If SMDR output programming forms a loop, the system will send the SMDR output to the node's backup SMDR port. For example, if the SMDR port on Node 1 routes to Node 2 and the SMDR port on Node 2 routes to Node 1, the configuration will cause an infinite loop of SMDR routing. SMDR reports for Node 1 would be printed to the backup serial port on Node2 and vice versa.
- **Voice Mail Messages:** To correct for time zone differences when message notification is sent from the Voice Processing Unit on one node to stations on other nodes, there is a Time Zone setting in each mailbox. If set properly, the voice mail message envelope will indicate the time the message was received using the mailbox's programmed time zone.
- **SMDR Buffering:** The Voice Processing Unit will buffer SMDR records that are sent from other nodes in the network.
- **Message Print:** Each node has its own message print programming, message print output port, and message print output port backup. There should be a message print terminal at each node to monitor node and network performance and aid in troubleshooting. (See page 5-313 for more information about message print.)
  - If a node's Message Print output port is a node, the network sends Message Print records to the specified node.
  - You cannot select a node as the message print output port backup.
  - If Message Print output programming forms a loop, the system will send the output to the node's backup Message Print port. For example, if the Message Print port on Node 1 routes to Node 2 and the Message Print port on Node 2 routes to Node 1, the configuration will cause an infinite loop. Message Print reports for Node 1 would be printed to the backup serial port on Node2 and vice versa.
  - A two digit node number is inserted in front of the sequence number to indicate the node where the message originated. The following is an example of the new message print string format for a message that originated on node 3:

    -03:001- 15:30 03-20 M6009 WRN EG CP Corrupted Queue Was Cleared: HISTORY QUEUE
  - Message Print will ***not*** route diagnostic messages that apply to another node, except during manual Message Print requests.
  - Message Print will not send messages during a startup or reset. The system will wait one minute before starting to send messages. Startup messages will only be printed locally.

— Messages may not be printed in time sequence order on remote nodes. The order of an incoming message print depends on how the message print was routed to the destination node. Therefore, a message print could possibly arrive at the destination message print node out of order.

- **System OAI:** System OAI Applications that support networking can be used in conjunction with the CT Gateway/Transceiver. (For detailed information about the Gateway/Transceiver, see page 3-243 in INSTALLATION.)

  — The Gateway allows a single System OAI application to support a multi-node network. A single RS232 connection is used for communication with every node in the network (each node also requires a Transceiver). Each application will require a separate Gateway.

NOTE: System versions 8.0 and later require version 3.0 or later CT Gateway software.

  — The Transceiver enables the Gateway to connect to each node of a network via the Local Area Network (LAN). Transceivers allow the Gateway to communicate with multiple nodes regardless of the number of serial ports available on the Gateway PC and regardless of the physical distance between the Gateway PC and the Inter-Tel switch.

NOTE: Some System OAI applications, that do not support networking, will not operate properly on an Inter-Tel system that is part of a network, even with a Gateway and Transceiver. This is because those applications are not designed to track calls that cross over from node to node within a network. Those System OAI applications can only be used on stand-alone Inter-Tel systems that are not part of a network.

- **Inside Track:** Inside Track does not recognize the "NET" SMDR call type and treats it as an invalid record. Otherwise, Inside Track will work exactly as it did for previous system versions.

8. NETWORK-WIDE ALARMS

8.1 To allow one administrator to monitor multiple nodes, the system provides both system alarms and network-wide alarms:

- **Network-Wide Alarms:** When an event occurs that generates a network-wide alarm, the alarm is broadcast to every node in the system.
- **System Alarms:** System alarms appear only on the node on which the alarm was generated.

A. SENDING AND RECEIVING NETWORK ALARMS

8.2 Each node has two system flags that determine whether the node broadcasts or receives network-wide alarms.

- The **Send Network Alarms** flag determines whether a node will broadcast alarms that occur on that node to the rest of the network.
- The **Receive Network Alarms** flag determines whether the node will receive and display alarms sent by other nodes in the network.

8.3 To differentiate between network-wide and local alarms, network-wide alarms appear on administrator phones preceded by "NET ALARM" and local system alarms are preceded with "SYS ALARM."

8.4 On remote nodes, network-wide alarms will indicate the name of the node on which the alarm occurred. The node name is obtained from the node username.

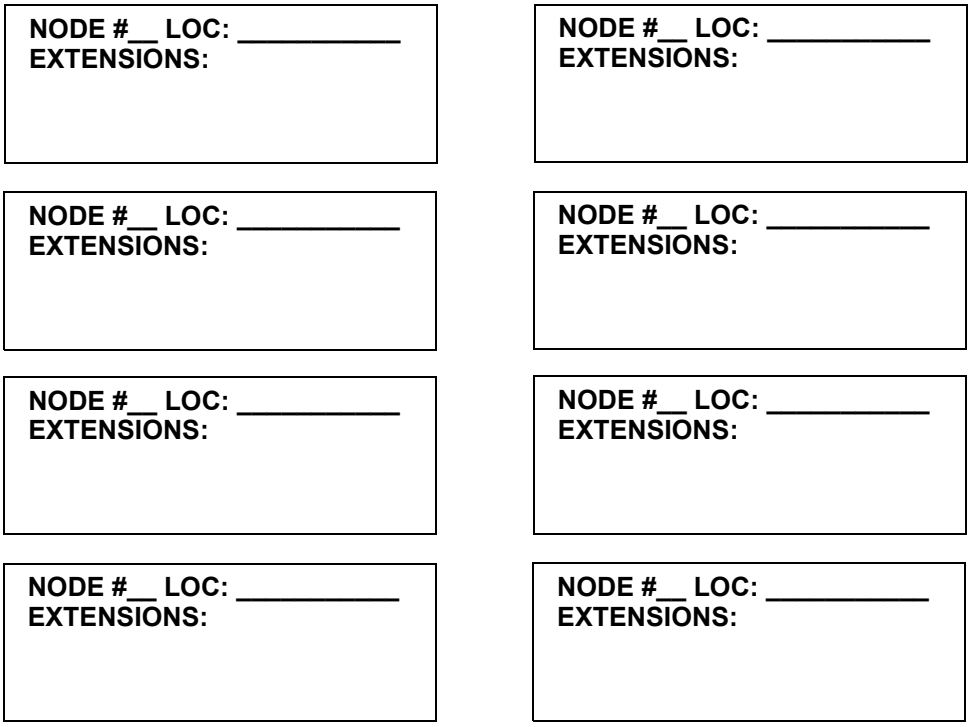
B. CLEARING NETWORK ALARMS

8.5 An administrator can clear a network-wide alarm on the local node only or on every node in the network.

- **Clear Network Alarm (9851):** This feature code clears network-wide alarms on every node in the network, but will not affect system alarms. The Clear Network Alarm feature code may be entered on any node in the network, but the Send Network Alarms flag must be set for the administrator to clear alarms on other nodes in the network.
- **Clear System Alarm (9850):** This feature code clears all local and network-wide system alarm displays on your node.

8.6 Network-wide alarms will override system alarms on an administrator's phone display.

Figure A-3. *Network Planning Sheet*



8.7 Use the drawing above, or one similar to it, to map your network. (See page A-19 for a sample network map.) Add information to the drawing as you answer the following questions:

- How many nodes are in the network? Draw additional Nodes if necessary.
- How are they numbered? Label each Node with its number.
- Where are they located? Label each Node with its location.
- How will the network be configured? Draw lines showing the connections between the Nodes.
- From where will calls ring in to the Voice Processing Unit and where will they be routed? Show the ring-in route on the map.
- Will call routing tables and/or ARS route calls to other nodes? If so, show these routes on the map.
- Where will the SMDR and Message Print output port(s) be located? If they cross between Nodes, show them on the map.
- Does each node have a local trunk group for emergency calls? Draw and label the local trunk groups for each node.
- Which nodes will have Voice Processing Units? Draw the Voice Processing Unit(s) on the appropriate Node(s). Record the Voice Processing Unit numbering plan in the worksheet on the next page.
- What is the numbering plan for the extensions, feature codes, hunt groups, node trunk groups, etc? Label your map and/or use the Numbering Plan Worksheet on the next page for recording this information.

Table A-12. *IP Networking Planning Worksheet*

The following is an example of a completed IP Private Networking Planning sheet. A blank sheet is provided on the following page for your use.

	Node 1	Node 2	Node 3	Node 4	Node 5	Node 6	Node 7	Node 8	Node 9
Extensions	1100-1199	1200-1299	1300-1399	1400-1499	1500-1599	1600-1699	1700-1799	1800-1899	1900-1999
Area Code	602	514	404	713	216	914	802	440	203
Nodes (Node Route Group)	97001	97002	97003	97004	97005	97006	97007	97008	97009
IP Connection Group	P6001	P6002	P6003	P6004	P6005	P6006	P6007	P6008	P6009
Node IP Connection Group*	P8000	P8001	P8002	P8003		P8004	P8005	P8006	P8007
IPRC (Networking)	192.168.0.41	192.168.0.42	192.168.0.43	192.168.0.44	192.168.0.45	192.168.0.46	192.168.0.47	192.168.0.48	192.168.0.49
IPRC (Station)	192.168.0.51	192.168.0.52	192.168.0.53	192.168.0.54	192.168.0.55	192.168.0.56	192.168.0.57	192.168.0.58	192.168.0.59
MGCP	192.168.0.61	192.168.0.62	192.168.0.63	192.168.0.64	192.168.0.65	192.168.0.66	192.168.0.67	192.168.0.68	192.168.0.69
IPPhone Plus	192.168.0.71	192.168.0.72	192.168.0.73	192.168.0.74	192.168.0.75	192.168.0.76	192.168.0.77	192.168.0.78	192.168.0.79
Subnet	255.255.252.0	255.255.252.0	255.255.252.0	255.255.252.0	255.255.252.0	255.255.252.0	255.255.252.0	255.255.252.0	255.255.252.0
Default Gateway	192.168.0.200	192.168.0.200	192.168.0.200	192.168.0.200	192.168.0.200	192.168.0.200	192.168.0.200	192.168.0.200	192.168.0.200

*The system automatically assigns the Node IP Connection Group extension numbers.

Table A-13. *Network Numbering Plan Worksheet*

Determine the following numbers for the network:

Voice Processing Unit:

Node Number:	
Voice Mail Application Number:	
Message Notification/Retrieval Application Number:	
Record-A-Call Application Number:	

EXTENSION NUMBERS:

Network Primary Attendant:	
Node 1 Range Of Extensions:	
Node 2 Range Of Extensions:	
Node 3 Range Of Extensions:	
Node 4 Range Of Extensions:	
Node 5 Range Of Extensions:	
Node 6 Range Of Extensions:	
Node 7 Range Of Extensions:	
Node 8 Range Of Extensions:	

Determine the following numbers for each Node:

NODE # _____ LOCATION _____

Local Extension Numbers:	
Local Trunk Group Numbers:	
Local Attendant:	
Off-Node Devices: (Include Voice Processing applications if they are not on this node)	
Off-Node Wildcard Numbers:	
Node Trunk Group Numbers:	

Appendix B — RS232

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1. INTRODUCTION

NOTE: RS232 has undergone several revisions over the last 30 years. The most current version of RS232 is EIA/TIA-232-E (July 1991), which includes the alternate use of RTS Running H/F 2-and CTS as character-by-character hardware flow control signals. The Inter-Tel® system serial ports conform to RTS/CTS flow control per EIA/TIA-232-E. Older versions of RS232, such as EIA-232-D (1987) and RS232-C (1969) are a subset of EIA/TIA-232-E. Hence, serial ports which conform to EIA/TIA-232-E are compatible with all RS232-C serial ports. However, if the RS232-C serial port does not support RTS/CTS flow control, then the connection cannot use RTS/CTS hardware flow control.

1.1 Connecting two different devices using RS232 can be very confusing as well as time consuming. The confusion arises because there are so many different factors that can vary on each type of RS232 connection, such as:

- Type of device
  - DTE (Data Terminal Equipment), e.g., PC Serial COM port, Printer, etc.
  - DCE (Data Circuit-Terminating Equipment), e.g., modems
- Type of connector
  - D-sub (a.k.a. “DB”)
  - Modular
- Male or female gender of connector
- Number of pins
  - 25 pin DB
  - 9 pin DB
  - 8 pin modular
  - 6 pin modular
  - 4 pin modular
- Different Cables
  - 25 wire straight-through cable
  - 9 wire straight-through cable
  - Reversing vs. non-reversing modular line cords
  - 4-wire, vs. 6-wire vs. 8-wire modular line cords
  - Custom cables
  - Cable length
- Special Adapters
  - 9-to-25 pin adapters
  - “Null-modem” adapters
  - Gender changers
  - Custom adapters

- Flow control settings
  - — No flow control
  - — Software flow control only (XON/XOFF)
  - — Hardware flow control only (RTS/CTS)
  - — Both hardware and software flow control
- Programmable settings
  - — Bit rate (e.g., 1200, 2400, 4800, 9600 bps, etc.)
  - — Number of data bits (e.g., 7 or 8)
  - — Parity (e.g., none, odd, even, mark, or space)
  - — Number of stop bits (e.g., 1 or 2)

1.2 Successfully connecting two devices using RS232 requires an understanding of all of the above and then requires you to program the proper settings, choose the correct cables and adapters, and make sure that the pins are wired up correctly.

2. INTER-TEL RS232 PRODUCTS

2.1 To make RS232 connection easier, Inter-Tel provides the following products and guidelines to help you with all of your RS232 connections to system serial ports.

- For modem connections, purchase the Axxess® Modem Kit (part number 550.3026). This kit provides you with the correct cable, adapter, a properly pre-configured and tested modem, and a set of installation instructions.
- For most other system-related RS232 serial connections, purchase the Inter-Tel Universal RS232 kit, part number 828.1282. This kit contains a variety of cables and adapters as well as documentation that shows you how to quickly and efficiently make virtually any type of RS232 connection you are likely to encounter, including:
  - — System to PC COM port
  - — PC COM port to PC COM port
  - — System to serial port printer
- For PCDPM connections, refer to the *Digital Phone PCDPM Installation Instructions*, part number 835.1823 (-5 or later), or the Model 8520 or 8560 PCDPM Installation Instructions, part number 835.2563 (-1 or later).
- For OAI connections, refer to the system *Card-To-PC System OAI Cable Kit Instructions*, part number 835.2007 (-2 or later)

2.2 These types of applications involve:

- DB Programming
- Diagnostics
- Inside Track
- Voice Mail
- Computer-Telephony Integration (CTI) Applications – Desktop OAI and System OAI

2.3 Although the system serial ports use 8-pin modular jacks and will support 4-wire, 6-wire, and 8-wire reversing line cords, ***Inter-Tel recommends that you always use 8-wire reversing line cords for all RS232 connections.*** Using 4-wire or 6-wire reversing line cords introduces limitations that, if not properly programmed for, will result in an inoperable RS232 serial connection. Inter-Tel ships **only** 8-wire reversing line cords with RS232 related products mentioned above.

2.4 Although you can custom wire your own adapters and cables, Inter-Tel recommends that you purchase the RS232 related adapters and cables you need, either individually or in one of the kits listed above. This will save you both time and frustration. Furthermore, all of Inter-Tel's RS232 related cables and adapters are clearly labeled for quick identification and proper usage and installation.

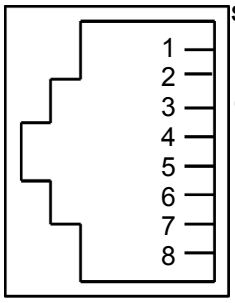
2.5 The basic set of cables and adapters that are needed for most RS232 connections, and which are included in the Inter-Tel Universal RS232 kit, include:

- Inter-Tel Custom DB9F-to-8P/8C modular adapter which supports hardware flow control and is designed to connect to a PC serial COM port (part number 804.2545)
- Inter-Tel Custom DB9F-to-8P/8C modular adapter which does not support hardware flow control and is designed to connect to a PC serial COM port (part number 804.2420 – replaces 804.2561)
- 14-foot, 8P/8C reversing modular line cord (part number 813.1682)
- Inter-Tel DB9F-to-DB9F "null modem" adapter which supports hardware flow control and is designed to connect a PC to a PC (part number 804.2605)
- Inter-Tel DB9F-to-DB9F "null modem" adapter which does not support hardware flow control and is designed to connect a PC to a PC (part number 804.2604)
- 12-foot DB9M-to-DB9F straight-through cable, part number 813.1683
- DB25F-to-DB9M adapter, part number 804.2574

3. SYSTEM SERIAL PORT PIN OUT

A. MODULAR JACK PIN OUT

3.1 Inter-Tel system RS232 serial ports (except for PCDPM) use a custom pin out as shown in the diagram below. See this diagram for all RS232 connections to the serial port.

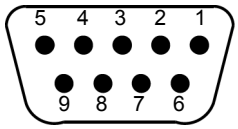


	SIGNAL TYPE	SIGNAL DIRECTION
1	No Connect	
2	No Connect	
3	Ground	
4	Data	Input To System
5	Flow Control	Input To System*
6	Data	Output From System
7	Port Ready	Output From System (always asserted)
8	Flow Control	Output From System

*The serial port on the CPU-112 card (not supported by system version 5.0 or later), does not have the Input Flow Control pin connected, but it can use the Output Flow Control pin. The two serial ports on a MEM card have both flow control pins connected; therefore, they can perform hardware flow control as the other Inter-Tel ports do.

B. DB9F PIN OUT

3.2 Inter-Tel PCDPMs use the de facto industry standard DB9F pinout configured as a DCE device. See the following diagram for all PCDPM connections.



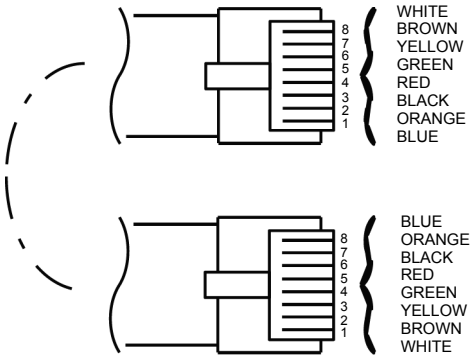
PCDPM DB9F Connector			
Description	**RS232 Equivalent Pin Name**	**Direction (DCE)**	**DB9F Pin No.**
Always True	DCD	Output from System	1
Output Data	RD	Output from System	2
Input Data	TD	Input to System	3
Input Flow Control (DTR)	DTR	Input to System	4
Ground	GND		5
Always True	DSR	Output from System	6
Input Flow Control (RTS)	RTS	Input to System	7
Output Flow Control	CTS	Output from System	8
Not Supported	RI	No Connection	9

4. CABLING

4.1 System serial ports (except for the PCDPM serial ports) use a modular jack for RS232 connections. Therefore, a modular line cord and modular adapter must be used for RS232 connections to the system serial port. ***Inter-Tel recommends that you always use 8-wire reversing line cords for all RS232 connections. Inter-Tel's modular adapters are designed to be used only with reversing line cords.***

NOTE: Although the system serial ports use 8-pin modular jacks and will support 4-wire, 6-wire, and 8-wire reversing line cords, ***Inter-Tel recommends that you always use 8-wire reversing line cords for all RS232 connections.*** Using 4-wire or 6-wire reversing line cords introduces limitations that, if not properly programmed for, will result in an inoperable RS232 serial connection. Inter-Tel ships ***only 8-wire*** reversing line cords with RS232 related products.

4.2 A reversing cable simply inverts the order of the wiring from one to the other. For example, on an 8-wire reversing line cord, pin 1 on one end of the cable goes to pin 8 on the other end, pin 2 goes to pin 7, etc. Reversing line cords can be identified by examining the modular plug connectors side-by-side as shown in the diagram below.



8-WIRE REVERSING LINE CORD (TOP VIEW)

NOTE: The colors shown in the diagram above can vary and are used for illustration purposes only.

4.3 For other RS232 connections, including PCDPM connections and PC serial COM port to PC serial COM port connections, use a 12-foot DB9M-to-DB9F straight-through cable (part no. 813.1683).

4.4 A straight-through cable does not change wiring from one end to the other. On an 8-wire straight-through cord, pin 1 on one end goes to pin 1 on the other end, pin 2 goes to pin 2, etc.

5. MODULAR ADAPTERS

5.1 Inter-Tel provides D-sub modular adapters that are designed specifically for certain types of RS232 connections to an system RS232 serial port modular jack. These include the following:

- Inter-Tel Custom DB9F-to-8P/8C modular adapter which supports hardware flow control and is designed to connect to a PC serial COM port (part number 804.2545). For further details, refer to the Inter-Tel Universal RS232 kit instructions.
- Inter-Tel Custom DB9F-to-8P/8C modular adapter which does not support hardware flow control and is designed to connect to a PC serial COM port (part number 804.2420 – replaces 804.2561). For further details, refer to the Inter-Tel Universal RS232 kit instructions.
- Inter-Tel Custom DB25M-to-8P/8C modular adapter which support hardware flow control and is designed to connect to a modem (part number 804.2543). For further details, refer to the Inter-Tel Modem Kit instructions, part number 835.1621 (-4 or later).

5.2 For your reference, wiring diagrams are printed on the labels of these adapters.

6. D-SUB ADAPTERS

6.1 Inter-Tel provides D-sub adapters that are designed specifically for certain types of RS232 connections. These include the following:

- Inter-Tel DB9F-to-DB9F “null modem” adapter (part number 804.2605) which supports hardware flow control and is designed to connect a PC to a PC using a 12-foot DB9M-to-DB9F straight-through cable (part no. 813.1683)
- Inter-Tel DB9F-to-DB9F “null modem” adapter (part number 804.2604) which does not support hardware flow control and is designed to connect a PC to a PC using a 12-foot DB9M-to-DB9F straight-through cable (part no. 813.1683)
- Inter-Tel DB9M-to-DB9F custom RTS/DTR hardware flow control adapter (part no. 804.2546) is required for older versions of the PCDPM when RTS/CTS hardware flow control is required. For more details, refer to the PCDPM Installation Notice (part no. 835.2040).
- Inter-Tel DB9M-to-DB9M “Inter-Tel Axxent Modem Adapter” (part no. 804.2556). For more details, refer to the Axxent Modem Kit Instructions, part number 835.1622 (-4 or later).
- DB9M-to-DB25F adapter (part number 804.2574) used for converting a nine-pin D-sub connector to a 25-pin D-sub connector. It is designed to be used on PC serial COM ports, when needed.

6.2 For your reference, wiring diagrams are printed on the labels of these adapters.

7. SETTINGS

7.1 There are five parameters that must be set correctly on each end of an RS232 connection in order for the connection to be successful:

1. Both devices must communicate at the same bit rate (e.g., bits per second).
2. Both devices must use the same number of data bits per character.
3. Both devices must use the same type of parity (for error checking).
4. Both devices must use the same number of stop bits.
5. Both devices must use the same type of flow control.

7.2 The system serial port end of the RS232 connection configures most of these settings as follows:

1. The bit rate, specified in bits per second (bps), can be configured and the allowed range of bit rates depends on a number of factors. (Examples of allowed bit rates include 1200, 2400, 4000, 9600, etc.)
2. System serial ports always uses 8 data bits per character (this is fixed and cannot be changed).
3. System serial ports always uses NO parity for error checking (this is fixed and cannot be changed).
4. System serial ports always uses 1 stop bit (this is fixed and cannot be changed).
5. System serial ports can be configured for the following types of flow control:
   - No flow control
   - Software flow control only (XON/XOFF)
   - Hardware flow control only (RTS/CTS)
   - Both hardware and software flow control

7.3 The PC serial communications (COM) port end of the RS232 connection must have the exact same settings (properties) as system serial ports, including:

- bits per second (e.g., 4800, 2400, 1200, etc.)
- 8 data bits
- no parity
- 1 stop bit
- same flow control

NOTE: If you plan on connecting the system serial ports to *anything other than a PC serial port* (e.g., to a serial printer), you will need to refer to the documentation provided with the device and set its RS232 serial port correctly.

8. FLOW CONTROL

8.1 Flow control tells each device (such as the system serial ports and the PC serial port) when it can transmit data and when it must stop transmitting data.

8.2 There are two types of hardware flow control commonly found in the industry:

- DTR/CTS hardware handshake (commonly used by printers and other devices, but not supported by any formal standard)
- RTS/CTS hardware handshake (now supported by the EIA/TIA-232-E standard)

8.3 When the system is in the default configuration, the Inter-Tel system RS232 ports are enabled for software flow control. Most output devices are designed for software flow control, which is the use of "X-ON" and "X-OFF" characters to indicate readiness to send or receive data. Refer to the device's operating manual to determine if it uses X-ON/X-OFF characters.

8.4 The PCDPM RS232 connection *supports only hardware flow control and hardware flow control is always enabled.* The reason that the PCDPM supports only hardware flow control, and that hardware flow control is enabled at all times, is because the data input buffer in the phone is very small; it is only 15 bytes. This means that the other device (typically a PC) connected to the PCDPM must also enable hardware flow control and observe the output flow control signal (CTS) from the PCDPM for the connection to work properly.

8.5 The flow control truth table for the most recent model PCDPM (marked with artwork number 826.5239-5 or later) is shown below:

Table B-1. *PCDPM Flow Control Truth Table*

PCDPM Flow Control Truth Table		
DTR	**RTS**	**Result**
No Connect	No Connect	Stop
No Connect	Off	Stop
No Connect	On	Go
Off	No Connect	Stop
Off	Off	Stop
Off	On	Stop
On	No Connect	Go
On	Off	Stop
On	On	Go

8.6 Note that RTS takes on the functionality of "Ready for Receiving."

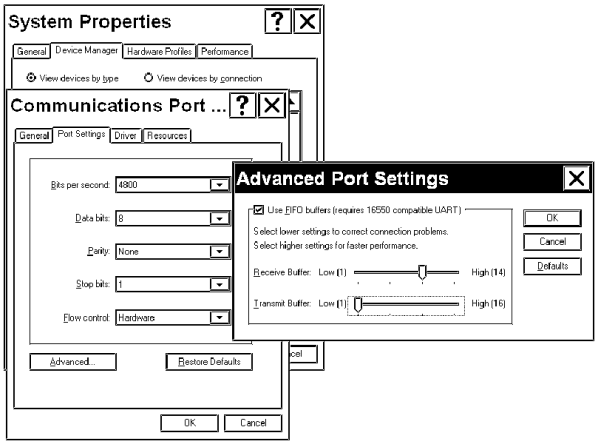
- Off = Negative voltage (less than -3V).
- On = Positive voltage (greater than +3V).
- No Connect = The pin is not connected to the anything and is therefore considered "floating" or "open."
- Go = The PCDPM transmits data to the PC.
- Stop = The PCDPM does not transmit data to the PC.

NOTE: If you plan on attaching the PCDPM to anything other than a PC serial port, you will need to make sure that the other device has hardware flow control turned on. The other device should be using RTS/CTS or DTR/CTS flow control. If the other device uses any other pins for hardware flow control, you will need to make your own custom adapter or cable to support such a non-standard form of hardware handshaking.

8.7 Windows 95/98 or Windows NT 5.0 Advanced Port Settings: If the PCDPM will be attached to a PC running Windows 95 or Windows 98, the PC's serial port transmit buffer FIFO must be set to "Low" (1 byte) since the PCDPM's data input buffer is very small (15 bytes).

NOTE: This procedure does not apply to PCs running Windows 3.1 or earlier or Windows NT 4.0. And, unless the 16550 UART FIFO option has been purposely enabled, it does not apply to PCs running Windows NT 3.51 or Windows for Workgroups 3.11.

1. From the Start Menu, select **Settings** and then **Control Panel**.
2. Double-click on **System**, and then select **Device Manager**.
3. Double-click on **Ports**, and then double-click on the ***appropriate*** COM port icon (or select **COM port** and then select **Properties**). Be sure to select the COM port physically attached to the PCDPM.
4. Select **Port Settings**, and then select **Advanced**.
5. In the Advanced Port Settings window (shown below), set the Transmit Buffer setting to **Low (1)** by moving the slide control all the way to the left.
6. Select **OK** to save the new setting, exit back out of Control Panel, and then reboot the PC.



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1. INTRODUCTION

1.1 With the Call Processing (CP) Server, call processing software has been moved from the telephone system (System) CPU Card to a Windows NT- or 2000-based server that communicates with the System via a 10 Base-T Ethernet connection.

1.2 The PC and equipment chassis are connected directly via a cross-connect Ethernet cable and use the User Datagram Protocol (UDP) to transfer data across a direct connection.

1.3 As of version 6.0, the following features are available:

- Security keys

  NOTE: If upgrading from v5.1 or earlier, you must use the security key (part no. 827.9775).

- Full support of System and Voice Processing Unit version 6.0 and later functionality
- NT and Windows 2000 Event Viewer support (see page C-38)
- Support for ATM-based nodes (Win 2000 CP Server only), which increases the number of supported devices from 1024 to 1792
- PCI Watchdog Card, which replaces the Audio Interface Card (AIC)
- ATM Adapter Card *(ATM systems only)*, either with OC-3 or UTP5 connectors

1.4 **Call Processing Server on Windows Server 2003:** Windows Server® 2003 (Telco version) operating system (OS) with Service Pack 2 (SP2) is now supported for the Call Processing Server (CPS). Field upgrades must be done by certified technicians only.

NOTICE

Only an A+ and Axxess Certified technician with thorough knowledge of the Windows 2000 and Windows Server platforms may perform this upgrade. If you do not have the recommended certification and need technical support, you are automatically charged a Technical Support Incident (TSI) credit.

Technical Support for the upgrade is available to those who have purchased an upgrade kit (upgrade kit part no. 828.1754 for non-ATM systems and 828.1755 for ATM systems). If you use Windows software from a source other than from an upgrade kit, Inter-Tel support is not available.

1.5 CPS platforms with the following Rev AR, dated July 2004 or later, with the following part numbers can be upgraded:

- Non-ATM part number 550.5360
- ATM (copper) part number 550.5361
- ATM (fiber) part number 550.5362

1.6 Verify that the CPS platform consists of the following hardware. If it does not consist of the following hardware, it is not the correct revision and cannot be upgraded:

- Intel® Pentium® 4 processor
- Rackmount chassis
- DVD drive

1.7 You cannot upgrade a CPS platform that is earlier than the Rev AR identified above. You need to purchase a new CPS if you want to upgrade to v11.0. Contact Inter-Tel for details.

1.8 The Windows Server 2003 hardware specifications for the CPS are the same as Windows 2000 Professional hardware specifications. There are no changes to CPS functionality for v11.0.

1.9 There are two installation images based on the CPS system configuration of ATM and non-ATM. Refer to the *Inter-Tel Axxess Call Processing Server Upgrade Instructions* (document part no. 835.3211) to upgrade the CPS from Windows 2000 Professional to Windows Server 2003 (Telco version). This document covers upgrading the following CPS configurations:

- ATM
- Non-ATM
- Redundant
- Non-Redundant

2. PC SPECIFICATIONS

2.1 Each CP Server PC meets the following minimum requirements:

- 850 MHz CPU
- 128 MB RAM
- 30 GB Hard Drive
- 52X CD-ROM
- 1.44 MB Floppy Drive
- 2 Intel 10/100 NICs—1 Intel and 1 Ericsson® (Marconi) ATM NIC for ATM-based Systems
- AGP Video
- 2 Serial Ports
- 1 Parallel Port
- 2 USB Ports
- Windows 2000 Pro with Service Pack 2 (Windows 2000-Based CP Servers)

3. SYSTEM CAPACITIES

3.1 The following table defines the capacities available when using a CP Server.

Table C-1. *CP Server System Capacities*

Devices	1792*
Forced and Standard Account Codes	5000
Hunt Groups	300
Page Zones	50 (9600-9649)
System Speed Dial Numbers	5000 (0-4999)
ACD Agent IDs	3000
ARS Route Groups	100
Number of Facility Groups in a Route Group	100
Off-Node Devices	25,000
Call Routing Table Patterns	2000+**
Extension List Capacity	40,000 devices
Maximum Page Zone Recommendation	120 phones
Maximum All-Ring Hunt Group Recommendation	60 stations
Networking Internet Protocol Resource Cards (IPRCs or IPRC-Ds)	28

* This value is based on an eight-chassis ATM System. If using the CP Server with a non-ATM system, the maximum number of devices is 1024.

** The maximum number of call routing table patterns depends on the number of digits that are translated (i.e., received from the CO). For example, if receiving ten digits, the maximum is approximately 2380 patterns; if receiving four digits, the maximum is approximately 2767.

NOTE: Even though a non-ATM database supports 1024 devices, the equipment chassis cannot. Therefore, the number of devices is only limited by the hardware configuration in the Database (DB) Programming.

4. SYSTEM APPLICATIONS PROGRAMMING CHANGES

A. DATABASE CONVERSION ALLOWS UPGRADE TO MAXIMUM DEVICES

4.1 The System database conversion program displays an option to upgrade to an ATM system, which is based on eight chassis. The database is then modified to allow the new database capacities shown on the previous page.

4.2 However, because the existing page zones are numbered 960-969 and the new page zones are numbered 9600-9649, only the first ten page zones are converted and none are added. This prevents possible extension conflicts between the new page zones and other extensions and feature codes. You can enable the additional page zones and select their extensions as needed.

B. DB PROGRAMMING

4.3 DB Programming allows you to change the CPU type to 1024 in local mode, if the current CPU is a CPU-PCM-F (Quad). When the CPU type is changed to 1024, the database is updated as if it were run through the conversion program. You cannot change the CPU type to a PCMA card (ATM) while in DB Programming. You must use DB Converter.

NOTE: ***Do not*** load System DB Programming software on the CP Server PC. This PC is intended for use only as a CPU.

5. CHASSIS COMPATIBILITY

5.1 The CP Server is responsible for ensuring the chassis in an ATM system are running compatible operating system (OS) or CP versions. When the CP Server first signs on to a chassis, the chassis sends a version response. If the version is not compatible, the CP Server will not allow that particular CPU to come online, and message print M5134 is submitted. If this occurs, you must flash the CPU with the correct OS version.

6. CPU020/PCM CARD AND ETHERNET ADAPTER CARD

6.1 For a non-ATM system, the CP Server uses a CPU020 Card with an Ethernet Adapter Card (EAC). The CPU020 has an 80-pin connector that attaches to the EAC, and the PCM faceplate has an extra cutout for the EAC's RJ45 connector.

6.2 The three components, CPU020, PCM, and EAC, are available in two configurations.

- CPU020/PCM-D with an EAC (550.9026)
- CPU020/PCM-F with an EAC (550.9025)

6.3 The EAC supports only 10 Base-T and is connected to the PC via an industry-standard category 5 crossed patch cable (813.1743). The cable is 14 feet in length, allowing some freedom of placement of the server. (See page C-34 for a pinout of the cable.) Per IEEE 802.3 10Base-T specification, this cable can be up to 328 feet (100 meters) long.

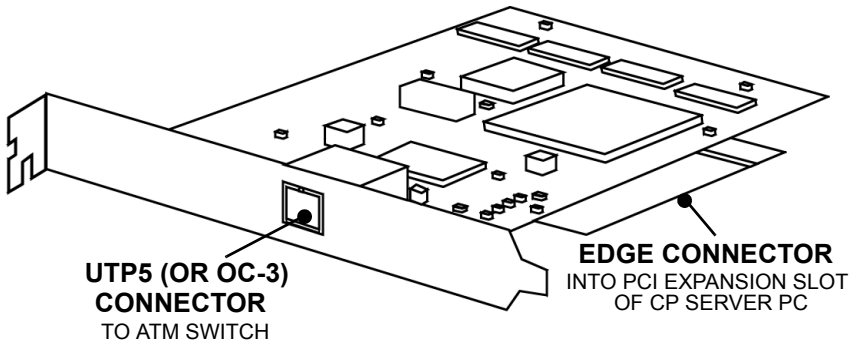
6.4 No changes to the Slave CPU are necessary.

NOTE: ATM Systems use PCMA Cards, which are placed in the equipment chassis and connected to the CP Server via the ATM Switch.

7. ATM ADAPTER CARD

7.1 For an ATM system (v6.0 or later), your CP Server comes equipped with an ATM adapter card. This card has a UTP5 or OC-3 connector that provides the interface for the ATM Switch (see the following illustration).

Figure C-1. *ATM Adapter Card*



7.2 This card comes pre-programmed for use with the ATM Switch.

8. PCI PC WATCHDOG BOARD

8.1 The hardware watchdog board is designed to reset the CP Server when it fails to operate for some unknown reason. This allows the PC to reboot and continue operation, which is very important for unmonitored systems. Previously, the Inter-Tel® Audio Interface Card (AIC) was used, but, due to compatibility issues, this card was replaced in v6.0.

8.2 The new card is the PCI PC Watchdog™ board. This PCI-based card is supported by Windows 2000 and Windows NT and has the ability to disable the watchdog timing. (For details about the board, refer to the manual included with the board.)

A. BASIC OPERATION

8.3 The PCI PC Watchdog™ board has eight DIP switches, which are all OFF by default. For proper operation of the CP Server, two of these switches, 5 and 7, must be ON.

8.4 Switch 5 represents the Power On Delay (POD) operation. If the POD is not ON, the card will wait 2.5 minutes after powering on before arming itself. After armed, it will expect an application to "tickle" the watchdog every five seconds; otherwise, the PC will reset. By enabling the POD switch, the card will not arm itself until it receives its first "tickle."

8.5 Switches 6, 7, and 8, determine the default amount of time before the PC will reset when it has not been tickled. By default, this is five seconds. For the CP Server, make sure switches 6 and 8 are OFF, but switch 7 is ON to change this time to 30 seconds.

8.6 After the switches are properly set, the Inter-Tel Watchdog Service will "tickle" the PCI PC Watchdog board every two seconds after the card is armed. If the Watchdog Service is stopped manually, the watchdog timing is disabled, and the PC will not reboot.

B. LED INDICATIONS

8.7 The PCI PC Watchdog™ has two LEDs on the card. When facing the back of the PC chassis, the LED on the right is the status light. During power-up and before the card is armed, this LED will blink at a rate of once per second. When the watchdog is armed, it will blink every 0.5 second. When the board is within four seconds of resetting the PC, it will flash every 0.1 second. If the watchdog resets the PC, the LED on the left will remain lit until the Inter-Tel Watchdog Service turns off the LED. The Watchdog Service will turn off this LED when the PC is reset under normal circumstances or when the software resets the PC.

9. SYSTEM CPU020 FIRMWARE

9.1 The Boot ROM (in socket U25) used for System CPU was modified to support the CP Server.

- Boot ROM for CPU020/PCM-D (827.8966)
- Boot ROM for CPU020/PCM-F (827.8965)
- Boot ROM for CPU020/PCM-F or D and CPU020/PCMA-F, FS, or C (827.9268)

9.2 In addition, the memory map PALs for both the dual and quad configurations have been combined into a single memory map PAL (socket U41). The part number for this PAL is 827.8967.

10. SOFTWARE SECURITY

10.1 The CP Server uses two methods of software security:

- **Software License:** The CP Server will not operate without the proper software license.

  NOTE: Versions 7.0 and earlier require a feature PAL on the CPU020 board.

- **Software Key:** The CP Server PC must have a security key attached to the parallel port. Without this security key, the software license will not work.

  NOTE: The basic call processing software key is only issued by Inter-Tel's Project Management team under special circumstances.

  For 8.0 systems, the base software security key is part no. 827.9775.

  For 6.0 and 7.0 systems, the software security key (part no. 827.9572) expands to include governing the premium features and the other special PAL features. This key not only replaces the standard set of security PALs, but it controls the system expansion beyond the normal two chassis pairs.

10.2 The software checks for the existence and validity of the key on power-up and every five minutes thereafter. If the key is missing, all audio is disabled, any slave chassis in a non-ATM system is off-line, and the phone system issues alarm #122, until a valid key is attached. If the key is replaced while system is running, the software re-enables audio and/or the appropriate features during the next periodic check of the software key.

11. INTER-TEL CALL PROCESSING AND WATCHDOG SERVICES

11.1 The Inter-Tel Call Processing service is started by the Watchdog service. The Watchdog service starts up automatically and does not require a user to log on to Windows in order to start it. The system administrator has the ability to start and stop the Inter-Tel Call Processing service through the Services applet in the Control Panel.

CAUTION

When the software is installed or upgraded, the Software Watchdog service is set for "manual" startup by default. It is very important for you to change the startup type to "automatic" after the new software is installed and is working properly.

11.2 The Software Watchdog is a service that runs independently of the Inter-Tel Call Processing service. Its job is to monitor the Inter-Tel Call Processing service through the use of the following:

- **Service Control Manager:** The software watchdog reads a list of target services it is to monitor from the registry. It then asks the Service Control Manager (SCM) the status of each target service. If the target service is stopped, it will ask the SCM to signal the service to start. This service starts automatically upon system power-up.
- **Event Signaling:** For each service the software watchdog monitors, it creates a watchdog event. If this event is not signaled for a given period of time, the target service is assumed stopped, and the software watchdog attempts to start the service again.
- **File Existence:** The watchdog service also checks for particular files to be in existence when checking the status of a target service. If the target service writes out a "User Terminated" file (**`~usertrm.svc`**), the watchdog service does not automatically start the target service, because the service was probably stopped manually for a software upgrade or other type of maintenance. There is also a "Restart" file (**`~restart.svc`**). This signals the watchdog service that the target service is waiting to be restarted as soon as possible. This is the file written during a Major Reset. The third file is the "Can't Connect" file (**`~cantcon.svc`**). This file signals the watchdog that the CP Server is unable to connect to the equipment chassis. If the CP Server is part of a redundant system, the "Can't Connect" file tells the watchdog service that the CP Server may be the backup server and that the watchdog service should attempt to connect to the other CP Server as a backup. If the system is not redundant, the watchdog service treats this file as a "Restart" file.
- **Hardware Watchdog:** The watchdog service also performs a global function of maintaining signaling to a hardware watchdog timer on the Audio Interface Card (AIC) or the PCI Watchdog Card. If the hardware watchdog is not signaled within a given time frame, the hardware device signals the reset circuitry on the motherboard and the system goes through a hard reset. This is used to safeguard against the PC from locking up and disabling the system for an extended period of time.

11.3 The software watchdog also keeps track of how many times it has had to do a soft restart of a service. If the service does not come up after a number of restarts, then a system shutdown is scheduled. Also, there are initial grace periods for allowing services to start up and begin signaling the software watchdog event.

12. MEMORY MAPPED FILES

12.1 The CP Server and the System track and store static information differently, as described below:

- The System uses static memory to store the message print log and buffer, reset log, exception log, history queue, and the database itself. Each of these is assigned, or mapped to, a specific predetermined memory address range.
- The CP Server follows the same concept. Address space uses memory-mapped files in lieu of the static memory allocation used on the System, with several subtle differences. The size of each individual buffer is determined at run time by registry settings. A file is then created on disk and its contents mapped into the processes memory address space.

12.2 When the CP Server first starts up, it either creates or checks for the existence of the following files:

- **Cp_queue.mmf:** This is the queue where all system commands are stored. This is equivalent to the **`.cph`** file you would see in a System history retrieval. The **`.cph`** file on the System is limited to 120 KB. On the CP Server, however, this file defaults to 1024 KB.
- **Exception_log.mmf:** This is the queue where all the exception data is kept. This includes the state of registers and call stack at the time the exception occurred. This is really no different than the exception data found in the upper half of the **`.dmp`** file you would see in a System history retrieval. The two differences are that the exception data is in a format more suited to Windows NT and Windows 2000 and that call stack addresses are translated to function calls directly. On the CP Server, the default is 256K.
- **Reset_log.mmf:** This is the queue where all the reset records are stored. It is equivalent to the lower half of the **`.dmp`** file you would see in a System history retrieval.
- **Msg_print_log.mmf:** This is the queue where message print messages flagged to be logged are kept. This is completely equivalent to the **`.log`** file you would see in a System history retrieval. On the System, however, the buffer is limited to 64 KB. On the CP Server, the default is 256 KB.
- **Msg_print_buffer.mmf:** This is the queue where message print messages waiting to be printed are stored until the message print device is available. This is equivalent to how the System works. On the System, however, the buffer is limited to 64 KB. On the CP Server, the default is 256 KB.
- **Static_memory.mmf:** The static memory region is essentially the database. All database changes are stored in this file. However, this file consists of more than just the database. All objects allocated into static memory are also located here. This includes many static structures within call processing. Deleting this file is the same as defaulting the database.
- **Network_print_log.mmf:** This is the queue where all of the network link information is stored.

13. BACKUP FILES

13.1 With access to a hard disk, call processing now has the ability to backup the memory-mapped files to disk. This gives the CP Server a tremendous amount of data collection ability.

13.2 For example, if the CP Queue memory mapped file is set to 1 MB, the default value of 20 backup files would give you 20 MB of CP history queue data. The number of backup files can be changed using the configuration utility.

13.3 The message print log, reset log, and exception log are backed up in a similar fashion. See "Retrieving CP History Information" below for information about specifying backup file location and numbers.

14. SYSTEM RECOVERY

14.1 Every CPS is shipped with a recovery CD/DVD located inside the chassis. If it ever becomes necessary to reinstall the operating system, you must use the recovery CD/DVD to return the unit to its original condition. Refer to the *Recovery Instructions for PC-Based Products* (part no. 835.2759) included with the recovery DVD for details.

14.2 If this unit was originally shipped with an E:\ partition and backup directories, you must recreate the E:\ partition and necessary backup directories on the unit as part of this recovery process.

14.3 To recreate the E:\ partition and required directories:

1. Open the Windows Disk Management utility.
2. Right-click the unpartitioned **`E:\`** to create a primary partition.
3. Follow the steps to format the partition with the NTFS file system.
4. When the partitioning process completes, verify that the partition is labeled "`E:\`" and then create the proper backup directories on this partition:
   - **`E:\backup`**
   - **`E:\backup\account`**
   - **`E:\backup\group`**
   - **`E:\backup\public`**

14.4 The automatic data backup process does not occur if you do not create the proper backup directories.

15. SYSTEM FREEZES

15.1 The ability to back up a file also gives power to system freezes. When a system freeze is performed, the memory-mapped files are copied into a freeze directory. You can program the number of freezes that will be saved, up to 100 freezes. Each freeze is tagged with a number. The freeze marked "00" is always the latest freeze. When another freeze is performed, the "00" freeze is copied to the next unused number. For example, if there are already freezes "01" through "23" in the freeze directory, when a freeze occurs, freeze "00" will be copied to freeze "24." When the number of freezes reaches the limit specified in the ICP Configuration Utility (30 by default), it wraps back to "01" and overwrites the previous "01" freeze.

15.2 Each freeze also saves copies of the backup files. Using the example in Backup Files above, a single freeze would contain 21 CP Queue files (the 20 backup files plus the one that was actively being written). This provides over 20 MB of system data for troubleshooting. On a large System, that is approximately 30-60 minutes of data, which varies based on the size and traffic of a particular system. The technician can tune the size and number of files to capture a typical time frame as needed on a per site basis.

15.3 Note that the system is never really "frozen" when using a the CP Server, so there is no longer a problem with forgetting to unfreeze; you will not be prevented from freezing the system again.

16. RETRIEVING CP HISTORY INFORMATION

16.1 On a phone System that is not equipped with a CP Server, you would use the "Error Information" option from the Operations menu in DB Programming.

16.2 CP History Retrieval on a phone System equipped with a CP Server is entirely different. All call processing logs are stored on the CP Server's hard drive. They are only accessible via the PC and cannot be retrieved through a DB Programming session.

16.3 **Active** histories are stored in memory mapped files (**`mmf`**). Five specific histories are separately logged: message print, reset history, exception history, network link information, and call processing command history. These files, by default, are stored in **`C:\ICP`**. To make the **`mmf`** files readable, they must be converted to text (**`txt`**) format (or, in the case of the call processing command history file, to a binary format).

16.4 **Backup** history files may exist. These are **`mmf`** files that have been automatically converted to the proper format and placed in **`C:\ICP\BACKUP`**.

16.5 Call processing history logs are used by field staff, Tech Support and Engineering to troubleshoot the System. In most cases, both active and backup logs are required. Only a system freeze will produce current usable history files. When a freeze is performed, all active history information is flushed from the mmf files, converted to the proper format, and placed in the freeze directory. All backup files are then renamed and moved to the **`C:\ICP\FREEZE`** directory. The device file, Windows system event logs, application event logs, and a copy of the customer's database (**`static_memory.mmf`**) are also copied to the freeze directory.

NOTE: In a system with redundant CP Servers, each CP Server stores freeze and CP history files in its ICP directory. If one of the CP Servers fail, make sure you retrieve the files from the failed server's ICP directory.

16.6 Freezing a System equipped with a CP Server does not actually "freeze" any history files as it does in a System equipped with a traditional CPU. A picture or snapshot is taken and placed in the freeze directory. There is no need to "unfreeze" the system. Each freeze is assigned a number. There can be 100 separate freezes, depending on configuration programming (see page C-27). **`Freeze_00`** is always the most current.

16.7 Freezes cannot be "retrieved" in the normal sense. History retrieval on a non-CP Server System must be done through DB Programming as the actual history information is stored in memory chips on the CPU card. On a CP Server system, the histories are files on a hard drive. These files can be viewed directly, while still on the PC. Or, they can be copied or moved to another media.

16.8 Because a full freeze can be large (20+ MB by default), WinZip software comes pre-installed on all CP Server PCs. Freezes, all or part of, can be zipped up and copied to floppy disks if needed. WinZip allows zip files to span multiple floppy disks.

16.9 If the CP Server PC is attached to a LAN/WAN and you have access to the LAN, files can be copied from another computer on the LAN. An understanding of Windows and networking are required for this.

16.10 Each CP Server PC comes with pcAnywhere 10.5 pre-installed. This makes it possible to access the PC remotely. An internal modem is also installed by default. This allows access to the PC over modem or LAN. After connected to the CP Server PC via pcAnywhere, WinZip can be used to compress the required files and pcAnywhere's File Transfer option to transfer the Zip file to your local PC. (A full freeze of 20+ MB compresses down to about 4 MB).

17. SERVICE LOGS

17.1 The CP Server writes various service and debug information to a service log file. Some of the data found in the service log are thread status, such as starting and stopping, socket thread logons, file read and write errors, mutex errors, etc. The default name is **icp.log**. This file and its backups are kept in the working directory by default. The name and location of the service logs are programmable through the configuration utility.

17.2 When a system freeze is performed, as described on page C-12, the **icp.log** file and all of its backups are frozen along with the CP History. The backup files are named **freeze_xx_ icp_yyy.log**, where *xx* is the freeze number (01-99) and *yyy* is the backup file number. The active **icp.log** file is named **freeze_xx_icp.log**.

17.3 If using an ATM Switch, you can use the chassis Log Utility to obtain log information, in a text file format, from the remote equipment chassis. This utility, which runs independently of the Inter-Tel Call Processing and Watchdog service, is automatically loaded with the CP Server software. Because it starts up automatically whenever the PC is turned on, you do not have to log on to Windows to start it. With this utility, you can maintain up to five different log files, up to 256 Kb each, for every remote chassis.

17.4 To obtain the log information, the chassis Log Service attempts to establish an ATM SVC (Switched Virtual Circuit) socket connection to the remote equipment chassis. After a valid SVC connection has been established, the equipment chassis Log Utility sends its login (carriage return). It then waits for a response from the remote equipment chassis. If the utility does not get a response within one second, it sends another login command. If the remote chassis does not respond to ten such login attempts, the utility assumes that the connection to the remote chassis has failed and tries to re-connect. If the connections to all the remote chassis fail, then the utility waits for 20 seconds before trying to re-establish SVC connections.

17.5 If the login to the remote equipment chassis is successful, the Utility attempts to log all the information it receives from that socket connection. The log is then stored in a file named **Remote_KSU_($Number)_log.log**, where *$Number* indicate the remote equipment chassis number. This log is located on the Call Processing Server (CPS) hard drive in the **Icp\KSULog** directory, which is where the service utility is installed. If the connection to a particular remote equipment chassis is lost, logging for that particular chassis is terminated until the utility can re-connect successfully.

NOTE: If the system is not ATM-based, the chassis Log Utility will start up and then immediately terminate.

18. TCP/IP SOCKET CONNECTIONS

18.1 The CP Server can use TCP/IP socket connections to Message Print, SMDR, System OAI, or the User System Monitor.

- **Message Print and SMDR:** These are the simplest socket threads. They are transmit-only sockets (from the CP Server). The CP Server will support up to ten of each type of socket.
- **System OAI Level 2:** These are the next simplest socket threads. They receive and transmit ASCII characters. The maximum number of OAI sockets is 30. These are primarily used by the CT Gateway and Transceiver applications to provide a high speed connection to third-party OAI vendors who are not socket-aware at this time.

NOTE: System OAI Level 1 sockets are no longer supported with system versions 6.0 and later.

- **DB Programming and Voice Processing Unit:** These sockets are essentially identical in that they still require the use of the ASAI protocol to communicate. These sockets are not available to developers and are used exclusively by the Voice Processing Unit and DB Programming for high-speed access to Call Processing. See the System Manual for Voice Processing Unit setup.
- **Desktop Interface:** These sockets provide a two-way TCP/IP link that is identical to an RS232 Desktop Interface connection.
- **System Monitor:** This is a socket-based diagnostic tool, used for performing diagnostic functions that are available through various interfaces in DB Programming or from an administrative Executive Display, Professional Display, or Model 8560 Phone. It gives an Inter-Tel engineer access to various call processing structures without the aide of a technician on site to enter feature codes.

18.2 Invalid client connections are rejected and denied any access. To ensure connections are available when needed, you should verify that the socket is enabled. You can also assign a password to the socket for security purposes. To access these programming options for the CPU card sockets, click on the CPU card slot while in system DB Programming. Then open the Cabinets folder and double-click on **Sockets**. Then program the following for the desired socket:

- **Password:** Right-click on the current network password, if applicable, and select **Edit Password** to assign/change the password for the socket. Enter the old password, if one exists. Then enter the new password in the New Password and Confirm Password text boxes. Click **OK** to save your changes.
- **Enabled:** Ensure that **Yes** is checked. If it is not checked, click **No** and place a check mark in the box.

NOTE: Only the CP Server uses the socket database.

18.3 If using TCP/IP socket connections on a system, a security key with Standard Features is also required (part no. 827.9360).

19. PART NUMBERS

A. CIRCUIT CARDS

Table C-2. *Circuit Card Part Numbers*

CPU020/PCM-F/EAC	550.9025
CPU020/PCM-D/EACI	550.9026
CPU020/PCM-F/EAC A-LAW U.K.	552.9025
CPU020/PCM-D/EAC A-LAW U.K.	552.9026
CPU020/PCMA-F	550.9041
CPU020/PCMA-C	550.9042
Ethernet Adapter Card (EAC)	550.3033

B. CP SERVER PC ASSEMBLIES

19.1 The following assemblies include the CP Server PC with two Network Interface Cards (NIC) or one NIC and one ATM Adapter Card, internal modem, monitor, cable, and software.

Table C-3. *CP Server PC Assembly Part Numbers*

Mini-tower model (Non-ATM)	550.5260 (IT133 in Europe)
Mini-tower model (with OC-3 ATM Adapter Card)	550.5257 (IT404)
Mini-tower model (with UTP5 ATM Adapter Card)	550.5256 (IT134)
Rackmount model (Non-ATM)	550.9089 (IT405)
Rackmount model (with OC-3 ATM Adapter Card)	550.5259 (IT138)
Rackmount model (with UTP5 ATM Adapter Card)	550.5258 (IT137)

C. INDIVIDUAL PC PARTS

Table C-4. *Individual PC Parts Part Numbers*

Base Security Key	827.9775
Network Interface Card (NIC) - 3COM	550.5304
Network Interface Card (NIC) - Intel	550.5314
ATM Adapter (UTP5)	550.5254 (IT129)
ATM Adapter (OC-3)	550.5255 (IT131)
PCI Watchdog Card	550.5250 (N/A)
56K U.S. Robotics Internal Modem - ISA	550.5193
56K U.S. Robotics Internal Modem - PCI	550.5243
14-inch SVGA Monitor	440.5011
10 Base-T CAT5 Cross-Connect Cable (14 ft./5 m)	813.1743 (N/A)
Multi-Mode Fiber (MMF) SC/SC Cable (14 ft./5 m)	813.1772 (C158)
Multi-Mode Fiber (MMF) ST/SC Cable (14 ft./5 m)	813.1773 (N/A)
Non-ATM Redundancy Kit	828.1549

NOTE: The NIC is a network communications processor card that can transfer data packets between the CP Server PC and the System or between the CP Server PC and a LAN. For a non-ATM system, two NICs are needed for LAN installations. The RJ45 connector on the NIC connects to the LAN via a customer-provided standard network interface cable. For an ATM-based system, one NIC and one ATM Adapter Card is required.

D. CP SERVER SECURITY KEYS (ATM SYSTEMS)

Table C-5. *CP Server Security Key Part Numbers*

Version 8.0 Base Key	827.9775
Version 8.0 Redundancy Backup Key	827.9776
Version 7.0 Base Key	827.9777
Version 7.0 Networking Feature	827.9506
Version 7.0 Redundancy Feature	827.9648
Third PCMA Option (v7.0)	827.9504
Fourth PCMA Option (v7.0)	827.9526
25-Feature Units (v7.0)	827.9509
50-Feature Units (v7.0)	827.9510
75-Feature Units (v7.0)	827.9511
100-Feature Units (v7.0)	827.9512
125-Feature Units (v7.0)	827.9513
150-Feature Units (v7.0)	827.9514
Additional 25 Feature Units (v7.0)	827.9649
Upgrade 6.X ATM to 7.X ATM	827.9639

NOTE: PALs are not required for system versions 8.0 and later. See REPLACEMENT PARTS for the software license part numbers.

E. SOFTWARE

Table C-6. *Software Part Numbers*

CP Server Installation SW (U.S./U.K.) CD-ROM (Includes Manual)	827.9806
CP Server CPU020 Flash Operating System SW Floppy*	827.9100

This is also included on the Installation CD-ROM (part no. 827.9806).

F. FIRMWARE

Table C-7. *Firmware Part Numbers*

Boot ROM for CPU-512, 256, or ATM (U25)	827.8966 or 827.9268
Memory Map PAL for CPU020 256, 512, or ATM (U41)	827.8768
Boot ROM for CPUX-Slave (U25)	827.8536* or 827.8764
Memory Map PAL for CPUX-Slave (U41)	827.8463* or 827.8769

827.8536 and 827.8463 are 4.X CPUX components, but will work on the CP Server. These must be used together and cannot be mixed with 5.X chips (827.8764 and 827.8769).

20. HARD DISK MIRRORING

20.1 The CP Server PC is equipped with an ARCO Dupli*Disk2*™ to provide disk mirroring (RAID 1). The Dupli*Disk2* resides between two mirrored hard drives and is attached to the host controller on the PC's motherboard. For information on installing the ARCO Dupli*Disk2*™, refer to the installation guide included with the unit or on the CD-ROM.

21. CPU020 AND CPU020A CARD INSTALLATION PROCEDURE

CAUTION

The cards and the software components are static-sensitive. Handle the cards by the edges only and keep the components in their protective case until they are ready to be installed. Do not bend or touch the pins of the components or subject them to a static discharge. When working with the cards, use a properly terminated anti-static wrist strap. Any static discharge (no matter how small the charge) must be discharged from the body before touching the cards or components. The warranty for this equipment does not cover damage caused by static or mishandling. Cards or components damaged in such a manner will not be replaced.

1. Place the CPU020/PCM/EAC or CPU020/PCMA Card on a clean, flat, anti-static surface.
2. Check the part numbers and socket numbers printed on the labels of the software components. The software license must match the software configuration purchased by the customer. The boot ROM and memory map PAL should be labeled with the part numbers and installed in socket numbers U25 and U41 respectively. If installing a PCMA Card for an ATM-based system, make sure an ATM PAL is present on the board (U13).

   NOTE: The v7.0 ATM PAL is not compatible with v6.0 software. Make sure the ATM PAL in U13 is the correct part number (827.9683 for 7.0 and 827.9413 for 6.0). Version 8.0, however, supports any ATM PAL.

3. Verify that all components on the card are seated securely in their sockets and that no pins are bent.
4. With the equipment chassis power turned OFF, insert the CPU020/PCM/EAC or CPU020/PCMA Card in its appropriate slot in the equipment chassis.
5. *If installing a CPU020/PCM/EAC Card,* connect one end of the cross-connect cable to the connector on the EAC. Connect the other end of the cross-connect cable to the Ethernet connector on the CP Server PC.

   If installing a CPU020/PCMA Card, connect one end of the CAT5 or fibre cable to the connector on the PCMA Card. Connect the other end to the ATM Switch. Connect another cable between the CP Server and the ATM Switch.
6. Plug both the equipment chassis and the CP Server into an uninterruptable power supply (UPS) on an isolated, dedicated circuit.
7. Attach the security key to the parallel port of the PC.
8. Connect the monitor, keyboard, and mouse to the CP Server PC.
9. Turn ON the AC POWER switches on the equipment chassis, the CP Server PC, and, if necessary, the ATM Switch.

10. Upload the CPU Operating System Software (part number 827.9100):

NOTE: As long as both the first (master) and third (slave) chassis in tri-/quad-chassis installations are properly connected and on line, the system software only needs to be uploaded to the master CPU020-Q/PCM-F Card. However, if for some reason the third (slave) chassis is not connected or is off line, the system software must also be uploaded to the slave CPU020-X/ PCM-F Card.

a. Attach an RS-232 cable from an available COM port on the remote PC (not the CP Server) to the top RS-232 port on the CPU Card.

b. Select **CPU/CPC** as the Board type.

11. Select the appropriate COM Port and the desired Speed (bit per second rate). Estimated upload times are as follows:

BPS Rate	Approx. Upload Time
9600	15 minutes
19200	12 minutes
38400	6 minutes
57600	4 minutes
115,000	2 minutes

NOTE: Be sure to select a bit rate that is compatible with the PC's capabilities. For example, only PCs equipped with high-speed serial communications chips (e.g., 16550 UART) can use the 38400 and 115,000 bit rates.

a. Enter the appropriate upload source file name in the File box or use the **Browse** button to locate the file (the file will end with **`.bin`**). Make sure the complete file name appears in the box.

b. *If the System has just been powered up,* wait at least three minutes to allow the system to stabilize (`Major Reset` should be displayed) before performing the next step.

c. Select **Start** (or the **Start** option from the File menu).

d. The software is automatically uploaded into flash memory on the CPU Card. A warning message will appear if you attempt to load CPU 512 Master software without a CPU 512 Slave Card installed and communicating. You can continue the upload by clicking **Yes** or abort the upload by clicking **No**. (If you click **Yes**, you must install software on the CPU 512 Slave Card before it will function.)

NOTE: If the upload fails to start, press the reset switch on the CPU Card ***six times within three seconds***, then retry the upload. If the upload fails while in progress, the red and yellow LEDs on the CPU Card blink rapidly (ten times per second) and the PC displays the reason for the failure. If possible, correct the problem and retry the upload.

e. When the upload is complete, the system resets and the PC will display a message indicating the operation was successful. Click **OK** and then exit the program.

12. Load the CP Server Installation software as described on page C-33.

22. CONFIGURATION PROCEDURE

22.1 The Configuration Utility allows you to change various aspects of the CP Server including:

- Default directories, backup file counts, and other user-configured information
- Fields necessary for the proper operation of the CP Server (For example, if the system is a quad system, then the quad configuration check box needs to be enabled.)
- IP Address of the remote chassis and Listening Socket PC
- MAC Address of the PCMA cards for ATM systems

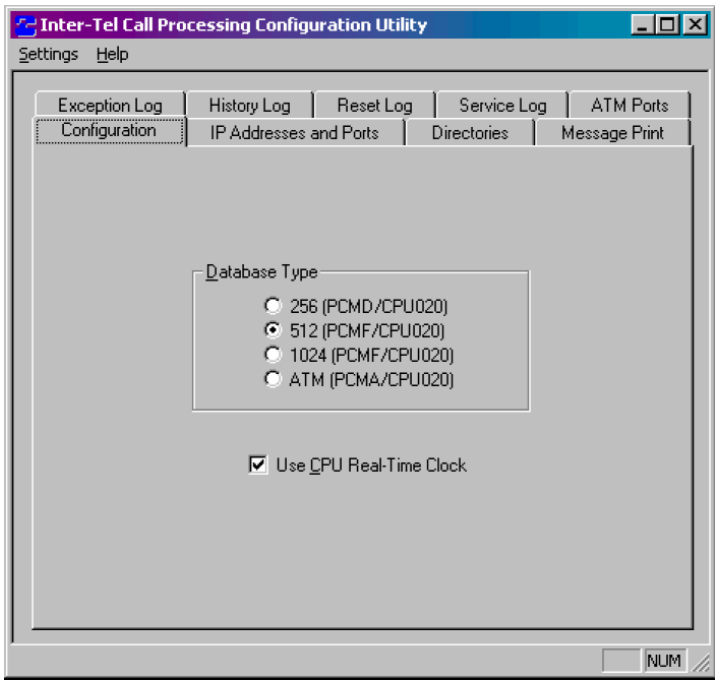
NOTE: You must restart the call processing unit to apply any changes made to the configuration except the MAC Addresses of the PCMA cards. However, because this will drop all calls in progress, do not restart the CPU during normal business hours.

22.2 The default path to the utility is: **`C:\ICP\Utilities\icpconfig.exe`**.

22.3 The Configuration Utility automatically runs during the CP Server installation. To make changes after the software is installed, double-click on the desktop icon. Or, select Start - Programs - Inter-Tel - Call Processing - **ICP Configuration Utility**. You can then use the Configuration Utility to program the necessary fields, as described below and in the following pages.

A. CONFIGURATION TAB

22.4 With the Configuration tab, you can program the device limit and specify the point-of-reference for the CP Server clock.



- **Database device limit:** Select the database device limit. Your options are dual-chassis (256), quad-chassis (512), CP Server (1024), or ATM. This field is available when you first install the software (or when the **`static_memory.mmf`** file has been deleted). If you select the 1024 database option, the maximum number of devices the database can support is increased to 1024. However, because the chassis cannot support 1024 devices, the total number of devices is limited by the hardware capacities. By default, this is set to 512.

NOTE: You cannot revert to a smaller database. For example, if you currently have a 1024 database configured, you cannot revert to a 512.

- **Use CPU Real-Time Clock:** Select this option to have the CP Server set and read the date and time from the remote chassis rather than from the PC Clock. If disabled (default), the CP Server reads the date and time from its PC Clock. If you select this option, make sure the battery strap is connected on the first PCMA board (ATM Systems only).

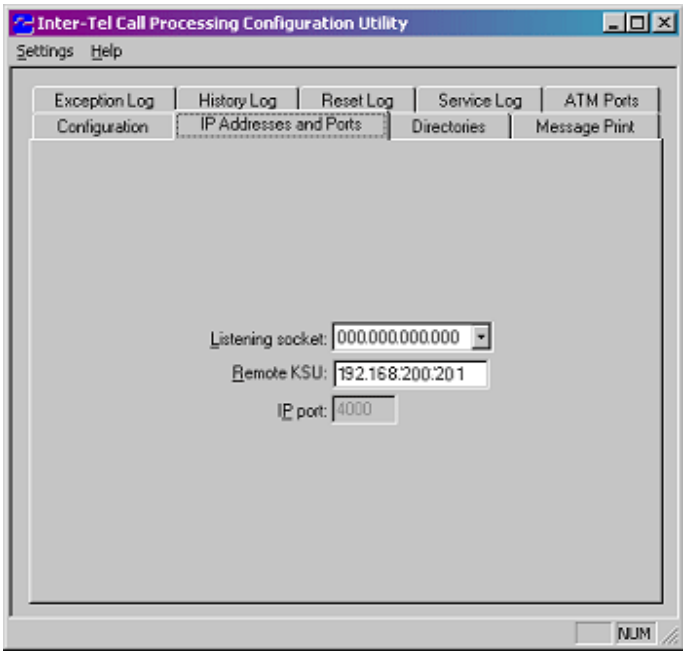
NOTICE

If the PC is currently synchronizing its time with an external clock (e.g., using Windows Time Service or some other utility), do *not* enable this option. Using a synchronization utility with this option enabled may result in unpredictable clock information. For best results, have the PC synchronize the time and disable the **Use CPU Real-Time Clock** option for the CP Server.

APPENDICES

B. IP ADDRESSES AND PORTS TAB

22.5 The IP Addresses and Ports tab allows you to configure the listening socket and remote chassis address to be used when the CP Server PC is installed on the customer's LAN.



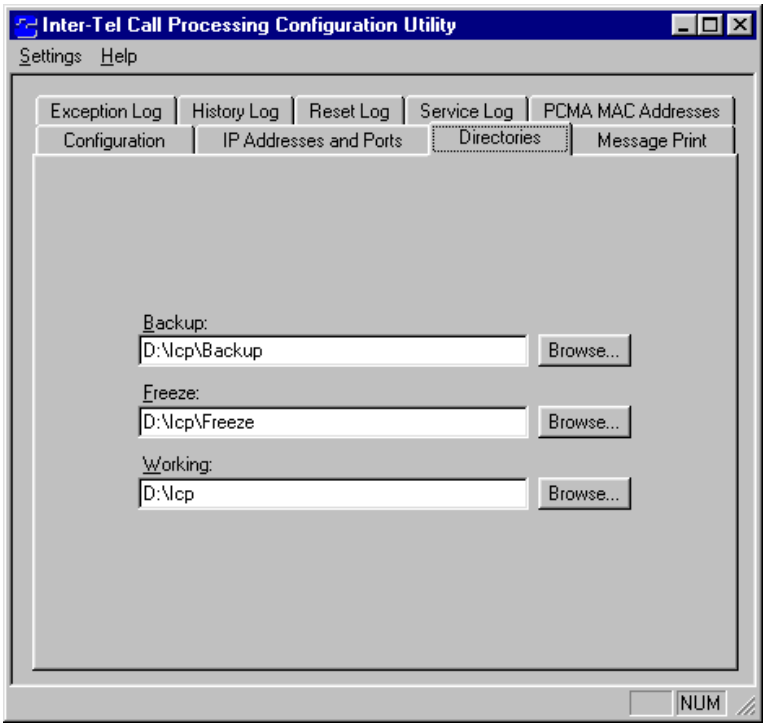
- **Listening Socket:** Enter the IP Address that the CP Server will use to processes incoming socket requests (e.g., voice mail, OAI, or DB Programming) from the LAN. Set this value to `0.0.0.0` (Any IP Address), which allows the listening socket to listen for a connection from the IP Addresses associated with any of the NICs.
- **Remote KSU:** Enter the IP Address of the equipment chassis the Inter-Tel Call Processing service is controlling. *Do not* change the default value (`192.168.200.201`).

> **NOTE:** This field is not available for ATM-based systems.

- **IP Port:** Indicates the port used by the CP Server's master listening socket. The CP Server expects all incoming TCP/IP connection requests to be made using this port. This field is shown for information purposes only and cannot be changed.

C. DIRECTORIES TAB

22.6 The Directories tab allows you to specify where the backup and CP history information is stored on the CP Server PC. Directory paths can be up to 200 characters long.



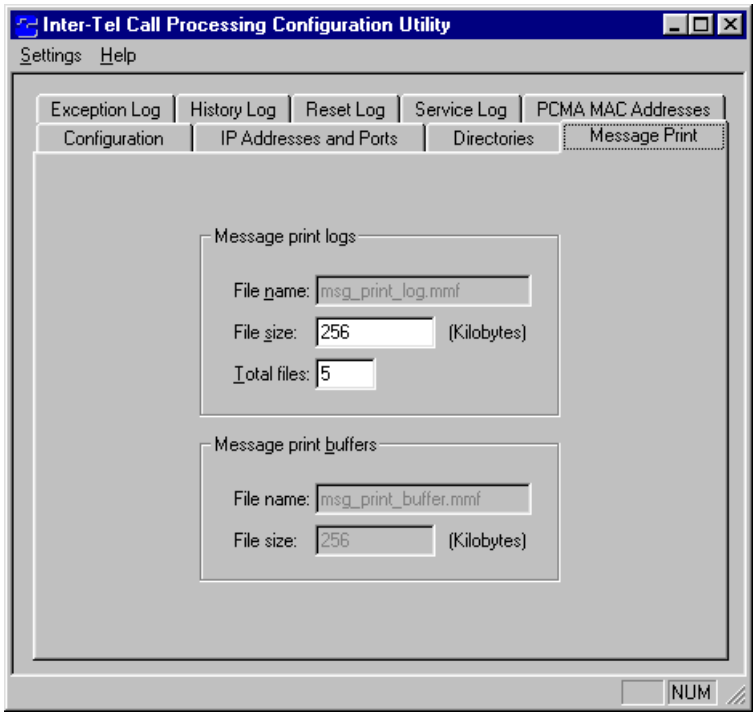
- **Backup:** Enter the system backup file directory, or click **Browse** to invoke the standard browse window. This is the directory where all the backup history, message print, exception, and reset files are kept. The default is **`C:\ICP\Backup`**.
- **Freeze:** Enter the system freeze directory, or click **Browse** to invoke the standard browse window. This is the directory where all the files are kept when a system freeze is performed, either manually or automatically. The default is **`C:\ICP\Freeze`**.
- **Working:** Enter the system working directory, or click **Browse** to invoke the standard browse window. This is the directory where the CP Server keeps its memory-mapped files. The default path is **`C:\ICP`**.

NOTE: *Do not* store any of this information on a network/shared drive.

APPENDICES

D. MESSAGE PRINT TAB

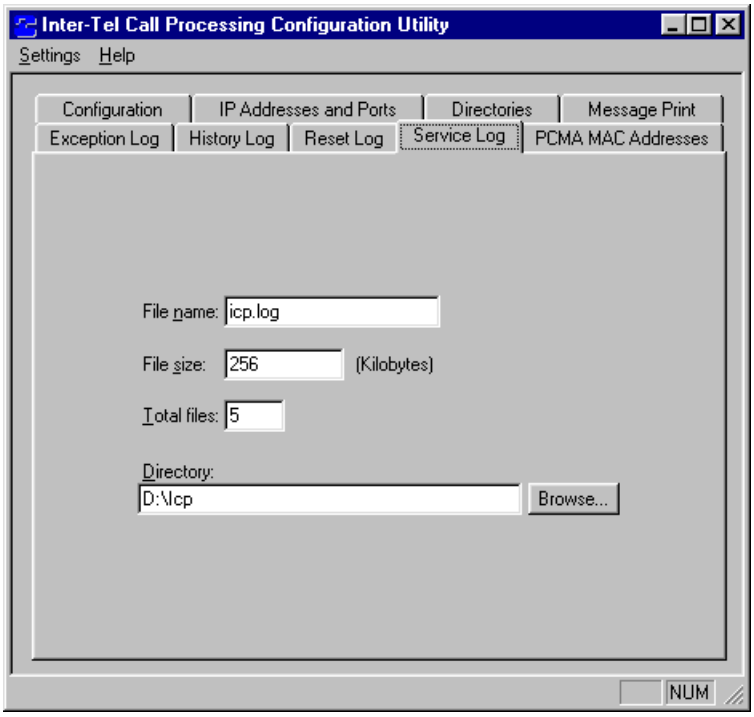
22.7 The Message Print tab contains fields that determine the file sizes and storage capacity for the Message print feature.



- **Message Print Logs**
  - — *File Name:* Specifies the name of the system message print log file. It is an internal buffer used to store messages after they are printed via the message print port. The buffer is saved when a freeze is performed. The information is shown for reference only and cannot be changed.
  - — *File Size:* Enter the size of the system message print log file in kilobytes (100-4,200,000). This determines how large the **`message_print_log.mmf`** file may grow before it is backed up in the backup directory (see Directories tab). The default is 256 KB.
  - — *Total Files:* Enter the number (1-99) of system message print log backup files that will be kept in the backup directory. The default is five files.
- **Message Print Buffer**
  - — *File Name:* Specifies the name of the system message print buffer file. This is an internal buffer used for storing messages that are waiting to be printed via the message print port. The information is shown for reference only and cannot be changed.
  - — *File Size:* Specifies the size of the system message print buffer file in kilobytes (100-4,200,000). It determines how large the **`message_print_buffer.mmf`** file may grow before pending message print messages are lost. This is shown for reference only and cannot be changed.

E. SERVICE LOG TAB

22.8 The Service Log tab fields determine the file sizes and storage capacity for the service logs.



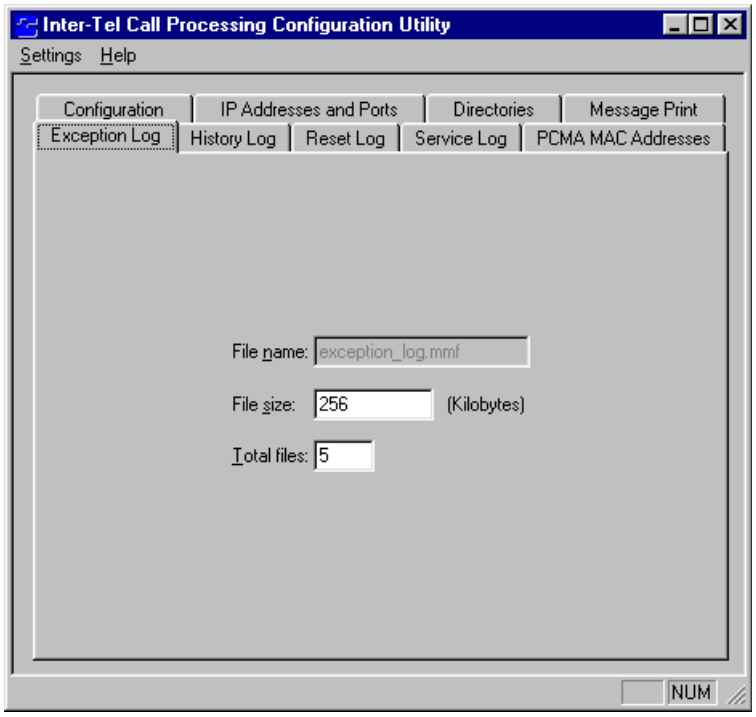
- **File Name:** Enter the name, up to 50 characters, of the system service log file. The default name is **`icp.log`** and is stored by default in the service log directory.
- **File Size:** Enter the size of the system service log file in kilobytes (100-4,200,000). It determines how large an individual log file may grow before it is backed up in the service log directory. The default size is 256 KB.
- **Total Files:** Enter the number (1-99) of system service log backup files that will be kept in the service log directory. The default is five files.
- **Service Log Directory:** Enter the directory path and name, up to 200 characters, where the service log files are kept, or click **Browse** to invoke the standard browse window. By default, this is the **`C:\ICP`**.

NOTE: *Do not* store the log files on a network/shared drive.

APPENDICES

F. EXCEPTION LOG TAB

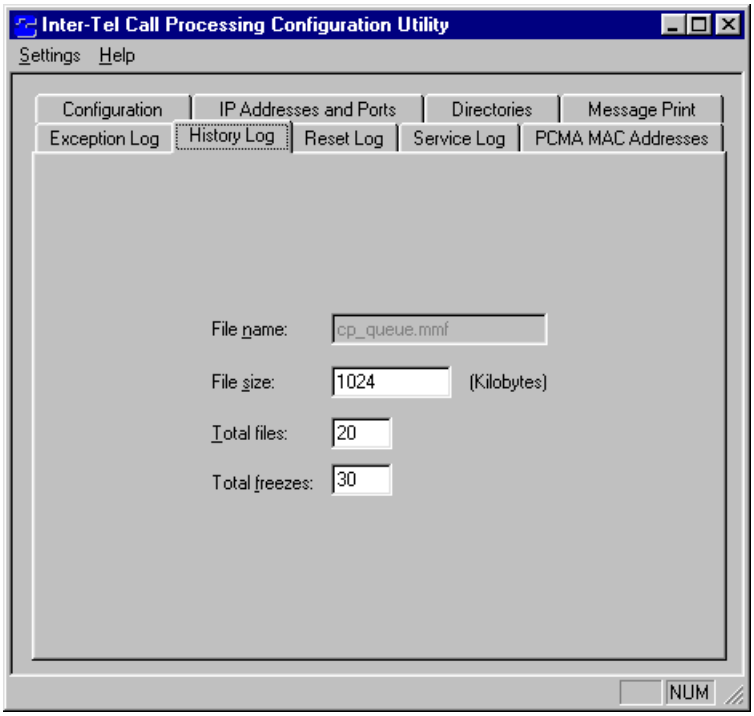
22.9 The Exception Log fields determine the file sizes and storage capacity for the exception logs.



- **File Name:** Specifies the name of the system exception log file. The information is shown for reference only and cannot be changed.
- **File Size:** Enter the size of the system exception log file in kilobytes (100-4,200,000). It determines how large the **`exception_log.mmf`** file may grow before it is backed up in the backup directory (see the Directories tab). The default size is 256 KB.
- **Total Files:** Enter the number (1-99) of system exception log backup files that will be kept in the backup directory. The default is five files.

G. HISTORY LOG TAB

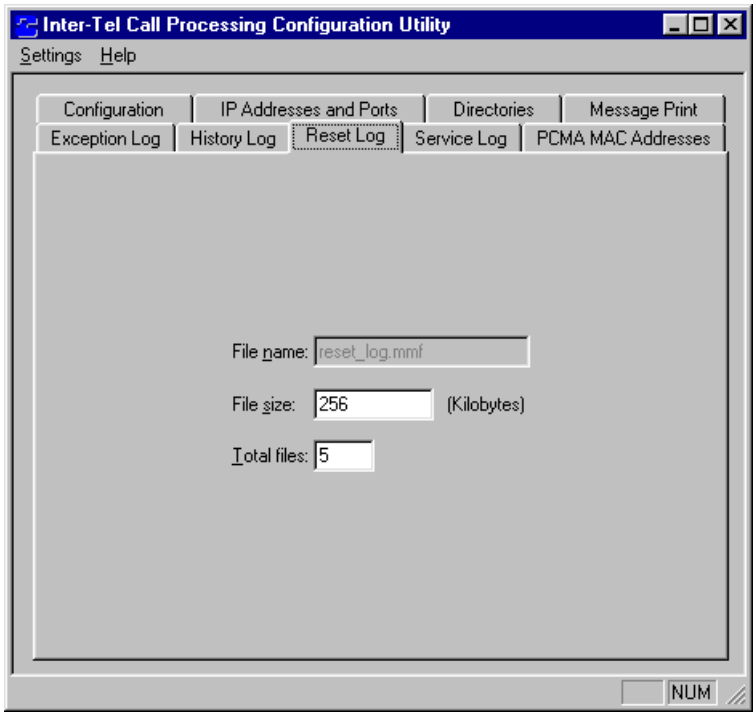
22.10 The History Log tab allows you to specify the file sizes and storage capacity for the CP history logs.



- **File Name:** Specifies the name of the system CP queue log file. The information is shown for reference only and cannot be changed.
- **File Size:** Enter the size of the system CP queue file in kilobytes (100-4,200,000). This determines how large the **`cp_queue.mmf`** file may grow before it is backed up in the backup directory (see the Directories tab). The default size is 1024 KB.
- **Total Files:** Enter the number (1-99) of system CP queue backup files that will be kept in the backup directory. The default is 20 files.
- **Total Freezes:** Enter the number (1-99) of freezes that will be saved in the freeze directory. The default is 30 freezes.

H. RESET LOG TAB

22.11 The Reset Log tab allows you to specify the file sizes and storage capacity for the reset logs.

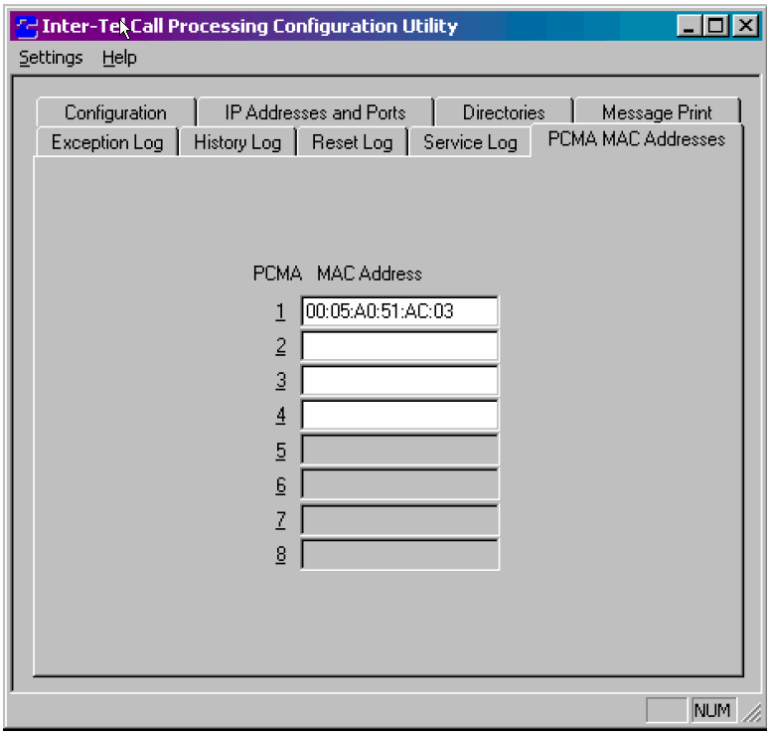


- **File Name:** Specifies the name of the system reset log file. The information is shown for reference only and cannot be changed.
- **File Size:** Enter the size of the system reset file in kilobytes (100-4,200,000). This is the limit for determining how large the **`reset_log.mmf`** file may grow before it is backed up in the backup directory (see the Directories tab). The default size is 256 KB.
- **Total Files:** Enter the number (1-99) of system reset log backup files that will be kept in the backup directory. The default is five files.

I. PCMA MAC ADDRESSES TAB

22.12 The PCMA MAC Addresses tab contains the fields used to associate a specific chassis's PCMA with a number.

NOTE: This tab is only available for an ATM-based system.



22.13 For each PCMA card (four maximum), enter the 6-byte MAC address, with each byte separated by a colon (:). Call Processing then uses this MAC address, which is located on the faceplate of the PCMA board, to associate the PCMA board with the MAC address of the chassis programmed in the database.

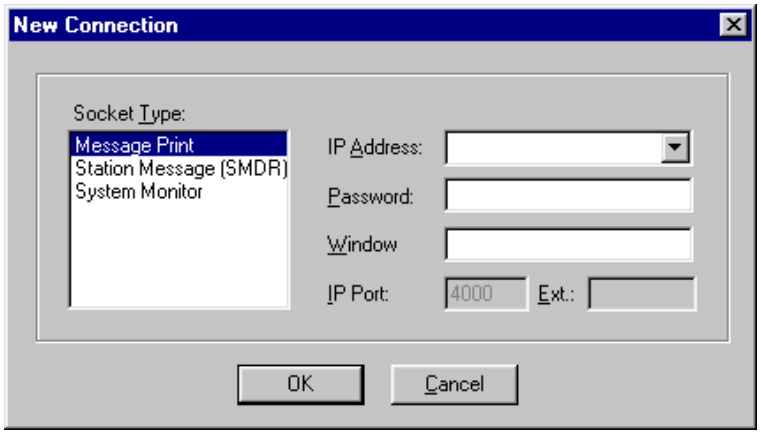
APPENDICES

23. DIAGNOSTICS MONITOR UTILITY

23.1 The Diagnostics Monitor (Diag Mon) utility can be used for monitoring the following types of socket connections:

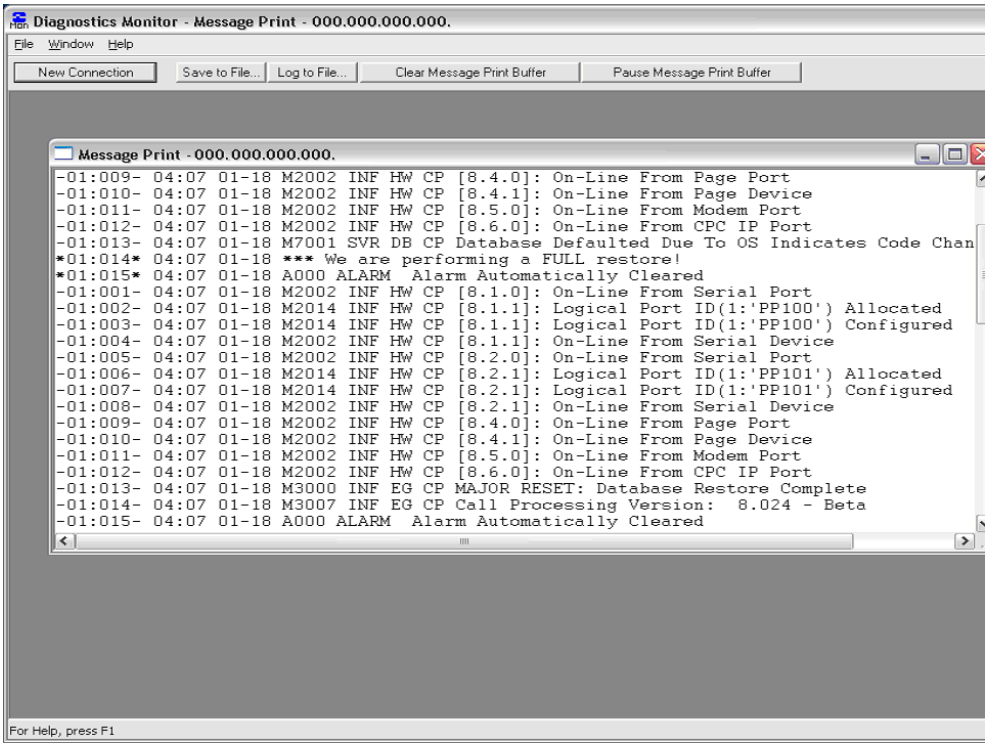
- **Message Print and SMDR:** These are the simplest socket threads. They are transmit-only sockets (from the CP Server). The CP Server will support up to ten of each type of socket.
- **System Monitor:** This is a socket-based diagnostic tool, used for performing diagnostic functions that are available through various interfaces in DB Programming or from an administrative Executive Display, Professional Display, or Model 8560 phone. It allows an Inter-Tel engineer access to various call processing structures without the aide of a technician on site to enter feature codes.

23.2 When Diag Mon is first started, you the New Connection Dialog appears, as shown below. This dialog facilitates connection to a CP Server.



- **Socket Type:** Select the desired socket type. Your options are Message Print, SMDR, and System Monitor. When you click **OK**, the highlighted socket type will attempt to connect to the CP Server.
- **IP Address:** Enter the full IP Address or fully qualified computer name. Diag Mon will take whatever name is entered and attempt to resolve that name to an IP Address. The IP Address entered here should be the IP Address of the NIC on the CP Server that connects to the LAN.
- **Password:** Enter the password for the socket type selected. Passwords are programmed in DB Programming in the Sockets Folder (**`System/Cabinets/Sockets`**). If the socket type is not enabled, then the connection will be refused.
- **Window Title:** Enter the socket window title that specifies the type of socket and the IP Address to which the CP Server is connected. Although this field automatically updates based on the selected socket type and the entered IP address, you can enter any string to replace the default title. For example, if you are monitoring message print for several systems, you might choose titles such as “Los Angeles—Message Print” and “Chicago—Message Print.” These titles may be clearer to the user than the IP Address.
- **IP Port:** Specifies IP Port 4000 and cannot be changed.

NOTE: The **Ext** field is currently not used and is grayed out.



23.3 When you click **OK**, the Diag Mon establishes the connection to the CP Server. When main screen appears, as shown above, it has the following menus:

- **File Menu:** When there is no active socket connection open, the only selectable menu options are New Connection, Print Setup, and Exit. Selecting New Connection or using **CTRL-N**, will present the New Connection Dialog to the user. When an active socket connection exists, the Save to File, Log to File, and Close Connection options are available.

  NOTE: Printing is currently not supported.

  — *New Connection:* Starts the New Connection dialog described on the previous page. This can also be accessed using **CTRL+N**.

  — *Save to File:* Saves the contents of the active socket connection to a file for later review. This can also be accessed using **CTRL+S**.

  — *Log to File:* Saves the current contents of the active socket connection to a file and logs any future inputs from the socket connection into the file. Selecting this option again will turn off the logging feature. If logging is active for a socket connection, a check mark will display next to the menu option. This can also be accessed using **CTRL+L**.

  — *Close Connection:* Closes the connection of the active socket.

  — *Print and Print Setup:* Not supported.

  — *Exit:* Terminates the Diag Mon application.

- **Window Menu:** The Window menu has no selectable options when no sockets are active. When at least one socket is active, the following options are available.
  - — *Cascade:* Arranges all windows in a cascade fashion from the upper left to the lower right and back again.
  - — *Tile:* Arranges all windows to be visible on the screen at one time. The size of the each window is dependent on the number of active connections.
  - — *Arrange Icons:* Not used in this application.
  - — *Active Window List:* Contains a list of all active windows in the application. Select a window to make it the active window.
- **System Monitor Menu:** This menu option is only available when a System Monitor socket is active. Because only trained personnel should use the options in this menu, the menu options are not discussed in this document.
- **Help Menu:** the Help menu has the following options:
  - — *About Diag Mon:* Displays a dialog box with version and copyright information about Inter-Tel Diagnostics Monitor.
  - — *How To Use Diag Mon:* Displays the online help file.
- **Context Menu:** You can access the Context Menu by right-clicking on a window. The Context menu provides quick access to the following four basic functions that you can perform on an active socket connection:
  - — *Clear Buffer:* Clears the entire contents of an active socket window. Any data currently logged to a file will remain logged in that file.
  - — *Pause Scrolling:* Pauses the scrolling of the buffer to allow you to look over the contents of the buffer. (On occasion, data may be coming out of the socket faster than you can read the data.) When Pause Scrolling is enabled, a check mark will appear next to the menu item.
  - — *Save to File:* Saves the contents of the active socket connection to a file for later review. This can also be accessed using **CTRL+S**.
  - — *Log to File:* Saves the current contents of the active socket connection to a file and logs any future inputs from the socket connection into the file as well. Selecting this option again will turn off the logging feature. If logging is active for a socket connection, a check mark will display next to the menu option. This can also be accessed using **CTRL+L**.

24. INTER-TEL CALL PROCESSING SERVICE UPDATE

24.1 To update the Inter-Tel Call Processing service, if necessary:

1. Perform a database save.

NOTE: Reset and exception log data are lost between updates. If maintaining this information is necessary, perform a system freeze prior to stopping the service. The freeze will have the latest exception and reset log data in it. Copy those files for safe-keeping.

2. Install the new software.
3. Set the Software Watchdog service to "automatic."

CAUTION

When the software is installed or upgraded, the Software Watchdog service is set for "manual" startup by default. It is very important for you to change the startup type to "automatic" after the new software is installed and is working properly.

4. Perform a database restore.

25. INSTALL INTER-TEL CP SERVER SOFTWARE

NOTE: Installation software is included on the Inter-Tel CP Server CD-ROM.

25.1 To install Inter-Tel CP Server software:

1. Run the setup file located in the CP Server PC Software folder.
2. Install the software to the default location (**`C:\ICP`**).

NOTE: If the C: drive is not formatted, cancel the installation and format the drive.

3. If you plan on using SNMP, review the SNMP section of this document (see page C-40) prior to selecting **Yes** when prompted to install SNMP. Otherwise, select **No** to continue. (You can add the SNMP option later by re-installing the Inter-Tel CP Server software.)
4. Select either **AIC card** or **PCI Watchdog** when prompted to select the watchdog hardware device, if using a PC supplied by Inter-Tel. If using a customer-provided PC, choose the appropriate device.
5. Set the listening socket to **0.0.0.0** while in the ICP Configuration Utility screen. This value allows listening on the installed NIC(s).

6. Select the desired database device limit (256, 512, 1024, or ATM). This value is determined by the card and the software flashed on the card. Because an Inter-Tel CP Server can either be a 256 or 512, you must configure the software for the correct size. The default is 512.

NOTE: When prompted to install Simple Network Management Protocol (SNMP), select **No** unless you have read the information starting on page C-40. You can install SNMP later if necessary.

7. Reboot the PC, and log into Windows.

CAUTION

When the software is installed or upgraded, the Software Watchdog service is set for "manual" startup by default. It is very important for you to change the startup type to "automatic" after the new software is installed and is working properly.

26. 10 BASE-T CAT5 CROSS-CONNECT CABLE PINOUT

26.1 A 14-foot 10 Base-T CAT5 Cross-Connect cable is supplied with the CP Server PC. However, if you need a longer cable, make sure that the pinout is as follows.

Table C-8. *10 Base-T CAT5 Cross-Connect Cable Pinout*

Pin	Pin
1	3
2	6
3	1
4	4
5	5
6	2
7	7
8	8

26.2 The cable has an 8-pin/8-conductor modular connector on each end.

NOTE: Additional cables are provided for ATM systems based on the type of connector (UTP5 or OC-30) used.

27. CLOCK CONFIGURATIONS FOR THE SYSTEM T1 AND E1 CARDS

NOTE: The following recommended configurations are optimized for fail-safe operation of the System. Although not required, failure to follow these recommendations can lead to intermittent voice degradation due to slips and clock synchronization problems and voice channel resource blocking. Although the new T1/E1/PRI card has no slot dependency, this document still recommends following the same guidelines as the previous T1/PRI Card to preserve backward compatibility with old cards and systems.

27.1 In the following guidelines, "Slave-to-Public Network," "Slave-to-Private Network," and "Master-to-Private Network" refer to the phone system's T1 or E1 cards (T1C, T1/PRI, E1C, E1/PRI, or T1/E1/PRI Card) and their software/hardware configuration.

A. QUAD CP SERVER NETWORKING HUB

27.2 The CP Server supports up to 28 PRI cards. Other System CPUs allow up to 28 T1 cards, but only eight of those can be PRI-equipped.

27.3 This configuration assumes that you will be installing a networking hub with eight or more PRI-equipped T1 or T1/E1 cards.

NOTE: Any non-T1 cards should be placed in a position where they would be least likely to be moved if additional T1s are added to the system.

B. SINGLE-CHASSIS CONFIGURATION

27.4 A single chassis can only support seven T1 cards. The T1 cards should be placed in descending order staring with slot 7. The table below shows a single chassis configuration. The "SLOT" row indicates the slot number on the equipment chassis. The "ORDER" row indicates the order in which T1 cards should be added to the system.

Table C-9. *Single-Chassis Configuration*

	Chassis 1							
Slot	1	2	3	4	5	6	7	8
Order	7	6	5	4	3	2	1	CPU

27.5 Slave-to-Public Network cards should be installed first, followed by Slave-to-Private Network cards, and finally, Master-to-Private Network cards. The clock configuration should start with the clock providing reference in slot 7, the next clock reference in slot 6, and so on with the card in slot 1 as the last card to provide reference.

C. DUAL-CHASSIS CONFIGURATION

27.6 A dual-chassis system can only support 14 T1 cards. The T1 cards should be placed in descending order starting with slot 7 in chassis 1 and ending with slot 4, as shown in the table below. Then cards should be added to chassis 2 starting in slot 9 and ending in slot 12. After that, cards should be alternated from chassis 1 to chassis 2 to balance the voice channel usage.

27.7 Any Slave-to-Public Network cards should be installed first followed by any Slave-to-Private Network cards. Any Master-to-Private Network cards should be installed last.

27.8 If you are expanding from a single chassis to a dual chassis, you should move the T1 cards from slots 1, 2, or 3, into slots 9, 10, and 11 to help balance the voice channel usage. Also, as many of the first four positions should be Slave-to-Public Network cards to improve clock switching in the event of losing the T1 span providing reference.

Table C-10. *Dual-Chassis Configuration*

	Chassis 1								Chassis 2							
Slot	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
Order	13	11	9	4	3	2	1	CPU	5	6	7	8	10	12	14	CPU

D. TRI-CHASSIS CONFIGURATION

27.9 A tri-chassis system can support up to 21 T1 cards. The T1 cards should be placed in descending order starting with slot 7 in chassis 1 and ending with slot 4. Then cards should be added to chassis 3, starting in slot 23 and ending in slot 19. Next, cards should be added to the first 4 slots in chassis 2. After 4 T1s have been added to each chassis, then alternate the addition of new T1 cards starting with chassis 1, then 3, and finally 2. Also, any Slave-to-Public Network cards should be installed first followed by any Slave-to-Private Network cards. Any Master-to-Private Network cards should be installed last.

27.10 If you are expanding from a single- or dual-chassis configuration, you should redistribute existing T1 cards to follow this configuration to help balance voice channel usage.

Table C-11. *Tri-Chassis Configuration*

	Chassis 1								Chassis 2							
Slot	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
Order	19	16	13	4	3	2	1	CPU	9	10	11	12	15	18	21	X

	Chassis 3							
Slot	1	2	3	4	5	6	7	8
Order	20	17	14	8	7	6	5	CPU

E. QUAD-CHASSIS CONFIGURATION

27.11 The same principles follow for a quad-chassis system as they did in the dual-chassis configuration. Cards should be added to the left chassis before being added to the right chassis.

27.12 Again, if you are expanding to a quad chassis configuration, you should redistribute existing T1 cards to follow this configuration to help balance voice channel usage.

Table C-12. *Quad-Chassis Configuration*

	Chassis 1								Chassis 2							
Slot	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
Order	25	21	17	4	3	2	1	CPU	9	10	11	12	19	23	27	X

	Chassis 3								Chassis 4							
Slot	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
Order	26	22	18	8	7	6	5	CPU	13	14	15	16	20	24	28	X

28. WINDOWS EVENT LOGGING

28.1 The ICP service registers itself with the Windows Event Viewer as an event source. This means that the ICP service can log events with the Application Event Log using Event IDs, and full string insertion capabilities allowed by the Event Viewer. The ICP service logs information specific to the service portion of the startup procedure. Currently, there are only six event IDs. The following is a break down of these events.

Source: Inter-Tel Call Processing
Type: INFORMATION
ID: 100
File: %System32%\ICPMsg.dll
Message: The Inter-Tel Call Processing service has %1
Strings: %1 = “started” or “stopped”
Cause: The service was either started or stopped. This could be done internally by the service itself (i.e., major reset), the Inter-Tel Watchdog Service, or by the end user starting and stopping the service via the Control Panel Services applet.

Source: Inter-Tel Call Processing
Type: ERROR
ID: 101
File: %System32%\ICPMsg.dll
Message: Could not open the Registry Key %1
Strings: %1 = <A Registry Key Name>
Cause: A registry key required for the proper operation of the service could not be opened. This usually means that the service was not properly registered.

Source: Inter-Tel Call Processing
Type: ERROR
ID: 102
File: %System32%\ICPMsg.dll
Message: Invalid Service Request
Cause: The system was issued an invalid service request. The only valid service requests are Stop, Shutdown, Pause, Continue, and Interrogate. The service ignores Pause, Continue, and Interrogate, but does not report this error for those requests.

Source: Inter-Tel Call Processing
Type: ERROR
ID: 103
File: %System32%\ICPMsg.dll
Message: Service Control Manager could not be opened
Cause: The service could not open a valid handle to the Service Control Manager.

Source: Inter-Tel Call Processing
Type: ERROR
ID: 104
File: %System32%\ICPMsg.dll
Message: The service could not be %1
Strings: %1 = “opened” or “created” or “deleted”

Type: ERROR
ID: 105
File: %System32%\ICPMsg.dll
Message: The Service Control Handler could not be installed

29. WINDOWS EVENT LOG FILE FREEZES

29.1 Windows provides an Event Log API. This API provides access to the raw data used by the Event Viewer to format the information. Using this API, the Applications and System Event Log raw data (with basic formatting) are saved whenever the system history is frozen. Occasionally, Inter-Tel Engineering will need to use these files for troubleshooting. The files are named **`freeze_xx_event_log.txt`**, where *xx* is 01-99 and *event* is either the "Applications" or "System" event file name.

29.2 The format for the information is as follows:

Record: 045
Source: ICP
Type: INFORMATION
Time: Thu Sep 28 02:00:02 2000
ID: 100
File: C:\WINNT\System32\IcpMsg.dll
Message: The Inter-Tel Call Processing service has started.

29.3 Where:

- **Record Number:** Specifies a sequential number representing this event's place in the overall log.
- **Source:** Identifies the name of the service or application that has registered itself with Windows as an event source.
- **Type:** Indicates if the event is informational, an error, or a warning.
- **Time:** Specifies the time that this event was logged.
- **ID:** Indicates the value, which is specific to the source, that is used to look up internal string tables.
- **File:** Identifies the name of the file that contains the internal string tables.
- **Message:** Specifies the actual formatted log string.

30. SIMPLE NETWORK MANAGEMENT PROTOCOL (SNMP)

30.1 The CP Server is designed to be used on a LAN/WAN where it can be monitored and managed from centralized network management stations, administered by MIS personnel, using Simple Network Management Protocol (SNMP).

30.2 SNMP on the version 5.2 and later CP Servers can be used to monitor and display the following information:

- **System Alarms** including all major and minor System and Voice Processing Unit alarms. It can also display warning messages and alarms from the operating system, hardware and database.
- **Error Statistics** are updated hourly and daily for the T1 cards.
- **Configuration information** includes System version, Windows version and System status.
- **Performance information** that includes network performance, status, statistics, capacity information etc.

30.3 The Simple Network Management Protocol (SNMP) is a standard that defines a method for remotely monitoring the performance of systems (routers, bridges, servers, etc.) connected to a Local Area Network (LAN) or Wide Area Network (WAN). The three components of SNMP are the Management Information Base (MIB), the Agent, and the Manager.

- The **Manager system** executes network management operations that monitor and control agent systems. The implementation of these network management operations is called a **manager**.
- The **Agent system** is a device, such as a router, bridge, or server, that has an **SNMP agent** responsible for performing the network management operations requested by the manager.
- The information on the agent is known as a **Management Information Base (MIB)**. It is a concept that encompasses the configuration and status values about the system being monitored. It contains a collection of standardized and non-standardized objects. MIBs are defined using an Internet-standard language called the Structure of Management Information, or SMI. A few examples of the types of information contained in a MIB are: the full name and version of the system, physical location of the managed node, the total number of bytes transmitted out of the interface, T1 Errors, capacity remaining in UPS batteries. To access the value of a MIB object, an SNMP manager sends a request to the agent representing the desired instance of the object. The request message contains MIB information (an OID – Object Identifier) that lets the agent identify the specific objects.

30.4 The Simple Network Management protocols communicate management information between the manager and the agent. SNMP enables the following:

- A manager to retrieve (**get**) information from the agent. The manager sends requests for information to the agent and the agent sends back replies containing the requested information.
- A manager to alter (**set**) information on the agent.
- An agent that sends an unsolicited response when it detects an alarm. The alarm transmission is referred to as a “trap.” Traps alert the manager of changes that occur on the agent without an explicit request from the manager.

30.5 The SNMP Installation assumes that the SNMP service is already installed on the CP Server server.

30.6 To verify/install this on a CP Server running Windows NT:

1. Open Control Panel, and double-click the **Network** icon.
2. Select the Services Panel in the Network screen and verify that the SNMP service is listed.
3. *If the SNMP service is present,* exit the Control Panel. No further programming is necessary.

   If the SNMP service is not present, click **Add** and choose SNMP service. Continue to the next step.
4. Double-click the **SNMP Service** in the Network Window/Services tab to set the appropriate properties once the SNMP service is installed. The Agent tab on the SNMP Service window should contain the Contact Name and Location (IP Address). If you need to view the traps from a remote location, then add the address of the remote machine to the Trap tab. (To do this, go to the Trap tab and enter the Community Name that will be public. Then, under the Trap Destination, add the IP address of the remote PC where you want the trap sent.)

NOTE: You need to re-install the Windows NT service pack after adding the SNMP service.

30.7 To verify/install this on a CP Server running Windows 2000:

1. Open Control Panel, and double-click the **Add/Remove Programs** icon.
2. Verify that the SNMP service is listed.
3. *If the SNMP service is present,* exit the Control Panel. No further programming is necessary.

   If the SNMP service is not present, click **Add/Remove** and choose Management and Monitoring Tools. Continue to the next step.
4. Select SNMP Service and follow the installation steps.
5. Double-click the **SNMP Service** in the Services tab to set the appropriate properties once the SNMP service is installed. The Agent tab on the SNMP Service window should contain the Contact Name and Location (IP Address). If you need to view the traps from a remote location, then add the address of the remote machine to the Trap tab. (To do this, go to the Trap tab and enter the Community Name that will be public. Then, under the Trap Destination, add the IP address of the remote PC where you want the trap sent.)

30.8 The three main components that are specific to SNMP Installation are as follows

- **SNMP Agent DLL (ICPSnmp.dll):** This is the agent specific to Inter-Tel SNMP. During service initialization, the CP Server SNMP service (**`snmp.exe`**) consults the registry to load the agent DLL. This registry is located under **`HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\Services\SNMP\Parameters\ExtensionAgents`**. The SNMP installation program takes care of loading ICP SNMP DLL to this registry. Thus, whenever Windows is started, the snmp.exe service will load the **`ICPSnmp.dll`**.

APPENDICES

- **COM Server (GenericSnmp.exe):** This is a server that serves COM interfaces and stores all the SNMP objects relating to the CP Server SNMP. Call Processing and the Inter-Tel SNMP Watchdog service constantly update the values stored in GenericSnmp.exe throughout the lifetime of the server. To allow several processes to share the information served by the GenericSnmp.exe server, it is configured to run under the **`it_GenericSnmp`** administrator account created by the installation software.
- **Inter-Tel SNMP Watchdog Service (ICPSnmpWatchdog.exe):** This is a service that takes care of SNMP table initialization. It can be located by double clicking on the Services Icon on the Control Panel. It is set to automatic startup mode so that it is automatically started every time Windows starts up.

30.9 The modules—SNMP Agent DLL, Inter-Tel SNMP Watchdog Service and Inter-Tel Call Processing—are implemented in such a way that it is unimportant as to what order the three cooperating modules start. Each module will wait until the required initialization has occurred before performing its tasks. Thus, you can start the three modules in any order, and they will work correctly.

30.10 To view all the SNMP alarms/ objects on the CP Server system, you need a management console like HP Openview. However, the MIB (Management Information Base) specific to the CP Server (**`ICPSnmp.mib`**) needs to be loaded to the management console to access any objects specific to the CP Server. The ICPSnmp.mib is located in the C:\ICP directory on the CPS/ATM system. Also, add the following sentence at the end of the paragraph: For detailed information about the trap.conf file, see Knowledge Base answer ID 2672. Also, to log the alarms into the respective categories on the management console the configuration file **`trap.conf`** on HP Openview needs to replaced with the Inter-Tel modified version of the file.

Appendix D — CPC64/CPU64 Configuration

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1. OVERVIEW

1.1 You can use specialized hardware to configure a smaller Inter-Tel system. Sometimes referred to as the "Axxess 64," this smaller system uses the standard Inter-Tel equipment chassis, cards, and software. It also adds three new components:

- **CPC64 or CPU64:** These CPU cards support up to 64 devices in a single-chassis configuration. Each has all of the features of the CPU128 card. You can use multiple T1/E1 cards with the CPC64 or CPU64 (one of which can be PRI-equipped). It can be used with any of the power supplies or voice mail options and it can be used as a node in an Inter-Tel network.

  NOTE: Versions 8.0 and later require a CPC for CPU64 systems.

- **Embedded Voice Mail Card (EVMC) and DSPM Card:** When installed in the equipment chassis, this OS/2-based voice mail card, equipped with one DSPM Card, will provide four Voice Processing Unit ports. With an additional DSPM Card attached, it provides eight ports. It can support 32, 64, or 200 mailboxes, depending on the PAL installed (see page D-5 for part numbers) and its 2.0 Gigabyte hard drive will support up to 180 hours of voice messages. The EVMC supports AMIS networking and database saves to a zip or floppy drive. It does not support Fax-On-Demand and Unified Messaging. It can be used only with the new 4-amp or the standard 9-amp power supply.

  NOTE: The 32- and 64-mailbox PALs can be used only with the CPC64 or CPU64. The 200-mailbox PAL requires system software version later than 5.270, but it can be used with any CPC or CPU. Currently, the 4-port EVMC is shipped with the 32-mailbox PAL.

- **4-Amp Power Supply:** This power supply will support any combination of Inter-Tel circuit cards, up to the 4-amp limit. It can be used with any CPU and any voice mail platform (EVMC or Windows-based Voice Processing Unit).

1.2 All of the standard system circuit cards can be used in the CPU64 configuration, as long as the device count and power requirements remain within limits. For example, you could create the following 64-device system:

- One 4-amp power supply
- Two DKSC-16 cards to provide 32 phones stations (without MDPMs)
- An SLC-16 to provide 16 single-line stations
- Two LSC-D cards to provide 16 trunks
- An OPC
- An EVMC to provide 4 or 8 Voice Processing Unit ports and 32, 64, or 200 mailboxes
- A CPU64

A. USING EXISTING SYSTEM HARDWARE

1.3 If you need more than eight Voice Processing Unit ports, you can use the standard Voice Processing Unit in place of the EVMC. Likewise, if you need more than 4 amps of power, you can use the standard Inter-Tel power supplies. (However, you cannot use the U.S. 6-amp power supply with the EVMC.)

B. SYSTEM DB PROGRAMMING

1.4 The system version 5.2 (and later) Database (DB) Programming software has been changed in the following ways to accommodate the CPU64 and EVMC:

- CPC64 is included in the Cabinets folder as a CPU card type (it is programmed like a CPU128).
- A slot can be programmed for the EVMC (no additional card programming is required).
- In Cabinets, only eight slots are displayed (one chassis). Slot 8 is used by the CPC64.
- Fax-On-Demand programming does not apply when the EVMC is installed.

1.5 There are five pre-programmed databases available on the CPU64 software CD. They are:

- Call Buttons with Voice Mail
- Individual Line Buttons with Voice Mail
- Call Buttons without Voice Mail
- Individual Line Buttons without Voice Mail
- Node on a Network

1.6 To use one of these databases, perform a Database Restore using the desired database. The pre-programmed databases have the following configurations.

Call Buttons or Individual Line Buttons with Voice Mail Configurations

Table D-1. *Call Buttons or Individual Line Buttons with Voice Mail Configuration*

Slot	1	2	3	4	5	6	7	8
Card Type	DKSC-16	DKSC-16	OPC	LSC-D	LSC-D	None	EVMC	CPU

Call Buttons and Individual Line Buttons without Voice Mail Configurations

Table D-2. *Call Buttons or Individual Line Buttons without Voice Mail Configurations*

Slot	1	2	3	4	5	6	7	8
Card Type	DKSC-16	DKSC-16	OPC	LSC-D	LSC-D	None	None	CPU

Node on Network Configuration

Table D-3. *Node on Network Configuration*

Slot	1	2	3	4	5	6	7	8
Card Type	DKSC-16	DKSC-16	OPC	T1/E1/PRI	LSC	None	None	CPU

Pre-Programmed Features

1.7 All of the default databases have the following pre-programmed features:

- Directories premium feature is licensed
- Serial ports set to 38400
- Allow Green LEDs flag enabled
- Three-digit extensions (unless networking, then use four digits)
- One hunt group with all telephones except 100 (or 1000)
- DSPs on the OPC are programmed to support Caller ID (U.S. version only)

1.8 Databases with Voice Mail have:

- Mailboxes have been created for each of the stations.
- Unassociated mailbox 500 (or 5000) is the Administrator
- Voice Processing applications have been created for Voice Mail, Message Retrieval/ Notification, Automated Attendant, and Auto Attendant Recall
- The Voice Mail application is extension 250 (or 2500)
- System Forwarding to voice mail is programmed for each station (except 100/1000) for outside calls, during day mode, using no answer and DND conditions
- Voice Mail is the Message Center for all stations
- Transfer prompts are disabled
- During day mode, calls ring 100/1000 (no system forwarding on this station).
- During night mode, calls ring to the automated attendant and recall to the auto attendant recall application
- In each keymap, the last button (8th or 12th) is programmed as a “Voice Mail” (250/ 2500) button to simplify transfers and access to voice mail

1.9 Individual Line Key Databases have:

- Hold button programmed as System Hold
- Eight buttons are programmed as individual trunks
- An Alternate Keymap is programmed with trunks 9-16
- A button is programmed to switch keymaps

1.10 The Networked Node has the following programmed:

- The T1/E1/PRI card is set for Private Networking and all 23 B-channels are activated

NOTE: A Network Clock Card (NCC) is required.

- Advanced CO Interfaces premium feature is licensed

C. PART NUMBERS

1.11 The part numbers for the CPU64 components are:

Table D-4. *CPU64 Part Numbers*

Component	U.S. (Europe) Part No.
CPC64	550.2030 (PB2030)
4-Port EVMC (with 1 DSPM)	550.5030*
Additional DPSM for 4-Port EVMC	550.5011
8-Port EVMC (with 2 DPSMs)	550.5018
4-Amp Power Supply	550.0121 (PS2747)

* This replaces part no. 550.5014, which does not include the 32-mailbox PAL.

NOTE: CPU64 PALs are not required for software versions 8.0 and later. All premium features are controlled by the software license. Mailbox PALs on the EVMC, however, are still required.

1.12 The following CPU64 PALs are available for non-networked systems:

Table D-5. *CPU64 PALs for Non-Networked Systems*

PAL	U.S. (Europe) PART No.
25-Unit Non-Networking PAL	827.9670 (N/A)
50-Unit Non-Networking PAL	827.9671 (DI228)
75-Unit Non-Networking PAL	827.9672 (DI229)
100-Unit Non-Networking PAL	827.9673 (DI230)
125-Unit Non-Networking PAL	827.9674 (N/A)
150-Unit Non-Networking PAL	827.9675 (N/A)

1.13 The following CPU64 PALs can be used for a networked system:

Table D-6. *CPU64 PALs for a Networked System*

PAL	U.S. (Europe) Part No.
25-Unit Networking PAL	827.9677 (N/A)
50-Unit Networking PAL	827.9678 (DI231)
75-Unit Networking PAL	827.9679 (DI232)
100-Unit Networking PAL	827.9680 (DI213)
125-Unit Networking PAL	827.9681 ((N/A)
150-Unit Networking PAL	827.9682 (N/A)

1.14 The part numbers for the PALs used on the EVMC are as follows:

Table D-7. *PAL Part Numbers for an EVMC*

PAL	U.S. (Europe) Part No.
32-Mailbox Voice Mail PAL (used with CPU 64 Only)	827.9144 (N/A)
64-Mailbox Voice Mail PAL (used with CPU 64 Only)	827.9145 (DI176)
200-Mailbox Voice Mail PAL (can be used with any CPU)	827.9150 (DI177)

NOTE: The 32- and 64-mailbox PALs can be used only with the CPC64 or CPU64. The 200-mailbox PAL can be used with any CPU, but requires system software version later than 5.270. Currently, the 4-port EVMC is shipped with the 32-mailbox PAL.

1.15 The part number for the CPU64 software is as follows:

Table D-8. *CPU64 Software*

Software	Part No.
CPC64 CD-ROM	827.9871

2. CPC

2.1 This section describes the CPCs.

A. SPECIFICATION

2.2 Version 8.0 and later software support a new CPU card called the Call Processing Card (CPC). This card, which replaces the CPU64, CPU128, and CPU256 cards, uses a software license that allows you to upgrade the capacity of the system without having to replace the card (see page 5-11 for details about the software license feature). Because this software license contains premium feature information, the CPC or CPU020 cards do not require a feature unit PAL. In addition, the CPC does not require an attached PCM card because the CPC houses the typical PCM functionality and hardware. See Figure 3-47 on page 3-96 for an illustration.

NOTE: To use version 8.0 or later software on 512 and higher systems, a Call Processing (CP) Server (Windows NT- or 2000-based CPU) is required. To use version 8.0 or later software on 64, 128, or 256 systems, the CPC is required.

2.3 Features of the CPC include the following:

- **IP Port (10/100 Base-T):** Provides IP connections directly to the system, which is similar to the functionality of the IP port found on the CP Server. In addition, you can use the IP Port to upload Call Processing software, upload the software license, or program the database. See page 6-178 for more information.

CAUTION

Security is the responsibility of the customer's network administrator. Inter-Tel is not responsible for network problems due to security violations involving the IP address exposure of the CPC. This includes, but is not limited to, interrupted network service. Recommendations remain to place the CPC in a private IP network. In addition, Inter-Tel recommends that you install the CPC behind a firewall or Network Address Translation (NAT) box.

- *** Button:** Allows you to bypass the backup database, verify the status of Call Processing, and place the CPC into the download state. See for details.

NOTE: The * button is **not** a reset button.

- **RS232 Jacks:** (two 8-pin, 8-conductor modular jacks) Connects to the RS232 port on the Voice Processing PC or EVMC, or connects SMDR/error message output devices, a personal computer for programming the system database, and/or a modem for remote programming/maintenance (see pages 2-99 to 2-104 for requirements).

NOTE: The system can have a maximum of 32 RS232 ports for system services. This includes the RS232 ports on the CPC, Options Card(s), and PC Data Port Modules. If an Inter-Tel Voice Processing PC or EVMC is installed, at least one of the system RS232 ports must be dedicated to Voice Processing.

- **Relay:** (6-pin, 6-conductor modular jack) Currently used for power failure transfer equipment only. A power failure transfer jack can be used for connecting one or more single-line sets to predetermined CO [local exchange] trunks in the event of processor card failure or a power failure. The output is one normally-open (deactivated) double-pole, double-throw (DPDT) relay (actually wired as SPDT) that is held closed (activated) during system operation. Additional equipment needed includes a customer-provided PFT relay card and card mounting case. The relay contact ratings are listed below. (See page 2-103 for additional requirements.)

  CPC Relay Contact Ratings

  0.6A at 125VAC
  0.6A at 110VDC
  2.0A at 30VDC

- **Music-On-Hold Port:** Connects an external music source (such as a radio, tape player, etc.) that can be heard by callers when they are placed on hold or camped on. It may also be used for phone background music. This is an 1/8-inch (3-mm), two-conductor (mono), mini-phone jack. For installation instructions, see page 3-240. In DB Programming, music can be enabled or disabled, and each trunk group (as well as transfer audio) can be assigned music, silence, or tick tones. See page 6-234 in PROGRAMMING. This port supports A-law or mu-law companding, which is configured in DB Programming (System\Flags\Companding Type).

- **External paging jack:** (RCA-type phono jack with an output impedance of 600 ohms) Connects external paging speaker equipment to allow paging to large areas, such as warehouses or loading docks. For installation instructions, see page 3-242.

- **Built-in Network Clock Card (NCC):** Provides the smooth, error-free clock transitions required for T1 applications. If any T1 Cards are installed in the equipment chassis, the network clock circuitry provides the clock function for the entire system.

  CAUTION

  Do not use a Network Clock Card (NCC) with the CPC. Using an NCC with this card causes erratic problems which may disable the system. If upgrading an existing CPU64 or CPU128 to a CPC, you must remove the NCC before installing the CPC. If the CPC detects an NCC, the system will continually reset.

- **Software Security and Feature Program:** Supports a software license that enables functionality on the card. This feature serves two purposes: 1) the PAL is no longer required; and 2) feature and/or device upgrades are possible without replacing the card. For details about the software security and feature program, see page 5-11.

- **Modem:** Provides a built-in modem that is capable of transferring data at speeds up to 33600 bps. This modem is always equipped and is basically a system serial port that you must "call." Because the connection is established via a call, each CPC modem in a chassis is assigned an extension (1500 by default). For details about the modem, see page 6-180.

- **Rechargeable Battery:** Replaces the lithium battery that is available on CPU cards and is rechargeable. If the power to the CPC is lost, this battery will store the system database for up to a month (30 days). After power is restored, the battery recharges itself. This battery can be recharged more than 400 times before it needs to be replaced.

- **Flash Memory:** Stores a backup copy of the system database. The system automatically backs up this database once a day or whenever scheduled in DB Programming. The backup database is then restored if the same version of Call Processing software is reloaded or if the system detects a corrupt database on power up.

NOTE: The following four system resources are provided by the single Digital Signal Processor (DSP) on the CPC. For additional and/or other system resources, one or more Options Cards (with up to four DSPs each) may be installed in the equipment chassis.

- **Conferencing resources:** Connect up to 12 parties in multiple conferences. There can be up to three simultaneous four-party conferences or up to four simultaneous three-party conferences on a PCM card (non-ATM). Conferencing is not amplified.
- **DTMF receiver circuits:** Provides circuits (two for non-ATM cards; six for ATM cards) that are used when single-line sets connected to MDPMs are dialing and when features such as DISA and analog voice mail are used. These receivers convert DTMF signals to the digital codes used by the system. They are in use only while the dialed number is being processed, not for the duration of the call. If all DTMF receivers are busy, the user may try again (DISA users are redirected to the primary attendant).
- **DTMF transmitter circuits:** Provides four circuits for generating DTMF tones whenever needed by the system. For example, when an Axxess Basic Digital, Eclipse™ Basic Digital, Standard Display, Associate Display, Model 8500, Model 8520, or Model 8560 Phone user manually dials a number while on a call or when a station user uses one of the system auto-dial features (ARS, call forward to the public network, station or system speed dialing, etc.). Executive Display and Professional Display Phones can generate their own DTMF tones.
- **Tone generating circuit:** Produces all of the system tones (busy, reorder, do-not-disturb, etc.) heard by phone, single-line set, DISA, and automated attendant users.
- **Caller ID [CLIP] receivers:** Provides four resources for receiving Caller ID [CLIP]. These resources are in addition to those available on the OPC (see page 2-67).

2.4 The CPC supports Call Processing versions 8.0 and later only. In addition, if the CPC is installed on a networked system (i.e., one with multiple nodes), each node on the system must be running version 5.2F9 or later. The CPC also supports the U.S. and European system versions of software.

B. INSTALLATION

2.5 A CPC supports only systems with 64, 128, or 256 devices. For 512 devices and higher, a CP Server is required. Before you install the CPC, however, you must consider the following:

- Because the CPC has an IP port, there is a risk of security violations. Inter-Tel strongly recommends that you install the CPC behind a firewall or NAT box.
- If upgrading from a CPU64 or CPU128, you must first remove any Network Clock Cards (NCCs) from the chassis's backplane before installing the new CPC. If you do not remove NCCs from the backplane, the system will continually reset.

2.6 To install the CPC in the chassis:

1. Make sure all power supply AC POWER switches are turned OFF. Installing the CPC with the power on may damage the circuitry on the card.
2. Measure the rechargeable battery voltage. The voltage test points are located on the top of the battery itself (+) and underneath the battery, on the opposite side of the card (-).

   To test the voltage:

   a. Insert the "common" voltmeter probe into the ground point (the hole underneath the battery).

   b. Place the other probe on the voltage test point (+).

   If the voltage is not greater than 3.3VDC, the battery may need recharging. Allow the battery to recharge for at least another five hours and then test it again. If the voltage is still not greater than 3.3VDC, replace the CPC.
3. Install the CPC with the components *facing right* in the last (8th) card slot in the chassis.

   NOTE: Dual-chassis systems require only one CPC to control the entire system. However, the CPC must be installed in the eighth slot of the left-hand chassis (as viewed from the front).

4. Turn ON the AC POWER switch and observe the LEDs on the front edge of the card. The top four LEDs should sequentially flash in an up and down pattern for approximately 10 seconds. Then the CPU Active LED should light.

   NOTE: When a new system is installed, the DATABASE ERROR LED remains lit until any changes are made to the system database. The MINOR ALARM LED may also be lit and can later be cleared from an administrator station using the Clear System Alarm feature code (default is 9850).

5. Upload the software license (see page 6-45).

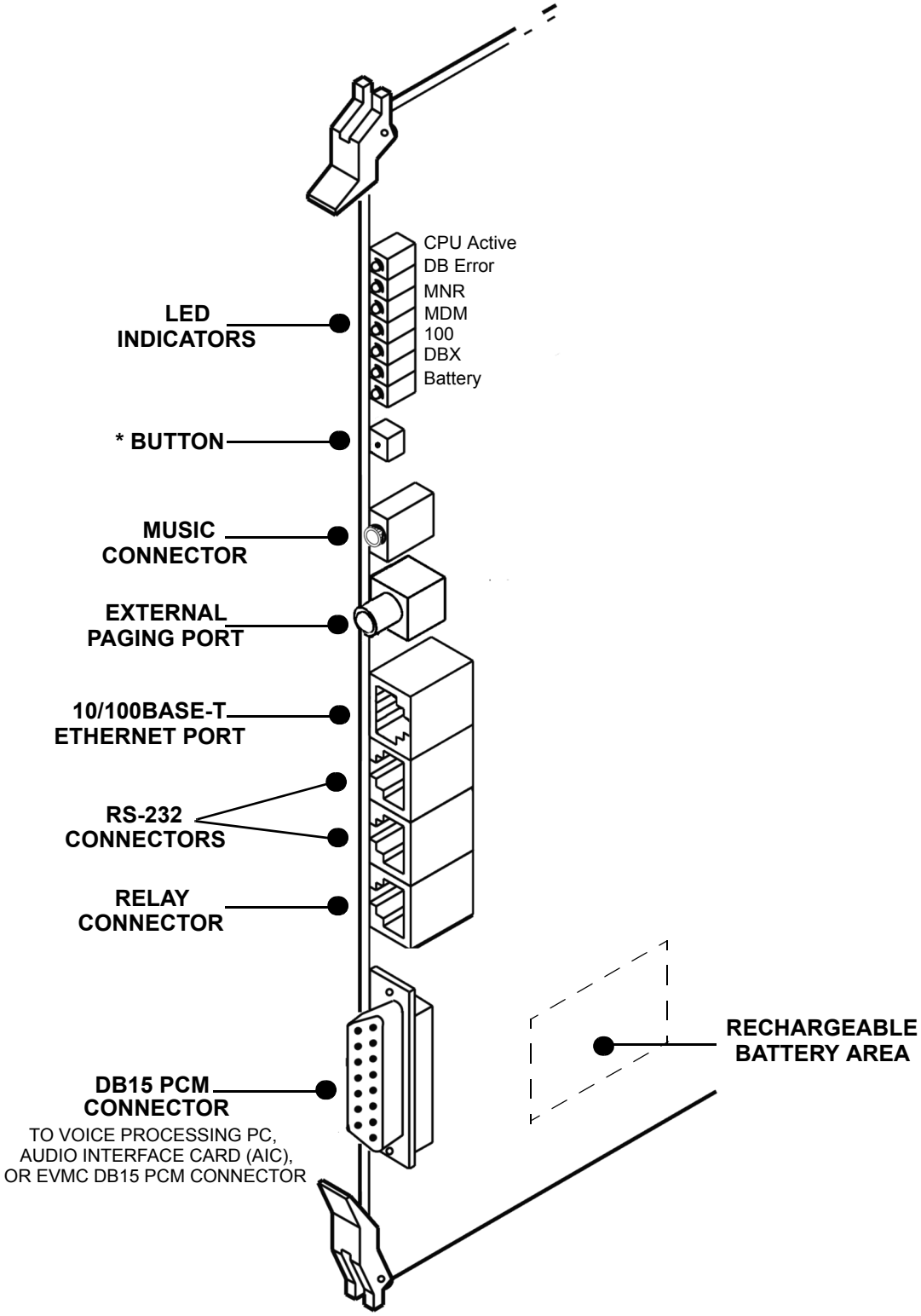
   NOTE: Inter-Tel highly recommends that you upload the software license before attempting to program the database. If you do not upload the software license, Alarm 125 will be generated, and "License Error" will display on administrator phones. In addition, the system will reset every four hours until a valid license is uploaded (see page 7-20).

6. *If the CPC does not have the appropriate system software already installed in flash memory,* use the Board Utility to upload the software from the programming PC to the card.

NOTE: If using the IP port for the Board Utility, the Web Server flag must be enabled (see page 6-179).

7. After installing the CPC, the optional music source, external paging equipment, Voice Processing PC, EVMC, power failure transfer equipment, output device, etc. may be connected to the CPC at any time.

Figure D-1. *Call Processing Card (CPC)*



NOTE: For purposes of detail, this drawing does not show the electromagnetic interference (EMI) shield attached to the front end of the card.

3. CPU64 CARD

3.1 This section describes CPU64 Cards.

A. SPECIFICATIONS

3.2 The CPU64 card is identical to the CPU128 Card, except that it only supports up to 64 devices and it can be used only in a single-chassis system. Like the CPU128, the CPU64 Card uses flash memory circuitry.

NOTE: Versions 8.0 and later require a CPC for CPU64 systems.

3.3 The CPU64 can be used with the 4-amp, 6-amp (U.S. only), or 9-amp power supplies.

3.4 The CPU64 can be used with any of the voice mail platforms (EVMC or Windows-based Voice Processing Unit).

NOTE: System version Beta 5.2 or later software is required if you are using the Windows-based Voice Processing PC.

3.5 If any T1/E1 Cards are to be installed in an equipment chassis using the CPU64 Card configuration, a single Network Clock Card must also be installed. Then, instead of using the clock circuitry on the CPU Card, the NCC becomes the clock for the entire system. The NCC provides the smooth, error-free clock transitions required for T1/E1 applications. You can use only one PRI-equipped card with the CPU64.

3.6 The CPU64 Card can be used in a node in an Inter-Tel network. However, the node should not be a hub and its voice mail system should not support any other nodes.

3.7 The CPU64 Card contains the following system resources (see Figure D-2 on page D-17):

- **DB15 PCM connector** (15-pin D-subminiature female connector) for connecting the first eight ports to the EVMC or the Audio Interface Card (AIC) in the Voice Processing PC.
- **RS232 jacks** (three 8-pin, 8-conductor modular jacks) for connecting to the EVMC's or Voice Processing PC's RS232 port, or for connecting SMDR/error message output devices, a personal computer for programming the system database, and/or a modem for remote programming/maintenance.

  NOTE: The system can have a maximum of 11 RS232 ports for system services. This includes the RS232 ports on the CPU Card, Options Card(s), and PC Data Port Modules. If an Inter-Tel Voice Processing PC is installed, at least one of the system RS232 ports must be dedicated to it.

- **LED indicators** show main processor operation (CPU ACTIVE - green), database in default state due to error (DATABASE ERROR - red), minor system alarms (MINOR ALARM - yellow), external clock activity (EXTERNAL CLOCK ACTIVE - green), and database back-up battery connection (BATTERY OFF - red). See TROUBLESHOOTING for detailed LED information.

- **Reset switch** is used to manually reset the system during troubleshooting. This is a software-controlled system reconfiguration. Pushing this switch does the following:
  - — Preserves the battery-backed database information, all calls in progress, outside calls being placed, pages in progress, camped-on calls, queue requests, system and user programming, inter-station messages, calls on system and individual hold, do-not-disturb messages, and reminder messages.
  - — Restores DTMF receivers, modem access, and speech channels, and resets the station clocks to match the system clock. Updates trunk button, DSS/BLF button, speed-dial button, and feature button lamp status.
  - — Disconnects calls ringing in.

NOTE: If necessary, the reset switch can also be used to place the CPU Card into "receive software upload" mode, as described in INSTALLATION.

- **Database back-up battery** (3.6V, ½AA lithium battery) that can support the database for at least two months of accumulated system down time.
- **Battery voltage test points and check button** for checking the database back-up battery charge. If the battery voltage is not greater than 3.3VDC, replace the battery.
- **Battery back-up jumper strap** for activating the database back-up battery. The jumper strap should be placed in the ON position (over the back two pins) before the CPU Card is installed. When the card is taken out of service for repair or storage, the jumper strap should be placed in the OFF position (front two pins) to preserve the battery charge.

NOTE: Placing the BATTERY jumper strap in the OFF position erases the database when the card is unplugged or loses power.

- **Music jack** for connecting an external music source (such as a radio, CD player, tape player, etc.) that can be heard by callers when they are placed on hold or camped on. It may also be used for phone background music. This is a 1/8-inch (3-mm), two-conductor (mono), mini-phone jack. In DB Programming, music can be enabled or disabled, and each trunk group (as well as transfer audio) can be assigned music, silence, or tick tones.
- **Relay jack** (6-pin, 6-conductor modular jack) currently used for power failure transfer equipment only. A power failure transfer jack can be used for connecting one or more single-line sets to predetermined CO trunks in the event of processor card failure or a power failure. The output is one normally-open (deactivated) double-pole double-throw (DPDT) relay (actually wired as SPDT) that is held closed (activated) during system operation. Additional equipment needed includes a customer-provided PFT relay card and card mounting case. The relay contact ratings are listed below.

CPU020/PCM Card Relay Contact Ratings

0.6A at 125VAC
0.6A at 110VDC
2.0A at 30VDC

- **Real-time clock** for date and time function.
- **External paging jack** (RCA-type phono jack with an output impedance of 600 ohms) for connecting external paging speaker equipment to allow paging to large areas, such as warehouses or loading docks.

- **Conferencing resources** for connecting up to 12 parties in multiple conferences. There can be up to three simultaneous four-party conferences or up to four simultaneous three-party conferences. Conferencing is not amplified.
- **DTMF receiver circuits** provide circuits (two for non-ATM cards; six for ATM cards) that are used when single-line sets connected to MDPMs are dialing and when features such as DISA and analog voice mail are used. These receivers convert DTMF signals to the digital codes used by the system. They are in use only while the dialed number is being processed, not for the duration of the call. If all DTMF receivers are busy, the user may try again (DISA users are redirected to the primary attendant).
- **DTMF transmitter circuits** (two) generate DTMF tones whenever needed by the system. For example, when a Basic, Standard, or Associate Phone user manually dials a number while on a call or when a station user uses one of the system auto-dial features (ARS, call forward to the public network, station or system speed dial, etc.). Executive Display and Professional Display Phones can generate their own DTMF tones.
- **Tone generating circuit** produces all of the system tones (busy, reorder, do-not-disturb, etc.) heard by phone, single-line set, DISA, and automated attendant users.

B. INSTALLATION

3.8 Install the CPU64 as follows:

1. Check the part number and socket number printed on the feature-unit PAL. They must match the software configuration purchased by the customer, their labels should have the correct part number, and they should be installed in socket number U4. (The boot ROM and memory map PAL are already installed on the card.)
2. Place the CPU64 Card on a clean, flat, anti-static, non-conductive surface.
3. Carefully install the feature unit PAL in the socket marked “Feature Unit PAL” on the card. (See Figure D-2 on page D-17.) The notch (or small dot in the corner) indicating pin 1 must match the notch of the outline silk-screened on the card.

NOTE: The label on the chip does not necessarily indicate its orientation.

4. Check that all components on the card are seated securely in their sockets and that no pins are bent.
5. Install the CPU64 Card in the equipment chassis as follows. See Figure D-2 on page D-17 for an illustration of the card.
   a. Ensure that all integrated circuits and components on the card are seated securely in their sockets and that no pins are bent.
   b. Ensure that the mu-law/a-law jumper (J14) is in the correct position for your installation (U.S. installations use mu-law).
   c. Place the battery back-up (BATTERY - J2) jumper strap in the ON position to enable database battery back-up. If the jumper strap is placed in the OFF position, the BATTERY OFF LED lights when the card is installed and AC power is on.

NOTE: The BATTERY OFF LED will not light if the jumper strap is missing.

d. While pressing the battery check button, measure the database back-up battery voltage. The voltage test points (TP1 to TP2) are located on the front edge of the card. Insert the "common" voltmeter probe into the ground point (TP2) and insert the other probe into the voltage test point (TP1). If the voltage is not greater than 3.3VDC, replace the battery.

e. Make sure all power supply AC POWER switches are turned OFF. Installing the CPU64 Card with the power on may damage the circuitry on the card.

f. Install the CPU64 Card with the components *facing right* in the last (8th) card slot in the equipment chassis.

g. Turn ON the AC POWER switch and observe the LEDs on the front edge of the card for the indications below. *If they are incorrect,* recheck the system voltages as outlined in INSTALLATION, and then contact Customer Support.

Table D-9. *CPU64 Card LED Indications*

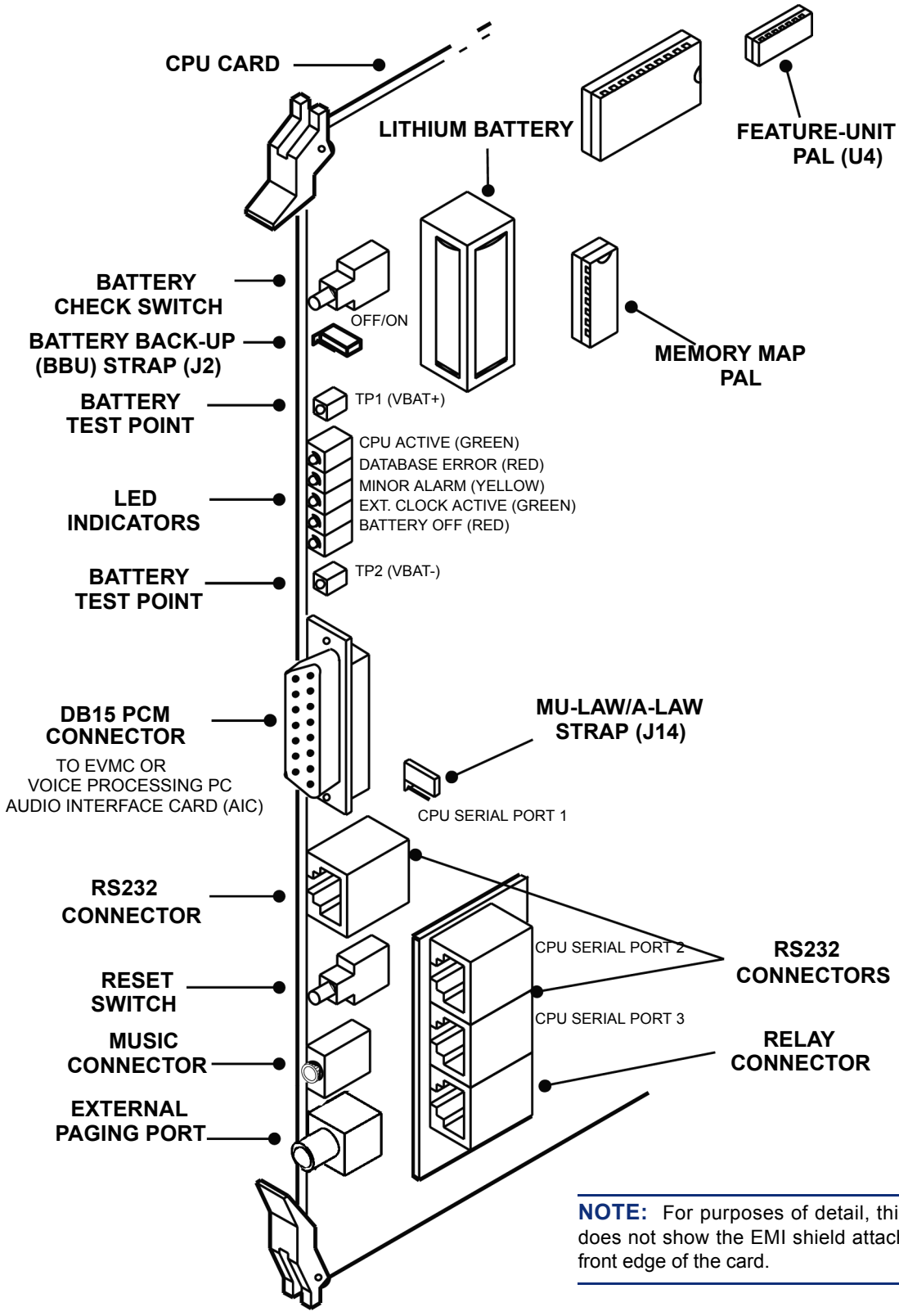
Name	Correct Indication
BATTERY OFF	Off (Red)
CPU ACTIVE	Lit (Green)
DATABASE	Off* (Red)
ERROR	Off (Green)
EXT. CLOCK	Off (Green)
ACTIVE	Off (Green)
REMOTE SYNC	
LOCAL SYNC	

* When a new system is installed, the DATABASE ERROR LED remains lit until the system database has been programmed. The MINOR ALARM LED may also be lit and can later be cleared from an administrator station using the Clear System Alarm feature code (default is 9850).

h. *If the CPU64 Card does not have the appropriate system software already installed in flash memory,* see INSTALLATION to upload the software from the programming PC to the card.

6. After installing the CPU64 Card, the optional music source, external paging equipment, EVMC, Voice Processing PC, power failure transfer equipment, output device, etc. may be connected to the CPU64 Card at any time.

Figure D-2. *CPU64 Card*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

APPENDICES

4. EMBEDDED VOICE MAIL CARDS (EVMC)

4.1 This section describes EVMCs.

A. SPECIFICATIONS

4.2 The Embedded Voice Mail Card (EVMC) was recently modified. Except where noted, the old EVMC and the newer EVMC are identical.

4.3 Both EVMCs come equipped with a single DSPM card to provide a four-port OS/2-based voice mail unit on a single circuit card. It is typically installed in slot 7 of the equipment chassis and it derives power from the chassis backplane.

4.4 The EVMCs can have an additional DSPM card to support an additional four voice mail ports. The 2.0 Gigabyte hard drive on the EVMC will support up to 180 hours of voice messages.

4.5 The PAL (U23) on the EVMCs determines how many mailboxes it supports (32, 64, or 200 mailboxes). Standard mailboxes and networking mailboxes (i.e., AMIS mailboxes) are both included when determining the total number supported. The 32- and 64-mailbox PALs can be used only with the CPC64 or CPU64. The 200-mailbox PAL can be used with any CPU or CPC, if the system is running software version 5.270 or later. If you exceed the supported mailbox limit, users will receive the "mailbox full" message when they try to leave a message in any mailbox. Currently, the 4-port EVMC is shipped with the 32-mailbox PAL.

NOTE: In system version 5.270, Extension IDs counted as mailboxes, thereby affecting the mailbox limit. However, in later 5.2 versions, only mailboxes are counted.

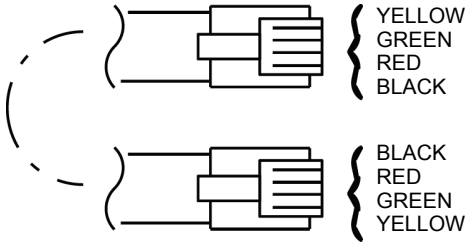
4.6 Timeslots for the EVMCs are supplied through the AIC ribbon cable. For example, the Options Card (OPC) could possibly need more than 16 time slots, depending on the number of Digital Signal Processors (DSPs) installed on the card. The OPC timeslots will not decrease the number of timeslots or ports provided by the CPU64 System.

4.7 The EVMC supports AMIS networking and database saves to a zip or floppy drive. It does not support Fax-On-Demand and Unified Messaging.

4.8 The parallel port on the EVMCs can be used, like the Voice Processing Unit's parallel port, for performing save and restore with a Zip drive. You can use a optional Zip Drive Backup Kit (828.1460) that contains an Iomega Zip Drive, Zip disks, a Save Boot floppy disk, and a Restore Boot floppy disk. To use this kit you will also need an external floppy drive, which is available in the Floppy Drive Kit (828.1486).

4.9 There are connectors on each EVMC for a VGA monitor and a keyboard (there is a connector for a mouse, but it is not supported in this EVMC version).

4.10 The old EVMC has two RS232 serial ports, however, only the first RS232 port is operational at this time. The RS232 interface consists of a standard, two-pair, reversing, mod-to-mod line cord. A reversing line cord can be identified by examining the modular plug connectors side-by-side as shown below.



TWO-PAIR REVERSING LINE CORD

4.11 The newer EVMC has one RS232 serial port and one DB9 serial port. The RS232 interface functions the same as it does on the old EVMC. The DB9 serial port provides a connection for Inside Track.

NOTE: If using the DB9 port, an SMDR buffer box is not required.

4.12 In addition, the newer EVMC no longer has the Reset button, which was for development purposes only and should not be used.

4.13 The EVMC can be used only with the new 4-amp or the standard 9-amp power supply.

4.14 The following table describes the he environmental requirements of the EVMC.

Table D-10. *EVMC Environmental Requirements*

Requirements	In Operation	In Storage
Temperature Relative Humidity (Non-Condensing)	41° to 104° F (5° to 40° C) 8% to 90%	-4° to 140° F (-20° to 60° C) 8% to 90%

B. DIAGNOSTICS

4.15 For advanced troubleshooting purposes, the Voice Processing Unit includes special diagnostics software. If an application problem cannot be resolved, Customer Support may instruct the technician to connect the monitor and keyboard and perform certain diagnostic procedures.

C. HARDWARE INSTALLATION

CAUTION

When handling the EVMC be very careful not to damage the hard drive. It is sensitive to vibration, pressure, and shock.

4.16 To install the EVMC, see Figure D-3 on page D-22 and do the following:

1. Make sure the proper security PAL is installed in socket U23. (The PAL determines the number of available mailboxes.)

   NOTE: Currently, the 4-port EVMC is shipped with the 32-mailbox PAL.

2. *If installing a second DSPM Card,* attach the DSPM card in the lower socket, below the existing DSPM card. Make sure that the connectors and the standoffs are properly aligned and seated.
3. Insert the EVMC into slot 7 of the equipment chassis.
4. Observe the LEDs on the front of the EVMC. The LED indications are as follows. (The left LED is marked "CARD ON-LINE" and the right LED is marked "OK TO REMOVE CARD.")

Table D-11. *EVMC LED Indications*

LED Indication	Meaning
Both LEDs off	If the system chassis is powered up, the OS/2 operating system is starting.
Left LED blinking slowly	Voice Processing Unit is attempting to connect to the CPC or CPU.
Left LED blinking rapidly	The OS/2 operating system and the Voice Processing Unit driver have started.
Left LED lit solid	Voice Processing Unit is initialized and connected to the CPC or CPU.
Right LED blinking rapidly	Voice Processing Unit is shutting down.
Right LED blinking slowly	Voice Processing Unit is shut down and the OS/2 operating system is shutting down.
Right LED lit solid	Voice Processing Unit and OS/2 are shut down. It is now safe to turn off power and remove the EVMC from the chassis.

You may see one of the following error indications.

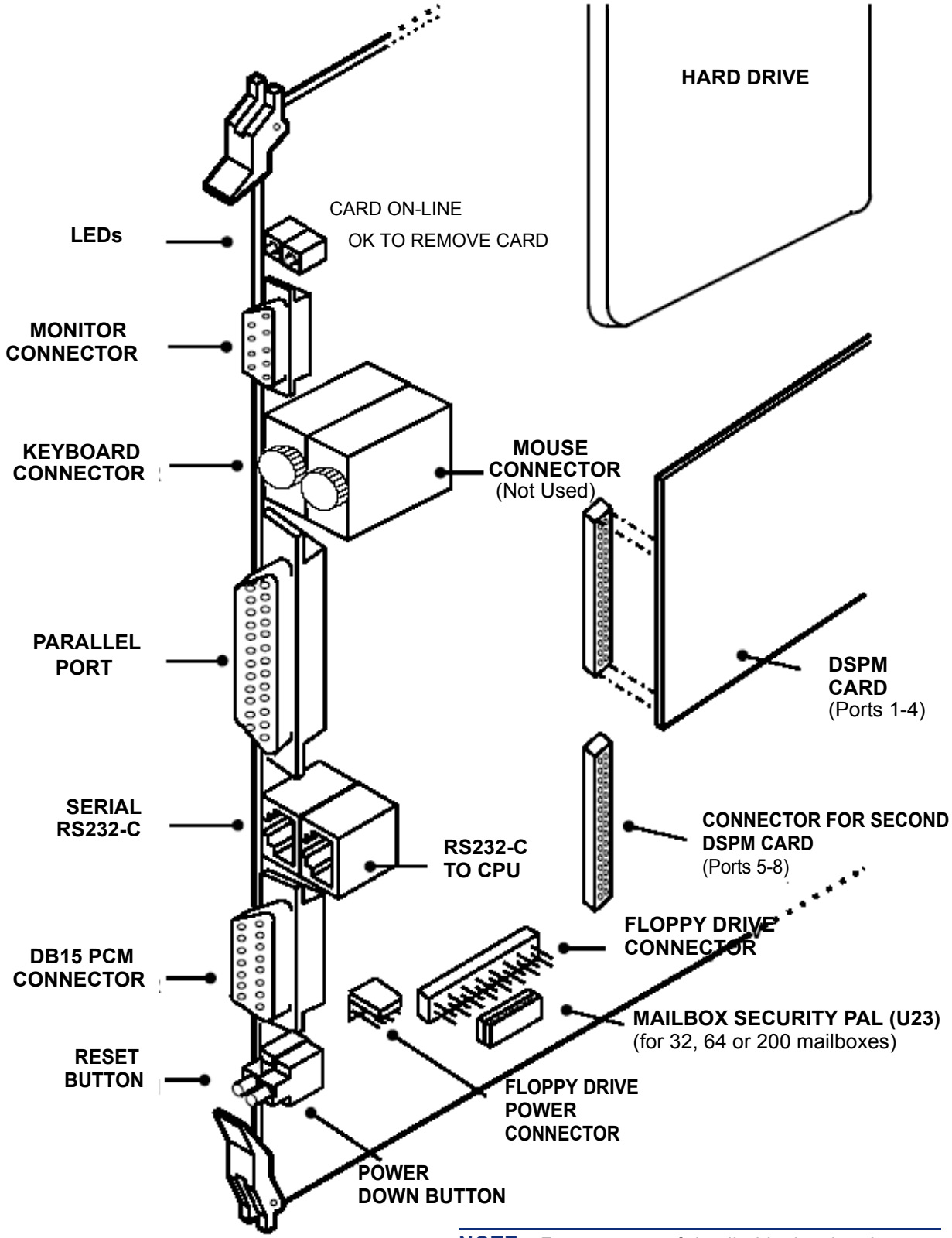
Table D-12. *Error Indications*

LED Indication	Meaning
Left LED blinking rapidly, right LED blinking three on/one off	No DSPM Cards are installed and Voice Processing Unit is shutting down. Press the POWER DOWN button and wait for the right LED to light before turning off the equipment chassis. Then check the EVMC. If there is at least one DSPM Card, make sure it is seated properly. If there are no DSPM Cards, install one or two. If using only one DSPM Card, make sure it is in the top socket, below the hard drive.
Left LED lit, right LED flashing slowly	The EVMC PAL not compatible with the CPC or CPU. Press the POWER DOWN button and wait for the right LED to light before turning off the equipment chassis. Then replace the CPC (or replace the CPU with a CPU64).
Left LED lit, right LED flashing very slowly	No PAL on the EVMC. Press the POWER DOWN button and wait for the right LED to light before turning off the equipment chassis. Then install the appropriate PAL on the EVMC. (See page D-5 for part numbers.)

5. To set up the RS232 interface between the CPC or CPU and the EVMC, plug one end of the mod-to-mod line cord into the RS232-C TO CPU jack. Then plug the other end of the cord into an RS232 port on the CPC or CPU Card. (The RS232-2 port is the preferred location.) Do *not* use the Options Card port.

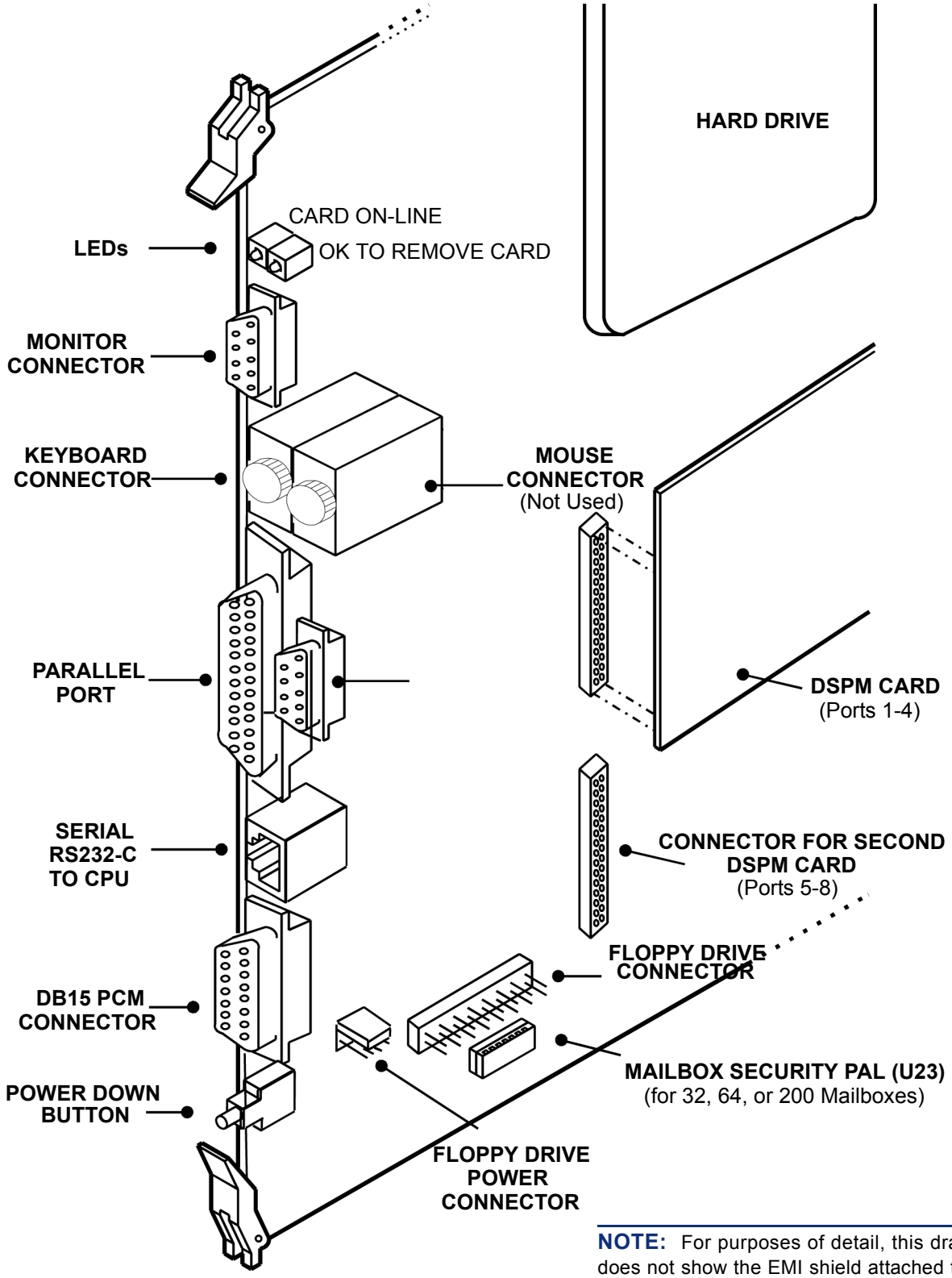
6. To enable voice path for the EVMC ports, attach one end of the DB15 interface cable (813.1739) to the DB15 PCM connector on the CPC or CPU Card. Then attach the other end of the cable to the PCM connector on the front of the EVMC.

Figure D-3. *Old EVMC*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

Figure D-4. *New EVMC*



NOTE: For purposes of detail, this drawing does not show the EMI shield attached to the front edge of the card.

D. VOICE PROCESSING SOFTWARE INSTALLATION

4.17 The EVMC is shipped with OS/2 and the Voice Processing applications software already installed on the hard disk drive. As a back-up, a CD containing the software is also available. If the software is upgraded at a later time, a new CD will be available.

4.18 To upload the software to the EVMC you will need:

- A set of Voice Processing software disks (make copies from the CD)
- An external floppy drive (available in the Floppy Drive Kit 828.1486)
- A keyboard and monitor

4.19 If you are using secondary language prompts, you will also need to do the following:

- Copy the secondary language prompt files to floppy disks. This will require five disks. The prompt files are located on the CPU64 software CD in the Axxess_64\CPU-SOFT\EVMCSOFT\PROMPTS directory.
- Make sure the Multilingual Support premium feature is licensed and select **Secondary Language** in system DB Programming.

4.20 Use the following procedure to install the Voice Processing applications software via floppy drive:

CAUTION

If the EVMC is not shut down properly before it is removed from the equipment chassis, the database will probably become corrupted, and the EVMC will go into an endless rebooting cycle.

1. Shut down the EVMC by using a small object to press the Power Down button on the front edge of the card (see illustration on the previous page for its location). The EVMC will go through a shutdown process and the "OK To Remove Card" LED will flash. When shutdown is complete, the LED will be solidly lit.

   NOTE: When the "OK To Remove Card" LED turns green, it is safe to remove the EVMC from the equipment chassis for the next ***10 seconds.*** If you do not remove the card within that amount of time, allow the card to reboot and wait for the "Card On-Line" LED to light. Then you can repeat the shutdown process and remove the EVMC.

2. Remove the EVMC from the chassis and lay it on a clean, flat, anti-static surface.

   CAUTION

   When handling the EVMC be very careful not to damage the hard drive. It is sensitive to vibration, pressure, and shock.

3. Attach the floppy drive cable to the floppy connector on the EVMC (see Figure D-3 on page D-22 for location).

   NOTE: When plugging in the floppy drive or power connector ribbon cable, the red stripe goes on the side closest to the power supply.

4. Attach the larger connector on the floppy drive power cable to the floppy drive.
5. Attach the other (smaller) connector on the floppy drive power cable to the floppy drive power connector on the EVMC.

6. Temporarily install the EVMC in slot 6 of the chassis (Or any slot that has a vacant space to the right, to allow room for the floppy ribbon cable. The slot you choose does not need to be programmed for an EVMC for this to work properly.)
7. Attach the monitor to the 15-pin connector on the front of the EVMC. Then, plug in the monitor's AC power cord and turn on the monitor's AC power switch.
8. Attach the keyboard to the keyboard connector on the front of the EVMC.
9. After some initial power-up information, the monitor eventually displays "Waiting 5 seconds. Press **ESC** to escape to the command line." Press **ESC** before the five seconds expire.
10. A flashing prompt will show "Shutting down in 5 seconds." Press **ESC** before the five seconds expire. The monitor displays the OS/2 command prompt "[OS/2 C:\AVDAP]".

NOTE: If you miss this five-second opportunity let the EVMC power up all the way up and then press the POWER DOWN button on the EVMC Card. Wait until you see "Waiting 5 seconds. Press <ESC> to escape to the command line." Then press **ESC** on your keyboard. The monitor displays the OS/2 command prompt "[OS/2 C:\AVDAP]".

11. Insert the EVMC Software Disk 1 in the PC's floppy disk drive.

NOTE: All disks must be installed. For optional secondary language voice prompts, additional disks will also be required.

12. At the [OS/2 C:\AVDAP] command prompt, enter "a:setup" and press **ENTER**. Then follow the installation instructions displayed on the monitor.
13. **To install secondary language prompts:** Insert the Secondary Language Disk 1 in the PC's floppy disk drive. At the [OS/2 C:\AVDAP] command prompt, enter "a:pmtinst" and press **ENTER**. Then follow the installation instructions displayed on the monitor.

NOTE: All five language disks must be installed.

14. When the installation is complete, remove the last disk.
15. Shut down the EVMC by using a small object to press the Power Down button on the front edge of the card (see illustration on page D-22 for its location). The EVMC will go through a shutdown process and the "OK To Remove Card" LED will flash. When shutdown is complete, the LED will be solidly lit.

NOTE: When the "OK To Remove Card" LED turns green, it is safe to remove the EVMC from the chassis for the next ***10 seconds.*** If you do not remove the card within that amount of time, allow the card to reboot and wait for the "Card On-Line" LED to light. Then you can repeat the shutdown process and remove the EVMC.

16. Remove the EVMC and disconnect the floppy drive.
17. Re-install the EVMC. The EVMC will reboot automatically when placed in the equipment chassis and the "Card On-Line" LED will light.
18. Program the Voice Processing Unit as described in PROGRAMMING.

E. EVMC VOICE PROCESSING SAVE AND RESTORE

4.21 To perform a Voice Processing save or restore operation on the EVMC you will need:

- An external floppy drive (available in the Floppy Drive Kit 828.1486)
- Zip Drive Backup Kit (part number 828.1460) and blank, formatted Zip Disks (The number required depends on the size of the Voice Processing database. Check the Voice Processing **Total Storage Used** statistics in DB Programming.)
- A keyboard and monitor

NOTE: The Zip Drive is supported only in software versions later than 5.270.

4.22 Use the following procedure to install the Voice Processing applications software via floppy drive:

CAUTION

If the EVMC is not shut down properly before it is removed from the equipment chassis, the database will probably become corrupted, and the EVMC will go into an endless rebooting cycle.

1. Shut down the EVMC by using a small object to press the Power Down button on the front edge of the card (see illustration on page D-22 for its location). The EVMC will go through a shutdown process and the "OK To Remove Card" LED will flash. When shutdown is complete, the LED will be solidly lit.

   NOTE: When the "OK To Remove Card" LED turns green, it is safe to remove the EVMC from the equipment chassis for the next ***10 seconds.*** If you do not remove the card within that amount of time, allow the card to reboot and wait for the "Card On-Line" LED to light. Then you can repeat the shutdown process and remove the EVMC.

2. Remove the EVMC from the equipment chassis and lay it on a clean, flat, anti-static surface.

   CAUTION

   When handling the EVMC be very careful not to damage the hard drive. It is sensitive to vibration, pressure, and shock.

3. Attach the floppy drive cable to the floppy connector on the EVMC (see Figure D-3 on page D-22 for location).

   NOTE: When plugging in the floppy drive or power connector ribbon cable, the red stripe goes on the side closest to the power supply.

4. Attach the larger connector on the floppy drive power cable to the floppy drive.
5. Attach the other (smaller) connector on the floppy drive power cable to the floppy drive power connector on the EVMC.
6. Temporarily install the EVMC in slot 6 of the equipment chassis (Or any slot that has a vacant space to the right, to allow room for the floppy ribbon cable. The slot you choose does not need to be programmed for an EVMC for this to work properly.)

7. Attach the monitor to the 15-pin connector on the front of the EVMC. Then, plug in the monitor's AC power cord and turn on the monitor's AC power switch.
8. Attach the keyboard to the keyboard connector on the front of the EVMC.
9. Plug in the Zip Drive AC transformer.
10. Connect the Zip Drive to the parallel port on the EVMC card using the cable supplied with the Zip Drive.

NOTE: The cable is marked to identify the end that connects to the Zip Drive and the end that connects to the parallel port connector. The parallel port connector is a 25-pin, female, D-sub connector and is located on the front of the EVMC Card.

11. Make sure the Zip Drive is installed properly and power up.
12. Insert a blank, formatted Zip Disk in the Zip Drive.
13. **To backup a database:** Insert the "Zip Drive Backup Kit Utilities—Save Disk" (supplied with the Zip Drive Kit) into the floppy drive.

    To restore a database: Insert the disk labeled "Zip Drive Backup Kit Utilities—Restore Disk" (supplied with the Zip Drive Kit) into the floppy drive.
14. After some initial power-up information, the monitor tells you that it is invoking the Voice Processing Backup System. The Backup or Restore will automatically take place. Follow the directions on the monitor screen.
15. When the Backup or Restore is complete, remove all software disks.
16. Shut down the EVMC by using a small object to press the Power Down button on the front edge of the card (see illustration on page D-22 for its location). The EVMC will go through a shutdown process and the "OK To Remove Card" LED will flash. When shutdown is complete, the LED will be solidly lit.

NOTE: When the "OK To Remove Card" LED turns green, it is safe to remove the EVMC from the equipment chassis for the next ***10 seconds.*** If you do not remove the card within that amount of time, allow the card to reboot and wait for the "Card On-Line" LED to light. Then you can repeat the shutdown process and remove the EVMC.

17. Remove the EVMC and disconnect the floppy drive and the Zip Drive.
18. Re-install the EVMC. The EVMC will reboot automatically when placed in the equipment chassis and the "Card On-Line" LED will light.

5. 4-AMP POWER SUPPLY

5.1 This section describes the 4-AMP Power Supply.

A. SPECIFICATIONS

5.2 The 4-amp system power supply unit, part no. 550.0121 [PS2TH7 for Europe] is installed in the far left slot of the equipment chassis.

5.3 Located on the front side of the power supply unit are the AC power cable connector, the ON/OFF power switch, three LED voltage indicators, and four voltage test points. During normal system operation, all three LEDs (+5V, -5V, and +24V) should be lit. If any of the LEDs are blinking or are unlit, the power supply is defective and the entire unit must be replaced.

5.4 Each power supply unit has one, 250V, 5A fuse. (This fuse may *not* be replaced in the field. If the fuse is blown, return the entire power supply unit for repair.) Each unit's maximum power outputs are +24VDC, 4A; +5VDC, 6A; and -5VDC, 0.5A.

5.5 Up to two 16-Port Digital Keyset Cards (DKSC-16s) can be installed in cabinets (chassis) equipped with this 4-amp power supply, depending on the number and variety of other circuit cards to be installed. If necessary, use the following chart to determine how many cards can be installed, while staying within the 4-amp limit.

Table D-13. *Required AMPs*

Card Type	AMPS Required	Card Type	AMPS Required
DKSC-16/16+	1.28	T1/E1/PRI	0.45
AKSC	1.28	LGC-D	0.30
EKSC	1.28	LSC-D	0.30
SLC	0.75	OPC	0.30
SLC-16	0.40	LGC	0.15
DKSC	0.64	LSC	0.15
T1C or E1C	0.40	IPC or IPRC	0.34
T1/PRI or E1/PRI	0.45	IPRC-D	0.48

5.6 The 4-amp power supply can support one EVMC.

Voltage Surge and Spike Protection

5.7 To reduce the effects of AC voltage surges and spikes that may cause system malfunctions, false logic, and/or damage to the electronic components, a surge/spike protector with the following specifications is recommended:

- Clamp voltage transients at 300VDC within 5 nanoseconds when exposed to waveforms as described in the ANSI/IEEE Standard C62.41-1980 (IEEE 587).
- Reduce RFI/EMI noise by at least 20dB at frequencies between 5 kHz and 30 MHz.
- Ground lug for grounding Single-Line Adapters, trunk gas discharge tubes, etc. to the chassis's AC outlet.

5.8 Contact Inter-Tel CommSource™ for surge/spike protectors that have these specifications.

Battery Back-Up

5.9 To provide back-up power in the event of a power failure or brownout condition, the power supply can have optional battery back-up using a customer-provided uninterruptable power supply (UPS) unit or a standby power supply (SPS) unit. (See SPECIFICATIONS for more information.)

B. INSTALLATION

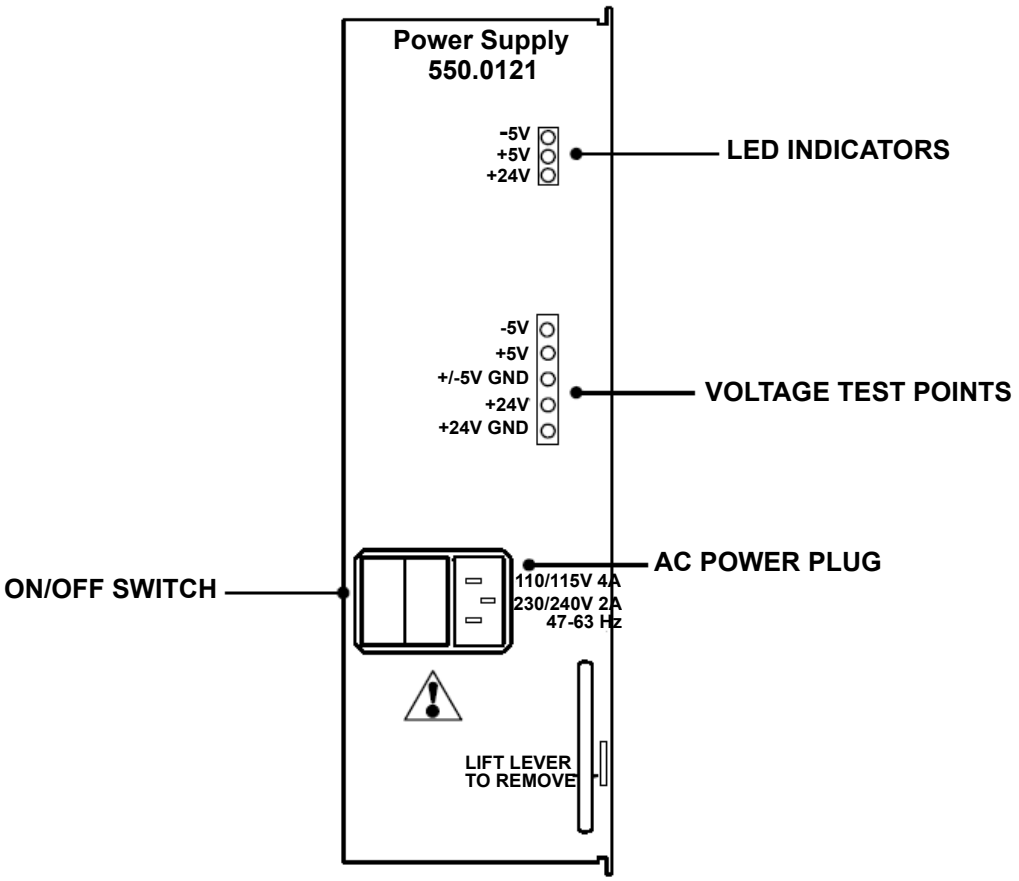
CAUTION

HIGH VOLTAGE. When the system power is turned on, the power supply unit contains 110VAC. Before inserting or removing the power supply unit, make sure that the AC POWER switch has been turned OFF for at least five minutes. Failure to allow the high voltage to dissipate can cause bodily injury.

5.10 Install the power supply unit in the chassis as outlined in the following steps (see Figure D-5 below):

1. Remove the four screws securing the panel on the front of the equipment chassis and set the panel aside.
2. Carefully slide the power supply unit into the far left slot in the equipment chassis until it locks securely into place.
3. Connect the AC power cable to the front of the power supply unit.

Figure D-5. *4-Amp Power Supply Unit*



Grounding Requirements

5.11 The equipment chassis must be plugged into a properly grounded AC power source. (See SPECIFICATIONS for additional AC power requirements.) Some good ground points include: the building's frame, a ground rod (driven a minimum of 8 ft/2.4 m), and a copper (not PVC) cold water pipe.

5.12 The grounding wire should be 10AWG (10 gauge thickness or larger -- smaller number gauge = larger thickness) or larger, and should not exceed 25 feet (7.5 meters) in length, to help provide RFI/EMI protection, or should not exceed 10 feet (3 meters) in length, to help provide both RFI/EMI and lightning protection.

5.13 To ground the system (see Figure D-6 below):

1. Ensure that the system's AC power source is properly grounded. *DO NOT* defeat (or leave floating) the ground wire in the AC power cord.
2. Install an AC surge/spike protector with a built-in ground lug and connect it to:
   a. The ground lug (indicated by the symbol ⏚) on the left, inside wall of the equipment chassis.

b. The ground lug (indicated by the symbol ⏚) on each Single-Line Adapter, if used.

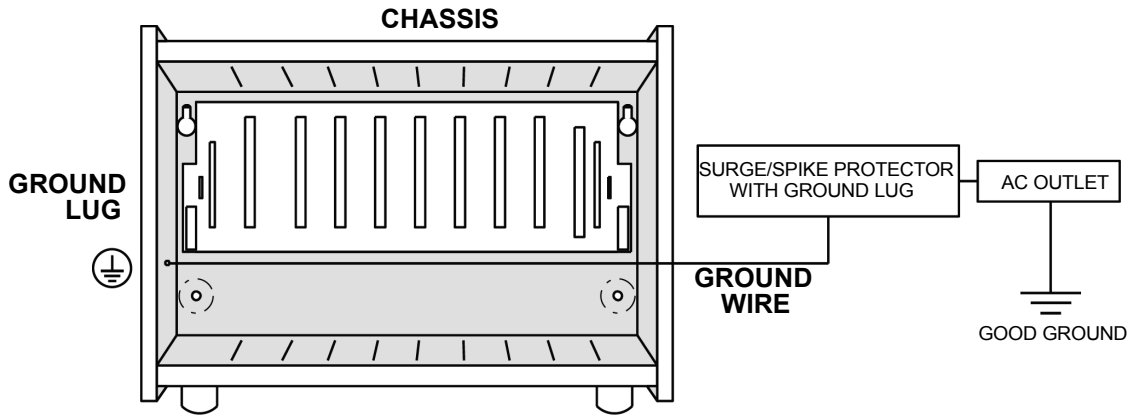
c. Battery compartments, if used.

d. Trunk gas discharge tubes, if used.

5.14 If AC power-related problems appear on the system while in operation, one of the following three methods can be used to test the ground.

- Use a digital multimeter to measure the difference of potential between the equipment chassis ground point and the power neutral, and between the chassis ground point and the third wire ground. If the voltage measurement exceeds 0.5V, noise may develop on the system. If this occurs, call an electrician.
- Use a Megger to test the ground point. (A Megger should be available through the local power company.) The reading should be 5 ohms or less.
- Unplug the AC power cord from the outlet and insert a ground isolation plug into the outlet. Plug the AC power cord into the ground isolation plug. If the problem ceases, call an electrician. When finished, remove the ground isolation plug to restore ground protection. *DO NOT* defeat (or leave floating) the ground wire in the AC power cord.

Figure D-6. *Equipment Chassis Grounding*



Power Supply Electrical Test

5.15 Perform the following electrical check on the power supply.

1. Ensure that the system AC power cable is not plugged in and the AC POWER switch on the front of the power supply is OFF.
2. Plug the recommended surge/spike protector provided into the proper power source:
   - **U.S.:** Use a 105-125 VAC, 57-63 Hz, 15A, single-phase commercial power source.
   - **Europe:** Use a 220-240 VAC, 50 Hz, 13A power source.

NOTE: This MUST be an isolated, dedicated circuit. DO NOT use an extension cord when installing the power supply. Also, DO NOT plug any other equipment into the same AC outlet.

3. Plug the system AC power cable into the surge/spike protector and turn ON the AC POWER switch.

4. After waiting five seconds, observe the three LEDs on the front of the power supply unit. All three LEDs +5V, -5V, and +24V) should be lit. *If any of the LEDs are blinking or are unlit,* the power supply is defective and the entire unit must be replaced.

NOTE: A blinking LED indicates that an overly high voltage has been detected. A lit LED merely indicates that there is some voltage present (as shown in the table at the right); it does not, however, indicate whether the voltage level is acceptable. To determine the exact voltage level present, perform the following step.

Voltage	LED	Output
+5VDC	OFF	1V
	ON	>1V & <6.5V
	BLINKING	6.5V
-5VDC	OFF	-1V
	ON	<-1V & >-6.5V
	BLINKING	-6.5V
+24VDC	OFF	1V
	ON	>1V & <29V
	BLINKING	29V

5. Referring to Figure D-5 on page D-30, measure the following system voltages on the front of the power supply unit. A digital voltmeter of ±0.5% accuracy is required. Insert the "common" voltmeter probe into the ground point and insert the other probe into the desired voltage test point. The measurements should be within the limits listed in the chart below.

Voltage	Tolerance
-5VDC	±5%
+5VDC	±5%
+24VDC	±5%

6. *If the voltages are within the specified limits,* turn OFF the AC POWER switch and proceed to the next section.

7. *If voltage measurements are not within the specified limits,* replace the defective power supply unit.

CAUTION

DO NOT attempt to adjust the power supply voltages. Damage caused by unauthorized voltage adjustment will not be covered by the warranty.

Appendix E — Upgrading to v11.0

CONTENTS PAGE

1. Introduction E-2
2. Version 11.0 Upgrade Instructions E-2
3. 99 Nodes Validation Test E-8
4. CPS Upgrade. E-8

1. INTRODUCTION

1.1 This chapter outlines the required steps for upgrading an existing system to version 9.0 or later. For details about specific steps, see the associated documentation. All of the requirements, capacities, features, and functionality for v9.1 are supported in v10.0 or later.

1.2 To support the 99 nodes functionality, ensure that you upgrade all nodes on the network to v10.0 or later and perform the validation test described on page E-8.

2. VERSION 11.0 UPGRADE INSTRUCTIONS

2.1 If you are upgrading a system to v9.006 or later, follow the procedures outlined below.

2.2 To upgrade to a v9.0 or later system:

NOTE: If you are upgrading a v8.x system to v9.0 or later, no further software licensing for premium features and security keys for CP Servers are required. However, you must upgrade the system license using the upgrade feature.

1. There are several items to consider, depending on the current version of software. These items include the following:

   - Axxess v11.0 now supports the Call Processing Server on Windows Server 2003. Not all hardware configurations can be upgraded to Windows Server 2003. See Table E-2 on page E-8 for the CPS upgrade method to Windows Server 2003. Axxess v11.0 continues to support Windows 2000 Professional OS for the CPS.
   - It is strongly recommended that you read this entire document to become familiar with the features and enhancements added for the current version before starting the upgrade.
   - The appropriate software license that includes all of the currently enabled premium features is required (see page 6-45 for details).
   - If upgrading a CPU64 to a 128 or higher system, you must purchase an upgrade for each feature. CPU64 features are not compatible with 128 or higher systems. (If you have an EVMC, you will need a 200 mailbox PAL.)
   - If upgrading from a version prior to v8.0, all CPU64, CPU128, and CPU256 cards must be changed to a CPC. CPU512 systems running software versions prior to v8.0 need to be upgraded to CPS systems.
   - The CPC IP port replaces one of the three system serial ports. If your system currently uses all three serial ports on the older CPU cards, determine which connection will use the new IP port. (Inter-Tel recommends that you use the IP port for DB Studio.)

     NOTE: You can use the modem as a third system serial port, if desired.

   - Because the CPC (or CP Server) has an available IP port, you can make multiple socket connections for applications, DB Studio, SMDR, etc. to the card (or server). The default connections for a CPC are: 1 DB Studio connection, 1 Voice Processing connection, 3 OAI connections, 3 Message Print clients (via Diagnostics Monitor), 3 SMDR clients, 3 System Monitor clients, 10 Desktop OAI clients, and 1 System Manager connection. **The sum of the socket connections cannot exceed 10 on a CPC.**

The default connections for a CPS are the same as the CPC except that a CPS can have 30 OAI connections, 30 Message Print clients, 30 SMDR clients, 30 System Monitor clients, and 255 Desktop OAI connections.

Inter-Tel strongly recommends that you use the socket connections whenever possible. Because IP connections are faster and more reliable than serial connections. Most OAI applications support IP connections.

NOTE: As of January 2004, Inter-Tel ceased development of Desktop Open Architecture Interface (OAI) applications and the Desktop OAI Toolkit was retired. Desktop OAI applications will run on v9.0 or later, but Inter-Tel will no longer develop new Desktop OAI applications.

- The IP port on the CPC supports OAI application connections. The number of applications connected to the CPC, however, may affect system speed.
- If upgrading a system running v7.x software or earlier that uses Call Processing Server redundancy, you must order new Master and Backup security keys (part no. 827.9775 and 827.9776 respectively). In addition, you must order the redundancy feature on the license.
- A software license is required only for the Master CP Server, not the Backup server.
- For a CPS 512 or ATM system, you must upgrade the firmware on the CPU020/EAC or PCMA card for use with the CP Server. The latest production builds are available on the *edGe* (www.inter-tel.com/software). For readers located in Europe, download the software from the Inter-Tel Europe FTP server.
- For Windows 2000-based CP Servers, you may need to uninstall the Watchdog Timer Card driver and install a new driver (see step 9). The new driver and instructions are available in Tech Central on the *edGe*. For readers located in Europe, contact the Inter-Tel Europe office for product information. The *Upgrade Instructions for the Berkshire PC Watchdog Timer Card Driver* (document part number 835.2735) includes procedures for identifying the server's current Watchdog Timer Card driver. The new driver is required to support Windows Service Pack 4. Currently, the Windows 2000-based CP Servers ship with Windows Service Pack 4.
- Versions 8.0 and later do not require feature PALs. In addition, the CP Server no longer requires PALs.
- As of version 8.0, the OS/2-based Voice Processing Unit is no longer supported. Axxess software versions 8.0 and later support the Windows NT/2000-based Voice Processing Units, as well as the OS/2-based and Linux-based EVMCs.
- To support more than 16 IP endpoints, the IPRC requires the daughter card (part number 550.2266) and v8.2.0 or later IPRC firmware.
- The following cards and devices require new firmware:
  - DKSC-16+s (see the *Digital Keyset Card Firmware Replacement Instructions* for details)
  - *If upgrading from v8.0 or earlier,* device IPRCs if connecting to SIP or IP endpoints (except the IP softphone)
  - *If upgrading from v8.0 or earlier,* hard IP endpoints

NOTE: Version 8.x T1Cs do not require firmware upgrades for v9.0 or later.

- The IP softphone requires v1.x firmware and is not supported by IPRCs that have v7.0.5 or later firmware. Other IP endpoints, such as the multi-protocol endpoints require IPRCs with v8.2.0 or later firmware to support point-to-point (PPP) audio. For these reasons, Inter-Tel recommends that you do not connect IP softphones and other IP endpoints to the same IPRC. If you do, you may lose feature or end-point functionality.
- If implementing networking, you need a universal numbering plan.
- System Manager has been updated to v1.1 for Axxess v9.0 or later. To use System Manager for all nodes on the network, it is not necessary to upgrade all nodes to v9.0 or later. However, the System Manager Call Processing Snap-In needs to be upgraded to v9.0 or later. It is also recommended that you upgrade to v9.0 or later DB Studio on the System Manger PC. (This upgrade is required if you are loading the CA certificate through the System Manager programming interface.)
- Version 9.0 or later software includes a new utility that zips all of the files in the Freeze directory into a single file for CPS systems. This utility makes it easier to transfer files to Technical Support for diagnostics and troubleshooting. The upper limit on the zip utility is 2 GB. Therefore, backup or remove the Freeze file contents from the CPS systems before installing the v9.000 or later CPS software.
- To use v9.0 or later features on a network containing CT Gateways, Inter-Tel recommends that you upgrade all CT Gateways to v4.2. (CT Gateway versions prior to v3.001 are incompatible with the v9.0 or later system; therefore, each CT Gateway must be running v3.001 or later.)
- Version 9.0 or later software does not support Attendant Console version 2.0 or earlier. Upgrade all Attendant Consoles to version 3.003.
- If implementing an ATM node for the first time, you must merge the individual node databases (using the Merge Utility), upgrade the appropriate CPU firmware, and install the appropriate licenses. In addition, the CP Server must be running Windows 2000. (For details about ATM installation and requirements, see page 3-166.)

NOTE: Make sure all of the system's baseline extensions are at default values before merging the databases.

- If running v9.0 or later software on a networked system, Inter-Tel recommends that all other nodes be running version 7.0 or later.

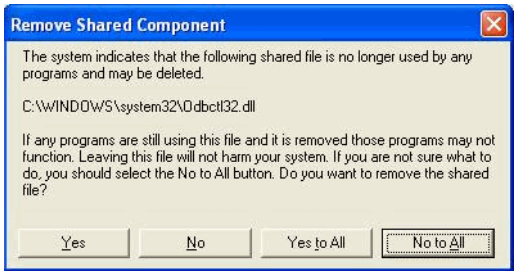
NOTE: For additional information about upgrades not covered in this document, see the latest version of the A*xxess Frequently Asked Questions* document available on the *edGe*. For readers located in Europe, contact the Inter-Tel Europe office for technical information.

2. Back up and save both the system and voice mail databases. See page 6-65 for detailed instructions.
3. Force a database backup (versions 8.0 and later only) to save the database in flash memory.

4. If the upgrade includes consolidating nodes via ATM and the system has a CT Gateway, delete the nodes that are being merged into a single node from the CT Gateway. For example, if you are consolidating nodes 1-4 into node 1, delete nodes 2-4 in the CT Gateway using the **Delete Connection** option under the File menu.
5. Stop the Attendant Console applications that are connected to the CT Gateway.
6. Stop all other CT applications that are connected to the CT Gateway.
7. Stop the CT Gateway. If the CT Gateway is running as a service, you must stop the Gateway using the Service Control Manager of the Control Panel.
8. Install the new DB Studio software on the programming PC and convert the saved database using DB Converter.

NOTE: Inter-Tel recommends that you convert the database using DB Converter whenever upgrading the system software. Do **NOT** rely on the backup database stored in the CPC flash memory. Using the flash database for upgrades may result in database conflicts. Run the DB Test utility to verify that the conversion was successful.

If you are uninstalling previous versions of Database Programming, the Remove Shared Component dialog box may appear prompting you whether you want to remove shared files that are no longer used by any programs. Click **No to All** to retain all the files. An example of the Remove Shared Component dialog box is shown below.



9. Upgrade the CP Server:
   - *If using an NT-based CP Server,* install the latest Service Packs and security updates from Microsoft®.

     NOTE: Inter-Tel does *not* support field upgrades from an NT-based server to a 2000-based server.

   - *If using a 2000-based CP Server,* uninstall the "PCI PC Watchdog" driver for the Watchdog Timer Card and install the new driver. After installed, you can load the latest Service Packs security updates and critical updates from Microsoft.

     NOTE: The "PCI PC Watchdog" driver for the Watchdog Timer Card does *not* support Service Packs 3 and 4. Attempting to install these Service Packs before upgrading the driver may result in a PC failure. See the latest version of the *Upgrade Instructions for the Berkshire PC Watchdog Timer Card Driver* (document part number 835.2735) available on the *edGe* for more information. For readers located in Europe, contact the Inter-Tel Europe office for technical information.

10. Install the CPC or CP Server:
    - *If installing a CPC,* flash the CPC with the new Call Processing software.
    - *If installing a CP Server,* load the new EAC or ATM operating system available on the *edGe* and add the Master security key (part no. 827.9775). If using a redundant system, add the Backup security key (part no. 827.9776). For readers located in Europe, contact the Inter-Tel Europe office for technical information.

    NOTE: Inter-Tel recommends that you install the CPC or CP Server behind a firewall to help prevent IP security problems.

11. Install the new voice processing software on the voice mail PC. If necessary, install the new Voice Processing Unit (i.e., if the site currently has an external OS/2 unit).
12. If upgrading from v7.x or earlier, upgrade the firmware chips on DKSC-16+s (part no. 841.0382), DKSC-16 (part no. 827.9202), SLC-16 (part no. 827.9075) and OPCs (part no. 827.9949 or 827.9950). See the *Digital Keyset Card Firmware Replacement Instructions* and *Options Card Firmware Replacement Instructions* respectively for details.
13. If upgrading from v7.x or earlier, upgrade the firmware on the OPCs using the appropriate OPC Firmware Upgrade Kit for the country that the equipment is in. The A-Law OPC Firmware Upgrade Kit (U.K.) part number is 828.1671, and the Mu-Law OPC Firmware Upgrade Kit (U.S. and Japan) part number is 828.1561. You do not need to upgrade the firmware on the OPC if you are upgrading from v8.x to v9.0 or later.
14. Upgrade the firmware on the 32-device IPRCs and networking IPRCs. If using v1.x firmware, upgrade to v1.5.2. If using v7.x or later firmware, upgrade to v9.0.0. The new firmware is available on the v9.0 or later software CD (part number 841.0130) or on the *edGe* (www.inter-tel.com/software). For readers located in Europe, download the firmware from the Inter-Tel Europe FTP server.

    NOTE: The IP softphone requires v1.x firmware and is not supported by IPRCs that have v8.1 or later firmware.

15. If your IP endpoints are connected to an IPC/IPRC configured as a DKSC-16 in DB Studio, leave the v1.5.1 firmware and PAL on the IPC/IPRC.

    NOTE: Peer-to-peer audio is not supported for IP endpoints connected to an IPC/IPRC with v1.5.1 firmware.

16. Upgrade the firmware on IPRCs. After you have upgraded the firmware, you can remove the PAL. Verify that the boot code is v1.x or later via OLM mode or Web interface.
17. If using a CP Server, install the new CP Server software on the CP Server PC.
18. Start a DB Studio session, using a socket connection, and load the software license.
19. Under System\Devices and Feature Codes\IP Ports\PPXXX, change the IP port settings on the CPC so that the CPC is available on the customer's network. (If the CPC does not have a static IP address available, view Message Print to determine the IP address assigned by the DHCP or BOOTP server.)

NOTE: A better way that does not require a reset would be to use a terminal serial connection port with the settings at 9600 bps, 8 bits, 1 stop bit, and no parity.

20. Start a new DB Studio session using a network connection that connects to the IP address (or hostname) programmed in step 19.
21. Run the DB Test Utility on the backed up database with the **Test/Repair All** option enabled.
22. Make sure the Voice Processing Unit is attached and restore the converted database.
23. If implementing IPRC networking, add the appropriate IPRCs.
24. If necessary, upgrade any existing IPRCs, T1Cs, IP endpoints, etc.
25. Program the system database to incorporate the desired v9.0 or later features. If you are expanding the system network to support more than 64 nodes, perform a 99 nodes validation test described on page E-8 after each of the new nodes is added to each site.
26. Test the system and voice mail to make sure everything is functioning properly.
27. If implementing IPRC networking, remove any unnecessary T1Cs and/or IP Networking Modules.
28. Install the CT Gateway upgrade software v4.201 or later. Do **not** uninstall the previous version; otherwise, you will have to reconfigure all CT Gateway settings. After you have installed the new version, start the CT Gateway and make sure all of the previously configured switch connections are properly communicating with the Gateway. To verify the connections, select **Switch Connections** from the View menu.
29. Because some applications, such as Contact Center Suite, (previously knows as Call Center Suite) require a database that does not contain duplicate extensions or trunks, use the "Find Duplicate Data" feature to identify redundant extensions. To find duplicate data, select View - Database - **By Node** and then click **Find Duplicate Data**.
30. Use the **New Switch Connection** option under the File menu to add any new switch connections that are required. In some cases, you can use the **Auto Configuration** option, but it is often just as easy to use the manual configuration option and click **Find** to add each new connection individually.
31. Upgrade Attendant Consoles to version 3.004 or later and start them.
32. Start any other CT applications that connect to the CT Gateway.

2.3 For details about specific steps, see the installation manual for the appropriate product (e.g., Axxess system, CT Gateway, Attendant Console, etc.).

APPENDICES

3. 99 NODES VALIDATION TEST

3.1 The following validation checklist allows you to evaluate the validity of the 99 nodes support on the system network. Perform the test after each of the new nodes is added to each site For complete information about 99 nodes support, see page 1-12.

Table E-1. *99 Nodes Validation Checklist*

Step	description	✓
1	Perform IC calls between Nodes 1, 2, 63, 64,75, and 99.	
2	Make local calls out T1, analog, or IP Gateway if appropriate.	
3	Make 911 calls to 911 operator.	
4	Place and receive calls to voicemail.	
5	Place calls to appropriate STAR applications or location specific applications as needed.	
6	Setup a conference call between same nodes as IC in step 1.	
7	Confirm that there is two-way audio and DTMF on all calls made	
8	Create an MSN hotmail account for Instant Message Network for communications while the nodes are converted. Each member is identified by "**`Node#_Location_First Name`** and **`Last Name Initial`** of person doing upgrade**`_Extension`**." For example, **`N02_chan_steves_29827@hotmail.com.`**	

4. CPS UPGRADE

4.1 Axxess v11.0 now supports the Call Processing Server on Windows Server 2003. Not all hardware configurations can be upgraded to Windows Server 2003. Axxess v11.0 continues to support Windows 2000 Professional OS for the CPS. Table E-2 shows the upgrade method to Windows Server 2003.

Table E-2. *CPS Upgrade Method to Windows Server 2003 Operating System*

Hardware Revision	Configuration	Can be Upgraded?	Upgrade Method
Rev AR and later	ATM and non-ATM	Yes	Follow the *Inter-Tel Axxess Call Processing Server Upgrade Kit Instructions* (part no. 835.3211)[a]
Rev AQ and earlier	ATM	No	Follow the *Inter-Tel Axxess Call Processing Hardware Server Upgrade Instructions* (part no. 835.3228)[b]
Rev AQ and earlier	non-ATM	No	Not applicable[c]

a. You must purchase a software upgrade kit (non-ATM part no. 828.1754, ATM part no. 828.1755), follow the upgrade instructions (835.3211) to reimage the existing CPS with Windows Server 2003, and then install the Axxess system software.

b. You must purchase a new ATM compatible CPS with Windows Server 2003 (part no. 550.5464), follow the hardware upgrade instructions (835.3228) to remove the ATM Adapter Card and cables from the existing CPS to install into the newly purchased CPS, and then install the Axxess system software.

c. The existing CPS does not support Windows Server 2003. You must purchase a new non-ATM CPS with Windows Server 2003, and then install the Axxess system software.

Appendix F – Network Topology

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1. INTRODUCTION

1.1 The Axxess system utilizes network communications for several purposes including administration, Internet Protocol (IP) endpoints, IP Trunks, and Inter-Tel private networking. The system can have one or more network connections shared among the different functions. This appendix discusses the security and associated network topology issues for the system. Network suitability (performance) for Voice over IP (VoIP) traffic is beyond the scope of this appendix. For information on VoIP traffic guidelines, refer to *Inter-Tel's VoIP Data Network Requirements* (part no. 835.2885).

2. TERMINOLOGY

2.1 The following terminology is used in this appendix:

- **ACL** (Access Control List): An ACL is a list of filters (restrictions) applied to communications passing through a router. Generally, each entry specifies a source, a destination, and a port. The source and destination can be individual IP addresses or subnets.
- **CPC** (Call Processing Card): A CPC is the card in an Axxess system that provides the resources necessary for call processing activities. The CPC card contains its own Ethernet port.
- **CPS** (Call Processing Server): An external Microsoft Windows-based PC that runs call processing in larger, processing-intensive applications. A CPS communicates with a CPC through a dedicated Ethernet connection.
- **DMZ** (Demilitarized Zone): A DMZ is a subnet that is "between" an organization's private network and the Internet. Communication between nodes in a DMZ, the Internet, and the internal network is controlled by a firewall. A DMZ provides some firewall protection for nodes that need to be available to the Internet, while also protecting the internal network from the DMZ nodes.
- **Enterprise® Conferencing**: Enterprise Conferencing is an Inter-Tel product that provides IP-based conferencing capabilities to Inter-Tel systems.
- **Firewall**: A firewall is a network device designed to provide limited connectivity between an organization's network and the Internet. A firewall generally prevents access from the Internet to an inside network while allowing access from the inside network to the Internet. Some firewalls also provide the capability for a special network, called a DMZ, that is placed between the Internet and the internal network.
- **Inter-Tel 5000 Communication Server:** This is a converged communications platform that provides and Internet Protocol (IP) solution for small- and medium-sized business environments.
- **Inter-Tel Private Networking**: This is the protocol used by the systems to provide networking functionality.
- **IPRC** (Internet Protocol Resource Card): This is a card for the Inter-Tel Axxess system that provides IP connectivity to Inter-Tel IP endpoints, trunk gateways, Inter-Tel Private Networking.
- **MGCP** (Media Gateway Control Protocol): MGCP is a standard protocol for commanding VoIP media gateways.
- **NAT** (Network Address Translation): A NAT is an Internet standard that enables a local-area network (LAN) to map between one set of IP addresses for internal traffic and a second set of addresses for external traffic.

- **NAT IP Address**: This is the public IP address that is statically mapped to a specific IPRC.
- **Native IP Address**: This is the native IP address that is statically mapped to a specific IPRC.
- **OAI**: (Open Application Interface): System OAI is a system-level communications protocol that allows command-oriented tools to be developed for the phone system. System OAI is available in two levels: Events and Third Party Call Control. System OAI Events provides call status information to an external device. Its primary use is for automatic call distribution products such as the Contact Center Suite. System OAI Third Party Call Control is a TSAPI protocol that allows system command-oriented tools to be used on the phone system. Contact Inter-Tel for more information regarding products and development materials for these features.
- **Port**: In Internet Protocol terminology, a port is a logical subaddress of an IP address. The port usually indicates the application layer protocol being used (for example, TCP port 80 indicates http).
- **Private**: In networking terminology, private refers to the portion of a network within an organization. This private network is considered trusted. The private network is separated from the public network (the Internet) by a firewall.
- **Public**: In networking terminology, public refers to the Internet. This network is considered untrusted because there are few controls about what it contains. The private network is separated from the public network (the Internet) by a firewall.
- **Router**: A router is a layer-3 device (OSI Model) that provides intelligent packet forwarding according to rules that describe the subnet topology. Routers usually make routing decisions based on IP address. Modern routers often contain some firewall functionality.
- **SIP** (Session Initiation Protocol): This protocol is an IETF standard for Internet telephony, conferencing, etc. It is a peer-to-peer text-based protocol focused on signaling setup and teardown. SIP is not a media protocol.
- **SIP Server**: SIP server is an Inter-Tel product that provides connectivity to SIP endpoints.
- **SPI** (Stateful Packet Inspection): SPI is provided by firewalls that allow IP communication to take place if it is initiated from the trusted network – the firewall keeps track of which sessions (conversations) are active (state).
- **Subnet**: A subnet is a grouping of devices that share common routing information.
- **TFTP** (Trivial File Transfer Protocol): TFTP is a simplified Internet standard for file transfers used primarily for loading configuration and software updates to IP devices. Unlike the more common FTP, TFTP provides no security mechanisms.
- **VPN** (Virtual Private Network): A VPN is a network that uses a public medium, typically the Internet, to provide connectivity to a logically private network. VPNs use encrypted tunnels to carry the private traffic in a secure manner.

3. NETWORK ADDRESS TRANSLATION OVERVIEW

3.1 Network Address Translation (NAT) is an Internet standard[1] that enables a local-area network (LAN) to use one set of IP addresses for internal traffic and a second set of addresses for external traffic. Generally, the NAT function is provided by a router or firewall.

3.2 The main purpose of NAT is to allow an organization to use a pool of (private) IP addresses that is separate from (public) Internet IP addresses. This alleviates a shortage of public IP addresses. Although NAT can provide a limited amount of security, it is rarely used stand-alone for security purposes[2].

3.3 To facilitate NAT, the Internet Assigned Numbers Authority (IANA) has designated certain IP addresses to be private[3]. This designation means that these IP addresses are not valid (routable) on the public Internet. This allows organizations to safely use these addresses within their networks. The designated private addresses are:

- 10.0.0.0 - 10.255.255.255 (One Class A Subnet)
- 172.16.0.0 - 172.31.255.255(16 Class B Subnets)
- 192.168.0.0 - 192.168.255.255 (256 Class C Subnets)

3.4 NAT operates by dynamically associating each internal private IP address with an external public IP address (and port). The NAT box (router or firewall) keeps track of the association between internal and external addresses and re-writes the IP packet header addresses as necessary. The association between internal and external IP addresses is generally short-lived based on activity.

3.5 Although NAT has been widely used throughout the Internet, some protocols do not work well with NAT. As described above, NAT translates the IP addresses in only the IP packet headers. The root of the problem is that some protocols carry IP addresses in the IP packet payload. As a result, private IP addresses are sometimes communicated out to the public Internet. By design, these private IP addresses are not accessible.

3.6 It is also possible to configure persistent or static NAT assignments in which a specific outside address is associated with a specific inside address. Although static NAT essentially allows an inside device to be accessible from the outside, it is still NAT and therefore problematic for some protocols when IP addresses are carried in the IP packet payload.

3.7 Some Internet applications (for example, IP telephony) do not allow use of NAT. Some firewall vendors offer the capability to "fix" NAT problems for specific protocols (for example, SIP).

3.8 Because NAT takes place where a private network connects to a public network, there are often two NAT operations taking place – one at each end of the communication. The NAT operations are the same at the two ends, but sometimes the impact on network protocols is different. For the purposes of this discussion, the NAT that takes place between the server (for example, an Axxess system) and the public network is referred to as near-end NAT. The NAT that takes place between IP endpoints and the public network (as in a home network) is referred to as far-end NAT.

---

1. Defined in RFC3022.
2. This is because the public address/port combinations can be easily guessed. Even with dynamic NAT, the address associations are open for long periods of (computer) time during which the internal computer is vulnerable to attack.
3. Defined in RFC1918.

3.9 There is a variation of NAT called PAT (Port Address Translation) in which the trusted IP address and port are translated dynamically to an outside IP address and (different) port. For the purposes of this appendix, the term NAT includes true PAT as well as true NAT.

4. INTERNET PROTOCOL RESOURCE CARD (IPRC)

4.1 Axxess systems support the use of IPRCs to provide VoIP communications for IP endpoints, IP gateway trunks, or IP networking. At least one IPRC is required to provide VoIP capabilities, but there may be applications for multiple IPRCs in a single system. Refer to the *IPC/IPRC and IP Endpoint Firmware Notice* (part no. 835.2720) for details about available IPRC firmware versions.

5. COMMUNICATION TYPES

5.1 The following communication types are discussed in this appendix.

- Administration
- IP Endpoints and Trunks
- Inter-Tel Private Networking
- Computer Telephony Integration (CTI)
- Enterprise Conferencing
- System Manager

A. ADMINISTRATION

5.2 The Axxess system provides administrative functions through its network connection. These functions include:

- Inter-Tel Database (DB) Programming
- Inter-Tel System Manager
- Web-based diagnostics
- Diagnostics Monitor
- TFTP: Trivial File Transfer Protocol
- SSH/Telnet/USB Console

B. IP ENDPOINTS AND TRUNKS

5.3 The Axxess system communicates to endpoints through IP. IP endpoints support an Inter-Tel proprietary protocol known as Inter-Tel Proprietary (ITP) mode as well as the industry-standards protocol SIP (Session Initiation Protocol), MGCP (Media Gateway Control Protocol), OAI (Open Architecture Interface), RTP (Real-time Transport Protocol), etc.

C. INTER-TEL PRIVATE NETWORKING

5.4 The Axxess system is capable of connecting with other Axxess systems and Inter-Tel 5000 platforms to form a network that supports many call processing features. This networking capability can be implemented using point-to-point T1 or PRI circuits, or it can be implemented using IP. This appendix is focused on IP-based Inter-Tel networking. See Appendix A — Networking for information on how to set up and program IP or T1 networking.

D. COMPUTER TELEPHONY INTEGRATION (CTI)

5.5 The Axxess system is capable of connecting with Computer Telephony Integration (CTI) applications, such System OAI (Open Architecture Interface, Contact Center Suite, etc.

E. ENTERPRISE CONFERENCING

5.6 Enterprise Conferencing runs on a stand-alone server to provide multiparty audio conferences. It utilizes ITP, OAI, and RTP, to communicate with the Axxess system as multiple Inter-Tel Protocol IP endpoints.

F. SYSTEM MANAGER

5.7 Inter-Tel System Manager is a server-based application that centralizes management functions for the Axxess system along with various peripheral components, such as Unified Communicator®, SIP Server, etc. For complete functionality, System Manager requires significant connectivity to and from the managed devices. These requirements are outside the scope of this appendix. They are described in the *System Manager Installation and Maintenance Manual* (part no. 835.2743).

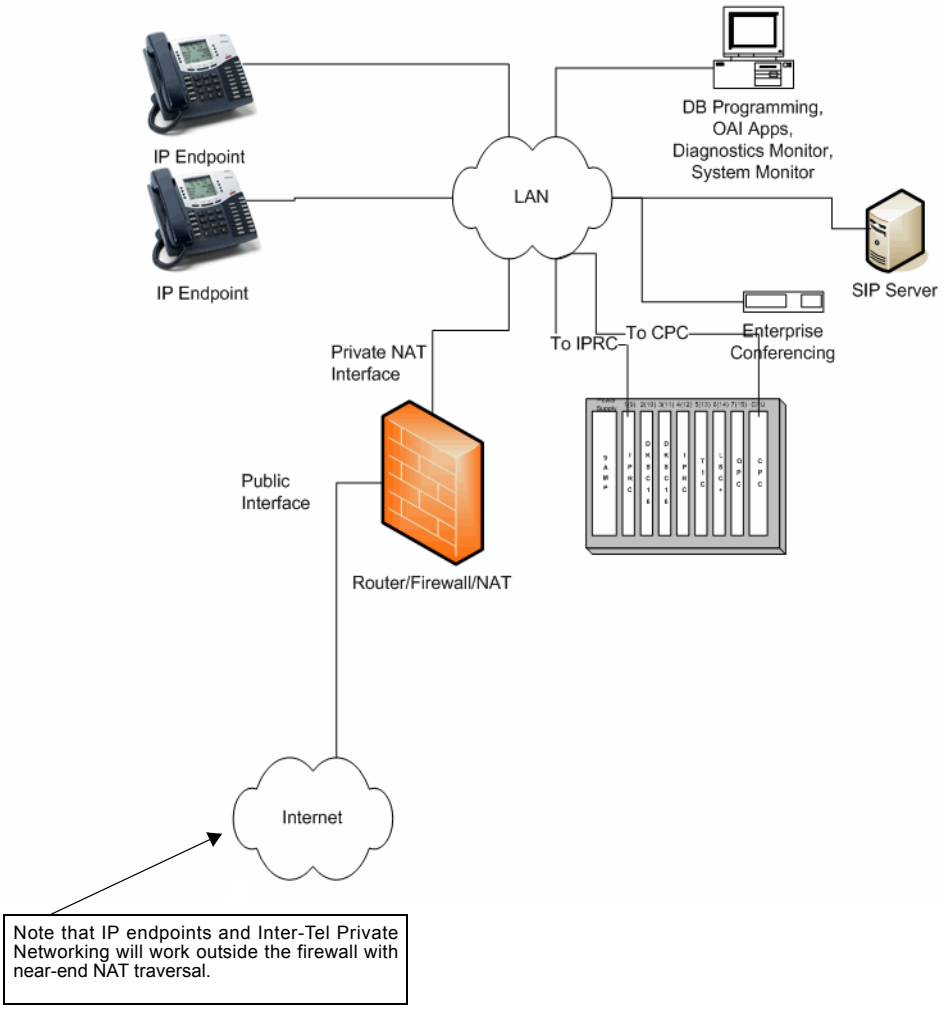
6. TOPOLOGY REQUIREMENTS

6.1 With regard to network topology discussed in this appendix, keep the following requirements in mind:

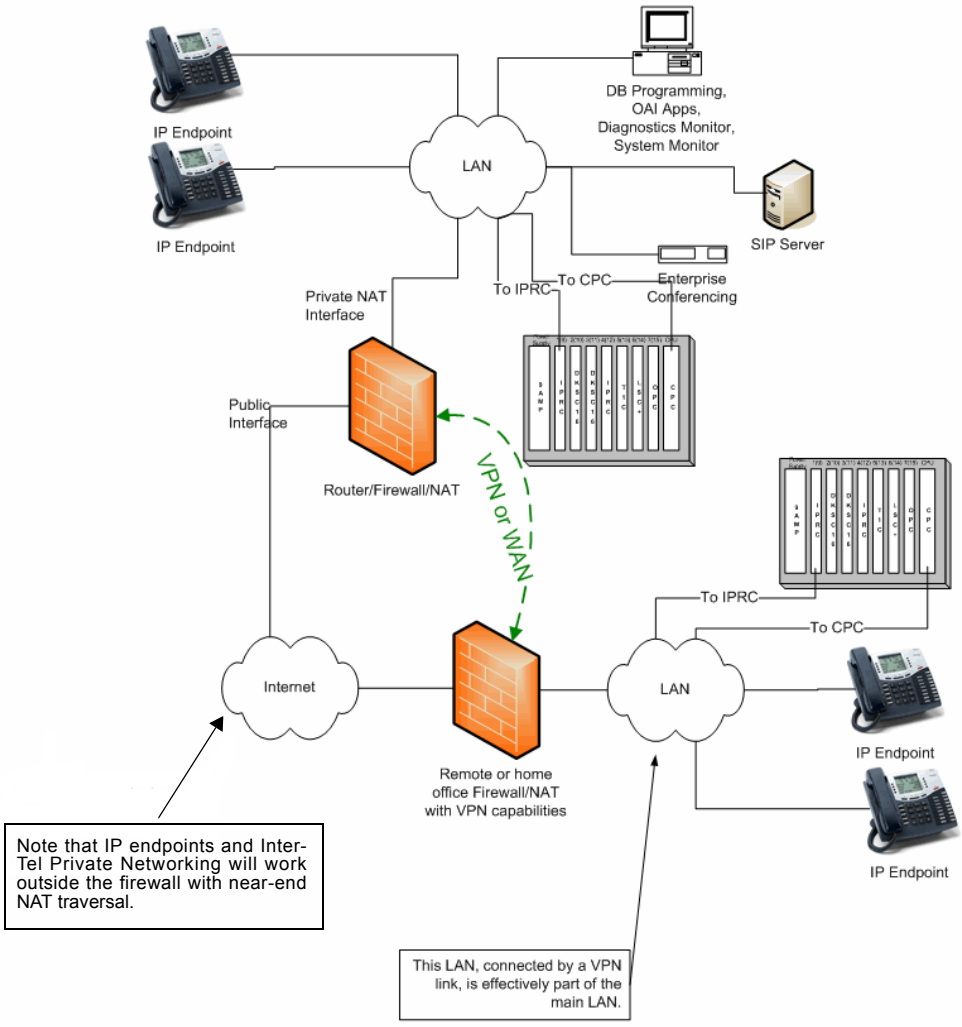
- An Axxess system has one CPC and optional IPRCs. In some configurations, a Call Processing Server (CPS) may be used. For nearly all VoIP purposes (except SIP Gateway Trunks), the network interface on the CPC/CPS is not used for voice.
- Inter-Tel Private Networking cannot operate through a NAT device.
- The communication between SIP endpoints and a SIP server must not pass through near-end NAT, unless it is performed by a SIP-aware firewall. The same requirement exists for a firewall performing far-end NAT.

7. PRIVATE (TRUSTED) NETWORKS

7.1 In the simplest form (a CPC and an IPRC), all IP devices, including endpoints, are on the same network as shown in the following illustration.



7.2 A Wide Area Network (WAN) provides the ability to connect two or more networks without necessarily passing through a firewall/NAT. It essentially makes one larger private network. A Virtual Private Network (VPN) uses public network facilities to provide private WAN connections. In either implementation, there are no restrictions on topology as long as all IP devices are on the same private network as shown below.



7.3 To allow IP endpoints on the Internet in the private networking scenario described above, you may want to consider the following options:

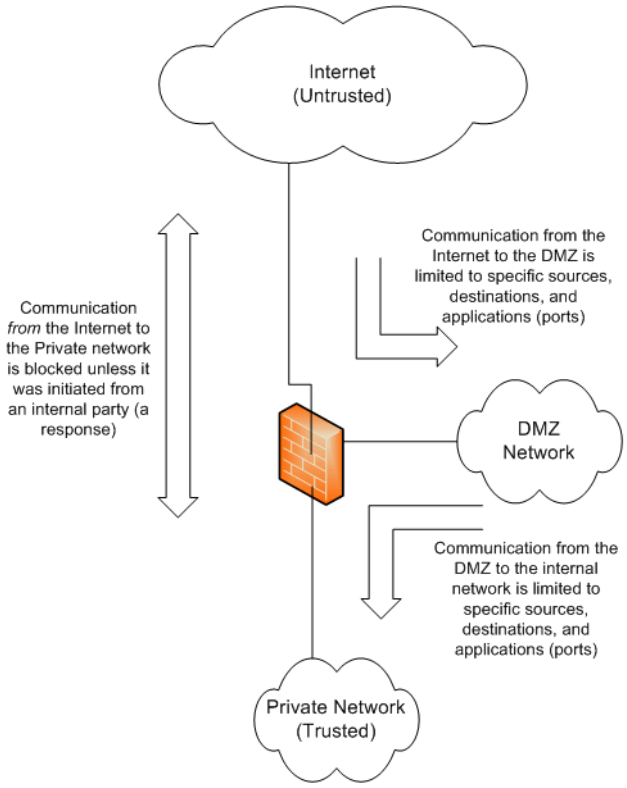
- Configure a static NAT assignment in the near-end NAT. Program the Axxess system to know IPRC's NATed address along with which endpoints are connecting from the Internet. See "Configuring Near-End NAT Traversal for IP Endpoints" on page F-14.
- Add an IPRC that connects outside the firewall (NAT). See "Using a Second IPRC to "Straddle" the Firewall (NAT)" on page F-13 for more information.
- Provide VPN connectivity[4] between the private network and the network containing the IP endpoint.
- Network an additional Axxess or Inter-Tel 5000 system using PRI networking, and place it in a non-NATed DMZ as described below.

4. Although VPN links avoid the security and NAT addressing issues, the encryption of VPN traffic sometimes introduces significant delays making it unsuitable for VoIP communications. This is a function of the network connections and the networking equipment.

8. PUBLIC (UNTRUSTED) NETWORKS

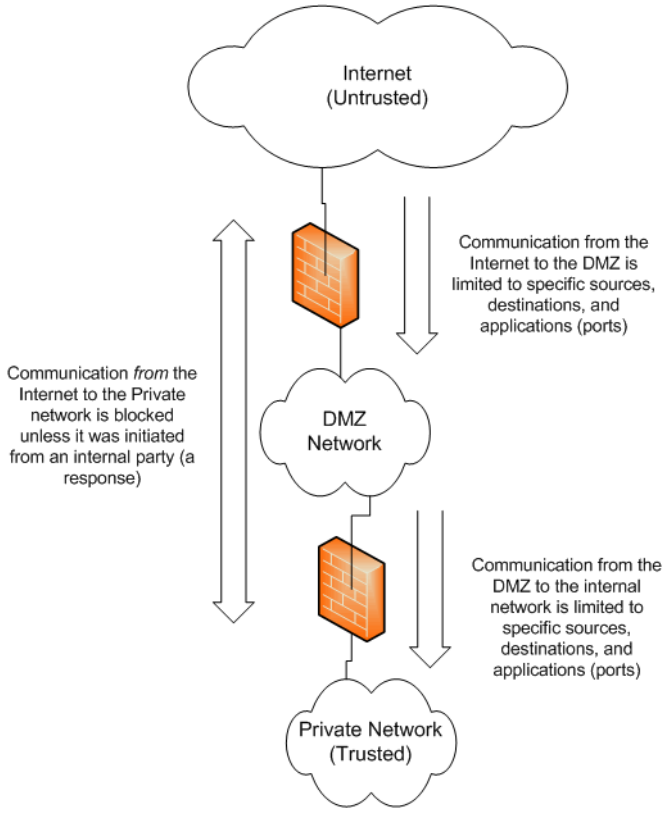
8.1 The private networking scenarios discussed in "Private (Trusted) Networks" on page F-7 impose very few restrictions on the topology of the IP telephony equipment because there are no security barriers between devices. However, one of the major benefits of IP telephony is the ability to leverage the existence of large public IP networks (for example, the Internet) to extend the communication network. Using public (untrusted) networks requires security measures to protect the private (trusted) network. This protection is usually provided through a firewall. Unfortunately, the functionality provided by a firewall impedes the communications necessary for IP telephony.

8.2 To use the public network in safe manner, it is necessary to provide some protection while allowing the necessary communications. In network terminology, this is usually configured using a topology known as a DMZ (DeMilitarized Zone)[5]. Devices in a DMZ network are accessible from the Internet in very specific ways according to the source and destination addresses and the applications that they support. However, the devices in the DMZ have very limited access into the private LAN to protect it should these devices be compromised. Devices in a DMZ network generally include Web servers, mail servers, etc.



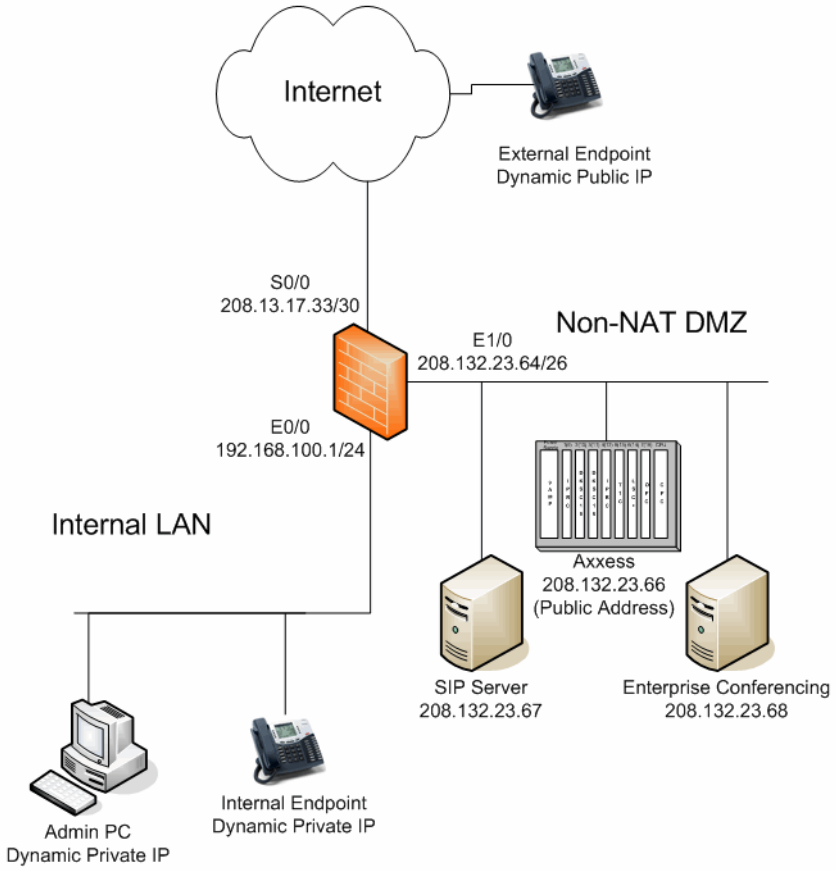
5. Note that many home router/firewalls (e.g., Linksys) inaccurately use the term DMZ to describe something else: an internal host accessible at a public address. This document uses the term DMZ as described in the definitions section (see page F-2).

8.3 In some environments, DMZs are implemented in what is referred to as a two-tier architecture using two firewalls as shown in the following example. In this environment, the DMZ is between the inside trusted network and the outside untrusted network. The security issues are still the same, but the configuration is split between the two firewalls.

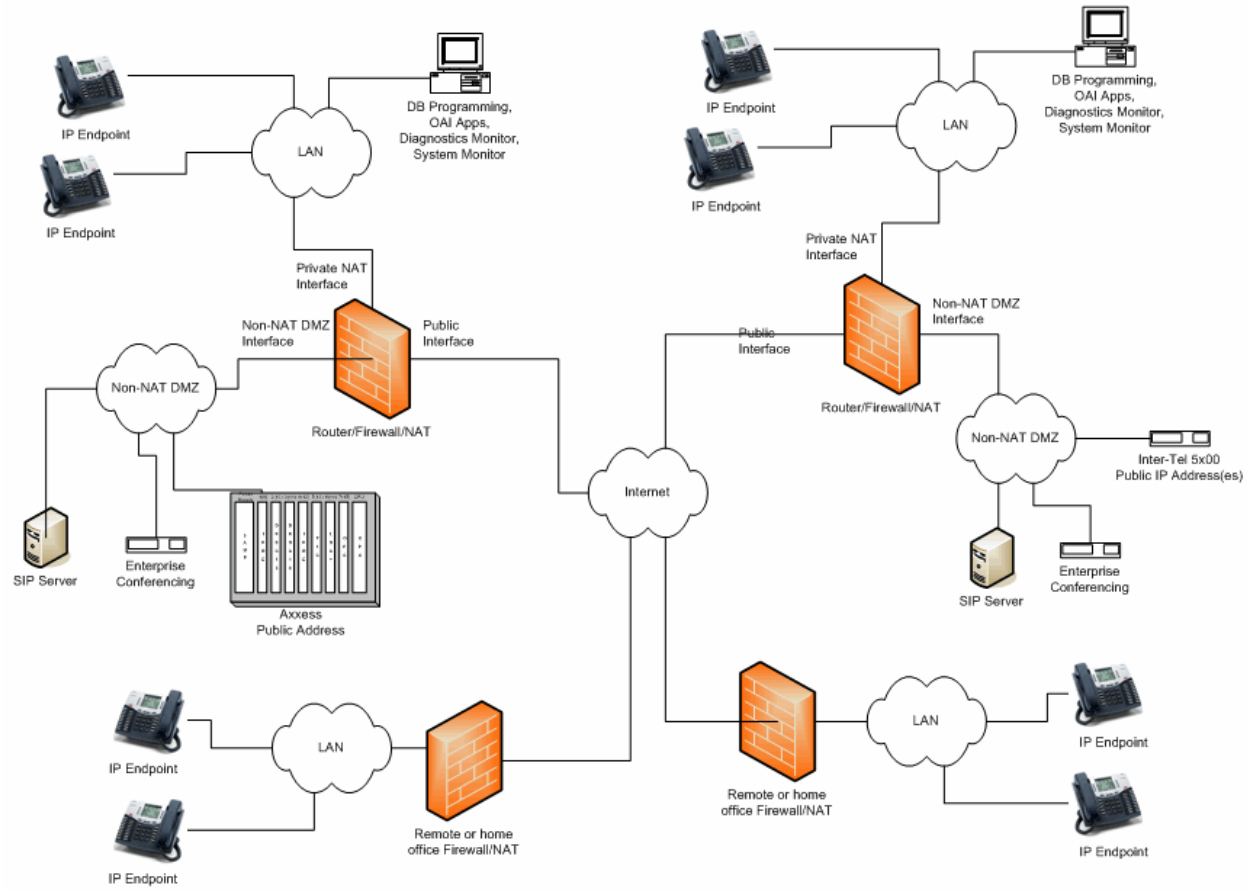


8.4 In addition to providing protection, firewalls also perform a Network Address Translation (NAT) function (described in "Terminology" on page F-2) in which private IP addresses are converted to public IP addresses as packets go out through the firewall. As a general rule, IP telephony devices do not perform well without special considerations when the communications must pass through a device performing NAT.

8.5 In order for the Axxess system and other IP devices to function properly in the DMZ, the DMZ must be a non-NAT network. This means that the devices in the DMZ use public IP addresses. The firewall can still provide some protection from the public Internet, but it does not translate the IP addresses or port numbers.

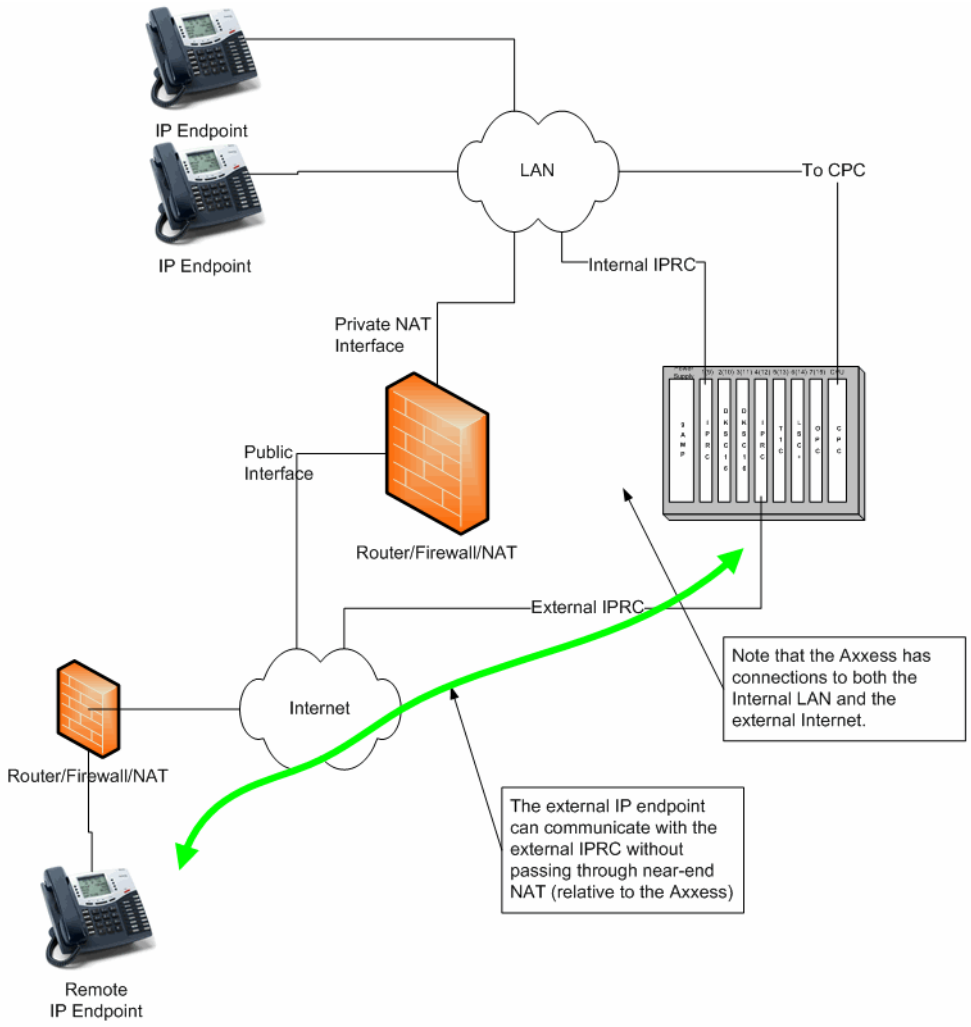


8.6 This topology uses public addressing. To support IP addressing means that Inter-Tel networking must be routed over the Internet. To use Inter-Tel Private Networking, set up each location in the same manner. An example is shown in the following illustration.



9. USING A SECOND IPRC TO "STRADDLE" THE FIREWALL (NAT)

9.1 Because the Axxess system supports multiple IPRCs, one option for supporting IP endpoints from the Internet is to add an IPRC and connect its Ethernet port outside the firewall (NAT) to a DMZ or the Internet itself. See the following diagram.



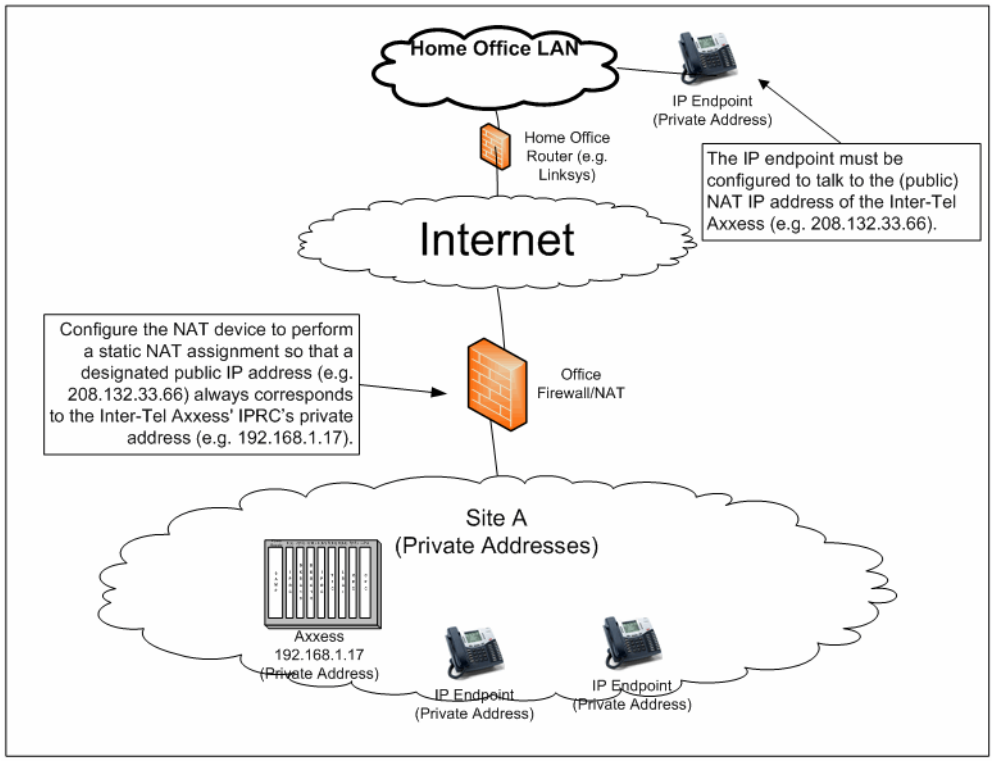
9.2 There are some considerations when using an additional IPRC:

- The recommended connection for the external IPRC is in a DMZ so that it is protected from security and denial-of-service (DOS) attacks from the Internet.
- All IPRCs used for IP networking must be on the same subnet.
- Although it may appear that this configuration introduces a security vulnerability by bypassing the firewall, there is no mechanism for Internet data to traverse the Axxess backplane.
- A "mobile" IP endpoint, such as the Model 8602 (sometimes inside and sometimes outside), may not work well with this topology.

10. CONFIGURING NEAR-END NAT TRAVERSAL FOR IP ENDPOINTS

10.1 The Axxess system supports traversing near-end NAT using statically assigned NAT addresses. In this configuration, a single IPRC can support internal IP endpoints, IP networking over the private network, and remote IP endpoints. An administrator configures the NAT device to always (statically) associate a specific public address with the Axxess IPRC's native address.

10.2 The following diagram illustrates the static NAT assignment and the IP endpoint configuration.



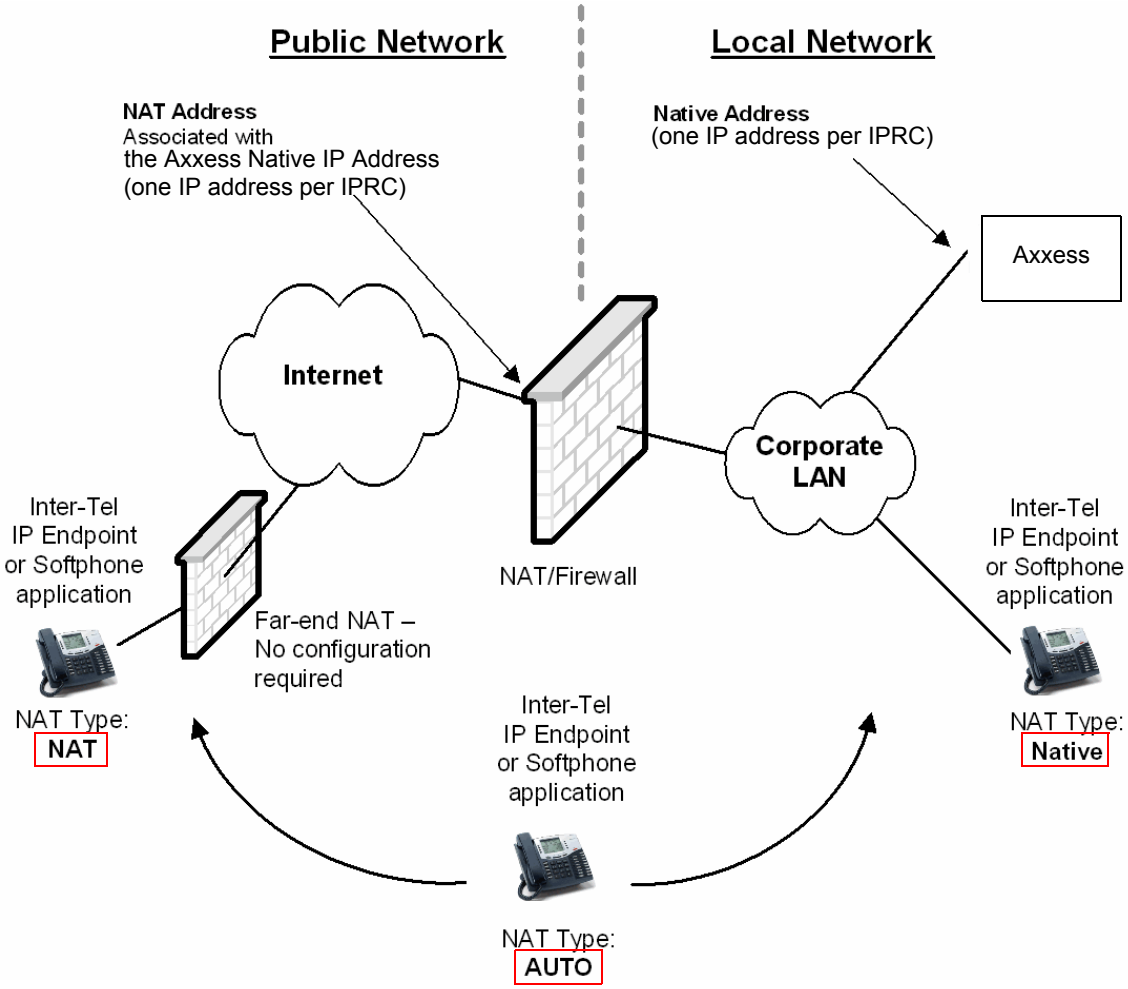
10.3 System v9.1 or later software can be programmed so that each IP endpoint could use either the Native address or the NAT address. In other words, the system can be programmed with corresponding IP addresses for communicating with each IP endpoint, whether the endpoint was located inside or outside the NAT/firewall. This is done in the NAT Address Type field in System\Devices and Feature Codes\Stations\Local\<*station*> when the <*station*> is an IP device (MGCP/SIP gateway and endpoints included). You can choose one of the following options:

- **Native:** Used when the endpoint is located inside the NAT (those that do not pass through near-end NAT).
- **NAT:** Used when the endpoint is located outside the NAT (those that do pass through near-end NAT).
- **Auto:** Allows the IP endpoints to be moved inside or outside the firewall (NAT) without programming intervention. This is the default value. See page F-16 for more information about Automatic NAT Detection.

NOTE: The "Auto" option is not applicable to SIP trunks. You must select either "Native" or "NAT" for SIP trunks.

10.4 Figure F-1 depicts the configurations of endpoints programmed for the three NAT Types in DB Programming.

Figure F-1. *Endpoint Configurations for Native, NAT, and Auto NAT Types*



10.5 To set up Near-End NAT Traversal for IP Endpoints on an Axxess system:

1. Configure the firewall/NAT device for a static mapping. This associates a public IP address with the IPRC native private address.

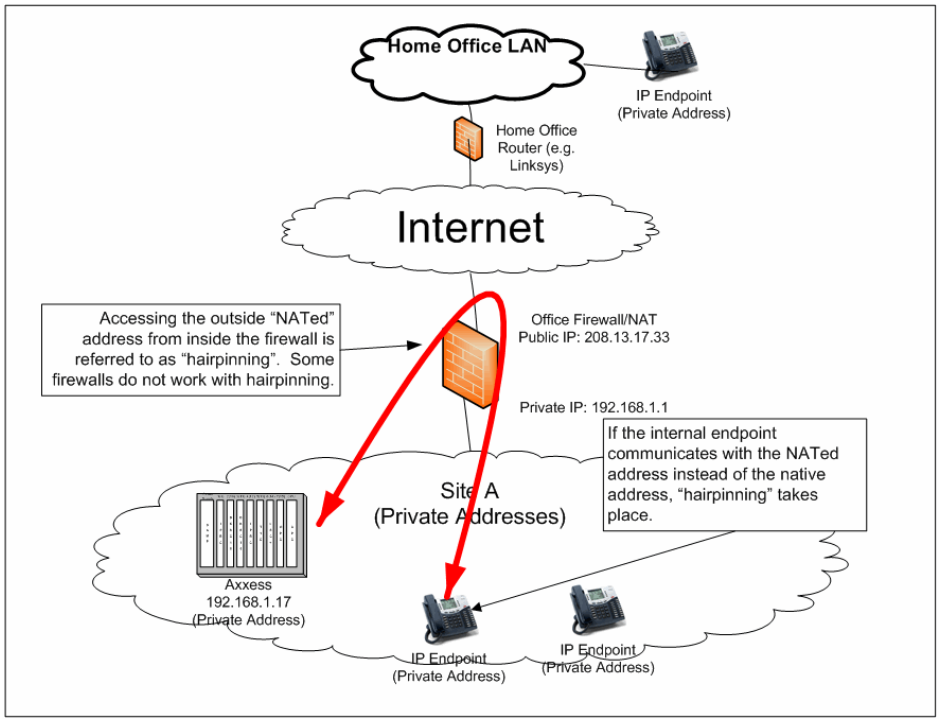
> **NOTE:** Configure the firewall security settings for the newly mapped IP address(es) as described in "Port Usage Summary" on page F-20. Depending on how the particular firewall defaults its security, failure to configure the security may result in no-audio conditions or security vulnerabilities.

2. In the System – Devices and Feature Codes – IP Connections folder, program the NAT IP Address to be the public address that was associated with the native private address in step 1 above.
3. In the System – Devices and Feature Codes – Stations folder, create an IP endpoint and specify the MAC address (or device ID).

APPENDICES

4. In the System\Devices and Feature Codes\Stations\Local\<extension>\IP Settings folder for the newly created IP endpoint from the Internet, set the NAT Address Type (Native, NAT, or Auto). The default is set to *Auto*.
5. Configure the IP endpoint (that is on the Internet) to communicate with the IPRC (public) NAT address. You can do this through the IP endpoint setup sequence or through the IP endpoint Web page. The address to use is the public address associated with the IPRC IP address.
6. *For SIP gateways only,* you must also program the System NAT IP Address field (in System\IP-Related Information) and the SIP Gateway Name field (in System\Trunk-Related Information\SIP Gateways).

10.6 Incorrect configuration may result in an undesirable effect called tromboning (also known as "hairpinning"). Not all firewalls support tromboning, so avoid this configuration, if possible. The following diagram illustrates tromboning.



10.7 **Automatic NAT Detection:** Automatic NAT detection is a method by which the Axxess and IP endpoint can communicate to determine whether the IP endpoint is inside the firewall (NAT) on the internal network or outside the firewall (NAT) on the Internet. This automatic capability is necessary for mobile IP endpoints that move between the internal network and the Internet. (This is a common scenario for the Model 8602 IP softphones). Note that this automatic detection capability is limited to IP endpoints in ITP mode; it does not apply to IP gateway trunks.

10.8 Under extremely rare scenarios, NAT autodetection might improperly detect a native condition instead of a NATed condition (possibly resulting in consistent one-way audio). If this situation occurs, contact technical support for further information to prevent this scenario.

10.9 If the "Auto" NAT Type is designated for an IP endpoint in DB Programming, the Auto NAT Detection feature determines the correct IP address to use when the endpoint comes online. After the NAT type is determined, the IP endpoint uses the setting for all calls. However, the NAT IP address associated with the system must be programmed manually in order for the NAT detection test to run.

10.10 Auto NAT Detection supports the following endpoints:

- Models 8600, 8620, 8622, 8662, and 8690 in ITP mode
- Model 8660
- Axxess/Eclipse IP PhonePlus
- IP SLA
- Model 8602 IP softphone

Auto NAT Detection does not support the following endpoints. The NAT setting must be set manually for:

- Inter-Tel IP endpoints in SIP mode
- Inter-Tel wireless IP endpoint Models 8664, 8665, and 8668
- Model 8601 SoftPhone for Pocket PC
- IP gateway trunks
- Non-Inter-Tel SIP phones

11. CONFIGURING NEAR-END NAT TRAVERSAL FOR IP GATEWAY TRUNKS

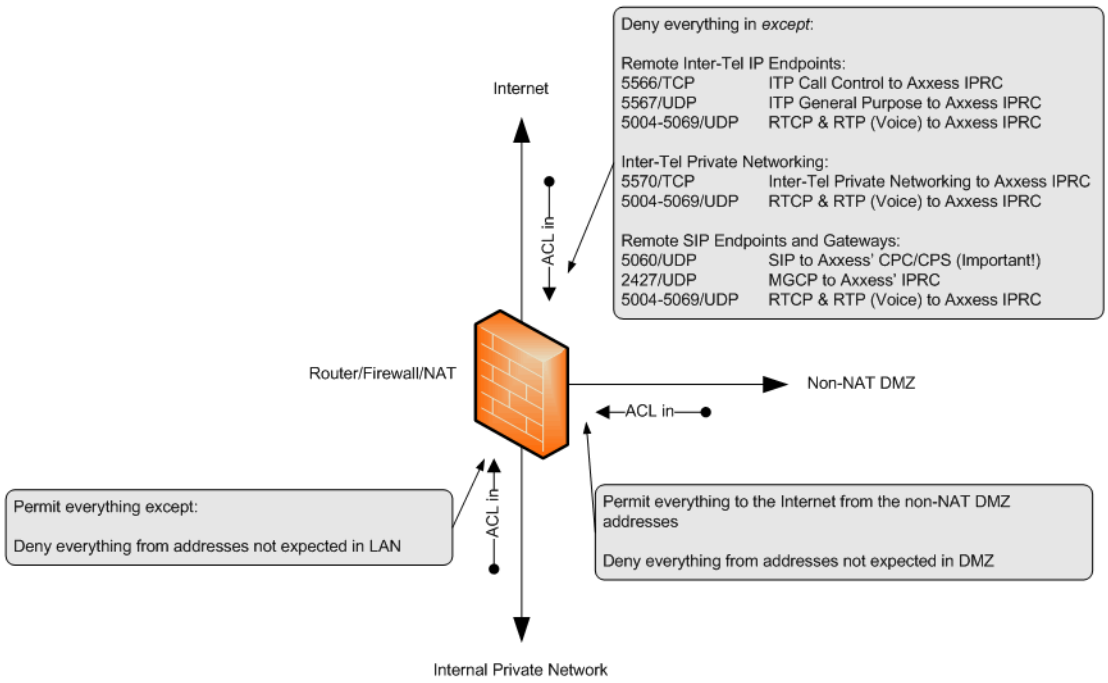
11.1 In addition to IP endpoints, the Axxess system also supports near-end NAT traversal for IP trunk gateways. Note that IP trunk gateways impose additional restrictions on *far-end* NAT.

11.2 To configure near-end NAT traversal for IP Gateway trunks:

1. Configure the static NAT assignments and the IPRC NAT IP Address assignments as described in step 1 on page F-15. This includes the information in the IMPORTANT notes.
2. In System\Devices and Feature Codes\Trunks, set the NAT Address Type to **NAT** for a MGCP Gateway and SIP Trunk gateway.
3. Repeat this process for each IP gateway trunk.

12. FIREWALL CONFIGURATION

12.1 In computer networks, a totally secure network is practically impossible. In general, the more types of communications allowed from the Internet, the less secure the network. Therefore, a network should be configured so as to allow only the minimum level of communication unless other capabilities are required. An example is shown in the following illustration.



12.2 In some environments, it will be necessary to allow additional types of communications through the firewall. As stated above, these communications should be allowed only when necessary. Although strong passwords are always a good idea, this is especially important when applications are opened to the Internet.

A. INTER-TEL PRIVATE NETWORKING

12.3 This is the protocol used to allow Axxess systems to communicate with each other (or to communicate with an Inter-Tel 5000 system). To allow Inter-Tel Private Networking to the Internet, allow TCP port 5570.

NOTE: Private networking cannot traverse NAT.

B. REMOTE ADMINISTRATION

12.4 These are the protocols that provide the capability to perform certain administration activities from outside the firewall. To enable remote administration including DB Programming over the Internet, enable TCP port 4000. To further tighten security, this communication can be filtered by source address[6].

6. Source address filtering accepts communication from specific IP addresses. While source address filtering provides some additional level of security, IP source addresses are easily spoofed. In addition, source addresses are often DHCP-assigned making it impractical to know the source addresses in advance.

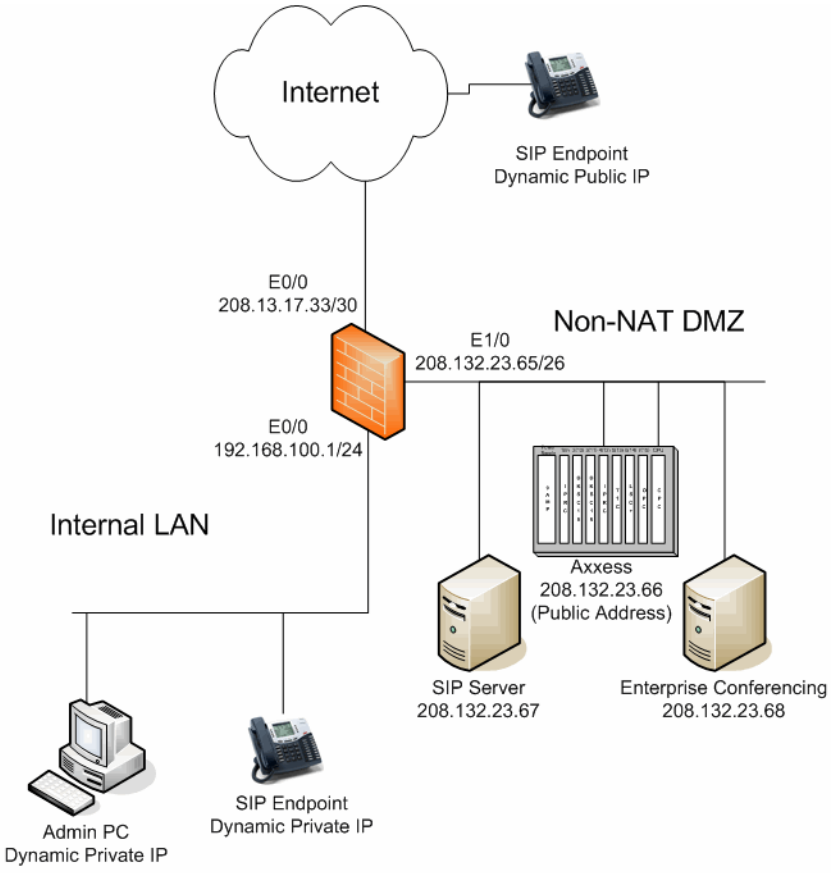
C. REMOTE MAINTENANCE AND DIAGNOSTICS

12.5 These protocols provide capabilities to diagnose problems, update software, etc. To enable remote maintenance and diagnostics over the Internet, enable the ports for DB Programming (TCP port 4000), Web interfaces (TCP port 80), SSH (TCP port 22), and AVDAP Monitor (TCP port 4444).

13. INTERNAL SESSION INITIATION PROTOCOL (SIP)

13.1 The SIP protocol presents some special challenges in that it does not function well when passing through NAT. In the diagram below, this means that the SIP server cannot communicate with the internal SIP endpoint. The SIP server must be in the DMZ because its communication with the Axxess system cannot pass through NAT. The only solution to this problem is to utilize a SIP-aware firewall that can perform the appropriate adjustments associated with NAT.

13.2 One other special characteristic of SIP on Axxess is that the call control communications go to the CPC (or CPS) while the audio goes to the IPRC. This needs to be taken into account for firewall configuration.



14. PORT USAGE SUMMARY

14.1 A port is a form of addressing that is subordinate to IP addressing. As such, port numbers serve to distinguish one protocol or application from another at the same IP address. For example, the primary protocol that browsers such as Internet Explorer use to access Web pages is HTTP, and HTTP uses port TCP 80. A single device with a single IP address can support many protocols simultaneously by using different port numbers.

14.2 Separate sets of ports exist for Transfer Control Protocol (TCP) and User Datagram Protocol (UDP).

14.3 Firewalls are devices used to promote network integrity and security by filtering data packets based on IP addresses and port numbers. For example, suppose a firewall is configured to allow communication with a Web server, which uses HTTP, TCP port 80. This configuration would prevent the server from being accessed as a mail server, which uses SMTP, TCP port 25.

14.4 For security reasons, all ports should be disabled except those that are specifically required. The default state for many routers is to disallow everything, but this practice varies from one manufacturer to another.

NOTE: This appendix assumes that everything is disallowed unless specifically allowed.

14.5 A firewall is designed to prevent access from an untrusted network such as the Internet while allowing communication initiated from within a trusted network. This capability is referred to as Stateful Packet Inspection (SPI) when the firewall is configured to keep track of the state of each session.

14.6 When enabled, SPI makes the firewall remember that an outbound TCP connection was initiated from behind the firewall and allows packets associated with this connection back through the firewall. SPI to or from the DMZ is required for satisfactory performance of Inter-Tel Private Networking.

NOTE: **IF:** You use Inter-Tel Private Networking over the Internet, and you have placed the Inter-Tel Axxess platform in a non-NATed DMZ to provide port-filtering protection from Internet threats. **THEN:** In addition to opening the ports specified in Table F-1, you must also enable SPI between the Internet and the DMZ.

14.7 Inter-Tel uses the port assignments shown in Table F-1 by default. Keep the following considerations in mind when opening ports:

- Many of these port associations are configurable through DB Programming.
- Installing Windows components on Inter-Tel products may open other ports that are not necessarily open.
- Installing non-tested and non-supported software on Inter-Tel products may open ports and cause security risk.
- Inter-Tel products can interface with third-party products that may use different port numbers.

Table F-1. *Port Assignments*

Port	TCP/UDP	Port Name	Inter-Tel Products	Access to:
21	TCP	FTP	• System Manager • Engineering File Transfer Protocol (FTP)	File downloads and file transfers, Board or Upload Utility
22	TCP	SSH	• Inter-Tel 5000 • Enterprise Conferencing (EC)	Used for remote access to IP products [1]
23	TCP	Telnet	• Call Processing Card (CPC) • IP Resource Card (IPRC) • IP Card (IPC) • Inter-Tel 5000 • System Manager • Unified Communicator (UC) • EC • IP endpoints • Web Conference	System programming and general information [1]
25	TCP	SMTP	• System Manager • AVDAP • Interactive Voice Response (IVR) • Contact Center Suite (CCS) • Enterprise Messaging (EM)	Used for sending mail
25	TCP	SMTP	EM (Outbound)	Simple Mail Transfer Protocol (SMTP) delivery for Unified Messaging (UM)
42	TCP/UDP	WINS	Most Inter-Tel products	Name resolution IP to Equipment Name
53	TCP/UDP	DNS	Most Inter-Tel products	Name resolution IP to Domain Name
67	UDP	DHCP	Most Inter-Tel products	Dynamic IP Allocation
67	UDP	BootP	Inter-Tel products are not BootP servers only clients except v1.X IPRC	Dynamic Allocation using the BootP process
68	UDP	BootP	Inter-Tel IPRC and IP endpoints	Client side of BootP
69	TCP	TFTP	IP products and IP endpoints	Used for uploading software and firmware updates
80	TCP	HTTP	• CPC • IPRC • IPC • Inter-Tel 5000 • System Manager • UC • EC • IP endpoints • Web Conference	Various features depending on the product

Table F-1. *Port Assignments (Continued)*

Port	TCP/UDP	Port Name	Inter-Tel Products	Access to:
110	TCP	POP3	• POP3 Exchange feature for Voice Mail products • IVR	Voice Mail e-mail features and UM
135	TCP	RPC	• Voice Mail systems using UM features • Messaging Application Programming Interface (MAPI) and faxing	Exchange services for e-mail features and UM
143	TCP	IMAP	EM (Outbound) • Voice Mail systems using UM features • MAPI and faxing	Internet Message Access Protocol (IMAP) login for E-mail Reader
161	UDP	SNMP	Network troubleshooting information for the Call Processing Server (CPS) servers	Using Simple Network Management Protocol (SNMP) management software with the CPS servers
162	UDP	SNMP	Network troubleshooting information for the CPS servers	Using SNMP management software with the CPS servers
389	TCP/UDP	LDAP	• UC • IVR	Database information
443	TCP	SSL Secure Web https	• System Manager • Automatic Diagnostics Delivery (ADD) feature on CPC and CPS • EC	• System Manager • File transfer for the ADD feature • EC features
993	TCP	IMAP	IMAP client on EM (Outbound)	SSL IMAP login for E-mail Reader
1433-1434	TCP/UDP	SQL	IVR database [2]	Structured Query Language (SQL) database
1800-1801	TCP/UDP	General	IVR MVEngine Service ports	IP messaging [1]
1863	TCP	General	UC	Required when using Session Initiated Protocol (SIP) server with UC and Microsoft Passports
1900	TCP/UDP		IVR	IP Telephony; Requires additional software [1]
2427	UDP	MGCP	Media Gateway Control Protocol (MGCP) Gateway	MGCP equipment
3001	UDP	General	Non ATM Call Processing Unit - CPU (EAC CPU to CPS Server)	512 CPS communications port between the CPS and EAC CPU
3307	TCP	SQL	IVR	SQL database
3389	TCP	Terminal Services	• UC • System Manager	Required for troubleshooting
3707	TCP	Secure Real-Time Connection	System Manager	SIP Server

Table F-1. *Port Assignments (Continued)*

Port	TCP/UDP	Port Name	Inter-Tel Products	Access to:
3986	TCP	General	CCS	CCS Server [1]
4000	TCP/UDP	General	• CPS • Between a CPS and Voice Processing Unit (VPU) • CT Gateway [3] • Desktop SoftPhone • Attendant Console • IVR • UC • EC	• Advanced communication platform to external products, DB Programming, System Open Architecture Interface (OAI), Diagnostics Monitor, Station Message Detail Recording (SMDR), and Message Print • IP sockets (used instead of RS232 ports) • Nodes, CPS and CPC, or applications [1]
4000	TCP	General	EM (Outbound)	ASAI connection to the CPC/CPS
4001	UDP	General	ATM ELAN	ATM ELAN; Port used by the ATM system for communicating between Redundant Servers
4444	UDP	General	VPU	AvdapMon [1]
4445	TCP	General	EM (Inbound/Outbound)	Networking Provider
4567	TCP	General	UC	UC communications for Client connections
5000-5018	UDP	General	Interprize 3200	Call Control for VoIP traffic [1]
5000-5099	TCP	General	UM (Inbound to EM)	UM server inbound file transfers
5004-5069	UDP	General	• IP devices • IPC • IPRC Networking	IP devices and IPRC Networking (IP Random Voice Ports) [1]
5004	UDP	General	• IPRC • IPC • IP endpoints [4]	Axxess Audio Stream Receive port [1]
5004	UDP	RTP	• IPRC • IP Resource Allocation (IPRA) • IP endpoints Ports must be odd numbers	IP Voice [1]
5005	UDP	RTPC	• IPRC • IPRA • IP endpoints Port must be odd numbers	IP Voice [1]
5060	TCP/UDP	SIP	SIP Server	Communications between the SIP server and IPRC or SIP devices
5070	TCP	General	IP Networking	TCP Call Control port [1]
5100-5149	TCP	General	UM	UM server outbound file transfers

Table F-1. *Port Assignments (Continued)*

Port	TCP/UDP	Port Name	Inter-Tel Products	Access to:
5444	TCP	General	EM (Inbound)	Default Operations, Administration, and Maintenance (OAM) Console admin interface
5445	TCP	General	EM (Inbound)	Default OAM Console event interface
5555-5556	TCP	General	IVR	Point Server
5566	TCP	General	• IP devices • IPRC Networking • IPC • Inter-Tel 5000	IP terminal TCP Call Control port [1]
5567	UDP	General	• IP devices • IPRC Networking • IPC • Inter-Tel 5000	IP terminal General Purpose port [1]
5568	UDP	General	• IPRC • IPRA • Inter-Tel 5000	Message server port for sending IP diagnostic information from Axxess or Inter-Tel 5000 [1]
5631	TCP	General	CPS Servers	pcAnywhere for remote access [1]
5632	UDP	General	CPS Servers	pcAnywhere for remote access [1]
6004	UDP	General	IP endpoints and Inter-Tel 5000 [4]	Inter-Tel 5000 Audio Stream Receive Port [1]
6004-6247	UDP	General	IP endpoints and Inter-Tel 5000 [4]	Inter-Tel 5000 pre-v2.0 Audio Steam
6004-6261	UDP	General	IP endpoints and Inter-Tel 5000 [4]	Inter-Tel 5000 v2.0 and later Audio Stream for the Processor Module (PM-1)
6604-6039	UDP	General	IP endpoints and Inter-Tel 5000 [4]	Inter-Tel 5000 v2.0 and later Audio Stream for the Processor Expansion Card (PEC-1)
7000	TCP	General	UM (Inbound to EM)	UM server
7777	UDP	General	UM on VPU	The UM clients [1]
8080	TCP	HTTP	IP products and IP endpoints	Programming information [1]
8886	TCP		SIP Server	SIP Server
16384	UDP	General	Interprize 3200	Voice Traffic (VoIP)
Various Ports [5]	TCP/UDP	General	Third-party or custom applications	Various features depending on the product

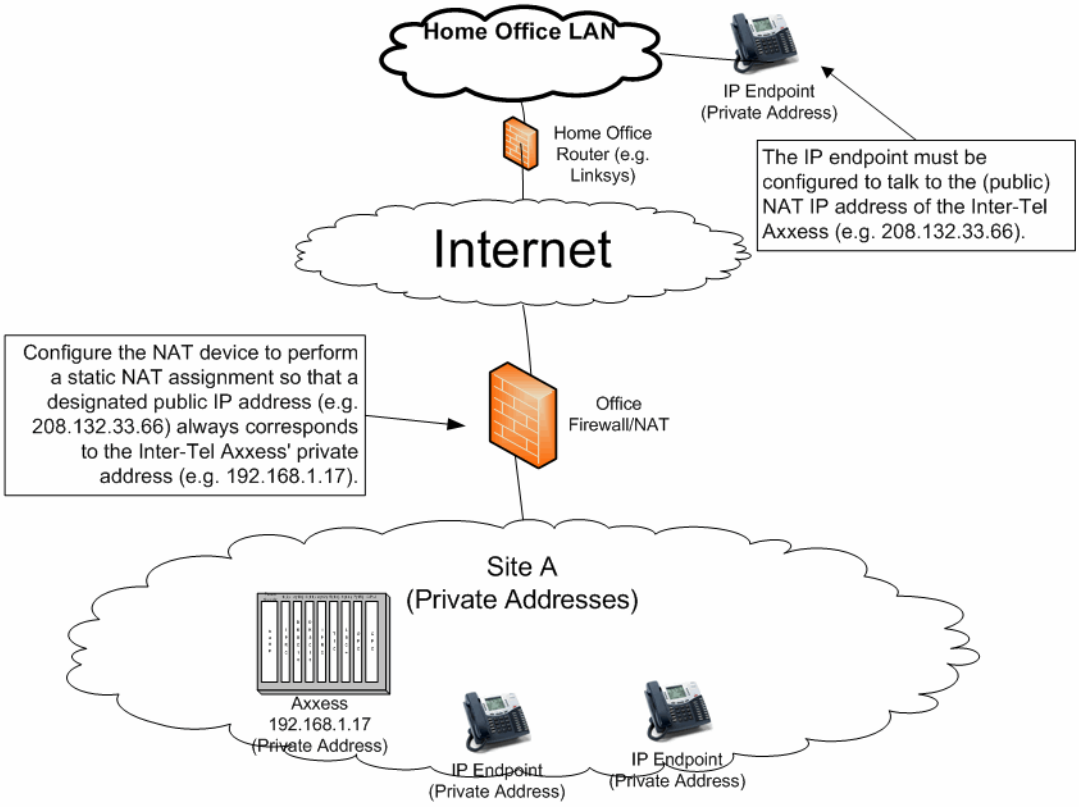
1. The listed port numbers for these applications are the default values and can be changed in DB Programming.
2. The IVR can use other types of database engines where other network ports can be used.
3. CT Gateway has two programmable ports - one for communicating to the nodes and one for communicating to the applications. Both of these ports are programmable and default to port 4000.
4. IP endpoints (Axxess/Eclipse IP PhonePlus and Model 8660) use port 5004 for audio. Newer IP endpoints (Model 86XX, excluding the Model 8660) use port 6004 for audio.
5. Third-party products that interface with Inter-Tel products may use port numbers other than what are listed in this document.

15. EXAMPLE CISCO CONFIGURATION COMMANDS

15.1 The following commands are provided as an example to further illustrate the recommended configuration with respect to security. Quality of Service (QoS) configuration is beyond the scope of this appendix. These configuration examples are based on the Cisco® IOS Firewall feature set. Adtran® routers use a very similar command set called AOS. See "Adtran NetVanta Switch/Router Example" on page F-30 for more information on Adtran routers. **These commands are not intended to be complete; they are only a guideline.**

A. PRIVATE NETWORK WITH NEAR-END NAT TRAVERSAL

15.2 The following diagram shows a configuration example.



15.3 The following commands set up the connection to the internal LAN. NAT is enabled, and it uses `access-group e0in` for traffic coming in (to the router) from the internal LAN.

```
interface Ethernet0/0
 description Internal LAN
 ip address 192.168.1.1 255.255.255.0
 ip access-group e0in in
 ip nat inside
 ip inspect inspector in
!
```

- The next section defines the access control list (the rules) for traffic coming from the internal LAN into the router. As a general rule here, you want to allow about everything to go out from a trusted LAN.

```
! Access Control List e0in
!
ip access-list extended e0in
 permit ip 192.168.1.0 0.0.0.255 any
 deny   ip any any
```

- The next section sets up the connection to the Internet. NAT is enabled between the Internet and the internal LAN. Traffic from the Internet is filtered using the access-group called *Internet*.

```
interface Serial0/0
 description connected to Internet
 ip address 208.13.17.33 255.255.255.252
 ip access-group s0in in
 ip nat outside
```

- The next section defines the access control list (the rules) for traffic coming from the Internet to either the Internal LAN or the DMZ. This is the first line of defense, so you want to filter as much as possible. Responses to communications initiated from inside (for example, http request for a Web page) are controlled by the firewall functionality through dynamic ACLs.

```
ip access-list extended s0in
 permit tcp any host 208.13.17.2 eq 5566
 permit udp any host 208.13.17.2 eq 5567
 permit udp any host 208.13.17.2 range 5004 5069
 deny   ip any any
```

- The next command associates the public address with the IPRC private address. This creates the "NATed" address. The ISP must be known to send packets for this public IP to the Axxess system.

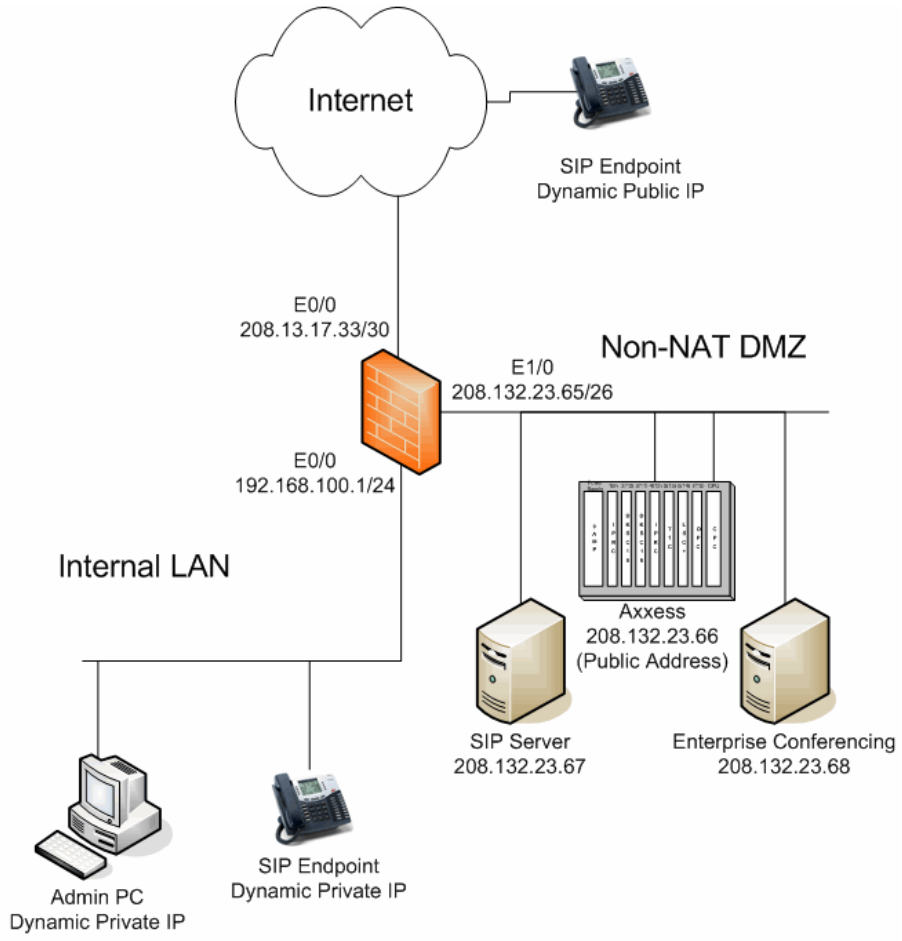
```
ip nat inside source static 192.168.1.2 208.13.17.2
!
```

- Finally, the following commands set up the dynamic NAT configuration to use a pool of public addresses. This is not specific to this example, but it is included here to contrast static versus dynamic NAT.

```
ip nat translation timeout 86400
ip nat translation tcp-timeout 86400
ip nat translation udp-timeout 300
ip nat translation dns-timeout 60
ip nat translation finrst-timeout 60
ip nat pool NatPool0 208.13.17.40 208.13.17.62 netmask
255.255.255.0
ip nat inside source list 1 pool NatPool0 overload
!
```

B. NON-NAT DMZ CONFIGURATION

15.4 The following diagram shows a configuration example.



Example 1: Single Node with ITP Endpoints

15.5 The following commands set up the connection to the internal LAN. In these commands, NAT is enabled and it uses `access-group e0in` for traffic coming in (to the router) from the internal LAN.

```
interface Ethernet0/0
 description Internal LAN
 ip address 192.168.100.1 255.255.255.0
 ip access-group e0in in
 ip nat inside
 ip inspect inspector in
!
```

- The next section defines the access control list (the rules) for traffic coming from the internal LAN into the router. As a general rule, this example allows just about anything to go out from the trusted LAN.

```
! Access Control List e0in
!
ip access-list extended e0in
 permit ip 192.168.100.0 0.0.0.255 any
 deny   ip any any
```

APPENDICES

- The following section sets up the connection to the Internet. NAT is enabled between the Internet and the internal LAN. Traffic from the Internet is filtered using the access-group called `Internet`.

```
interface Serial0/0
 description connected to Internet
 ip address 208.13.17.33 255.255.255.252
 ip access-group s0in in
 ip nat outside
```

- The next section defines the access control list (the rules) for traffic coming from the Internet to either the Internal LAN or the DMZ. This is the first line of defense, so filter as much as possible. Responses to communications initiated from inside (for example, http request for a Web page) are controlled by the firewall functionality through dynamic ACLs.

```
ip access-list extended s0in
 permit tcp any host 208.132.23.66 eq 5566
 permit udp any host 208.132.23.66 eq 5567
 permit udp any host 208.132.23.66 range 5004 5069
 deny   ip any any
```

- The next section sets up the connection to the DMZ. NAT is not enabled between the Internet and the DMZ. Traffic from the Internet is filtered using the access-group called DMZ. The “inspect” statement enables the stateful firewall functionality.

```
interface Ethernet 1/0
 description Site DMZ LAN
 ip address 208.132.23.66 255.255.255.192
 ip inspect dmzinspector in
 ip access-group e1in in

 ip inspect name dmzinspector udp
 ip inspect name dmzinspector tcp
 ip inspect name dmzinspector sip
```

- The next section defines the access control list (the rules) for traffic coming from the DMZ to either the Internal LAN or the Internet. Limit the communications between the DMZ and the internal LAN as much as possible in the event one of the DMZ nodes is compromised.

```
ip access-list extended e1in
 deny   ip any 192.168.100.0 0.0.0.255
 permit ip any any
!
```

Example 2: ITP Endpoints and Inter-Tel Networking

15.6 To add support for Inter-Tel networking, expand the ACL to allow the Inter-Tel Private Networking port to be accessible from the Internet to the Axxess system. Responses to communications initiated from inside (for example, http request for a Web page) are controlled by the firewall functionality through dynamic ACLs.

```
ip access-list extended s0in
 permit tcp any host 208.132.23.66 eq 5566
 permit udp any host 208.132.23.66 eq 5567
 permit tcp any host 208.132.23.66 eq 5570
 permit udp any host 208.132.23.66 range 5004 5069
 deny   ip any any
```

Example 3: ITP Endpoints, Inter-Tel Networking, and Remote Administration/ Diagnostics

15.7 To allow remote administration and diagnostics, open up additional ports shown in bold in the following example. Each of these ports has a distinct function, so they should be evaluated individually to determine the need.

```
ip access-list extended s0in
 permit tcp any host 208.132.23.66 eq 5566
 permit udp any host 208.132.23.66 eq 5567
 permit tcp any host 208.132.23.66 eq 5570
 permit tcp any host 208.132.23.66 eq 4000
 permit tcp any host 208.132.23.66 eq 4444
 permit tcp any host 208.132.23.66 eq 80
 permit tcp any host 208.132.23.66 eq 22
 permit udp any host 208.132.23.66 range 5004 5069
 deny   ip any any
```

Example 4: ITP Endpoints, Inter-Tel Networking, Remote Administration/Diagnostics, and SIP Endpoints

15.8 To allow SIP endpoints from the Internet, open up the SIP port to the SIP server. SIP endpoints on the "internal LAN" may require additional "fixup" if NAT is performed between the internal LAN and the DMZ.

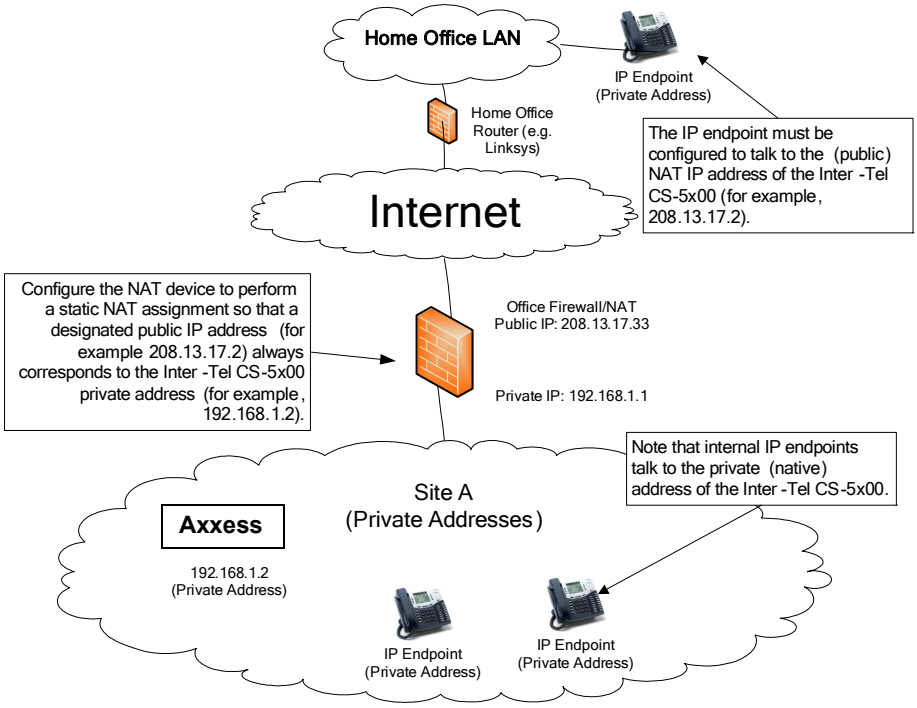
```
ip access-list extended s0in
 permit tcp any host 208.132.23.66 eq 5566
 permit udp any host 208.132.23.66 eq 5567
 permit tcp any host 208.132.23.66 eq 5570
 permit tcp any host 208.132.23.66 eq 4000
 permit tcp any host 208.132.23.66 eq 4444
 permit tcp any host 208.132.23.66 eq 80
 permit tcp any host 208.132.23.66 eq 22
 permit udp any host 208.132.23.67 eq 5060
 permit udp any host 208.132.23.66 range 5004 5069
 deny   ip any any
```

16. ADTRAN NETVANTA SWITCH/ROUTER EXAMPLE

16.1 Adtran® provides a family of products (NetVanta®) that are a combination of switch, router, and firewall[7]. The security considerations are independent of the router vendor, but the implementation of the DMZ using Adtran NetVanta products is slightly different. Instead of simply designating an Ethernet port on a router as the DMZ, you must first create a new VLAN containing one or more Ethernet ports. The VLAN creates a logical partition of the switch that can only communicate with the rest of the switch using the router/firewall functionality. Each set of VLAN Ethernet ports function as a single router interface. In addition, the following example makes use of access policies that allow you to apply multiple access control lists to a specific interface. **These commands are not intended to be complete; they are only a guideline.**

A. PRIVATE NETWORK WITH NEAR-END NAT TRAVERSAL

16.2 The following diagram shows an example configuration (this is the same configuration used in the Cisco example on page F-25).



16.3 The following commands define the VLANs within the switch. This is necessary to be able to route/firewall between the LAN and the Internet.

```
vlan 1
  name "Default"

!
```

7. Although the Adtran AOS is common among all of their switch/router platforms, this example is based on an Adtran NetVanta 1224R.

16.4 This is a set up of some of the ports in the default VLAN. Repeat for as many ports as you need to configure.

```
interface eth 0/1
  no shutdown
  switchport access vlan 1
!
interface eth 0/2
  no shutdown
  switchport access vlan 1
!
```

16.5 Each VLAN performs as a port on the router with its own IP address. Access policies are also applied to each interface. The access policies are defined as follows:

```
interface vlan 1
  ip address  192.168.1.1  255.255.255.0
  access-policy Private
  no shutdown
```

16.6 In this example, the connection is to the Internet with a T1. Set up the IP address and apply the access policy (defined further below). In this example, assume that the ISP is configured with this address as a gateway for the (public) NATed addresses.

```
interface t1 1/1
  clock source line
  tdm-group 1 timeslots 1-24 speed 64
  ip address  208.13.17.33  255.255.255.0

  access-policy Public
  no shutdown
```

- The following commands define access lists for the different traffic types. These defined lists do not have any effect until they are applied to interfaces.
  - The following commands define an access list so that we can communicate with the internal trusted network.

    ```
    ip access-list extended self
      remark Traffic to NetVanta
      permit ip any  any    log
    !
    ```

  - The next commands define the set of communications coming from any internal hosts *except* the Axxess system.

    ```
    ip access-list standard PrivateHosts
    deny host 192.168.1.2
    permit any log
    ```

  - The next commands define the set of communications coming from the Axxess system.

    ```
    ip access-list standard OutAXXIPRCMain
    permit host 192.168.1.2 log
    !
    ```

16.7 The following commands defines the communications that we want to allow to the Axxess system. In this example, this is limited to the communications required for ITP endpoints.

```
ip access-list extended InAXXIPRCMain
  permit tcp any  host 208.13.17.2 eq 5566  log
  permit udp any  host 208.13.17.2 eq 5567  log
  permit udp any  host 208.13.17.2 range 5004 5069   log
```

16.8 These commands define the policy for the interface to the private (trusted) network. Each policy can specify more than one access list. From a security perspective, allow everything from the trusted network. The "NAT" commands define the behavior of outbound NAT. If the packet is from the Axxess system, translate the source address to the specified public address. The second NAT command uses PAT on IP 208.13.17.33, the address on the public interface, for any other internal node.

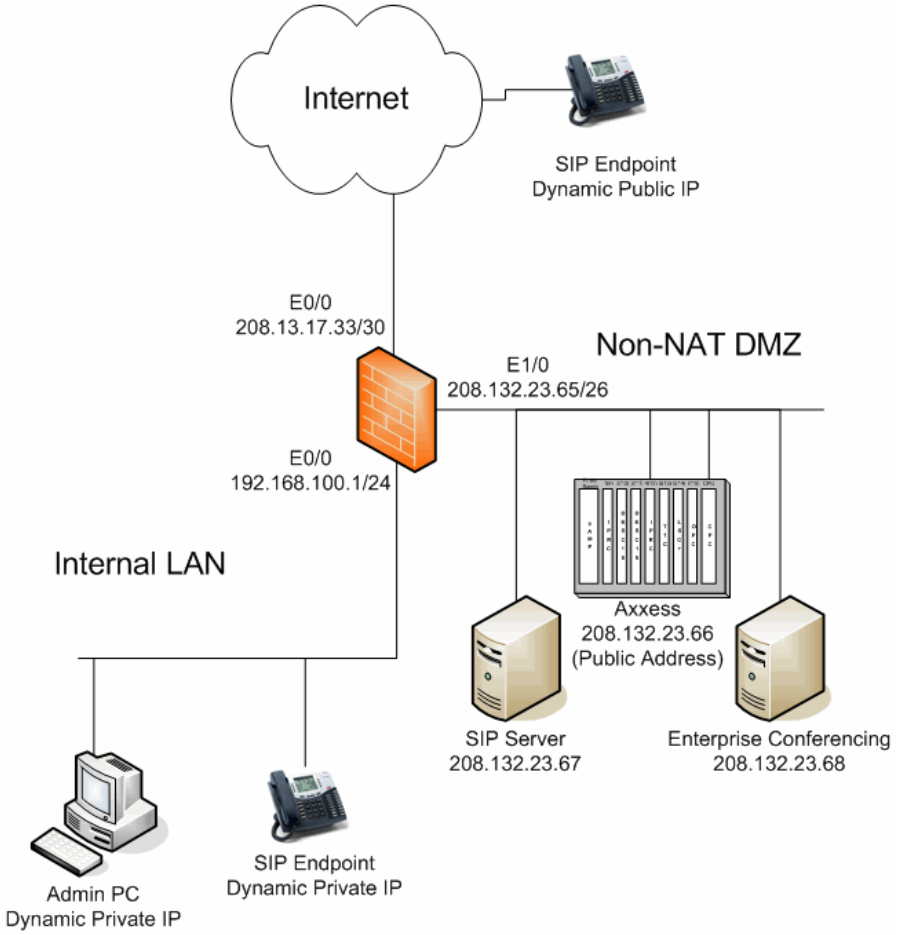
```
ip policy-class Private
  allow list self self
nat source list OutAXXIPRCMain address 208.13.17.2 overload
nat source list PrivateHosts address 208.13.17.33 overload
!
```

16.9 The following commands define the policies for the interface to the public (untrusted) network. In the Public policy-class, you are only translating (and therefore allowing) the ports required for ITP (endpoint) as defined in the access list `InAXXIPRCMain`.

```
ip policy-class Public
  nat destination list InAXXIPRCMain address 192.168.1.2
!
```

B. NON-NATED DMZ CONFIGURATION

16.10 The following example illustrates the configuration for ITP endpoints and Inter-Tel Private Networking (this is the same configuration used in the Cisco example on page 27).



16.11 The following commands define the VLANs within the switch. This is necessary to be able to route/firewall between the LAN and the DMZ.

```
vlan 1
  name "Default"
vlan 2
  name "DMZ"
!
```

16.12 The following commands set up the Ethernet switch ports for specific VLANs. In this example, take the first two ports and associate them with the DMZ VLAN.

```
interface eth 0/1
  no shutdown
  switchport access vlan 2
!
interface eth 0/2
  no shutdown
  switchport access vlan 2
!
```

Set up some of the ports in the default VLAN. This example shows two ports.

```
interface eth 0/3
  no shutdown
  switchport access vlan 1
!
interface eth 0/4
  no shutdown
  switchport access vlan 1
!
! and so on...
```

16.13 Each VLAN performs as a port on the router with its own IP address. Access policies are applied to each interface. The access policies are defined further below.

```
interface vlan 1
  ip address  192.168.1.1  255.255.255.0
  access-policy Private
  no shutdown
interface vlan 2
  ip address  208.132.23.64  255.255.255.192
  no shutdown
!
```

- In this example, the connection is to the Internet with a T1. Set up the IP address and apply the access policy (defined further below).

```
interface t1 1/1
  clock source line
  tdm-group 1 timeslots 1-24 speed 64
  ip address  208.13.17.33  255.255.255.252
  access-policy Public
  no shutdown
```

- The following commands define access lists for the different traffic types. Defining the lists does not have any effect until they are applied to the interfaces.

```
ip access-list extended web
  permit ip any  any

ip access-list extended Voice
  permit udp any  host 208.132.23.66 range 5004 5069 log

ip access-list extended ITP
  permit tcp any  host 208.132.23.66 eq 5566  log
  permit udp any  host 208.132.23.66 eq 5567  log

ip access-list extended InterTelNetworking
  permit tcp any  host 208.132.23.66 eq 5570  log
```

16.14 The following commands define the policies for the different interfaces. Each policy can specify more than one access list. In the Public policy-class, allow the voice ports, the ITP (endpoint) ports, and the Inter-Tel Networking ports.

```
ip policy-class DMZ
  allow list web
!
ip policy-class Private
  allow list self self
  allow list wizard-ics
!
ip policy-class Public
  allow list Voice
  allow list ITP
  allow list InterTelNetworking
```

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- **MD5**: Encryption code provided with the Inter-Tel Axxess Converged Communications Platform (see page G-14 for details).

1.3 To request open source code modified by Inter-Tel, Inc., see "Requests for Source Code" on page G-14.

B. ABBREVIATIONS

1.4 This appendix refers to the selected abbreviations listed in the following table.

Table G-1. *Selected Abbreviations*

Abbreviation	Meaning
GNU	GNU's Not Unix
GPL	General Public License
LGPL	Lesser GPL

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Trunk Access Codes

FEATURE NAME	CODE (U.S./EUROPE)
Automatic Route Selection (ARS)	92000
Trunk Group Access 1-208	92001-92208
Emergency Calls	911/999
Outgoing Call	8

Station Feature Codes

FEATURE NAME	CODE
Account Code – All Calls Following	391
Account Code – Optional	390
ACD Agent Login ACD Agent Logout ACD Agent Login/Logout Toggle	326 327 328
ACD Agent Wrap-Up Terminate	329
Agent Help	375
Agent Help Reject	376
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Automatic CO Access On/Off	360
Automatic IC Access On/Off	361
Automatic Trunk Answer	350
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Barge-In	386
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Call Forward If Busy	357
Call Forward If No Answer	356
Call Forward If No Answer/Busy	358
Change Language	301
Call Logging	333
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Conference	5
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Directory	307
Display Outside Party Name On/Off	379
Display Time/Date (ITP) Show IP Address (SIP)	300
Do-Not-Disturb Do-Not-Disturb Cancel Do-Not-Disturb On/Off	370 371 372

Station Feature Codes (Continued)

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Enhanced Speakerphone Enable	310
Feature Key Default	395
Group Listen	312
Handsfree On/Off	319
Headset Enable Headset Disable Headset On/Off	315 316 317
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Hold – System	335
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System Speed Dial	381
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Administrator Feature Codes

FEATURE NAME	CODE (U.S./ EUROPE)
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Clear Network Alarm	9851
Clear System Alarm	9850
Compression On/Off	9982/9182
Compression Statistics	9981/9181
Diagnostics On/Off	9900/9100
Enable Network Day	9862
Enable Network Night	9861
Modem Disable	9867
Modem Enable	9866
Modem Reset	9869
Night Ring On/Off	9860
Periodic Diagnostics On/Off	9825
Program Database	9932/9132
Program System Speed Dial	9801
Set Network Date/Time	9810
Set Time/Date	9800
Synchronize Network Time	9811

The Diagnostics Mode feature code (9900 or 9100) must be entered before the following feature codes can be used.

Diagnostics Mode Feature Codes

FEATURE NAME	CODE (U.S./ EUROPE)
Diagnostic - ASAI Snoop Off	9926/9126
Diagnostic - ASAI Snoop On	9927/9127
Diagnostic - Dump Extension	9922/9122
Diagnostic - Dump Node Information	9936/9136
Diagnostic - Dump System Diagnostics	9933/9133
Diagnostic - Heap Dump	9943/9143
Diagnostic - Heap Statistics	9947/9147
Diagnostic - ISDN View	9948/9148
Diagnostic - Major Reset	9962/9162
Diagnostic - Mark As Leaks	9945/9145
Diagnostic - Mark As Quiescent	9946/9146
Diagnostic - Minor Reset	9964/9164
Diagnostic - Network Freeze Zone System Histories Diagnostic - Network Unfreeze Zone System Histories	9939/9139 9989/9189
Diagnostic - Network Groups	9963/9163
Diagnostic - Print Auxdata	9972/9172
Diagnostic - Print Message Log	9975/9175
Diagnostic - Print Network Log	9976/9176
Diagnostic - Query Node Traffic	9978/9178
Diagnostic - Show Version	9928/9128
Diagnostic - SIP View	9987/9187
Diagnostic - Spare 1-3	9910-9912/ 9110-9112
Diagnostic - System History	9974/9174
Diagnostic - View Displays	9983/9183
Seize Device	9973/9173
System History—Freeze System History—Unfreeze	9993/9193 9998/9198

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