communication

MERLIN LEGEND®
Communications System

Release 3.0
Equipment and Operations
Reference

555-630-115
August 1994

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Notice

Every effort was made to ensure that the information in this book was complete and accurate at the time of printing. However, information is subject to change.

See Appendix A, "Customer Support Information," for important information.

Security of Your System: Preventing Toll Fraud

As a customer of a new telephone system, you should be aware that there exists an increasing problem of telephone toll fraud. Telephone toll fraud can occur in many forms, despite the numerous efforts of telephone companies and telephone equipment manufacturers to control it. For important information regarding your system and toll fraud, see Appendix A, "Customer Support Information."

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This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at his own expense. For further FCC information, see Appendix A, "Customer Support Information."

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This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the radio interference regulations of the Canadian Department of Communications.

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The ordering number for this document is 555-630-115. To order this document, call the AT&T Customer Information Center at 1-800-432-6600 (in Canada, 1-800-255-1242). For more information about AT&T documents, refer to the section entitled, "Related Documents" in "About This Book." The *Pocket Reference*, listed in that section, provides full ordering information for replacement parts, accessories, and other compatible equipment; or, contact your AT&T representative.

Support Telephone Number

In the continental U.S., AT&T provides a toll-free customer helpline 24 hours a day. Call the AT&T Helpline at 1-800-628-2888 if you need assistance when installing or using your system.

Outside the continental U.S., contact your local AT&T representative.

Warranty

AT&T provides a limited warranty on this product. Refer to "Limited Warranty and Limitation of Liability" in Appendix A, "Customer Support Information."

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The exclamation point in an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the product.

IMPORTANT SAFETY INSTRUCTIONS

When installing telephone equipment, always follow basic safety precautions to reduce the risk of fire, electrical shock, and injury to persons, including:

- Read and understand all instructions.
- Follow all warnings and instructions marked on or packed with the product.
- Never install telephone wiring during a lightning storm.
- Never install a telephone jack in a wet location unless the jack is specifically designed for wet locations.
- Never touch uninsulated telephone wires or terminals unless the telephone wiring has been disconnected at the network interface.
- Use caution when installing or modifying telephone lines.
- Use only AT&T-manufactured MERLIN LEGEND Communications System circuit modules, carrier assemblies, and power units in the MERLIN LEGEND Communications System control unit.
- Use only AT&T-recommended/approved MERLIN LEGEND Communications System accessories.
- If equipment connected to the analog extension modules (008, 408, 408 GS/LS) or to the MLX telephone modules (008 MLX, 408 GS/LS-MLX) is to be used for in-range out-of-building (IROB) applications, IROB protectors are required.
- Do not install this product near water, for example, in a wet basement location.
- Do not overload wall outlets, as this can result in the risk of fire or electrical shock.
- The MERLIN LEGEND Communications System is equipped with a 3-wire grounding-type plug with a third (grounding) pin. This plug will fit only into a grounding-type power outlet. This is a safety feature. If you are unable to insert the plug into the outlet, contact an electrician to replace the obsolete outlet. Do not defeat the safety purpose of the grounding plug.
- The MERLIN LEGEND Communications System requires a supplementary ground.

- Do not attach the power supply cord to building surfaces. Do not allow anything to rest on the power cord. Do not locate this product where the cord will be abused by persons walking on it.
- Slots and openings in the module housings are provided for ventilation. To protect this equipment from overheating, do not block these openings.
- Never push objects of any kind into this product through module openings or expansion slots, as they may touch dangerous voltage points or short out parts, which could result in a risk of fire or electrical shock. Never spill liquid of any kind on this product.
- Unplug the product from the wall outlet before cleaning. Use a damp cloth for cleaning. Do not use cleaners or aerosol cleaners.
- Auxiliary equipment includes answering machines, alerts, modems, and fax machines. To connect one of these devices, you must first have a Multi-Function Module (MFM).
- Do not operate telephones if chemical gas leakage is suspected in the area. Use telephones located in some other safe area to report the trouble.

WARNING:

- *For your personal safety, DO NOT install an MFM yourself.*
- *ONLY an authorized technician or dealer representative shall install, set options, or repair an MFM.*
- *To eliminate the risk of personal injury due to electrical shock, DO NOT attempt to install or remove an MFM from your MLX telephone. Opening or removing the module cover of your telephone may expose you to dangerous voltages.*

SAVE THESE INSTRUCTIONS

About This Book

The MERLIN LEGEND Communications System is an advanced digital switching system that integrates voice and data communications features. Voice features include traditional telephone features, such as Transfer and Hold, and advanced features, such as Group Coverage and Park. Data features allow both voice and data to be transmitted over the same system wiring.

Intended Audience

This book provides detailed information about system and telephone hardware. It also provides overview information for software applications which may be used with Release 3.0. It is intended for use as a reference by anyone needing such information, including support personnel, sales representatives, and account executives. It is also intended for technicians who are responsible for system installation, maintenance and troubleshooting.

How to Use This Book

This document describes system components and capabilities, modes of operation, lines and trunks, applications, and data communications support.

Refer to the following documents for additional information:

- *Feature Reference* provides detailed information about system features and telephone features.
- *System Planning* is used for setting up the system configuration.
- *System Programming* gives procedural instructions for programming system features.
- User's Guides and Operator's Guides give procedural instructions for programming and using telephone features.

"Related Documents," later in this section, provides a complete list of system documentation together with ordering information.

In the U.S.A. only, AT&T provides a toll-free customer Helpline (1-800-628-2888) 24 hours a day. Call the Helpline, or your authorized dealer, if you need assistance when installing, programming, or using your system.

Terms and Conventions Used

The terms described here are used in preference to other, equally acceptable terms for describing communications systems.

Lines, Trunks and Facilities
Facility is a general term that designates a communications path between a telephone system and the telephone company central office. Technically a trunk connects a switch to a switch, for example the MERLIN LEGEND Communications System to the central office. Technically, a line is a loop-start facility or a communications path that does not connect two switches, for example, an intercom line or a Centrex line. However, in actual usage, the terms line and trunk are often applied interchangeably. In this book, we use line/trunk and lines/trunks to refer to facilities in general. Specifically, we refer to digital facilities. We also use terms such as personal line, ground-start trunk, DID trunk, and so on. When you talk to your local telephone company central office, ask them what terms they use for the specific facilities they connect to your system.

Some older terms have been replaced with newer terms. The following list shows the old term on the left and the new term on the right.

trunk module	line/trunk module
trunk jack	line/trunk jack
station	extension
station jack	extension jack
analog data station	modem data station
digital data station	7500B data station
analog voice and analog data station	analog voice and modem data
digital voice and analog data station	MLX voice and modem data
analog data only station	modem data only station
digital data only station	7500B data only station
digital voice and digital data station	MLX voice and 7500B data station

Typographical Conventions

Certain type fonts and styles act as visual cues to help you rapidly understand the information presented:

Example	Purpose
It is *very* important that you follow these steps. You *must* attach the wristband before touching the connection.	Italics indicate emphasis.

Example	Purpose
If you press the **Feature** button on an MLX display telephone, the display lists telephone features you can select. A programmed Auto Dial button gives you instant access to an inside or outside number.	The names of fixed-feature, factory-imprinted buttons appear in bold. The names of programmed buttons are printed as regular text.
Choose `Ext Prog` from the display screen.	Plain constant-width type indicates text that appears on the telephone display or PC screen.
To activate Call Waiting, dial ***11**.	Constant-width type in italics indicates characters you dial at the telephone or type at the PC.

Product Safety Labels

Throughout these documents, hazardous situations are indicated by an exclamation point inside a triangle and the word *caution* or *warning*.

WARNING:
Warning indicates the presence of a hazard that could cause death or severe personal injury if the hazard is not avoided.

CAUTION:
Caution indicates the presence of a hazard that could cause minor personal injury or property damage if the hazard is not avoided.

Security

Certain features of the system can be protected by passwords to prevent unauthorized users from abusing the system. You should assign passwords wherever you can and limit knowledge of such passwords to three or fewer people.

Nondisplaying authorization codes and telephone numbers provide another layer of security. For more information, see Appendix A, "Customer Support Information."

Related Documents

In addition to this book, the documents listed below are part of the documentation set. Within the continental United States, these documents can be ordered from the AT&T Customer Information Center by calling 1-800-432-6600.

Document No.	Title
	System Documents
555-630-117	*Introduction*
555-630-118	*System Manager's Guide*
555-630-110	*Feature Reference*
555-630-115	*Equipment and Operations Reference*
555-630-116	*Pocket Reference*
555-630-111	*System Programming*
555-630-112	*System Planning*
555-630-113	*System Planning Forms*
	Telephone User Support
555-630-122	*MLX-10D™, MLX-10DP™, MLX-28D™, and MLX-20L™ Display Telephones User's Guide*
555-630-150	*MLX-10D Display Telephone Tray Cards (5 cards)*
555-630-153	*MLX-28D and MLX-20L Telephone Tray Cards (5 cards)*
555-630-124	*MLX-10™ Nondisplay Telephone User's Guide*
555-630-151	*MLX-10 Nondisplay Telephone Tray Cards (6 cards)*
555-630-120	*Analog Multiline Telephones User's Guide*
555-630-126	*Single-Line Telephones User's Guide*
555-630-138	*MDC 9000 and MDW 9000 Telephones User's Guide*
	System Operator Support
555-630-134	*MLX Direct-Line Consoles Operator's Guide*
555-630-132	*Analog Direct-Line Consoles Operator's Guide*
555-630-136	*MLX Queued Call Console Operator's Guide*
	Miscellaneous User Support
555-630-130	*Calling Group Supervisor's Guide*
555-630-129	*Data User's Guide*
	Documentation for Qualified Technicians
555-630-140	*Installation, Programming, & Maintenance (IP&M) Binder*

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Introduction

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Introduction

1

The MERLIN LEGEND Communications System is an advanced digital switching system that integrates voice and data communication features. Voice features combine traditional telephone features, such as Transfer and Hold, with advanced telephone features, such as Group Coverage and Park. Data features enable the transmission of voice and data over the same system wiring.

This chapter describes the following aspects of the system:

- **Components.** This section lists the required and optional equipment that makes up the system.
- **Functional Description.** This section discusses the functional units that make up the control unit and describes how they work together. It also discusses signaling.
- **Modes of Operation.** This section describes the three modes for which you can configure the system: Key, Hybrid/PBX, and Behind Switch.
- **Programming.** This section provides general information about programming the system and telephones.
- **System Capacities and Requirements.** This section describes system capacities for hardware and software and technical requirements for environment, power, and grounding.
- **Release Differences.** This section explains the system enhancements offered in Releases 1.1, 2.0, 2.1, and 3.0.

Components

The system consists of basic components (control unit and telephones) and auxiliary components (adjuncts, adapters, and applications).

Control Unit

The control unit consists of the *basic carrier* and up to two *expansion carriers*. The basic carrier contains the *processor module*, *power supply module*, and *line/trunk and extension modules*. Each expansion carrier contains a power supply module and line/trunk and extension modules.

IMPORTANT:
Facility is a general term that designates a communications path between a telephone system and the telephone company central office. Although the terms *line* and *trunk* technically refer to different *facilities*, the difference is not as clear as it once was. Technically, a *trunk* connects a switch to a switch, for example, the MERLIN LEGEND Communications System to the central office. Technically, a *line* is a loop-start facility or a communications path that does not connect switches, for example, an *intercom line* or a *Centrex line*. However, in actual usage, the terms *line* and *trunk* are often applied interchangeably. This book, and other documentation for this communications system, uses the term *lines/trunks* and *line/trunk* to refer to facilitiers in general. Specifically, we refer to digital *facilities*. We also use terms such as *DID trunks*, *tie trunks*, and *personal lines* when referring to these facilities. When you talk to your local telephone company central office, ask them what terms they use for the specific facilities they connect to your system.

Telephones

The system supports MLX (digital) telephones, analog multiline telephones, and single-line telephones. See the lists of supported models below.

MLX Telephones

You can use the following MLX telephones with the system:

- MLX-10D (10 buttons with display)
- MLX-10DP (same as the MLX-10D, except that it has an interface in the back for connection to the PassageWay™ Direct Connect Solution software application.)
- MLX-10 (10 buttons, no display)
- MLX-20L (20 buttons with display, interface for PassageWay Direct Connect Solution)
- MLX-28D (28 buttons with display, interface for PassageWay Direct Connect Solution)

Analog Multiline Telephones

You can use the following analog multiline telephones with the system:

- BIS-10 (10 buttons, built-in speakerphone)
- BIS-22 (22 buttons, built-in speakerphone)
- BIS-22D (22 buttons, built-in speakerphone, 16-character display)
- BIS-34[1] (34 buttons, built-in speakerphone)
- BIS-34D (34 buttons, built-in speakerphone, 16-character display)
- MDC 9000 (6-line, cordless)
- MDW 9000 (6-line, cordless, wireless)
- MLC-5 Cordless (5-button multiline, cordless)
- MERLIN® II System Display Console
- MERLIN PFC™ (Phone Fax Copier) telephone (requires two analog extension jacks)
- 5-button[1] (5 buttons, membrane, no adjuncts supported)
- 10-button[1] (10 buttons, membrane)
- 34-button[1] (34 buttons, membrane)
- 34-button Deluxe[1] (34 buttons, membrane)
- 10-button HFAI[1] (10 buttons, hands-free answer, no adjuncts supported)
- 34-button BIS[1] (34 buttons, built-in speakerphone)
- 34-button BIS/DIS[1] (34 buttons, built-in speakerphone, 16-character display)

Single-Line Telephones

These single-line telephones are specifically designed for Release 3.0 of the system:

- 8101 (desk or wall-mount telephone, data/fax jack, selectable positive disconnect)
- 2500 YMGL (desk telephone, selectable positive disconnect)
- 2500 MMGL (basic desk telephone, selectable positive disconnect). When the positive disconnect is on, system features cannot be used. When the positive disconnect is off, the switchhook is used to access system features using feature codes.

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1 Vintage telephone, no longer available for sale or lease.

The following single-line telephones were designed for earlier releases of the system:

- 2500MMGB (desk telephone)
- 2554MMGJ (wall telephone)
- 2500YMGK[1] (desk telephone, Message light, **Recall** button)
- 2500SM (desk telephone used with 4A speakerphone)
- 2514BMW (desk telephone with built-in headset jack)
- 2526BMG (outdoor telephone used with waterproof enclosure)
- 5200[1] cordless telephone
- 5320[1] cordless telephone
- 5500[1] cordless telephone
- 7101A[1] (desk telephone, Message light, **Recall** and **Disconnect** buttons, no adjuncts supported)
- 7102A (desk telephone, Message light, **Recall** button, supports 101 and 201 speakerphones and 500 headsets)
- 710 memory telephone (auto dial buttons)
- 715 memory telephone (auto dial buttons)
- 725 memory telephone (auto dial buttons, speakerphone)
- 730 memory telephone (auto dial buttons, speakerphone, display)
- CS6402U01A[1] (desk telephone, built-in speakerphone, memory, redial)
- 2500MMGJ (desk telephone)
- 2500MMGK (desk telephone, timed **Recall** button action activates Hold and Transfer)
- 8102 (desk telephone, data jack for connecting a modem, slot for headset adapter, and jack for speakerphone adjuncts)
- 8110 (desk telephone, built-in speakerphone with volume control, auxiliary power jack for improving quality of built-in speakerphone, **Mute** button with LED indicator, and data jack for connecting a modem)
- VideoPhone 2500 (for small-screen video)
- Picasso™ Still-Image Phone (for interactive display of still images)
- 500MM, 554BMPA, 500SM (rotary dial)

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1 Vintage telephone, no longer available for sale or lease.

- 3129-WTWA (touch-tone outdoor telephone equipped with cast aluminum housing and armored handset cord with bell ringers)
- 3129-WRWA (rotary-dial outdoor telephone equipped with cast aluminum housing and armored handset cord with bell ringers)
- 3129-WAWA (auto dial outdoor telephone equipped with cast aluminum housing and armored handset cord with bell ringers)
- 3129-WNWA (nondial, automatic ringing on dedicated circuit outdoor telephone equipped with cast aluminum housing and armored handset cord with bell ringers)

Adjuncts

Adjuncts are optional pieces of equipment that connect directly to the control unit or to a telephone through an adapter (see "Adapters," below). Answering machines, credit card verification terminals, modems, speakerphones, Station Message Detail Recording (SMDR) printers, fax machines, and alerts are all adjuncts.

Adapters

Adapters let you connect adjuncts or, in the case of a Channel Service Unit (CSU), Digital Signal 1 facilities to the control unit. There are two types of adapters: *system adapters* connect directly to the control unit, and *telephone adapters* connect to telephones.

The system supports the following adapters:

- System Adapters
  - Loop-Start Trunk Adapter (for loudspeaker paging)
  - Universal Paging Access Module (UPAM, for loudspeaker paging)
  - PagePal (for loudspeaker paging)
  - ACCULINK™ 3150 CSU and 3160/3164 DSU (data service unit)/CSU
  - Extended Super Frame (ESF) T1 CSU (no longer available)
  - 551 T1 LI CSU (no longer available)
- Telephone Adapters
  - Multi-Function Module (MFM) for MLX telephones, for attaching a tip/ring (T/R) device to an MLX telephone

WARNING:
The Multi-Function Module (MFM) must be installed or repaired only by a qualified technician. To avoid the risk of electrical shock, do not disassemble the MLX telephone.

— General Purpose Adapter (GPA) for analog multiline telephones, for connecting T/R devices

— Supplemental Alert Adapter (SAA) for connecting external alerts (such as a horn or strobe) to analog multiline telephones

— ISDN (Integrated Services Digital Network) 7500B Data Module for connecting digital data equipment

Applications

The following applications consist of software and/or hardware and add optional capabilities to the system. See Chapter 4, "Applications," for details.

- PassageWay Direct Connect Solution (for Release 3.0 and later) and Passage Way Solution (for release 2.1)
- MERLIN MAIL™ Voice Messaging System (VMS)
- AT&T Attendant
- MERLIN Identifier
- Call Accounting System (CAS) Plus V3 for DOS
- Call Accounting System (CAS) for Windows™
- Call Accounting Terminal (CAT)
- Call Management System (CMS)
- System Programming and Maintenance (SPM) for DOS
- Integrated Solution II (IS II) for UNIX System
  - — Integrated Voice Power Automated Attendant (IVP AA IS II)
  - — AUDIX Voice Power™ IS II (AVP IS II)
  - — Call Accounting System IS II (CAS IS II)
  - — System Programming and Maintenance IS II (SPM IS II)
- Integrated Solution III (IS III) for UNIX System
  - — Integrated Voice Power Automated Attendant (IVP AA IS III)
  - — AUDIX Voice Power IS III (AVP IS III)
  - — Call Accounting System IS III (IS CAS)
  - — System Programming and Maintenance IS III (SPM IS III)

  - Fax Attendant™ System
- Primary Rate Interface (PRI) Applications
  - Group IV (G4) fax
  - Video conferencing
- MERLIN PFC
- Automated Document Delivery System (ADDS)
- CONVERSANT®

Functional Description

This section describes the functional units that make up the control unit. It also discusses how the system processes signals.

The control unit contains the following functional units:

- **Processor Module.** This module controls the operation of the system and its features.
- **Power Supply Module.** This module supplies power to the control unit. Release 2.1 (and later) provides ferrite cores around the AC power cord and ground wire to comply with FCC Part 15 requirements. The Release 2.1 power supply module can serve Release 3.0 and earlier.
- **Carrier and Backplane.** This assembly contains the input/output (I/0) bus and the time-division multiplex (TDM) bus. A *bus* is a physical communication link between the microprocessor (on the processor module) and the other system components plugged into the control unit.
- **Line/Trunk and Extension Modules.** These modules connect outside telephone company facilities and inside extensions to the control unit.

All modules connect to the backplane, which provides common circuitry for the I/O bus, the time-division multiplex (TDM) bus, and power distribution. The processor module connects through the I/O bus to intelligent ports on the line/trunk and extension modules, using the digital switch element (DSE) on each line/trunk and extension module. The TDM bus also connects to the digital switch element (DSE) on each line/trunk and extension module.

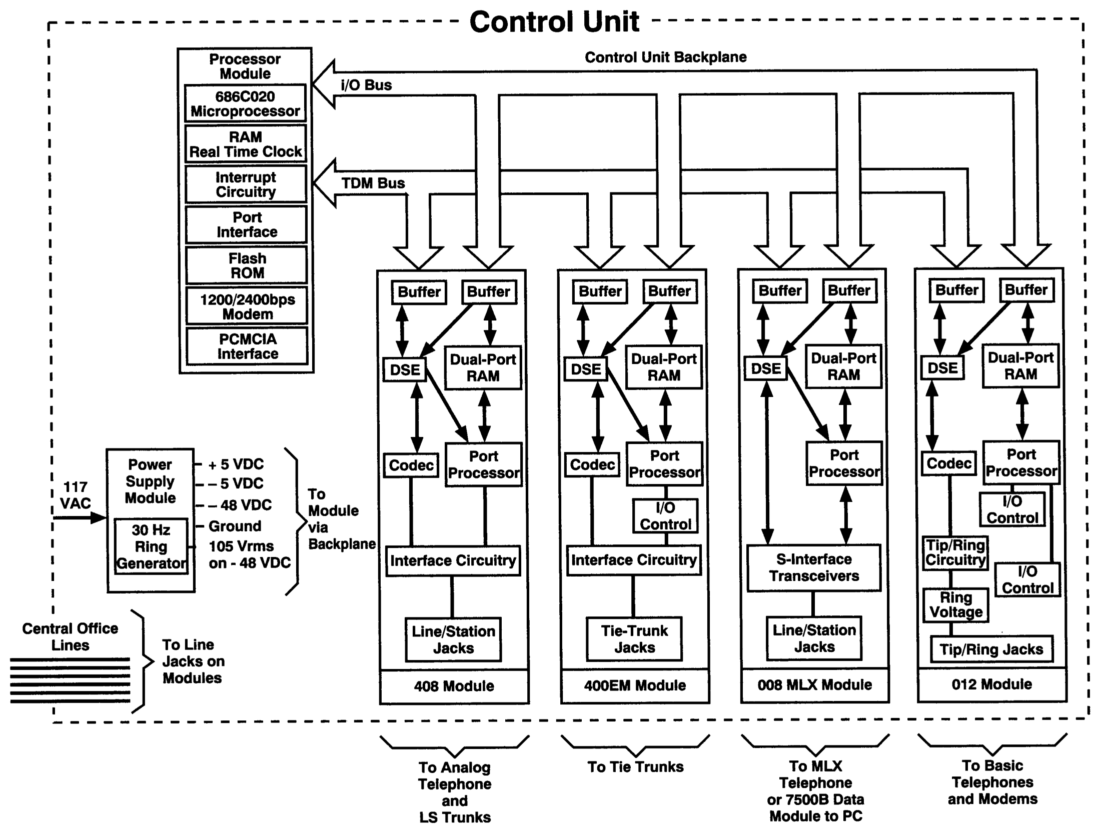

Figure 1–1. Functional Units

The following two buses, shown in Figure 1–1, are on the backplane:

- **Input/Output (I/O).** The I/O bus contains an address bus and a data bus. The address bus selects the module that receives instructions from the 32-bit 68EC020 (Release 3.0 and later) microprocessor in the processor module. The microprocessor provides instructions to the port processors and DSEs through the data bus. The microprocessor operates at 16 MHz.
- **Time-Division Multiplex (TDM).** The TDM bus connects the DSEs to allow voice or data to flow in and out of the system. The TDM bus is parallel, 8 bits wide, and runs at 2.048 MHz (256 time slots x 8 kHz = 2.048 MHz). Each TDM cycle has 256 time slots for voice, data, and tones. The frame repetition rate is 8 kHz, providing a 64-kbps channel on each time slot (8-bit bus x 8 kHz = 64 kbps).

The built-in modem connects to the TDM bus; this permits access from a local or remote PC or workstation equipped with a 1200/2400-bps modem. The TDM bus connects with the built-in diagnostics that enable the processor to read and write to dedicated TDM test slots.

The TDM bus carries analog signals encoded in Mu-Law 255 pulse code modulation (PCM) format for domestic use. The system provides a circuit-switched connection for transmission of digital data signals at up to 64 kbps.

Digital Switching

Because the system is internally digital in a world of both analog and digital devices, it must accurately translate analog signals. The translation involves signal conversion and switching. *Codecs* provide analog-to-digital and digital-to-analog conversion. These digitally encoded signals are routed from one interface port to another interface port by assigning source and destination information to specific time slots on the TDM bus. In this way, the system can transmit signals to one or several destinations and reconstruct them at the original amplitude. This results in no signal loss during switching and transmission from one extension to another.

The TDM bus allows many users to communicate over a common electrical connection, because it is physically distributed across the backplane of the control unit and connects all line/trunk and extension modules.

The system uses digital switch elements (DSEs) to interface codecs or digital transceivers to the TDM bus. The DSE specifies time slots for various functions. For example, during a conversation between Extension A and Extension B, the system reserves a time slot for Extension A to transmit on and for Extension B to receive from. The actual digital switching occurs when the system's I/0 bus programs the DSE to transmit data on or receive data from the TDM bus in specific time slots. In addition, the DSE can sum digital signals from designated TDM slots to provide conferencing for up to five parties.

Using time slots 0 to 39, a digital tone plant in the processor module provides touch-tone and call progress signals to extensions. Unlike other bus configurations, the DSEs on the TDM bus receive all transmissions. If a DSE is not assigned to any of the time slots, it ignores the data.

Modes of Operation

You can program the system to operate in Key, Hybrid/PBX, or Behind Switch mode. The mode of operation determines the following:

- The types of outside trunks that you can connect to the system
- How telephone users access outside trunks

- The types of system operator consoles allowed
- The features available and how they work

Table 1–1 summarizes each mode of operation, and the following sections describe each mode in more detail.

Table 1–1. Modes of Operation

	Key	Hybrid /PBX	Behind Switch
Trunks connected directly to the control unit			
KF registration (FCC)	Yes	No	Yes
MF registration (FCC)	Yes	Yes	Yes
PF registration (FCC)	No	Yes	No
Ground-start	Yes	Yes	Yes
Loop-start	Yes	Yes	Yes
PRI	Yes	Yes	No
DS1	Yes	Yes	No
Tie	Yes	Yes	Yes
FX	Yes	Yes	Yes
WATS	Yes	Yes	Yes
DID	No	Yes	No
Trunk pools	No	Yes	No
ARS	No	Yes	No
ICOM* buttons	Yes	No	Yes
SA* buttons	No	Yes	No
Line buttons (outside trunks assigned to telephone buttons)	Yes	Yes	Yes
Shared trunks	Yes†	Yes‡	Yes†
Prime lines	No	No	Yes
Queued Call Console (QCC)	No	Yes	No
Number of extensions			
Less than 50	Good	Good	Good
Greater than 50	Not recommended	Good	Good up to 80

* **ICOM** stands for *Intercom*, a type of line button used in Key and Behind Switch modes to make or receive inside calls. **SA** stands for *System Access*, a type of line button used in Hybrid/PBX mode to make or receive inside or outside calls.

† Outside trunks only

‡ Outside trunks and **SA** buttons

Key Mode

A system operating in Key mode is a Key system. Key mode is the simplest way to provide people with more than one line from a single telephone. Older Key systems have telephones that look like single-line telephones except for a row of buttons, illuminated by incandescent lights when active, across the bottom. The leftmost button is labeled **Hold**, and the other buttons are labeled with telephone numbers.

When the system operates in Key mode, line buttons can work in three different ways:

- **Line Buttons** (or Keys). These buttons are associated with specific *outside lines* for making or receiving calls to telephone numbers other than system extensions ("outside" the system). Line buttons allow you to see activity on other telephones, join conversations, and make and receive outside calls.
- **Intercom Buttons.** These buttons allow you to make and receive *inside calls* to or from system extension numbers ("inside" the system).
- **Programmed Buttons.** These buttons are programmed to activate features that you need frequently.

Key mode accommodates the following kinds of outside lines/trunks:

- Loop-start and emulated loop-start on T1 facilities, including basic, Wide Area Telecommunications Service (WATS), and Foreign Exchange (FX)
- Ground-start (only if registered as MF and if the switch is not set for Permanent Key mode, as described below) and emulated ground-start on T1 facilities
- Digital Signal 1 (DS1) facilities, T1 and Primary Rate Interface (PRI)
- Tie and emulated tie on T1 facilities

A standard Key system's trunks are all *loop-start.* A loop-start trunk introduces a slight delay between the time the telephone company's central office (CO) recognizes a call attempt and the time the system processes the call. This delay is minimal and virtually unnoticeable. Most residence and small business telephones have loop-start trunks.

You configure the system for Key mode through system programming, but you should take into account how the system is registered with the Federal Communications Commission (FCC), as described later in this chapter. If the system is switched or strapped for Permanent Key operation in the processor module, you cannot connect ground-start trunks to it. However, the KF classification allows ground-start emulation on a T1 facility.

NOTES:

1. The default programmed mode is Hybrid/PBX.
2. On initialization of a Release 1.0 system, programming for all loop-start and ground-start trunks reverts to loop-start. In Releases 1.1 and later, if the system is programmed for Key mode, the switch or strap is checked on initialization. If the switch or strap is set for Permanent Key operation, all trunks revert to loop-start. If the switch or strap is not set, any programmed designation of ground-start trunks is retained.
3. Beginning with Release 3.0, Caller ID is available to identify incoming calls on the displays of MLX display telephones, as long as the calls are from supported telephone company jurisdictions. This service is available on loop-start trunks in systems equipped with 800 GS/LS-ID modules, but the local telephone company must supply the service.

The following features are not available in Key mode:

- Direct Inward Dialing (DID) trunks
- Trunk pools (more than one line available from a single button)
- Automatic Route Selection (ARS)
- Queued Call Console (QCC) operation
- **SA** buttons

Line Access

In Key mode (whether Permanent by using the switch or strap or programmed only), you must assign each outside line to a line button on at least one telephone. As a result, the telephones most commonly used in Key mode are multiline telephones. Figure 1–2 shows how lines are assigned in Key mode.

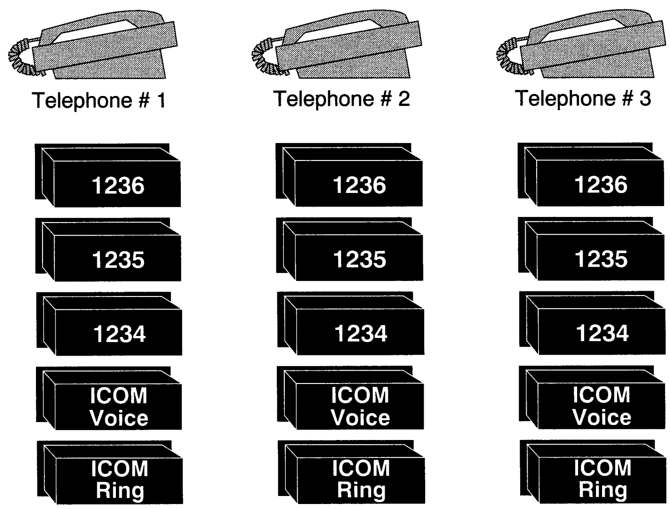

Figure 1–2. Lines Labeled for Key System Telephones

You select an outside line by pressing a *line button*—a button labeled with a telephone number. When you hear a dial tone, you can dial out. When a line is in use, the green light-emitting diode (LED) next to the corresponding line button lights on any telephones that share that line.

The telephones in a Key system also have Intercom buttons, labeled **ICOM**, that allow you to make and receive calls to and from other extensions within the system. When you press **ICOM**, the system provides an inside talk path and you hear a system dial tone, which you can program, if necessary for equipment or applications, to sound the same as an outside dial tone. The factory setting is to provide a different inside dial tone.

You can use the following types of **ICOM** buttons to make and receive inside calls in Key mode:

- **ICOM Ring.** Use this button to make inside calls and to receive inside and outside calls transferred from another extension. When you use an **ICOM Ring** button to make an inside call, the telephone at the destination extension rings once per ring cycle to indicate an inside call.

- **ICOM Voice.** Use this button to make inside calls and to receive inside and outside calls transferred from another extension. When you use **ICOM Voice** to make an inside call, the user at the destination extension hears the caller's voice on the speakerphone after a beep, rather than after ringing. (If the user has a single-line telephone or a telephone that does not have a speakerphone, or has disabled voice announcements, the telephone rings just as if the call was made on an **ICOM Ring** button.)
- **ICOM Originate Only.** Use this button to make inside calls. You cannot receive inside or outside calls on **ICOM Originate Only** buttons. This type of button ensures that you always have a button available to make or transfer a call, establish a conference call, answer a call-waiting call, or pick up parked calls. You can program this button for either voice or ring operation.

You can assign any combination of up to 10 **ICOM Voice**, **ICOM Ring**, and **ICOM Originate Only** buttons to each telephone on line buttons 1 through 10. The number of line buttons that you can assign to a telephone is limited only by the number of lines/trunks in the system and the number of buttons available on the telephone. See *System Planning* for button diagrams.

Key System Configurations

You can configure Key systems either as square, modified, or hybrid Key systems, as described in the following sections. (System operation is not affected by these configurations. The initial mode setting does affect system operations.)

In Key mode, the first eight trunks connected to the system are automatically assigned to the same eight buttons on all multiline telephones; all trunks are automatically assigned to each Direct-Line Console (DLC).

Square Key System

In a square Key system, every outside line in the system terminates on a line button on every telephone in the system.

Figure 1–2 shows a square Key system in an office with three outside lines and three telephones. Each of the three lines is assigned to the same button on each telephone. When a line is assigned to a line button on more than one telephone, it is considered a *shared personal line*.

Modified Key System

You can modify a Key system through system programming to provide line access for special business needs. For example, some businesses do not require every user to have access to a tie trunk, so you can program the system so that some telephones do not have access to all outside lines.

Figure 1–3 shows an example of line button assignments in a modified Key system that includes two outside lines and one tie trunk. Not every line/trunk is assigned to a button on every telephone.

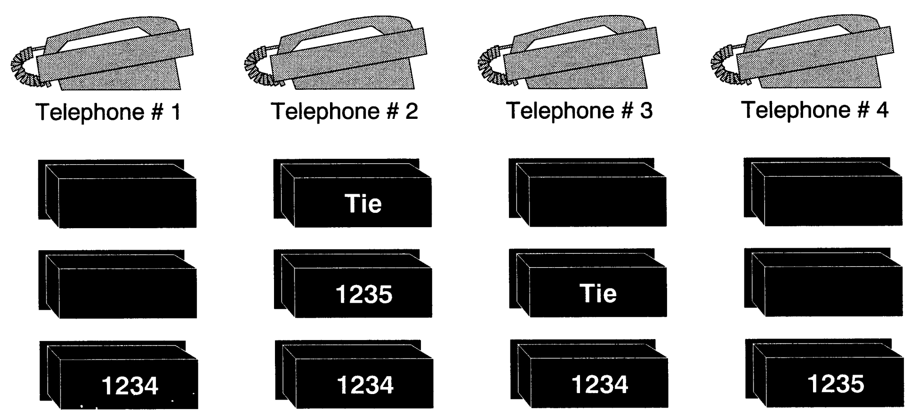

Figure 1–3. Lines Labeled for Modified Key System Telephones

Hybrid Key System

A hybrid Key system allows you to connect ground-start trunks directly to the control unit. If ground-start trunks are used in a Key mode system, the system's switch or strap *cannot* be set for Permanent Key operation and it *must* be registered with the FCC under an MF classification (as described in "FCC Registration" later in this chapter). In this configuration, all outside lines/trunks—including ground-start—are assigned to line buttons on each telephone.

Key Mode Considerations

Review the following operational considerations for Key mode:

- The multiline telephones most commonly used in a Key system provide easy access to outside lines. To get a dial tone, the user simply lifts the handset; the system automatically selects an outside line.
- Key mode has the flexibility to provide line access according to user needs. For example, you can assign tie trunks to the telephones of only those users who need them.
- The loop-start trunks usually associated with Key mode operation often cost less than the trunks used in the other modes.

- Key systems are best for smaller businesses. Key systems are not as easily to expand as other types of systems. As more outside lines are connected, it becomes harder and harder for telephones to have enough buttons for users' access to the lines.
- To take advantage of the features and functionality of the Key system, all users should have multiline telephones.
- To make efficient use of outside trunks by grouping them into pools for shared use, or to use Automatic Route Selection (ARS), you must program the system for Hybrid/PBX operation.
- If there are 20 or more trunks connected to the system, consider using Hybrid/PBX mode and a Queued Call Console (QCC).

Hybrid/PBX Mode

Originally, a private branch exchange (PBX) was a large switchboard installed at a customer's office; it functioned as a small, self-contained telephone company. The system operators physically connected calls by plugging cords into the board's jacks. Today's PBX is a processor in the control unit. The major distinction of Hybrid/PBX mode is that you can make both inside and outside calls on the same button. In Hybrid/PBX mode, this button is called a *System Access* button, and is labeled **SA**.

Although there is no longer a person handling cords, when set up for Hybrid/PBX mode, the system still requires you to request an outside trunk. On an **SA** button, you do this by dialing a dial-out code (usually **9**) and the telephone number; the system routes the call to an available outside trunk.

Hybrid/PBX mode accommodates the following kinds of outside lines/trunks:

- Loop-start (including basic, WATS, and FX) and emulated loop-start on T1 facilities
- Ground-start (including basic, WATS, and FX) and emulated ground-start on T1 facilities
- DS1 facilities (including T1 and PRI)
- Tie and emulated tie on T1 facilities
- DID (Direct Inward Dialing) and emulated DID on T1 facilities

Programming the system for Hybrid/PBX mode automatically arranges outside trunks in functional groups, or *pools*, within the control unit (see Figure 1–4). The system can have up to 11 separate trunk pools. The number of pools programmed depends on both the kinds of trunks and the special needs of the users.

Since the outside trunks are pooled, outside numbers are not associated with individual telephones. When you assign a pool to a line button during system programming, it becomes a ***Pool*** *button.* You request specific trunk pools by dialing the trunk pool number (70, 890–899) for the pool or by pressing a **Pool** button, which gives you one-touch access to a group of trunks. A system operator normally answers outside calls and transfers them to the appropriate system user.

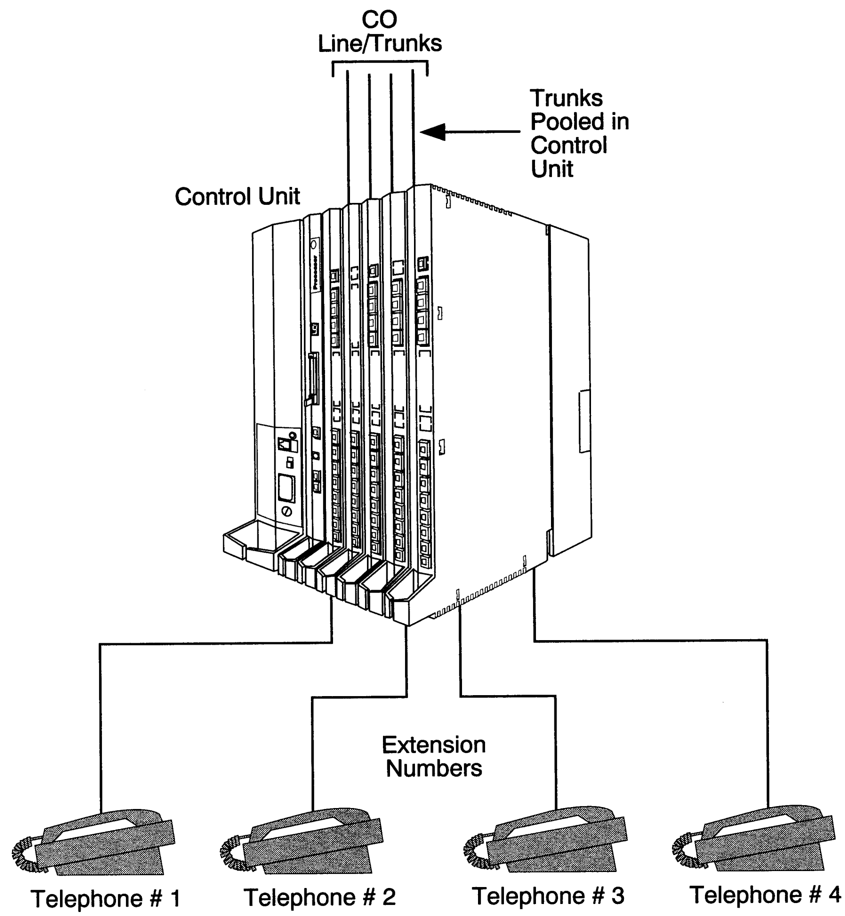

Figure 1–4. Hybrid/PBX Mode

A feature commonly used in Hybrid/PBX mode is Automatic Route Selection (ARS). When you press an **SA** button to make an outside call and dial the ARS dial-out code, the system connects you to the next available trunk from the type of pool that is most cost-effective for the call.

The system provides three types of trunk pools and automatically assigns trunks to the appropriate pool type:

- **Loop-Start Pool** (or main pool). By default, the system assigns loop-start trunks to pool number 70, the loop-start pool.
- **Ground-Start Pool.** By default, the system assigns ground-start trunks to pool number 890, the ground-start pool.
- **Tie Pool.** By default, the system assigns tie trunks to pool number 891, the tie pool.

  NOTE:
  When a Release 1.0 system is initialized, all loop-start and ground-start trunk programming reverts to loop-start. (The ground-start pool never has trunks assigned to it automatically, and you have to program the trunks after the ground-start ports are designated.) In Release 1.1 and later, ground-start trunks are assigned to the ground-start pool on initialization, except in a system switched or strapped for Permanent Key mode operation.

Through system programming, you can rearrange the three automatically assigned pools and create special-function or special-user pools. For example, you can divide the main pool and assign smaller pools of loop-start trunks to different groups of users.

Line Access

To make an outside call, the single-line telephone user dials a pool access or ARS dial-out code, and the system automatically selects an outside trunk. In addition, **SA** buttons on multiline telephones allow you to make different kinds of calls from the same button, including outside calls on basic loop-start, ground-start, or tie trunks or on special service facilities such as WATS, as well as inside calls to other system extensions.

You can assign the following types of buttons to multiline telephones:

- **SA Ring.** Use this button to make and receive inside and outside calls. When you use an **SA Ring** button to make an inside call, the telephone at the destination extension rings once per cycle to indicate an inside call.
- **SA Voice.** Use this button to make and receive inside and outside calls. When you use an **SA Voice** button to make an inside call, the user at the destination extension hears the caller's voice on the speakerphone after a single beep, rather than after ringing. (If the user has a single-line telephone, has a telephone without a speakerphone, or has disabled voice announcements, the telephone rings just as if the call were on an **SA Ring** button.)

- **SA Originate Only.** Use this button to make inside and outside calls. You can receive neither inside nor outside calls on **SA Originate Only** buttons. The purpose of this type of button is to ensure that you always have a button available to make or transfer a call, establish a conference call, answer a call-waiting call, or pick up parked calls. For inside calls, you can program the button for either voice or ring operation.
- **Shared SA.** Use this button to allow two or more users to answer each other's calls, join conversations, or make or receive inside or outside calls on each other's **SA Ring** or **SA Voice** buttons. In a shared System Access arrangement, one extension is the *principal* (or primary) extension. This extension is the telephone whose **SA Ring**, **SA Voice**, and/or **SA Originate Only** buttons are assigned as **Shared SA** buttons on other extensions.

  Shared SA buttons are often used by secretaries and their bosses, as well as people who work closely together as in a customer service department. For inside calls, you can program the button for either voice or ring operation. A DLC may be a principal extension but cannot have **SSA** buttons.
- **Pool.** Use this button to make outside calls on a specific trunk pool. To make an outside call, press the appropriate **Pool** button; no dial-out code is necessary.
- **Personal Line.** Use this button to dedicate the use of a specific outside trunk to one or more telephones in the system. You can use the personal line button to make and receive only outside calls. To make a call, press the appropriate personal line button; no dial-out code is necessary.

You can assign any combination of up to 28 **SA Voice**, **SA Ring**, **SA Originate Only**, and **Shared SA** buttons to each telephone (but *not* to a QCC) on buttons 1 through 28. (See *System Planning* for button diagrams.) All of these can be **Shared SA** buttons. An **SA** button on a principal extension may be shared by up to 16 extensions, but an extension may have only one **Shared SA** button for a given principal extension. The number of personal line buttons that you can assign to a telephone is limited only by the number of trunks in the system and the number of buttons available on the telephone.

Operator Consoles

The other modes of operation support only Direct-Line Consoles (DLCs) operator console. Hybrid/PBX mode supports both DLCs and Queued Call Consoles (QCCs).

Queued Call Console

Available only in Hybrid/PBX mode is the Queued Call Console (QCC), which allows calls to come to the operator one at a time. This is especially useful when the number of outside trunks connected to the control unit exceeds the number of buttons available on a Direct-Line Console (DLC).

A QCC holds a call in queue until a QCC operator is available. After the call reaches a **Call** button on a QCC, the operator transfers it to the desired inside extension. The operator can switch between the caller and the called person to screen calls.

Direct-Line Console

This type of operator console, discussed in the sections covering the other two operating modes, differs from the QCC in that all outside lines are automatically assigned to the operator position as personal lines (you can reprogram the line buttons, however). Line buttons can be programmed with features, while the QCC button assignments cannot be changed. Calls are not queued, and more than one call can ring at a time. In Hybrid/PBX mode, this type of console is often useful for calling supervisors or departmental receptionists.

Hybrid/PBX Mode Considerations

Review the following operational considerations for Hybrid/PBX mode:

- Hybrid/PBX mode provides the most efficient use of outside trunks, since they can be pooled and are more readily available to users. You can program the ARS feature for more cost-effective use of pools.
- Hybrid/PBX mode provides greater functionality for single-line telephones than other modes of operation. You can make both inside and outside calls with a single **SA** button assigned to the phone.
- The QCC, available only in Hybrid/PBX mode, ensures efficient call handling and is especially useful when the number of lines exceeds the number of buttons available on a DLC system operator position.

Behind Switch Mode

Behind Switch mode simply means that the communications system control unit is connected to (or is *behind*) another communications system, which acts as the switch to the CO. The other system is referred to as the *host*, and can be either a private branch exchange (PBX) or Centrex service. Centrex is a telephone company service that provides PBX-like features but is located at the CO.

Figure 1–5 illustrates a simple Behind Switch configuration in which the outside trunks connect to the host system. The lines connecting the two control units are similar to extensions that provide users with access to the host system's trunks, as if through a trunk pool.

You assign each extension number from the host system to individual telephones as a *prime line.* The prime line rings when a person receives an outside call. Unless the person manually selects a different line, he or she is automatically connected to this line (even when the line is in use).

You can assign a prime line to additional telephones as a *secondary line* so that users can see activity on other telephones sharing the button and join their co-workers' conversations.

The system automatically assigns all prime lines to DLC system operator positions. The first line assigned is the system operator's prime line, and the rest are assigned as secondary lines. This setup allows the system operator to answer calls received by other people on their prime lines.

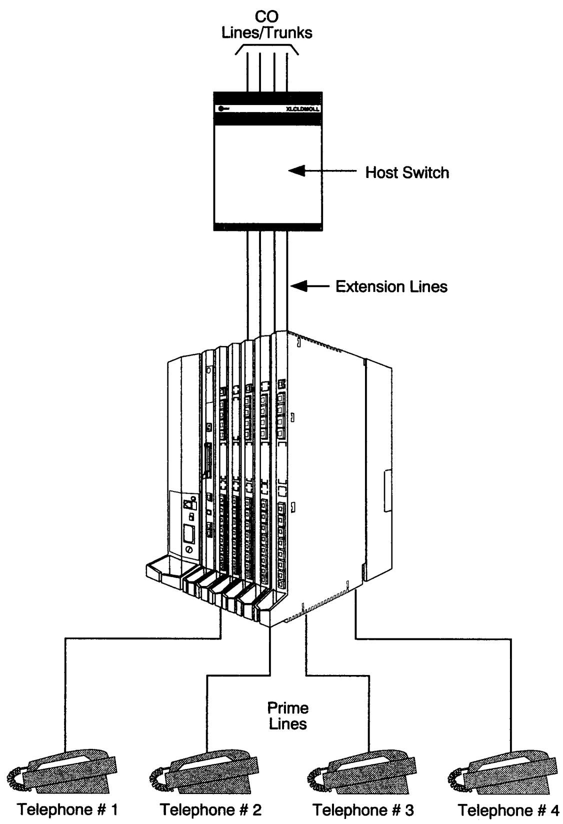

Figure 1–5. Behind Switch Mode

Modified Configurations

In addition to accessing the host's outside trunks, you can modify the Behind Switch system to bypass the host and provide direct access to outside trunks, for example, to connect WATS lines directly to the control unit so that they are available only for the local system's users. Figure 1–6 illustrates the modified configuration.

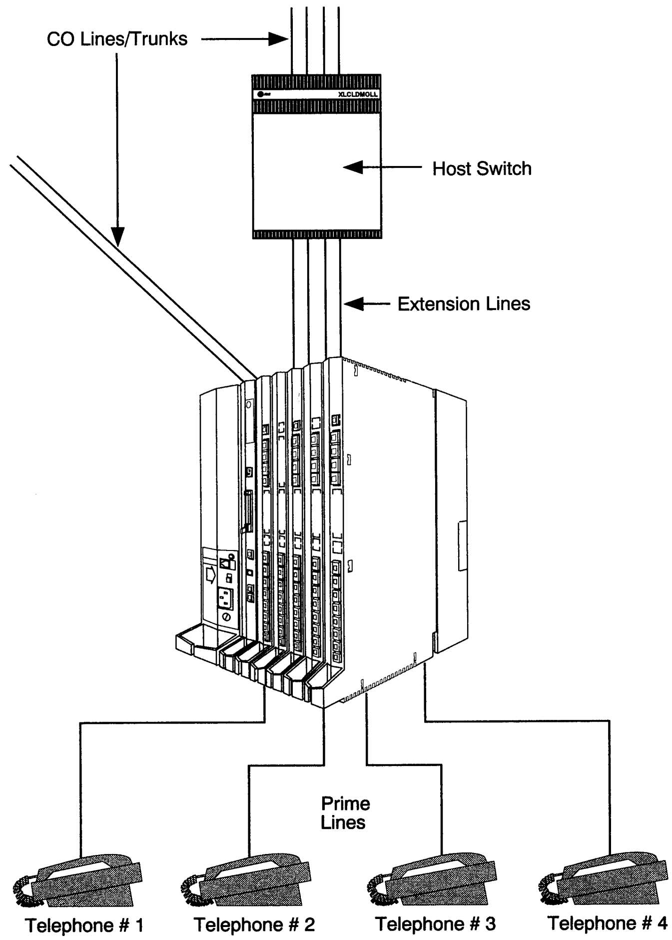

Figure 1–6. Behind Switch Mode with Direct Outside Trunks

Depending on your business needs, you can add the following kinds of direct outside trunks:

- Loop-start (including basic, WATS, and FX)

- Ground-start (only if not registered as KF and not switched or strapped for Key mode)
- DS1 facilities
- Tie

You must terminate the direct outside trunks on individual telephones, and the trunks must appear on the telephones' line buttons. For example, if you assign tie trunks to buttons on Telephones 1 and 2, the buttons on those telephones appear as shown in Figure 1–7.

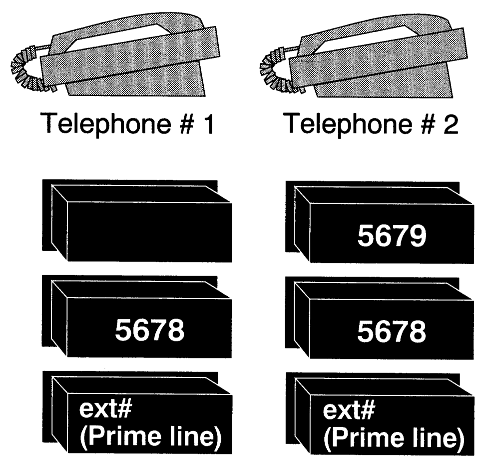

Figure 1–7. Labeled Line Buttons for Behind Switch Telephones

One button on each telephone is reserved for the prime line from the host system. The outside lines appear on the labeled line buttons above the prime line. In Figure 1–7, the first line (5678) is assigned to buttons on both telephones, and the second (5679) is assigned to a button only on the second telephone.

The following features are not available in Behind Switch mode:

- Trunk pools
- Automatic Route Selection
- QCC
- **SA** buttons

Line Access

Multiline Telephones

In Behind Switch mode, the most commonly used telephones are analog and MLX multiline telephones.

When you program the system for Behind Switch mode, the system assigns a single prime line, an **ICOM Ring** button, and an **ICOM Voice** button to each multiline telephone. When you lift the telephone handset, the prime line is selected automatically (even when it is busy) unless you have first selected a different button. The prime line connects only to the host system, not directly to an outside trunk.

To call another person connected to the host system, you dial the *host system* extension number assigned to that person. To access an outside trunk, you dial the host system's dial-out code (usually a **9**), and the host system selects an available outside trunk.

In Behind Switch mode, **ICOM** buttons allow you to call other people connected to the system. When you press an **ICOM** button, you reach an inside talk path and dial tone from this system (not from the host). You can then reach co-workers without tying up a prime line.

You can use the following types of buttons to make and receive inside calls in Behind Switch mode:

- **ICOM Ring.** Use this button to make inside calls and to receive inside calls and outside calls transferred from another extension. When you use an **ICOM Ring** button to make an inside call, the telephone at the destination extension rings with one burst to indicate an inside call.
- **ICOM Voice.** Use this button to make inside calls and to receive inside calls and outside calls transferred from another extension. When you use an **ICOM Voice** button to make an inside call, the person at the destination extension hears your voice on the speakerphone after a single beep, rather than after ringing. (If the person has a single-line telephone, has a telephone without a speakerphone, or has disabled voice announcements, the telephone rings just as if the call had been made on an **ICOM Ring** button.)
- **ICOM Originate Only.** Use this button to make inside calls only. Neither inside nor outside calls can be received on an **ICOM Originate Only** button. This button ensures that you always have a button available to make or transfer a call, establish a conference call, answer a call-waiting call, or pick up a parked call. You can program the button for either voice or ring operation.

You can assign a combination of up to 10 **ICOM Voice**, **ICOM Ring**, and **ICOM Originate Only** buttons to each multiline telephone, on buttons 1 through 10. (See *System Planning* for button diagrams.) The number of prime line buttons that can be assigned to a telephone is limited only by the number of trunks in the system and the number of buttons available on the telephone. When both systems have common features, you must decide which system to use for those features.

When you press a fixed-feature **Conference**, **Drop**, or **Transfer** button, the respective host features are activated, not those of the communications system. However, an unused line button on a telephone can be programmed for the communications system's own **Conference**, **Drop**, or **Transfer** feature. Each system must program to meet your needs, and you must give users the appropriate access instructions.

Single-Line Telephones

You can also use single-line telephones in Behind Switch mode. They can be set up in two ways. In one configuration, you have constant access to a prime line (the default setting for Behind Switch mode) but cannot make inside calls or use system features. In the second configuration, the single-line telephone may be programmed, through system programming of Automatic Line Selection (ALS), to access an intercom line. With this second configuration, you can make and receive inside and prime line calls and can use system features. A single-line telephone has access only to the host system's (not the communication system's) Conference, Drop, and Transfer features, using a switchhook flash.

Behind Switch Considerations

Behind Switch mode is appropriate for users who are part of a large organization. For example, a department might not want (or be able) to support a large-capacity PBX. Programming the communications system for Behind Switch operation provides the advantage of the host's features and capabilities. A business with multiple locations can use Centrex services to provide the appearance of a single system at all locations.

NOTE:
Ground-start trunks have become available to Centrex systems in some regions. The MERLIN LEGEND Communications System does not support ground-start Centrex lines/trunks.

FCC Registration

The account representative who planned the system's mode of operation provides the FCC registration number that you report to the local telephone company. Depending on the mode of operation and the hardware switch or strap in the processor module, this number includes the letters *KF*, *MF*, or *PF*, loosely corresponding to *key function, multifunction*, or *PBX function*, respectively. (The FCC has no Behind Switch classification.) Table 1–2 lists the registration number(s) used for each mode of operation. The guidelines described in the next section determine the classification.

NOTE:
The system's modes of operation (Key, Behind Switch, and Hybrid/PBX) do not correspond directly to these designations.

Table 1–2. FCC Registration Numbers

Mode of Operation	Registration Number
Key or Behind Switch	AS593M-72914-KF-E
Key, Hybrid/PBX, or Behind Switch	AS593M-72682-MF-E
Hybrid/PBX	AS5USA-65646-PF-E

KF Classification

This classification is applicable only to Key and Behind Switch modes of operation. If any of the conditions listed below are met, the system is registered under the KF classification.

Key Mode

- The system is switched or strapped for Permanent Key operation.
- All outside trunks terminate on one or more telephones.
- All outside trunks are loop-start, tie, DS1 facilities, or a combination of these.
- No trunks are pooled.

Behind Switch Mode

- The system is switched or strapped for Permanent Key operation.
- No outside trunks connect directly to the control unit. The communications system accesses only trunks connected to the host system.
- No ground-start trunks connect directly to the control unit.
- No trunks are pooled.

MF Classification

This classification is applicable to all three modes of operation, Key, Hybrid/PBX, and Behind Switch. If any of the conditions listed below are met, the system is registered under the MF classification.

Key Mode

One or more ground-start trunks connect directly to the control unit. These trunks connect to a 400 GS/LS, 408 GS/LS, 800 GS/LS, 800 GS/LS-ID (Release 3.0 and later only), or 408 GS/LS-MLX module.

Hybrid/PBX Mode

All outside trunks are pooled; no trunks are terminated directly on a telephone. The only directly terminated trunks are personal lines, which can be shared and appear on more than one telephone.

Behind Switch Mode

One or more ground-start trunks connect directly to the control unit. These trunks connect to a 400 GS/LS, 408 GS/LS, 800 GS/LS, 800 GS/LS-ID (Release 3.0 and later only), or 408 GS/LS-MLX module. The processor module must not be switched or strapped for Permanent Key operation.

PF Classification

This classification is applicable only to Hybrid/PBX mode. If either of the following conditions is met, the system is registered under the PF classification:

- All outside trunks are pooled; no trunks are terminated directly on a telephone.
- The only directly terminated trunks are personal lines, which can be shared and appear on more than one telephone.

Programming

Programming customizes a communications system with the appropriate features and options required by your company for optimal performance. There are three types of programming:

- **System Programming.** Allows the system manager to program features that affect all or most system users.
- **Centralized Telephone Programming.** Allows the system manager to program any feature that can be programmed by individual telephone users or system operators.
- **Extension Programming.** Allows telephone users and system operators to tailor their telephones to meet personal needs.

System Programming

You can program the following system options:

- Basic operating conditions
- System renumbering
- Settings for lines/trunks
- Telephones and operator consoles
- Adjuncts
- Applications
- Optional features

Program system options from:

- An MLX-20L telephone connected to one of the first five extension jacks on the first MLX module in the control unit. The first MLX module in an expansion unit cannot be used to connect the telephone for programming.
- The built-in modem in the processor module, which permits remote programming through the public network. For example, support personnel can access the system by using a PC with a modem and SPM software; support personnel call the system to gain access.
- A PC with SPM software directly connected to the programming RS-232 jack on the processor module.

You access programming options from display screen menus. (For more information, see *System Programming*.) To use SPM to program your system on a PC, you need SPM software and a PC with version 3.3 (or a later version) of DOS.

The PC must have:

- At least 640 kbytes of random access memory (RAM)
- A floppy disk drive that can accommodate the SPM diskette
- A serial port that can use either a DB-9 or DB-25 connector
- Either a 355AF modular adapter (if a male connector is on the interface cable) or a 355A modular adapter (if the connector is female)
- A 4-pair modular cord (D8W)
- Either a monochrome or color monitor

In addition, the following equipment is useful:

- A parallel printer (the PC needs a parallel port for the connection)

- A 1200- or 2400-bps modem for local or remote connection

NOTES:

1. For a DB-9 connector, use a 9- to 25-pin adapter to convert the 25-pin connector to a modular connector.
2. SPM uses Interrupt 4 and I/O address 3F8 for COM1. It uses Interrupt 3 and I/O address 2F8 for COM2.
3. When you use a PC with SPM to program the system, the maximum number of MLX-20L telephones that you can connect to the system is reduced by one.

Centralized Telephone Programming

Centralized telephone programming is an option you can choose from the System Programming menu to program any feature onto a telephone. Although individual telephone users and system operators can also program many features, the following features can be programmed *only* through centralized telephone programming:

- Barge-In
- Headset Hang Up
- **ICOM** buttons, all types (Key and Behind Switch modes only)
- **SA** buttons, all types (Hybrid/PBX mode only)

Extension Programming

Extension programming allows telephone users and system operators to tailor their telephones to meet personal needs. Multiline telephone users can assign a wide range of features to multiline telephone buttons. In addition, users can program settings that do not require button assignment (such as Call Waiting) on both multiline telephones and single-line telephones.

To program telephones, users dial programming codes or, on MLX display telephones, select features from the display. When a telephone is in programming mode, the system considers it busy; therefore, no incoming calls ring at that telephone until the programming session is over.

System Capacities and Requirements

This section details the technical requirements and capacities of the system:

- Hardware and software capacities for the system
- Environmental requirements for placement of the control unit
- Power and grounding requirements for operating the system

Hardware and Software Capacities

You can configure the system as a stand-alone unit or as part of a private network. Maximum system capacities are as follows:

- Up to 108 simultaneous two-party conversations
- Up to 80 line/trunk jacks, including loop-start, ground-start, DID, and tie
- Up to 255 extension endpoints that support a combination of the following:
  - Up to 144 physical extension jacks for telephones and adjuncts
  - Up to 127 logical digital data ports (through 7500B data modules connected to jacks on the MLX module) providing RS-232 connections to data terminals and personal computers
- System call-handling capability of 3828 hundred call seconds per hour (ccs/hr)
- Up to three 100D DS1 modules

The system has a total capacity of 224 jacks (80 outside lines/trunks plus 144 extensions); however, each MLX module extension jack supports two logical endpoints (extension devices that can operate simultaneously and independently of each other). For example, an MLX telephone with a Multi-Function Module (MFM) plugs into one extension jack, but the jack supports both the telephone and the equipment (for example, a fax or answering machine) connected to the MFM.

Similarly, although the 100D module has only one jack, it can serve up to 24 endpoints (emulated lines/trunks or PRI lines/trunks). Thus, you can configure the system to connect up to 80 lines/trunks and 255 extension endpoints, a total of 335 endpoints.

IMPORTANT:
The system has a time-slot capacity of 216. If more than 216 endpoints are in use at the same time, blocking can occur.

Table 1–3 lists the hardware and software capacities of the system. Constraining Factors appear with a checkmark (✔) and are explained at the end of the table.

Table 1–3. Hardware and Software Capacities

	Limit	Constraining Factor
Allowed/Disallowed Lists		
Number of lists	8	
Entries per list	10	
Digits per entry	7	
Automatic Route Selection		
Number of ARS tables	16	
Subpatterns per table	2	
Routes per subpattern	6	
Entries per table	100	
Entries across all tables	1600	
Default tables	4	
Callback		
Number of calls in queue	64	
Calling Groups		
Number of groups	32	✔
Members per group	20	
Groups per member	1	
Delay announcements per system	32	
Delay announcements per group	1	
Groups per delay announcement	32	
External alerts per group	1	
Coverage groups per group	1	
Carriers	3	
Line/trunk and extension module slots per basic carrier	5	✔
Line/trunk and extension module slots per expansion carrier	6	
Maximum slots available for line/trunk and extension modules	17	
Coverage Groups		
Number of groups	30	✔
Senders per group	144	
Groups per sender	1	
Receiver buttons per group	8	
Groups per QCC receiver	30	

Continued on next page

Table 1–3, *Continued*

	Limit	Constraining Factor
Data Hunt Groups		
Number of groups	32	
Members per group	20	
Groups per member	1	
Direct Inward Dialing		
Number of blocks	2	
Number of trunks	80	
Directories		
System Directory	1	
Listings	130	
Extension Directory	1	
Listings	144	
Personal Directory (MLX-20L only)	48	
Listings	50	
100D Module (maximum 2 per carrier)	3	
Endpoints (devices)	255	
Extensions		
Total physical jacks	144	
Total endpoints	255	
Fax Machines with Message Waiting	16	✔
Lines/Trunks	80	
Night Service		
Groups	8	
Members per group (including one group calling number)	144	
Groups per member	8	
Emergency Allowed List entries	10	
System Operating Consoles		
DLCs		
MLX-20L or MLX-28D	8	✔
BIS-22D, BIS-34, BIS-34D, or MERLIN II System Display Console	8	✔
QCCs	4	✔
DSSs	16	✔
Combination of DLCs and QCCs	8	
Number of consoles per module	2	
Park Codes (number of codes)	8	
Personal Lines	64	

Continued on next page

Table 1–3, *Continued*

	Limit	Constraining Factor
Pool Buttons	64	
Ports (not simultaneously)		
Total extensions	224	
Voice and data (physical pools)	144	
Voice Announce to Busy extensions	127	
Voice-mail interface	20	✔
7500B data module data	127	
Paging	3	
Delay announcements	32	
Remote Access		
Number of barrier codes	16	
Digits per code, systemwide	4–11	
Shared System Access Buttons		
No. of buttons per principal extension	27	
Speed Dial		
Personal Speed Dial		✔
Entries per telephone	24	
Entries per system	1200	
Digits per entry	28	
System Speed Dial		
Entries per system	130	
Digits per entry	40	
System Programming Equipment		
MLX-20L	1	✔
RS-232 jack (to connect PC w/SPM)	1	
Modem (built-in processor module)	1	

Continued on next page

Table 1–3, *Continued*

	Limit	Constraining Factor
Telephones (not simultaneously)		
Single-line	144	✔
Analog multiline		
Without Voice Announce to Busy	136	✔
With Voice Announce to Busy	68	✔
MLX-20L	48	✔
All other MLX telephones (with/without 7500B data module/MFM)	127	✔
Power failure transfer	20	
Traffic (100 call seconds/hr/system)	3828	✔
Two-Party Conversations	108	✔
Voice-Messaging Systems	1	

Constraining Factors

This section describes the constraining factors that limit the capacities supplied in the table above.

Calling Groups
Members of groups. QCCs cannot be members of calling groups because the QCC position is set up as a system operator and has its own queue different from the group's queue.

Carriers
The first slot of the basic carrier is used for the processor module.

Coverage Groups
Senders per group. QCCs cannot be senders, because they don't have coverage available and use Position-Busy instead.

Fax machines with message-waiting
The system can support more than 16 fax machines, but those in excess of 16 cannot use the fax message-waiting indication.

System operator consoles
DLCs. Two consoles are allowed for each MLX or analog module, with a maximum of eight per system. Up to two DSSs can be attached to an MLX operator console, and one is built into the MERLIN II System Display Console.
QCCs. Two consoles are allowed for each MLX module with a maximum of four per system.

Ports (not achievable simultaneously)

Voice-mail interface	Although the system software supports up to 24 voice-mail interface (VMI) ports, all the VMI ports must be in the same calling group, and the maximum number of extensions in a calling group is 20.

Speed Dial
Personal Speed Dial. Single-line and 5- or 10-button telephones only.

System programming equipment
Remote access overrides on-site programming except during backup or restore.

Telephones (not achievable simultaneously)

Single-line	RAM limit
Analog multiline without Voice Announce to Busy	17 slots x 8 ports/board.
MLX-20L	RAM limit and the total includes the MLX-20L telephone used for system programming.
All other MLX telephones	RAM limit
Power failure transfer	1 for each 4 LS/GS trunk jacks.

Traffic (hundred call seconds/hr/system)
Assumes 20 percent internal traffic.

Two-party conversations
216 time slots

Environmental Requirements

The control unit requires a regulated environment and can be located in any room or closet that is temperature-controlled and clean. Do not mount the control unit where it will be exposed to direct sunlight.

In addition, the control unit should not be co-located with air conditioning or ventilation units, compressors, fans and blowers, heaters, arc welders, or other machinery that produces electrical interference.

The control unit is mounted on an AT&T Pre-drilled backboard.

Once installed, it is important to keep the control unit site clear of hazards, such as stacked paper or boxes, that block ventilation. Installing any machinery in the vicinity of the control unit should be avoided. If any pollution-producing work (such as sanding or spray painting) is to be done in the area, care should be taken to protect the unit.

Table 1–4 gives the environmental specifications for the control unit.

Table 1–4. Environmental Specifications

Control Unit

Fully loaded basic carrier
Weight: 45 lb (20.4 kg)
Dimensions: 14 in. wide x 23 in. high x 12 in. deep
(35.6 cm x 58.4 cm x 30.5 cm)

Fully loaded 2-carrier system (basic carrier plus one expansion carrier)
Weight: 90 lb (40.8 kg)
Dimensions: 25 in. wide x 23 in. high x 12 in. deep
(63.5 cm x 58.4 cm x 30.5 cm)
Fully loaded 3-carrier system (basic carrier plus two expansion carriers)
Weight: 135 lb (61.2 kg)
Dimensions: 37 in. wide x 23 in. high x 12 in. deep
(94 cm x 58.4 cm x 30.5 cm)

Mean Time Between Failures

2.1 years
(mean/average time the system is expected to operate before any type of failure occurs)

Mean Time Between Outages

4.0 years
(mean or average time the system is expected to operate before a failure affecting more than 25 percent of extensions or lines for more than 15 seconds occurs)

Backboard Mounting Hardware Requirements

This list identifies the type of wall construction the backboard is mounted to.

Type of material	Mounting Hardware
Wood surface	Wood screws
Concrete surface, brick, cinder block	Masonry anchors
Plaster, plasterboard	Toggle bolts
Sheet-metal surface	Sheet-metal screws

Hardware has a combined pullout force of 650 lb (294.8 kg). When mounting to sheet-metal walls, attach to structural members.

Location

Within 5 ft. (1.5 m) of dedicated AC power outlet (1 plug per carrier)
Within 1000 cable ft. (304.8 m) of telephones

Continued on next page

Table 1–4, ***Continued***

Heat Dissipation

Fully loaded basic carrier	500 Btu/hr	(35 cal/sec)
Fully loaded 2-carrier system (basic carrier plus one expansion carrier)	1000 Btu/hr	(70 cal/sec)
Fully loaded 3-carrier system (basic carrier plus two expansion carriers)	1500 Btu/hr	(105 cal/sec)

Power Requirements

Basic carrier	117 VAC	60 Hz +/- 5%	3 A
2-carrier	117 VAC	60 Hz +/- 5%	6 A
3-carrier	11 7 VAC	60 Hz +/- 5%	9 A

Temperature/Humidity Range

40°–104°F (4°–40°C)
20%–80% relative humidity

Ventilation Clearances

1 in. (2.5 cm) on right and left sides

Radio Frequency Interference, Tolerance 1.0 V/m

In most cases, electrical noise is introduced to the system through trunk or telephone cables. However, electromagnetic fields near the control unit may also induce noise in the system. Therefore, the control unit and cable runs should not be placed in areas where a high electromagnetic field strength exists. Radio transmitters (AM or FM), television stations, induction heaters, motors (with commutators) of 0.25 horsepower (200 watts) or greater, and similar equipment are leading causes of interference. Small tools with universal motors are generally not a problem when they operate on separate power lines. Motors without commutators generally do not cause interference. Field strengths below 1.0 volts per meter are unlikely to cause interference.

The field strength produced by radio transmitters can be estimated by dividing the square root of the emitted power in kilowatts by the distance from the antenna in kilometers. This yields the approximate field strength in volts per meter and is relatively accurate for distances greater than about half a wavelength (150 m for a frequency of 1000 Hz).

Continued on next page

Table 1–4, *Continued*

Electromagnetic Interference (EMI)

See Appendix A, "Customer Support Information." To reduce electromagnetic interference emissions (possible interference problems with handheld telephones) check the date of manufacture of the CPU (517A27) units. If they were manufactured before April 1993, replace with a later version.

CAUTION:

For the control unit, do not use an AC outlet that is controlled by a wall switch or some other switch.

Use an approved ground (AC receptacle for 3-prong plug).

Do not install the control unit outdoors.

Do not place the control unit near extreme heat (furnaces, heaters, attics, or direct sunlight).

Do not expose the control unit to devices that generate electrical interference (such as arc welders or motors).

Do not place anything on top of carriers.

Do not install the control unit under any device that may drip fluid, such as an air conditioner.

Each auxiliary power unit requires one outlet.

Do not expose the control unit to moisture, corrosive gases, dust, chemicals, spray paint, or similar materials.

Power and Grounding

Proper power and grounding are essential for correct and safe functioning of the system.

Power Specifications

The system control unit plugs into a 117-VAC outlet. To avoid accidental disconnection of the system, this outlet should not be controlled by a wall switch.

Each carrier unit requires its own power supply. Each power supply requires a maximum current of 3 amps. Therefore, if expansion carrier units are added to the system, extra AC outlets may be needed.

Grounding Requirements

Proper grounding of the installation site protects the system against the following:

- Lightning
- Power surges
- Power crosses on outside lines/trunks
- Electrostatic discharge (ESD)

The telephone company is responsible for providing protection of outside lines/trunks at the entrance to the site. The protection should consist of the following:

- Carbon blocks or gas discharge tubes connected to an approved ground
- Adequate bonding of the outside line/trunk protector ground and the power company ground

> ⚠ **WARNING:**
> *An improper ground can result in equipment failures and service outages. Verify that the AC power uses an approved ground for its primary ground, that all voltage-limiting devices are grounded to an approved ground, and that the ground is one of the approved grounds below.*

The following is a list of approved grounds, starting with the most preferred:

- Building steel
- Acceptable water pipe, must be a metal, underground water pipe at least ½ in. (30.4 cm) in diameter, and in direct contact with the earth for at least 10 ft. (3 m).

  It must be electrically continuous so that the protector ground is connected. (Check for insulated joints, plastic pipe, and plastic water meters that might interrupt electrical continuity.)

  A metallic underground water pipe must be supplemented by the metal frame of the building, a concrete-encased ground, or a ground ring. If these grounds are not available, one of the following types of grounds can supplement the water pipe ground:

- Other local metal underground systems or local underground structures such as tanks and piping systems
- Rod and pipe electrodes, a 5/8-in. (1.6-cm) solid rod or ¾ in. (1.9-cm) conduit or pipe electrode driven to a minimum depth of 8 ft. (244 cm)
- Plate electrode, a minimum of 2 square ft. (61 square cm) of metallic surface exposed to the exterior soil

■ Concrete-encased ground, which must be an electrode, consisting of one of the following:

- At least 20 ft. (6.1 m) of one or more steel reinforcing rods, each being at least 0.5 in. (1.27 cm) in diameter.
- 20 ft. (6.1 m) of bare copper conductor not smaller than #4 AWG, encased in 2 in. (5 cm) of concrete. This electrode must be located within and near the bottom of a concrete foundation or roofing that is in direct contact with the earth.
- Ground ring, consisting of at least 20 ft. (6.1 m) of bare copper conductor not smaller than #2 AWG, encircling the building. The ground ring must be in direct contact with the earth and buried at least 2.5 ft. (77 cm) below the earth's surface.

WARNING:
Do not use a metal underground gas piping system. This is a safety risk.

For most power surges, the following standard grounding requirements provide adequate lightning and surge protection:

- Properly wired/grounded/bonded outside line protectors
- Properly wired/grounded AC outlet
- Properly grounded single-point ground bar
- Properly wired connection between single-point ground and power supplies

Additional Power Surge Protection

The 391A1 power supply has built-in AC line protection. This built-in protection handles almost all situations.

Occasionally, additional protection may be needed if the customer is located in a heavy lightning area. A 147A surge protector can be connected to the system to limit surges from the AC lines and outside lines. One 147A protector provides protection for four outside lines. Up to three 146A protectors can be added to the 147A to provide protection for a maximum of 16 outside lines. For more than 16 lines, additional 147A protectors are required.

NOTE:
The 147A protector is usually not needed with the 391A1 power supply. It may be needed with the older 391A power supply module in heavy lightning areas.

Complete installation instructions are provided with the protectors.

Unit Loads

A unit load is a measure of power (1.9 watts) used to determine the electrical load that the following components have on each carrier's power supply:

- Telephones and adjuncts
- 800 DID modules

Only the telephones and adjuncts that connect to the analog and digital ports on the control unit require unit load calculation. Do not include any equipment with its own power supply, for example, a fax machine, an MFM, or an answering machine, in the unit load calculation.

Before installation, unit load and auxiliary power requirements for a new system are computed by qualified service personnel, and any necessary auxiliary power equipment is ordered automatically. However, in the event of maintenance or equipment changes, unit loads should be calculated to ensure proper operation under all conditions.

The 391A1 power supply provides 54 unit loads to each carrier (the 391A provides 45). If the unit load requirement for a carrier exceeds 54 (or 45) unit loads, an auxiliary power unit is needed to allow that carrier to support an additional 27 unit loads.

CAUTION:
Running the system with more than 54 or 45 (depending upon the power supply) unit loads per carrier may not appear to do harm. However, this can cause the system to malfunction, creating "No Trouble Found" situations, such as malfunctioning LEDs on multiline telephones, or power unit failure.

An auxiliary power unit redirects the power requirements from the last two slots on the carrier. Any telephone connected to the modules in the last two slots receives power from the auxiliary power unit instead of from the power supply module. There is more information about auxiliary power units in Chapter 2, "Auxiliary Power Units."

Checking Unit Loads

In the event of maintenance or equipment changes, recalculate the unit loads for each carrier resulting in a different configuration (use the worksheet in *Installation*, if available). As a general rule, if you can distribute the 800 DID modules and telephone modules equally across the carriers, you will prevent unnecessary drain on any one carrier.

Also, depending on the system's mode, the rules vary. The next two sections provide the rules for calculating unit loads in various modes.

Unit Loads for Hybrid/PBX Mode

The power supply module generally supports six modules of any type in a Hybrid/PBX system, without requiring an auxiliary power unit.

If, however, both of the following conditions are true, the unit loads on a carrier can exceed the 54-unit maximum, and therefore require auxiliary power:

- All six carrier slots are occupied by MLX telephone or analog multiline telephone modules.
- The carrier has a total of more than 45 MLX-20L telephones or 34-button analog multiline telephones installed.

Unit Loads for Key or Behind Switch Mode

In a Key or Behind Switch system with four or fewer modules, no calculation is needed. The power supply module generally supports four modules of any type in Key or Behind Switch mode.

Release Differences

Release 1.1 Enhancements

Refer to *Release 1.1 Notes* for detailed descriptions of Release 1.1 enhancements. Release 1.1 includes all Release 1.0 functionality plus the enhancements described in the following sections.

Language Selection

This selection allows you to program the system for the display of prompts, menus, and messages on MLX display telephones in English, French, or Spanish. You can also program the following options in any of these languages, independently of the system language:

- Individual extensions with MLX telephones
- System Programming and Maintenance (SPM)

- System programming reports
- SMDR report headers

MLX-10D, MLX-20L, and MLX-28D display telephones and MLX-10 nondisplay telephones are available in three separate versions, with factory-set buttons in English, Spanish, or French. (The MLX-10DP is available in the English version only.) In addition, user and operator guides and telephone tray cards are available in all three languages.

Programming and Maintenance

Programming and maintenance enhancements include the following:

- Additional Inspect capability in system programming
- Editing capability (Backspace selection) in extension programming
- Improvements to system reports
- An access log that records the last 20 times maintenance or system programming has been accessed
- Longer (20-second) gap between ring cycles for programming mode and Forced Idle tone

Operational

System operational enhancements include the following:

- Automatic selection of an **SA** button when Conference is invoked (Hybrid/PBX mode)
- Prompting through Conference feature on MLX display telephones
- Relocation of the More prompt on the MLX-20L display
- Display of the number saved on a programmed Last Number Dial or Saved Number Dial button when the button is inspected

SPM

SPM enhancements include operation in English, French, or Spanish, faster backup and restore, and automatic on-screen display of reports as they are created, with a Browse capability for reading the reports.

Equipment

Additional equipment includes the 8102 and 8110 analog telephones, four headsets, two headset amplifiers, and a transparent protective cover for the MLX-10 and MLX-10D telephones. The 8102 and 8110 telephones are also compatible with Release 1.0.

PF Registration

PF registration number AS5USA-65646-PF-E is assigned by the FCC for operating the MERLIN LEGEND Communications System in Hybrid/PBX mode in the United States. (The PF registration is also applicable to Release 1.0 systems.)

Release 2.0 Enhancements

Refer to *Release 2.0 Notes* for detailed descriptions of Release 2.0 and later enhancements. Release 2.0 includes all Release 1.1 functionality plus the enhancements listed below.

Programming and Maintenance

Programming enhancements include the following:

- Extension Copy is a feature that reduces programming time by allowing the use of any extension as a template for programming another extension or block of extensions through centralized telephone programming.
- Integrated Administration provides a single interface through Integrated Solution III (IS III) for programming entries common to the MERLIN LEGEND Communications System and AUDIX Voice Power.
- Any SPM Version 2.xx (where xx is replaced by numbers) provides a Convert function for use in upgrading the system from Release 1.0 or 1.1. This function converts a backup file from a Release 1.0 or 1.1 system to Release 2.0 and later format, allowing reuse of existing system programming on the upgraded system.
- Forced idle reductions keep system interruptions at a minimum. In general, the smallest necessary component is forced idle during programming activities. For example, renumbering a single extension idles only one extension. Only a few systemwide programming activities, such as setting the system mode and system renumbering, idle the entire system.

Maintenance enhancements include the following:

- Clear descriptions of module test failures
- Optional printing of hard copy of error logs
- Display that correlates extension numbers to slot/port and logical ID
- Display showing which slots, trunks, and extensions are maintenance busy
- Internal digital switching element (DSE) loopback test for all modules
- B-channel loopback test for MLX modules

- B-channel line or call service states display
- Error log entries for dual-port RAM errors

Operational

System operational enhancements include the following:

- Coverage VMS is a feature that prevents incoming outside calls from going to voice mail. (All other Coverage remains active as programmed.) The feature is programmed extension by extension, either through extension programming or through centralized telephone programming.
- A Night Service group can be programmed to include either extensions or a calling group as members. However, you should not program both individual extensions and a calling group into the Night Service group, because individuals will not have a chance to answer before calling group members do.
- When AUDIX Voice Power sends a Leave Message notification to an extension, the system identifies the voice mail system as the sender of the message. When the voice mail subscriber uses the Return Call feature, the call goes to any available voice mail port, not just to the specific port that generated the message. This reduces the chance of getting a busy port.
- Coverage receivers can call Coverage senders and have the call receive Coverage treatment. If a receiver calls a sender for whom he or she is covering, and the sender is busy or unavailable, the call proceeds to other points of Coverage. It does not come back to the receiver who originated the call.
- Enhancements to display prompts include automatic posting of a `Do Not Disturb` message when a user activates the Do Not Disturb feature, and confirmation messages when a user activates Hold, Privacy, Saved Number Dial, and Transfer.
- Direct Inward Dialing (DID) trunk emulation on a T1 facility provides up to 24 DID channels on a single DS1 interface, instead of requiring 24 separate physical trunks.
- A telephone user can send a timed flash (switchhook flash) on a loop-start trunk call at a System Access (**SA**) button.

Fax Attendant System

Fax Attendant is an application for sending and receiving fax messages; its interface is similar to the voice mail interface provided by AUDIX Voice Power. Fax Attendant System, which co-resides with AUDIX Voice Power on the IS III platform, provides the following services:

- **Fax Call Coverage.** Receives and holds messages for subscribers whose fax machines are busy or out of paper. This service also allows a subscriber to have a personal fax number without having a fax machine.
- **Fax Mail.** Allows subscribers to create and use fax distribution lists, send and receive fax messages, and record personal greetings for incoming fax calls.
- **Fax Response.** Prompts callers to select and receive faxes from a customer-created menu of choices, using touch-tone responses.

408 GS/LS-MLX Module

The 408 GS/LS MLX module (Releases 2.0 and higher only) combines four line/trunk jacks for ground-start or loop-start trunks and eight extension jacks for MLX telephones on a single module in the control unit.

Primary Rate Interface (PRI)

Primary Rate Interface (PRI) enhancements include the following:

- Connectivity to the 5ESS® Generic 6
- Multiple incoming calls to directory number
- Call-by-Call Service Selection
- Password handling for FTS2000
- Extension ID as Calling Party Number for Automatic Number ID (ANI)

Release 2.1 Enhancements

Refer to *Release 2.1 Notes* for detailed descriptions of Release 2.1 enhancements. Release 2.1 includes all Release 2.0 functionality plus the enhancements listed below.

Operational

System operational enhancements include the following:

- When a call is forwarded to a multiline telephone that has an Auto Dial or DSS button programmed for the forwarding telephone, the green light next to the Auto Dial or DSS button for the forwarding telephone does not flash.
- People answering calls received on **Cover** buttons are allowed to generate touch tones if their telephone is not outward- or toll-restricted.
- Calls received on personal lines with Do Not Disturb on go immediately to Coverage instead of waiting for the coverage delay interval.

- A call put on hold at a **Cover** button can be added to a conference by someone who has a personal line for the call.
- A call put on hold at a **Cover** button can be picked up by any person who has a personal line for the call.
- Calls that have been put on hold at a **Cover**, **SA**, **Shared SA**, or **Pool** button can be picked up by a person who has a personal line button for the call.
- An inside call on hold at an **SA** button can be picked up and transferred by any person with a **Shared SA** button corresponding to the button with the held call.
- Calls that are on hold awaiting transfer can be picked up by any user who has a personal line for the call.
- Beginning with Integrated Solution III Version 1.2, the automatic reconciliation program that was run automatically at 3:00 a.m. has been disabled and can be invoked manually from the User Maintenance menu.
- When a telephone is programmed for Forced Account Code Entry, account codes do not have to be entered when using a programmed Loudspeaker Paging button. In addition, an SMDR record is no longer generated for calls made to paging ports.
- When an MLX telephone, other than an MLX-20L, is plugged into an MLX port and the Personal Directory does not contain any entries, the allocation of the Personal Directory resource is released. If there are any entries in the Personal Directory, the Personal Directory allocation and the entries in the Personal Directory are saved in the MLX port.
- SMDR call records for calls made on PRI facilities are more accurate than SMDR call records for calls made on non-PRI facilities. Outgoing calls made on PRI facilities receive "answer supervision." Consequently, SMDR timing for calls made on PRI facilities begins when the call is answered. Timing for calls made on non-PRI facilities begins when dialing is completed. Therefore, an SMDR call record is not generated when a call made on a PRI facility is not answered at the far end.
- The Call Type field and the Called Number field on the SMDR report have been changed for both the Basic and ISDN report formats.
- An 012 port that is programmed as a *generic* voice messaging interface (VMI) port can transfer an outside call to an outside number.
- In a system where the Transfer Audible option is programmed for Music On Hold and a music source is provided, outside callers who are transferred to a calling group and are waiting in the queue, or outside callers who are parked or camped-on, hear music while they are waiting. Internal callers never hear music on hold while waiting in the calling group queue, or when they are parked, camped-on, or being transferred to another extension.

Installation

Installation enhancements include the following:

- The control unit covers for the MERLIN LEGEND are the same easy-to-use covers as used for the MERLIN II Communications System.
- A new 012 (T/R) module [apparatus code 517G13 (28) or higher letter] contains a built-in ring generator. The maximum ring equivalency number (REN) supported is 2.2, and the module will ring four ports at one time. Bridging of single-line telephones is not supported due to poor transmission quality. It rings four ports at a time.
- A new 008 OPT module (Apparatus Code 517D28) (labeled "with RING GEN.") contains a built-in ring generator. It rings four ports at a time.
- Ferrite cores for the power supply modules are shipped from the factory to comply with FCC Part 15 requirements.

Equipment

Equipment and operations enhancements include the following:

- A new release (Version 2.16) of the System Programming and Maintenance (SPM) software to support international use.
- Support of PRI connection to DEFINITY® Communications Systems.
- Four special-purpose outdoor telephones available in rotary, touch-tone, nondial (to a dedicated number with a ringdown, dedicated circuit), and programmable auto dialing.
- MLX-10DP telephone, identical to an MLX-10D, except that it provides a jack for access to the PassageWay Solution and PassageWay Direct Connect Solution application.

Additional Application Packages, Telephones, Adjuncts, and Adapters

Additional application packages, adjuncts, and adapter enhancements include the following:

- An AT&T Digital Announcer Unit, compatible with all call management systems and tip/ring applications currently available for the MERLIN LEGEND Communications System.
- The HackerTracker™ system software enhancement to the Call Accounting System (CAS) detects abnormal calling activity by allowing monitoring of facilities or authorization code usage.
- A new digital Magic on Hold® unit is available in three configurations:
  - Basic Prerecorded Package
  - Personalized Package

   — Custom Production Package

- The MERLIN Identifier application enables people to receive, store, and use information provided by Custom Local Area Signaling Services (CLASS), specifically, the telephone number of a caller in an area where the service is also supported.
- An Off-Premises Range Extender (OPRE) supports off-premises operation with an off-premises extension capability and extended range operation for tip/ring devices as well as variable gain to improve voice transmission levels.
- PagePac® Plus Loudspeaker Paging System replaces the discontinued (but still supported) PagePac 20 system; PagePac Plus requires no system adapter (it uses the PagePal interface), nor does it require Zonemate; instead, it comes equipped with 8 built-in zones, expandable to as many as 48 zones with 16-zone zone expansion modules.
- PassageWay Solution (Release 1.0)software consisting of five applications that run with Microsoft® Windows 3.1 or later and provide an interface between an IBM®-compatible personal computer and the MERLIN LEGEND system.
- Four single-line telephones with memory buttons: 710, 715, 725, and 730.
- Four specialty handsets compatible with all MLX telephones and the 3101-series, 3178-NHL, 8102, and 8110 single-line telephones. See Chapter 2, "Specialty Handsets," for details.

Release 3.0 Enhancements

Release 3.0 includes all Release 2.1 functionality plus the enhancements listed below.

Equipment

New hardware includes a variety of components. Additional details are included elsewhere in this book.

- CPU modifications include:
  - — A processor running at 16 MHz with a 32-bit wide data bus
  - — 1.5 Mbytes of non-volatile (battery-backed) RAM
  - — 4.0 Mbytes of Flash ROM
  - — PCMCIA Memory Card Interface
  - — A full-duplex 1200/2400 bps modem
  - — Error/Status Code display for maintenance diagnostics

- An 800 GS/LS-ID line/trunk module delivers the calling party's telephone number to the customer premises (MLX display telephones only) if the CLASS service is subscribed to by the customer and if it is supported by the caller's telephone company.
- Pre-fabricated and pre-drilled backboard
- VideoPhone 2500 single-line phone with interactive video display
- MDC 9000 (6-line, cordless)
- MDW 9000 (6-line, cordless, wireless)
- 8101 (single-line telephone, desk or wall-mount, data/fax jack, selectable positive disconnect)
- 2500YMGL (single-line desk set telephone, selectable positive disconnect)
- Picasso Still-Image Telephone (for interactive display of still images)
- PictureTel 4000 (video conferencing system)

Installation, Upgrade, and Backup

There are two ways to convert previous versions of the MERLIN LEGEND Communications System to Release 3.0 as well as doing back up and restore programming. They are:

- PC based tool under SPM feature interface available in both DOS and UNIX® System versions
- PCMCIA Memory Card Interface (a Release 3.0 processor board required)

Operational

Security

The Remote Access feature allows people to dial into the system by dialing the number of a line or trunk designated for remote access. The remote user can optionally be forced to dial a barrier code (a type of password) after reaching the system. In earlier versions, the systemwide barrier code length is fixed at 4 digits. Release 3.0 allows a systemwide barrier code length ranging from a minimum of 4 digits to a maximum of 11 digits, with a default setting of 7 digits. SMDR records are enhanced to provide information for remote access calls. If the remote access call is received on a facility providing Caller ID information, the SMDR report can help trace the call.

Caller ID

Caller information (telephone number) is furnished to MLX display telephones by a 800 GS/LS-ID module. This allows customers to screen calls prior to answering the phone, as well as providing calling party information for use with various applications. This function is only available when the customer subscribes to caller identification service from the telephone company and when the caller's telephone company supports that service.

Shared System Access (SSA)

A telephone may have up to 27 **Shared SA** buttons.

Authorization Codes

The Authorization Code feature allows a user who is at an extension other than his or her *home extension* to use the calling restrictions associated with the home extension. When the visitor enters an authorization code (ranging from 2 to 11 characters and unique across the system), the extension in use takes on the calling priviledges of the home extension and the feature overrides the current restrictions at the host extension. This includes toll restriction, outward restriction, Facility Restriction Level (FRL), Allowed Lists, Disallowed Lists, Night Service Exclusion List, and Dial Access to Pools. All other functions on the telephone are those of the local telephone, not the home extension.

Authorization codes can also be used for the purposes of call control and call accounting through the SMDR printout. The SMDR account code field can hold the Authorization code extension number or the Authorization code itself.

Direct Voice Mail

If your company has voice mail, this feature allows you to dial a co-worker's voice mailbox without calling that person. Direct Voice Mail is especially useful for transferring calls when a co-worker is not available.

Additional Application Packages, Adjuncts, and Adapters

PagePal

PagePal is an enhancement which allows loudspeaker paging. It replaces the Loop-Start Trunk Adapter and the UPAM.

Fax Attendant 2.1.1

Fax Attendant Release 2.1.1, which co-resides with AUDIX Voice Power on the IS III Release 1.2 platform, provides the same functionality as earlier versions, plus the following enhancements:

— **Personal Fax Messaging.** Inbound faxes can be stored until the subscriber asks that they be printed, at any fax machine he or she specifies, on company premises or off-site (when the subscriber retrieves fax messages remotely).

- **Fax Mail.** Allows subscribers to send fax messages, get fax messages, record personal greetings, program outcalling (stand-alone configuration with AVP only: create fax distribution lists, change account passwords, deliver report settings, and autoprint setting).
- **Fax Broadcast.** Provides a simple way to send one fax to as many as 1000 fax numbers.

Call Accounting System (CAS) for Windows

This stand-alone version of CAS takes advantage of the easy-to-use graphical environment offered by Microsoft Windows. Through data communications, it also allows one CAS system to serve multiple business sites.

PictureTel Video Conferencing

PictureTel's System 4000™ video conferencing system is supported over DS1 (Digital Signal Level 1) facilities with PRI (Primary Rate Interface).

Hardware Components

2

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Hardware Components

2

This chapter describes the system's basic hardware. It includes descriptions of the control unit, MLX telephones, analog multiline telephones, single-line telephones, system operator consoles, adjuncts, and adapters for system telephones, and power-related accessories.

Control Unit

The control unit connects telephone company line/trunk to telephones and adjuncts (such as answering machines and fax machines). It includes the following components:

- Carriers (up to three, at least one required)
- Processor module (one required per system)
- Power supply module (one required per carrier)
- Line/trunk and extension modules (up to 17 in three carriers)
- Cover that protects the unit

Carriers

The basic carrier has seven slots, holding these modules:

- Power supply module
- Processor module (slot 00)
- Up to five line/trunk and extension modules (slots 01–05)

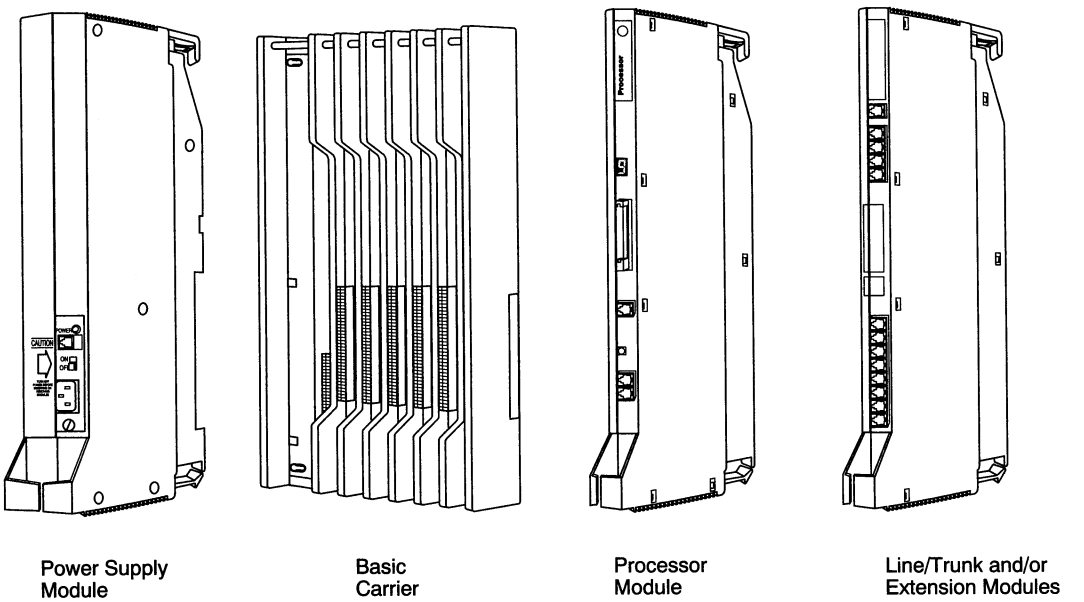

Figure 2–1. Release 3.0 Control Unit Required Components

To increase trunk and/or extension capacity, you can connect up to two expansion carriers to the basic carrier. Like the basic carrier, the expansion carrier's leftmost slot holds a power supply module; the remaining six slots hold line/trunk and extension modules.

The basic and expansion carriers have a *backplane*, which uses an input/output (I/O) bus to interface to the system modules. Figure 2–2 shows both carriers.

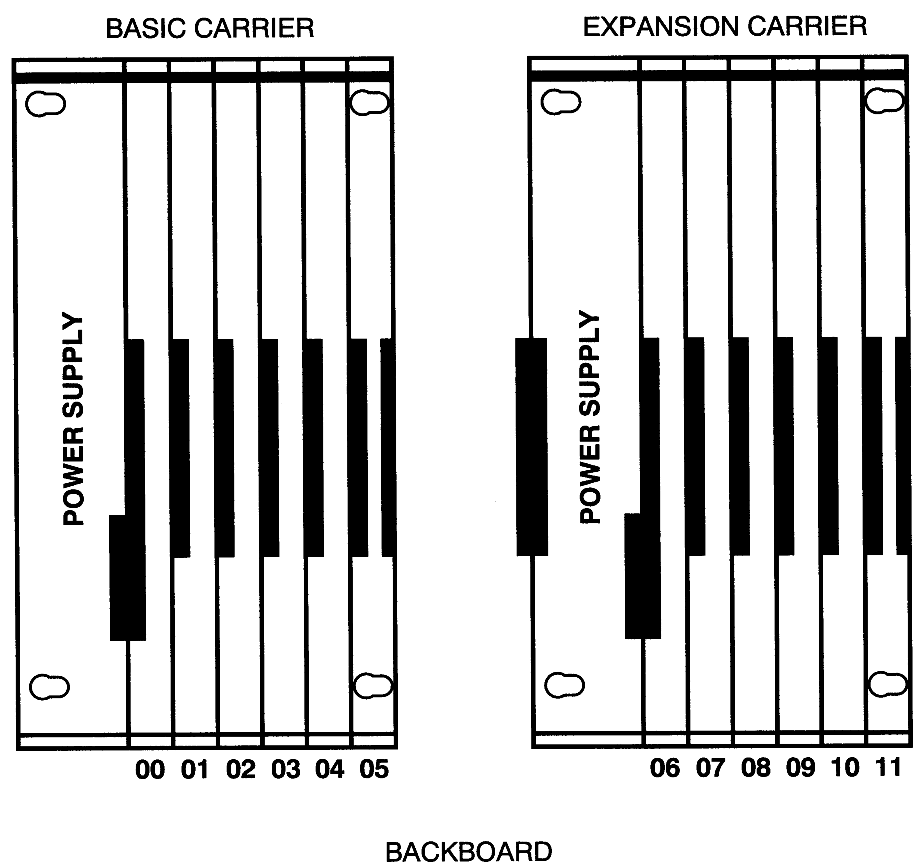

Figure 2–2. Carriers

Processor Module

The processor module is the “brains” of the system, a miniature computer that controls the traffic among the modules, as well as system features and the system’s diagnostics. The processor module provides three jacks, one for Station Message Detail Recording (SMDR), one for system programming and maintenance using a PC, and one for software maintenance by AT&T technicians only. Also, a Personal Computer Memory Card International Association (PCMCIA) interface slot provides access for a PCMCIA card. A PCMCIA card, functionally very similar to a floppy diskette used in a PC, can store backup programming, retrieve backup programming, or upgrade the system.

Release 3.0 Enhancements

Release 3.0 includes the following changes to the processor module:

- Eliminates the feature module and provides a PCMCIA (Personal Computer Memory Card International Association) memory card interface slot. There are three types of PCMCIA cards (each of which is about the size of a credit card) used by customers and field technicians.
  - **Forced Installation.** This card is used for initial installation of system features as well as reinstalling system software if the system has been corrupted.
  - **Translation.** The backup and restore procedures available through SPM can be performed using this card, as an alternative to SPM. These procedures take less time when the PCMCIA card is used.
  - **Upgrade.** This card is used for upgrading the system and upgrading the 800 GS/LS-ID module (if it exists on your system).
- 4 MB of on-board flash ROM (replacing 1.5 MB of EPROM in earlier releases). Flash ROM is a storage medium that acts as read-only memory (ROM), but is capable of being erased and written over multiple times. The PCMCIA card reads the system codes and features onto the Flash ROM for storage. Flash ROM technology makes it easier to perform functions such as initial feature installation, upgrades, and maintenance.
- Replaces the 10-MHz, 16-bit, 68000 microprocessor with a 16-MHz, 32-bit 68EC020 microprocessor.
- Increases RAM from 890 KB to 1.5 MB, nonvolatile and battery-backed.
- Replaces the 1200-bps modem with a full-duplex, 1200/2400-bps modem.
- Adds two (for a total of four) EIA ports for use with SMDR, programming, software maintenance by qualified technicians, and the internal modem. (The port for the modem has no external interface.)
- Adds an Error/Status code display for diagnostic and status conditions.

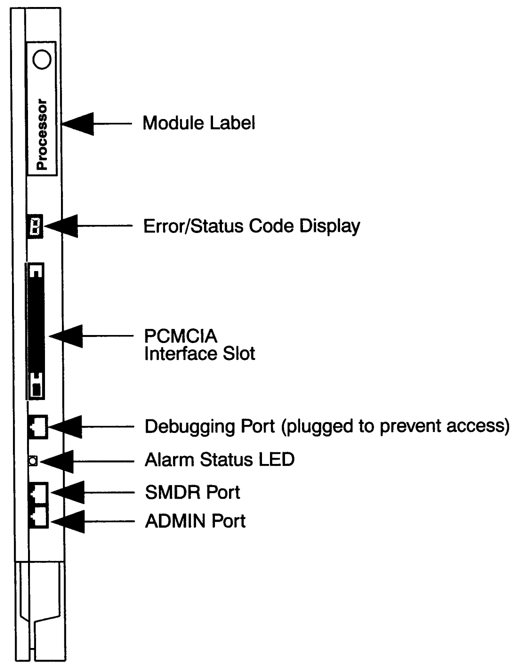

Figure 2–3. Release 3.0 Processor Module

A super capacitor in the processor module provides backup power for up to 5 days for translations in case of a power failure or system shutdown.

Power Supply Module

The power supply module provides power to the carrier, to each telephone, and to adjuncts (except for adjuncts that come with their own power supply, such as answering machines and fax machines). Each carrier requires its own power supply module, which is installed in the leftmost slot of the carrier.

The power supply module converts 117-VAC line voltage to these outputs:

- **+5 VDC and -5 VDC.** All modules use +5 VDC and -5 VDC for logic circuits.
- **-48 VDC.** Most line/trunk and extension modules use -48 VDC for power to the extensions. The Direct Inward Dialing (DID) and Off-Premises Telephone (OPT) line/trunk and extension modules also provide -48 VDC on the tip/ring (T/R) interface to the telephone company or OPT extension.

The 012 basic telephone module provides 21 VDC (battery-powered) to single-line telephones and equipment.

If the system contains an old 012 (PEC 61487) or 008 OPT module (Apparatus Code 517C28 or earlier), you may need to install a 129B frequency generator (or ring generator) in the power supply module of each carrier that houses one or more of these modules. However, if you have one of the newer 012 (PEC 61494) (factory-labeled “with RING GEN.”) or 008 OPT modules (factory-labeled “with RING GEN.”) available beginning with Release 2.1, installing a ring generator is not necessary. Beginning with Release 2.1, a 012 or 008 OPT module includes a built-in ring generator. These modules are compatible with all releases of the communications system and can be added to existing systems with an 012 module and ring generator already in place.

Beginning with Release 2.1, each power supply module comes with ferrite cores that must be installed around the AC power cord and ground wire to comply with FCC requirements.

A green light-emitting diode (LED) on the power supply module remains on as long as the module is receiving power. The power supply module also has an on/off switch and a modular telephone jack for connecting an auxiliary power unit as needed (see Figure 2–4).

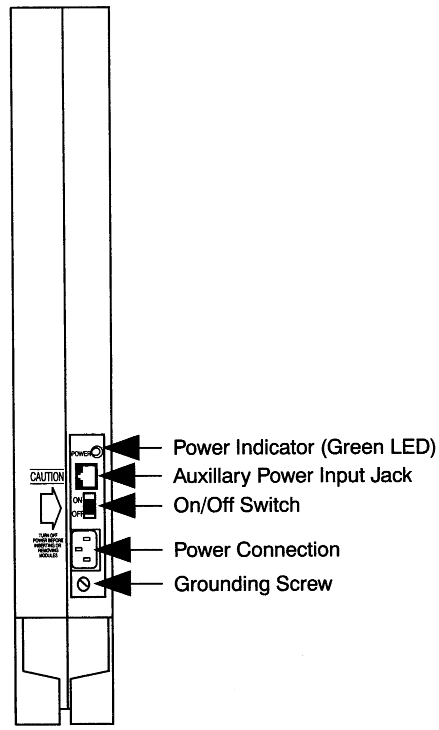

Figure 2–4. Power Supply

Line/Trunk and Extension Modules

Line/trunk and extension modules have jacks for connecting telephone company line/trunk and extension wires to the control unit. Extension wires connect to individual telephones and to adjuncts such as answering machines and fax machines.

A basic carrier has five slots for line/trunk and extension modules. Each expansion carrier provides six additional slots for line/trunk and extension modules.

Each line/trunk and extension module supports a certain type of telephone company line or trunk. Supported types are listed below; for more information, see Chapter 3, "Lines and Trunks."

- **Loop-Start** (Incoming and Outgoing Calls). This trunk is the simplest and most common in the nationwide telephone network. It provides incoming and outgoing calls and is intended primarily for single-line telephones and older Private Branch Exchanges (PBXs).
- **Ground-Start** (Incoming and Outgoing Calls). AT&T introduced this type of trunk specifically to solve the problems that PBXs encountered on loop-start trunks. Ground-start trunks provide an immediate signal when the trunk is seized and when the call is completed and disconnected. You can use a ground-start trunk only if it is registered with the FCC (see "FCC Registration" in Appendix A and Chapter 1 for details).
- **Tie.** This type of trunk directly connects two communications systems. Thus, a caller on one system can call an extension on another system by dialing an access code and the extension number or by simply selecting the line. In more complex tie-trunk configurations, people can access a facility on the other system that does not exist on their own system.
- **Direct Inward Dialing** (Incoming Calls Only). This type of trunk provides fast access to people at specific extensions. The system routes an incoming call on a DID (Direct Inward Dialing) trunk directly to the called extension, a calling group, or an outgoing trunk without system operator assistance. You must coordinate the installation and maintenance of DID trunks with the telephone company.
- **Digital Signal Level 1 (DS1).**Programmed for either T1 or Primary Rate Interface (PRI) operation (incoming and outgoing calls), DS1 provides two-way connection and high-speed transmission of analog and digital signals simultaneously. T1 operation enables the system to transmit and receive analog voice and analog signals; the PRI service arrangement enables the system to transmit and receive digital voice, analog, and digital data signals. PRI provides a wide range of benefits not available on any other single trunk; for example, access to services on the channels can be on a call-by-call basis, with the system selecting the most efficient or cost-effective channel for that call. Beginning with Release 2.1, the system supports direct PRI connections to DEFINITY systems (Generic 1.1, 2.1, 3i, 3r, and 3s).

The system supports 14 types of line/trunk and extension modules. Figure 2–5 shows each module; Table 2–1 lists the number of available jacks for each module. The module's name is a number that identifies the module's connectivity and port capacities. The first digit always describes the number of line/trunk jacks supported, and the next two digits always describe the number of extension jacks supported. For example, the 408 GS/LS module provides four line/trunk jacks supporting ground-start or loop-start trunks and eight extension jacks; the 012 module supports 12 tip/ring connections.

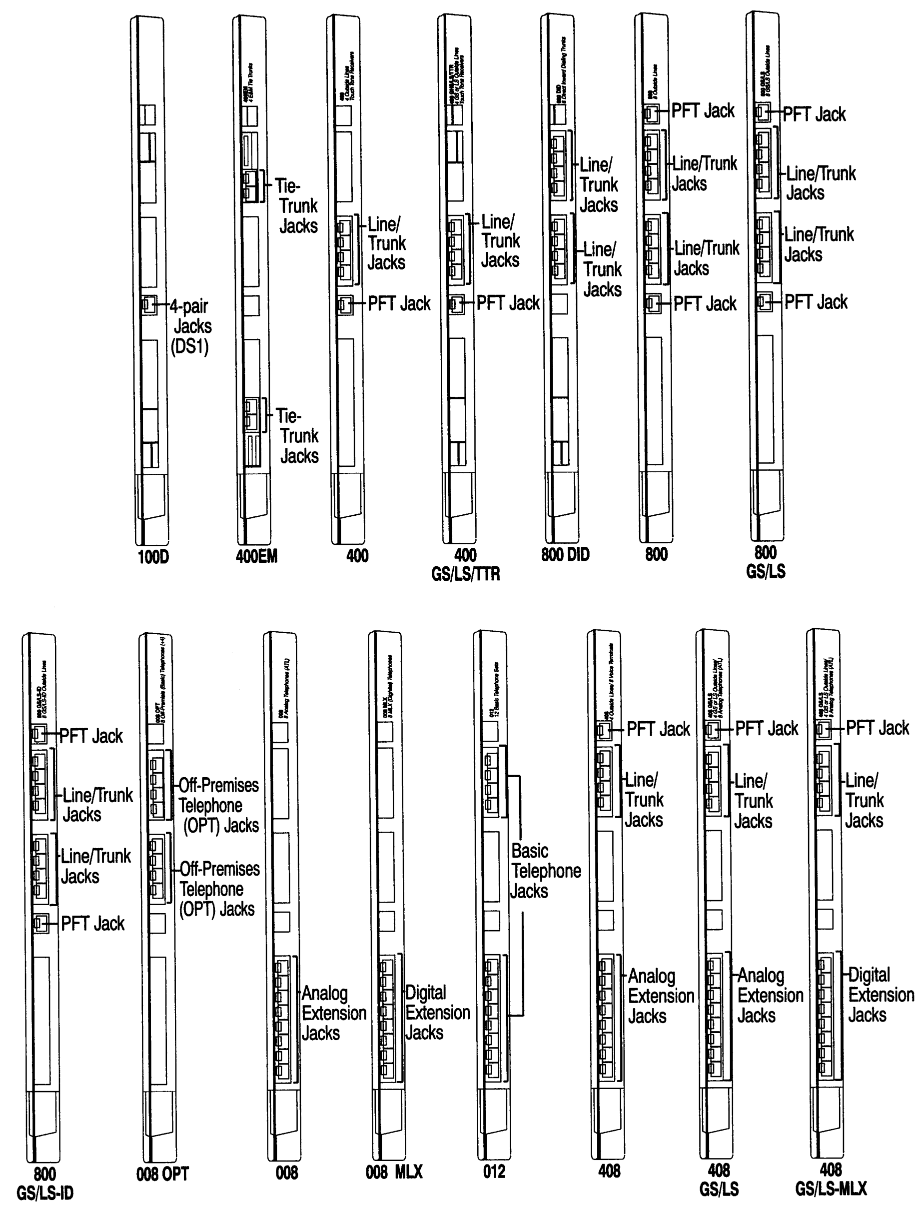

Figure 2–5. Line/Trunk and Extension Modules

Table 2–1. Line/Trunk and Extension Modules

Module	Lines/Trunks	Extension Type	Specifications
008	N/A	Analog multiline telephone; Call Management System (CMS)	**Capacity:** 8 analog extensions, no TTRs **Signaling:** Analog multiline telephone protocol (40 kbps) **Loop range:** 1000 ft. (305 m). For In-range Out-of-Building (IROB) service, use analog IROB protectors, over 1000 ft. (305 m), use local power adapter up to 200 ft (610m).
008 MLX	N/A	MLX telephone; digital data device (such as 7500B data module)	**Capacity:** No TTRs, 8 digital extension jacks, each with 1 or 2 extensions (each extension is assigned an individual extension number), including the following extension types: - MLX voice only - MLX voice and 7500B data module - MLX voice and Multi-Function Module (MFM) - 7500B data module only **Signaling:** BRI S protocol (two 64-kbps B-channels, one 16-kbps D-channel) on a passive bus **Power:** 48 VDC phantom power to telephone; 48 VDC over a separate pair (7–8) to a Direct Station Selector (DSS) console **Loop range:** 1000 ft. (305 m). For In-range Out-of-Building service, use MLX IROB protectors.
008 OPT*	N/A	On-premises or off-premises single-line telephone	**Capacity:** 8 T/R extensions on 2-way voice transmission path with support for telephones with message-waiting LEDs, 2 TTRs. **Notice to telephone company:** Meets FCC Class C **Ringing current:** 105-Vrms, 30-Hz sinusoidal ringing superimposed on -48 VDC; a ring generator must be installed in the power supply of each carrier that has a 008 OPT module, unless the module is labeled "with RING GEN." (Apparatus code 517C28 or earlier 008 OPT modules do not have ring generators.) **REN:** < 1.0 per port **Disconnect signal:** 900 ms (T/R short for answering machines, Group III fax, etc.) **Switchhook flash detection:** 300–1200 ms **Loop resistance:** Serves 2-wire loops to 1300 ohms, including extensions.

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Table 2–1, *Continued*

Module	Lines/Trunks	Extension Type	Specifications
012	N/A	Single-line telephone; AT&T Attendant; MERLIN MAIL Voice Messaging System; T/R adjunct (such as answering or fax machine); analog data device (such as modem)	**Capacity:** 12 T/R extensions on 2-way voice transmission path with support for telephones with message-waiting LEDs, 2 TTRs. **Power:** 21 VDC, 600-ohm battery source **Ringing current:** 105-Vrms, 30-Hz sinusoidal ringing superimposed on -48 VDC; a ring generator must be installed in the power supply module of each carrier that has a 012 module with an apparatus code of 517F13 or lower letter. Current 012 modules [apparatus code 517G13 (28) and higher letters] have a built-in ring generator. **REN:** < 2.2 per port **Disconnect Signal:** 900 ms (T/R short for answering machines, Group III fax, etc.) **Switchhook flash detection:** 300–1200 ms **Loop range:** 1000 ft. (305 m), in-building only.
100D	T1 or PRI		**Capacity:** 24 lines/trunks for voice and analog data (T1) or 23 lines/trunks for voice and data with 1 channel used for signaling (PRI). No TTRs **Mode:** Multiplexes 24 or 23 lines/trunks into one facility and demultiplexes one facility into 23 or 24 lines/trunks **Speed:** Up to 64 kbps per channel (1.544 Mbs total) **Signaling:** DS1 over 4-wire; T1 uses robbed-bit signaling (RBS); PRI has 23 B- and 1 D-channel and uses common channel signaling (CCS).
400†	LS and TTR	Power failure transfer (PFT) telephone	**Capacity:** 4 lines/trunks, 4 TTRs, 1 PFT telephone **Signaling:** Loop-start
400EM	Tie trunk		**Capacity:** 4 tie trunks, no TTRs **Method of Completion:** Automatic or dial-repeating start; immediate start, wink-start, or delay dial-start **Signaling:** E&M type 1S, type 1C, type 5
400 GS/LS/ TTR	LS or GS and TTR	PFT telephone (GS button needed for PFT telephone)	**Capacity:** 4 lines/trunks, 4 TTRs, 1 PFT telephone with ground-start button **Signaling:** Loop-start or GS, optioned per port.

Continued on next page

Table 2–1, *Continued*

Module	Lines/Trunks	Extension Type	Specifications
408†	LS	Analog multiline telephone; CMS; PFT telephone	**Capacity:** 4 lines/trunks, 8 extensions, 1 PFT telephone, no TTRs **Extension Signaling:** Analog multiline telephone (40 kbps) **Lines/trunks signaling:** Loop-start lines/trunks; analog voice **Loop range:** 1000 ft. (305 m). For In-range Out-of-Building service, use analog IROB protectors, over 1000 ft. (305 m), use local power adapter.
408 GS/LS	LS or GS	Analog multiline telephone; CMS; PFT telephone (GS button needed for PFT telephone)	**Capacity:** 4 lines/trunks, 8 extensions, 1 PFT telephone with ground-start button, no TTRs **Extension signaling:** Analog multiline telephone (40 kbps) **Lines/trunks signaling:** Loop-start or ground-start lines/trunks (optional per port), voice **Loop range:** 1000 ft. (305 m). For In-range Out-of-Building service, use analog IROB protectors, over 1000 ft. (305 m), use local power adapter.
408 GS/LS-MLX‡	LS or GS	MLX telephone; digital data device (such as 7500 B Data Module); PFT telephone (GS button needed for PFT telephone) Release 2.0 and later	**Capacity:** 4 lines/trunks, 1 PFT telephone with ground-start button, no TTRs, 8 digital extension jacks, each with 1 or 2 extensions (each extension is assigned an individual extension number), including the following extension types: - MLX voice only - MLX voice and 7500B data module - MLX voice and Multi-Function Module (MFM) - 7500B data module only **Signaling:** BRI S protocol (two 64-kbps B-channels, one 16-kbps D-channel) on a passive bus **Power:** 48 VDC phantom power to telephone; 48 VDC over a separate pair (7–8) to a DSS console **Loop range:** 1000 ft. (305 m). For In-range Out-of-Building service, use MLX IROB protectors.

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Table 2–1, *Continued*

Module	Lines/Trunks	Extension Type	Specifications
800†	LS	PFT telephone	**Capacity:** 8 lines/trunks, 2 PFT telephones, no TTRs **Signaling:** Loop-start
800	DID		**Capacity:** 8 lines/trunks, 2 TTRs **Protocol:** Incoming calls only; 2-wire (one pair) fixed impedance to DID trunks; no outgoing calls (no 2-way DID). **Signaling:** Loop-reverse battery; wink-start or immediate-start; accepts touch-tone dialing
800 GS/LS	LS or GS	PFT telephone (GS button needed for PFT telephone)	**Capacity:** 8 lines/trunks, 2 PFT telephones with ground-start buttons, no TTRs **Signaling:** Loop-start or ground-start
800 GS/LS-ID§	LS or GS	Calling number identification on MLX display telephones over loop-start lines/trunks only	**Capacity:** 8 lines/trunks, 2 PFT telephones with ground-start buttons; 2 TTRs **Signaling:** Loop-start or ground-start **Protocol:** Requires CLASS calling number identification feature from CO.

* The system software recognizes the OPT module as an 012 module. Even though the OPT module only has 8 jacks, it uses 12 ports of capacity, thereby decreasing overall extension capacity by 4 extensions for every OPT module.

† Although these MERLIN II modules are supported, the following are recommended modules for the system: 400 GS/LS, 408 GS/LS, 408 GS/LS-MLX.

‡ This module is not compatible with releases prior to Release 2.0.

§ This module is not compatible with releases prior to Release 3.0.

408 GS/LS-MLX Module

The 408 GS/LS-MLX module (Release 2.0 and later only) is similar to the 408 GS/LS module in providing four trunk and eight MLX extension jacks. MLX port operation is the same as the 008 MLX module; GS/LS jack operation is the same as the 400 GS/LS module. The first line/trunk jack on each module provides power-failure transfer (PFT) functionality. The module does not provide touch-tone receivers.

You can replace a 400 GS/LS and a 008 MLX module with one 408 GS/LS-MLX module to obtain slot savings. You can install 408 GS/LS-MLX modules in *any* of the 17 slots.

IMPORTANT:
The 408 GS/LS-MLX module is not compatible with releases prior to Release 2.0.

012 (T/R) Module

Release 2.1 provides a new 012 (T/R) module for the system. This module contains a built-in ring generator. The 012 module can ring four ports simultaneously.

If you are using 012 ports for an automated attendant or voice messaging system that requires 8 ports, do not use any additional ports on the module. If fewer than 8 ports are required for the application, you can use the remaining ports. If the module is used for other tip/ring devices that ring as often as normal telephones, you can use all 12 ports. Each device must have a REN (ring equivalency number) equal to or less than 2.2.

New installations with the 012 module [apparatus code 517G13(28) or higher letter] do not need the 129B ring generator installed in the power supply module.

The new 012 module is compatible with all releases of the MERLIN II and MERLIN LEGEND Communications Systems and can be added to existing systems that have an earlier 012 module and ring generator already in place.

800 GS/LS-ID Module

Release 3.0 (and later) provides an 800 GS/LS-ID circuit module capable of providing eight analog ground-start or loop-start jacks assigned on a jack-by-jack basis so that both loop-start and ground-start trunks can use the module. Caller identification information (a Custom Local Area Signaling Service available from COs in some areas) is available on loop-start trunks only. The caller's number is available at extensions equipped with MLX display telephones. Analog display telephones cannot display caller information.

The 800 GS/LS-ID port module has the same programming options as the 800 GS/LS-ID port module, plus the LS-ID Delay option, programmable on a per-trunk basis. The LS-ID Delay option can avoid the loss of caller ID information when certain features are activated (such as Auto-Answer). Any application or adjunct that answers a loop-start line on the first ring causes the loss of caller ID information, unless the application or adjunct is set to answer the call on the second or later ring or the LS-ID Delay option is activated.

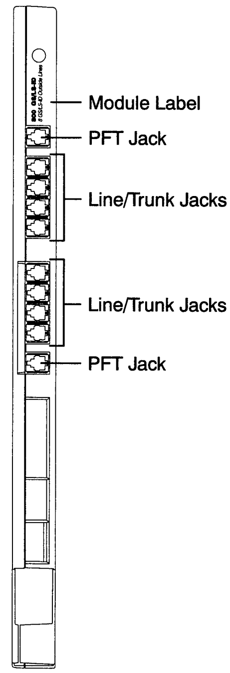

Figure 2–6. 800 GS/LS-ID module

The 800 GS/LS-ID module's firmware and initialization data stored in the port processor flash and boot ROM can be upgraded using a forced installation PCMCIA card inserted into the processor module by an AT&T technician.

The module also provides two power-failure transfer (PFT) ports, which enable incoming and outgoing service through a single-line telephone during a commercial power failure.

Additionally, the 800 GS/LS-ID module provides two touch-tone receivers (TTRs), which are used as system resources.

IMPORTANT:
The 800 GS/LS-ID module is compatible with Release 3.0 and later and not with earlier releases of the system.

Cover

The control unit cover is a plastic cabinet that protects the equipment. The size of the cover accommodates the expansion carriers in addition to the basic carrier. Figure 2–7 shows how the cover fits around the control unit. Release 2.1 and later systems have a cover identical to the MERLIN II cover.

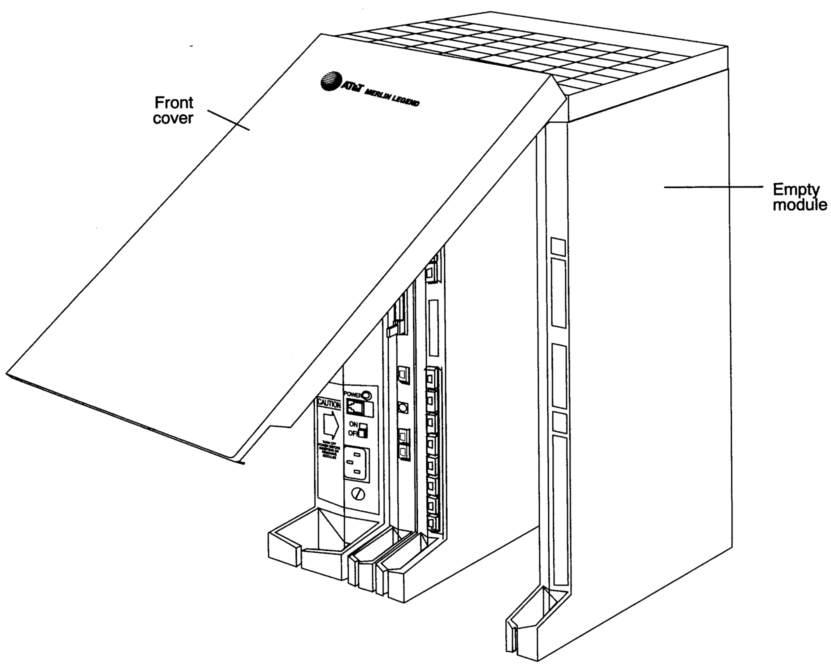

Figure 2–7. Control Unit Cover

MERLIN II Reusable Equipment

You can use the following MERLIN II Release 3 modules in this system:

- 391A power supply module
- 800 line/trunk module
- 400 line/trunk module
- 400 E&M line/trunk module

- 012 basic telephone module
- 008 analog extension module
- 408 analog line/trunk and extension module

NOTE:
While you can reuse the 391A power supply module in this system, it *does not* supply as much power as the 391A1 power supply module. You should replace the 391A if you add extensions to the system because the 391A also has less protection against power surges than the 391A1. Also, under certain circumstances, electromagnetic interference (EMI) can interfere with hand-held transceivers. In such cases, the 391A2 power supply is recommended, because it is equipped with a copper shield and has ferrite cores on the AC power leads.

Table 2–2 lists reusable MERLIN II modules and their apparatus codes. Table 2–3 lists reusable MERLIN II hardware and associated apparatus codes or price element codes (PECs).

Table 2–2. Reusable MERLIN II Modules

Type	Apparatus Code	Comments
008	517A3	Fully compatible
	517B3	Fully compatible
012	517A13	Compatible but does not support the downlink disconnect needed for voice mail; does not meet Megacom® transmission requirements.
	517B13	Compatible but does not support the downlink disconnect needed for voice mail; does not meet Megacom transmission requirements.
	517C13	Compatible but can be used for Megacom only when the customer does not have to meet EIA transmission standards.
	517D13	Compatible but can be used for Megacom only when the customer does not have to meet EIA transmission standards.
	517E13	Fully compatible
100D	517A15	Supports only tie-trunk emulation
	517B15	Fully compatible

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Table 2–2, *Continued*

Type	Apparatus Code	Comments
400	517A12	No lightning protection; 146A surge protector required. Fully compatible
	517B12	
400EM	517A14	Fully compatible
408	517A1	No lightning protection; 146A surge protector required. Fully compatible
	517B1	Fully compatible
	517C1	
800	517A4	No lightning protection; 146A surge protector required. Fully compatible
	517B4	

Table 2–3. Reusable MERLIN II Hardware

Type	Apparatus Code or PEC	Comments
Power supply module	391A	No surge protection; 147A protector recommended.
	391AA	For Canadian use only; no auxiliary power jack.
	391A1	Fully compatible
Basic carrier	403A	Compatible but must order system cover separately (part 16A); required spring clips for the system cover are provided with the upgrade package.
	403C	For Canadian use only; must order system cover separately (part 16A); required spring clips for the system cover are provided with the upgrade package.
	403E	Fully compatible

Continued on next page

Table 2–3, *Continued*

Type	Apparatus Code or PEC	Comments
Expansion carrier	403B	Compatible but must order system cover separately (part 17A).
	403D	For Canadian use only; must order system cover separately (part 17A); required spring clips for the system cover are provided with the upgrade package.
	403F	Fully compatible
Ring generator	129B	Fully compatible
Auxiliary power	335A	Compatible but can be used only when the unit loads do not exceed the 335A's capacity; an Auxiliary Power Unit 9024 is recommended.
	9024	Fully compatible
Music coupler	61398	Fully compatible

Telephones

In addition to MLX (digital) telephones, you can use several analog and single-line telephones with the system.

NOTE:
An MLX, analog, or multiline telephone located in a different building but within 1000 ft. (305 m) of the control unit requires an In-Range Out-of-Building (IROB) protector at each building entrance. If a single-line telephone is located in a different building from the control unit or is in excess of 1000 ft. (305 m), you must use either an Off-Premises Range Extender (OPRE) or a 008 off-premises telephone (OPT) module.

MLX Telephones

The communications system supports the following MLX telephones:

- MLX-20L
- MLX-28D
- MLX-10D
- MLX-10
- MLX-10DP

MLX telephones are available in black or white with factory-set buttons in English, French, or Spanish. (The MLX-10DP is available with English-language buttons only.) In addition, all models have the following features in common:

- Line buttons programmable with features, with red and green lights
- Fixed-feature buttons (including **Feature**, **HFAI**, **Mute**, and **Speaker**)
- Red Message light
- Built-in speakerphone
- Support PRI services
- Separate volume controls for speakerphone, handset, and ringer
- Telephone card tray for easy reference to frequently used features
- Two-position adjustable desk stand
- Four-pair modular line cord
- Optional MFM to connect T/R equipment and alerting devices

NOTE:
An MLX-20L telephone used as a Queued Call Console (QCC) has no programmable buttons. Additionally, it cannot have a Multi-Function Module (MFM).

MLX display telephones have the following two additional features:

- Large display
- Display-associated buttons

A list of features specific to each MLX telephone follows.

MLX-28D

The MLX-28D telephone provides the following features:

- Direct-Line Console (DLC) operation
- Display (2 lines by 24 characters)
- 28 line buttons
- 8 fixed display buttons and 8 fixed-feature buttons
- Support for one or two Direct Station Selectors (DSSs)
- Support for PassageWay Direct Connect Solution

MLX-28D telephones cannot be wall-mounted.

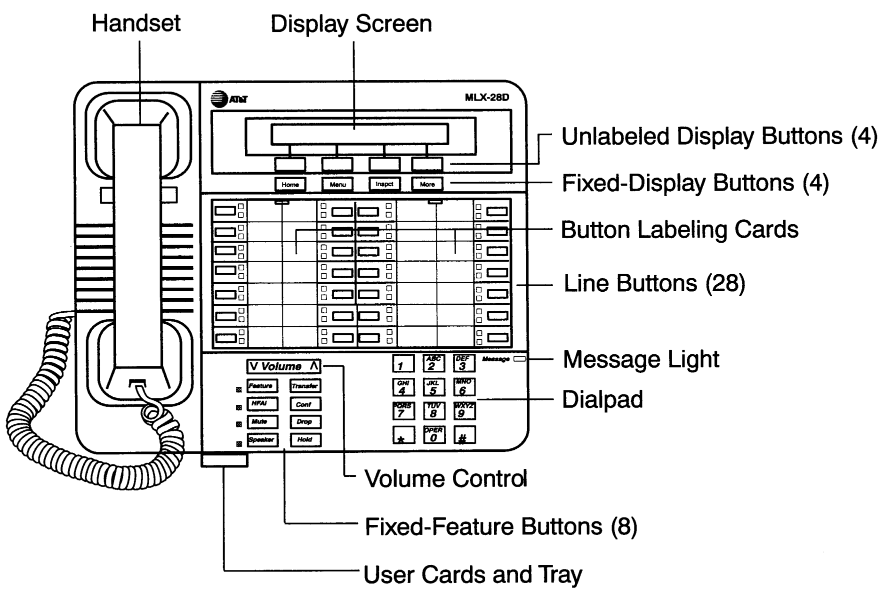


Figure 2–8. MLX-28D Telephone

MLX-20L

The MLX-20L telephone provides the following features:

- System programming and DLC or QCC operation
- Display (7 lines by 24 characters)
- 20 line buttons
- Display buttons and 8 fixed-feature buttons
- Support for one or two DSSs
- Support for Passageway Direct Connect Solution

MLX-20L telephones cannot be wall-mounted.

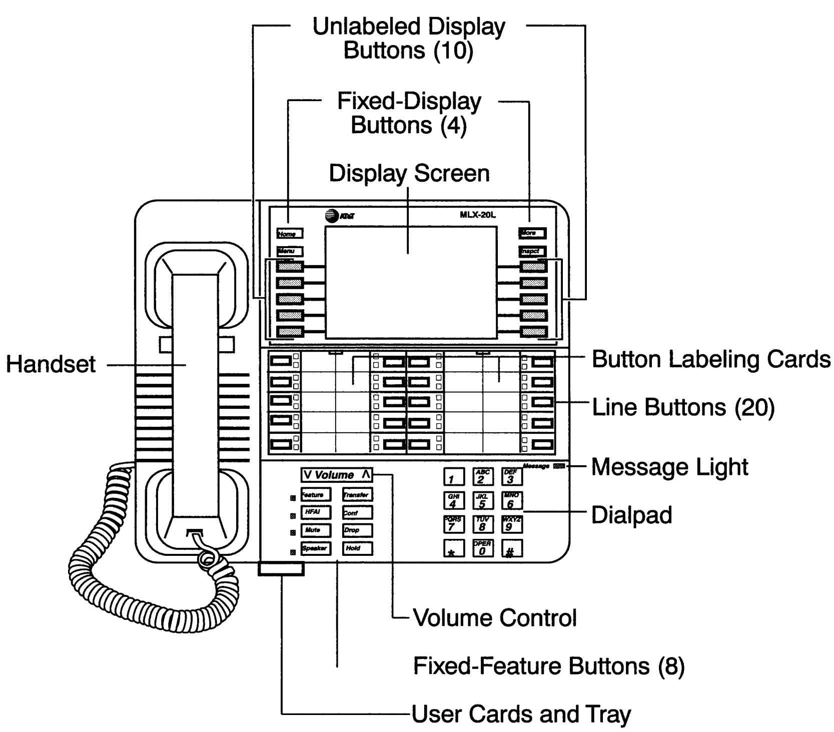


Figure 2–9. MLX-20L Telephone

MLX-10D

The MLX-10D telephone provides the following features:

- Display (2 lines by 24 characters)
- 10 line buttons
- 8 fixed display buttons and 8 fixed-feature buttons

MLX-10D telephones can be wall-mounted, but doing so makes the display hard to read.

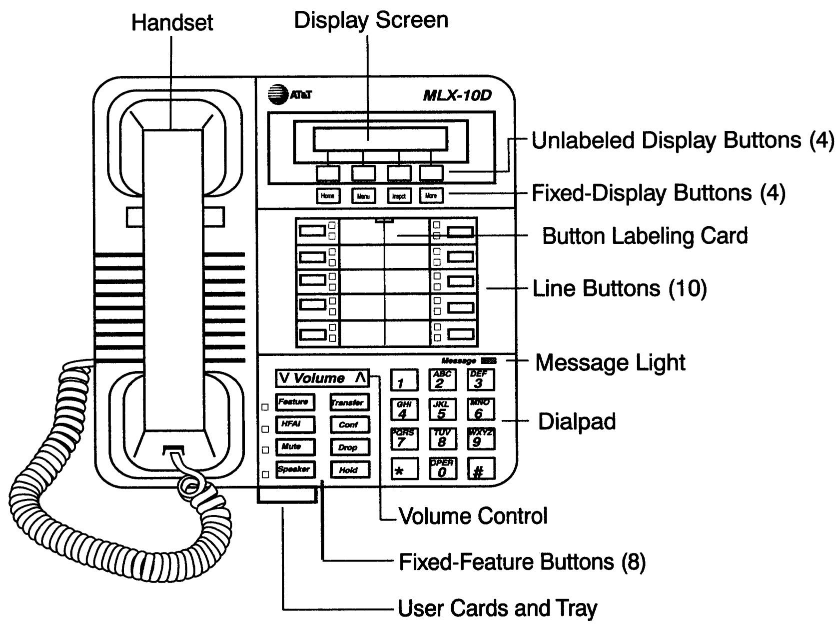


Figure 2–10. MLX-10D Telephone

MLX-10DP

The MLX-10DP telephone is the same as the MLX-10D telephone, except that the MLX-10DP has an adjunct jack in the back of it for connecting the PassageWay Direct Connect Solution application.

MLX-10

The MLX-10 telephone provides the following features:

- 10 line buttons
- 8 fixed-feature buttons

MLX-10 telephones can be wall-mounted.

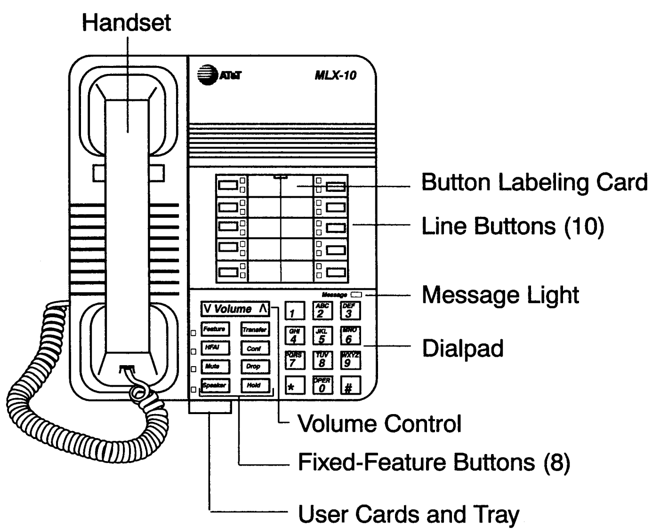


Figure 2–11. MLX-10 Telephone

Direct Station Selector

The Direct Station Selector (DSS), shown in Figure 2–12, is an adjunct that you can connect to an MLX-20L or an MLX-28D telephone programmed as an operator console (it cannot connect to any other telephone). DSSs enhance the capabilities of both DLCs and QCCs and, when connected to an MLX-20L telephone, facilitate programming. The DSS has 50 buttons, all of which have lights.

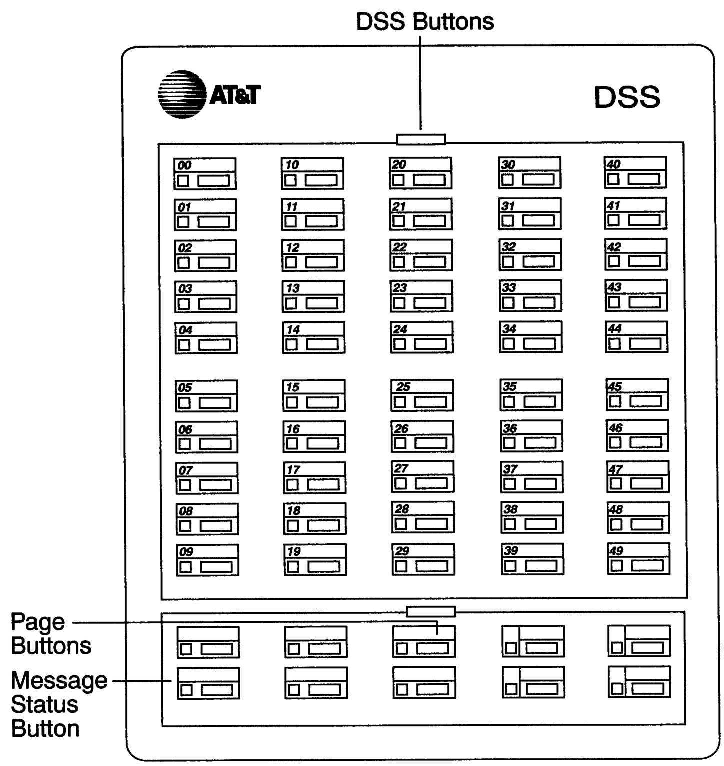


Figure 2–12. Direct Station Selector

The system operator can use DSS buttons for one-touch dialing and Transfer. You can program the buttons with the following numbers:

- Extension numbers
- Line/trunk numbers
- Pool dial-out codes
- Calling group extension numbers
- Paging group extension numbers
- Park zone access codes
- Automatic Route Selection (ARS) access codes
- Remote Access dial code
- Listed Directory Number (the extension for the QCC queue)

Ten fixed-feature buttons with green LEDs are at the bottom of the DSS. The first three (from left to right) on the top row are **Page** buttons, which you use to select the range of extension numbers represented by the **DSS** buttons. Each DSS button can represent up to 3 extension numbers, and each set of 50 extension numbers is called a *page*. The DSS can have up to 3 pages of numbers, for a total of 150 extensions handled by one DSS.

Page buttons act like the Shift key on a PC or typewriter. Each of the 3 **Page** buttons activates a set of 50 numbers. For example, the **Page 1** button may access extensions 1–50, the **Page 2** button may access extensions 51–100, and the **Page 3** button may access extensions 101–150.

If two DSSs are connected together, the total extension capacity of the console increases to 300. Each **Page** button then handles a range of 100 extensions. See "Telephone Power Units" and Table 2–9 for more information about connecting DSSs and consoles.

A fourth button (lower leftmost) is the **Message Status** button, which changes the mode to and from message status operation. The remaining six buttons in the bottom two rows are not used.

NOTE:
DSSs are shipped without auxiliary power supplies; if two DSSs are connected to a console, you must order auxiliary power (329A unit) separately. Also, if a DSS is out of the building, you need to order an auxiliary power unit.

Analog Multiline Telephones

In addition to MLX telephones, the system supports the analog multiline telephones listed in Table 2–4.

Table 2–4. Analog Multiline Telephones

Model	Description
5-line button*	5-line button telephone with membrane; no adjuncts supported
10-line button*	10-line button telephone with membrane

* Vintage telephone, no longer available for sale or lease

Continued on next page

Table 2–4, *Continued*

Model	Description
34-line button*	34-line button basic telephone with membrane
34-line button Deluxe*	Deluxe 34-line button telephone with membrane
10-line button HFAI*	10-line button hands-free-answer telephone; no adjuncts supported
34-line button BIS*	34-line button telephone with built-in speakerphones
34-line button BIS/DIS*	34-line button telephone with 16-character display and built-in speakerphone
BIS-10	10-line button telephone with built-in speakerphone
BIS-22	22–line button telephone with built-in speakerphone
BIS-22D	22–line button telephone with 16-character display and built-in speakerphone
BIS-34*	34-line button telephone with built-in speakerphone
BIS-34D	34-line button telephone with 16-character display and built-in speakerphone
MLC-5 Cordless	Cordless 5-line button analog telephone. Cannot use Conference feature
MDC 9000 Cordless	Cordless 6-line button telephone
MDW 9000 Cordless/Wireless	Cordless/wireless 6-line button telephone
MERLIN PFC Telephone	Analog multiline phone, fax machine, and copier; requires two analog multiline extension jacks

* Vintage telephone, no longer available for sale or lease

Single-Line Telephones

Releases prior to Release 3.0 also support the single-line telephones listed in Table 2–5. These telephones can work with Release 3.0, but single-line telephones with positive disconnect are preferable.

Table 2–5. Single-Line Telephones

Model	Description
2500MMGB	Basic desk telephone
2554MMGJ	Basic wall telephone
2500YMGK*	Basic desk telephone with Message light and **Recall** button; **Recall** button is used instead of the switchhook for features that require a switchhook flash, such as Transfer and Hold.
2500SM	Basic desk telephone used with 4A speakerphone
2514BMW	Basic desk telephone with built-in headset jack
2526BMG	Outdoor telephone used with weatherproof enclosure
7101A*	Basic desk telephone with Message light and **Recall** and **Disconnect** buttons. No adjuncts supported.
7102A	Basic desk telephone with Message light lamp and **Recall** button. The 101 and 201 speakerphones and the 500 headsets are supported. Can be used for power-failure transfer (PFT) extensions.
CS6402UO1A*	Basic desk telephone, Feature Phone Model 420. Has built-in speakerphone, memory, and redial.
2500MMGJ	Basic desk telephone
2500MMGK	Basic desk telephone with **Recall** button; **Recall** button is used instead of the switchhook for features that require a switchhook flash, such as Transfer and Hold.
8102†	Basic desk telephone with jack to support headset adapters and speakerphone adjuncts.
8110	Basic desk telephone with a built-in speakerphone with volume control and **Mute** button with LED indicator.

* Vintage telephone, no longer available for sale or lease

† Although the model 8102 can be connected to a speakerphone and the model 8110 has a built-in speakerphone, neither can be used for group paging, which is not supported on single-line telephones. The auto answer function on the model 8110 must be disabled for operation on the system.

Continued on next page

Table 2–5, *Continued*

Model	Description
500MM 554BMPA 500SM	Basic telephone with the following limitation: equipped with rotary dials so no system features requiring * and # can be used. Telephones with neon Message lights are not supported.

* Vintage telephone, no longer available for sale or lease

† Although the model 8102 can be connected to a speakerphone and the model 8110 has a built-in speakerphone, neither can be used for group paging, which is not supported on single-line telephones. The auto answer function on the model 8110 must be disabled for operation on the system.

Single-Line Telephones in Release 3.0 and Later

In Release 3.0 and later, only single-line line telephones with positive disconnect are supported as system telephones (any single-line telephones, equipped with a ground-start button if necessary, can be used as PFT telephones). Table 2–6 lists the single-line telephones supported in Release 3.0 and later.

Table 2–6. Single-Line Telephones Supported in Release 3.0 and Later

Model	Description
2500 YMGL	Basic desk telephone with Message light and **Flash** button; **Flash** button is used instead of the switchhook for features that require a switchhook flash, such as Transfer and Hold.
8101	Basic desk telephone with Message light and **Flash** button; **Flash** button is used instead of the switchhook for features that require a switchhook flash, such as Transfer and Hold.
VideoPhone 2500	Provides interactive, small-screen video when both parties use one
Picasso Still-Image Phone	Allows 2 parties with Picasso Still-Image Phones to transmit and simultaneously discuss full-color images which they view over a customer-supplied TV monitor or flat-panel LCD (liquid crystal display) monitor.

Continued on next page

Table 2–6, *Continued*

Model	Description
3129-WTWA	Touch-tone outoor telephone equipped with cast aluminum housing, armored handset cord with bell ringers
3129-WRWA	Rotary dial outdoor telephone equipped with cast aluminum housing, armored handset cord with bell ringers
3129-WAWA	Auto dial outdoor telephone equipped with cast aluminum housing, armored handset cord with bell ringers
3129-WNWA	Non-dial, automatic ringing on dedicated circuit outdoor telephone equipped with cast aluminum housing, armored handset cord with bell ringers

Telephones and Adjuncts Not Supported

You cannot use the following telephones and adjuncts with the system.

WARNING:
Connecting the adjuncts listed in Table 2–7 can damage the telephones, adjuncts, and system.

Table 2–7. Telephones and Adjuncts Not Supported

Model	Notes
510D Personal Terminal	Uses Digital Communications Protocol (DCP)
DCP telephones	7400 telephones and adjuncts (asynchronous data units and multiple asynchronous data units) that use DCP and are supported on MERLIN II
MET telephones	Multibutton electronic telephones (MET) and adjuncts that are used with the Dimension® PBX and Horizon® communications systems
Single-line telephone with neon Message light	Cannot support voltage required for neon light.
Analog telephone adjuncts	Basic telephone modem interface (BTMI and BTMI-2); Off-premises extension (OPU) unit; System 25 direct extension selector (DXS); DSS attached to a 34-button deluxe membrane

Operator Consoles

Operator consoles are telephones that you program for call handling and other system operator duties. With one exception, these telephones themselves are no different from the ones already described. In most cases, the telephone's *programming* is the only distinction that makes the telephone an operator console. An operator console is available in two configurations: Queued Call Console (QCC) and Direct-Line Console (DLC). QCCs are available only in Hybrid/PBX mode.

A system operating in Hybrid/PBX mode can include both QCCs and DLCs. Table 2–8 shows the maximum number of both types of system operator positions.

Table 2–8. Maximum Number of System Operator Positions

Position Types	Type of Telephone	Maximum Positions
QCC	MLX-20L	4
DLC	MLX-20L MLX-28D	8
DLC	Analog multiline telephones	8
QCC and DLC, or all QCCs, or all DLCs	Any	8 (see note below)

NOTE:
The system cannot have more than eight operator positions of any combination (QCCs and/or DLCs); if you use a combination of consoles, no more than four can be QCCs.

Queued Call Consoles

The Queued Call Console (QCC) is available only in Hybrid/PBX mode. In a QCC configuration, the system holds incoming calls in a queue and directs them to a QCC as a position becomes available. Only one call rings at a time.

The MLX-20L telephone is the only telephone that you can assign as a QCC. The QCC system operator *cannot* use feature codes to activate features; however, the operator can use the features available from the display, as well as any fixed features that have been assigned to the console buttons.

The 7-line, 24-character display also provides the system operator with descriptive information about incoming and outgoing calls. This information includes extension numbers and any programmed labels (such as names), trunk identifiers, reasons for call return and redirection, and the number of unanswered calls waiting in the queue.

QCC Buttons

The system automatically sets the buttons on the QCC with fixed features, and they are not programmable by the system operator or through centralized telephone programming. The QCC has the following fixed-feature buttons:

- **Call.** Five buttons used to answer incoming calls and make inside and outside calls.
- **Start.** Initiates the call-directing process by putting a caller on hold at the **Source** button and providing an internal dial tone to the system operator.
- **Source.** Reconnects the system operator to the original caller before the call is connected to (released to) its destination.
- **Release.** Releases the system operator from a call and/or completes the call-directing process, making the system operator available for another call.
- **Destination.** Reconnects the system operator to the destination before a new call is released to its destination.
- **Cancel.** Cancels call directing and reconnects the system operator with the caller (source).
- **Join.** Connects the system operator with the caller (source) and the person being called (destination) in a three-way conference. All three parties are connected on one **Call** button.
- **Headset Mute** (Headset/Handset Mute). Activates or deactivates the headset or handset microphone.
- **Headset Status.** Activates and deactivates the headset operation of the console.
- **Headset Auto Ans** (Headset Auto Answer). Activates or deactivates the Headset Auto Answer feature when headset operation is activated by pressing the **Headset Status** button.
- **Send/Remove Message.** Turns on the telephone Message LED to indicate a waiting message and turns off the Message LED when all system operator messages are delivered.
- **Position Busy.** Temporarily takes the system operator console out of service.
- **Night Service.** Activates or deactivates Night Service.

- **Alarm.** Provides visible indication of a system alarm. When a system alarm has occurred, the red LED next to the button is on and the system operator can use the Inspect feature to determine the number of alarms present.
- **Pool Status.** Provides the system operator with the status of all trunk pools (a maximum of 11). The information includes the number of trunks and the number of busy trunks in each pool.
- **Forced Release.** Disconnects the system operator from an active call and makes the system operator available to receive another call.

QCC Capacities and Requirements

You can attach one or two DSSs to a QCC. The system operator can use the DSS buttons during call handling, for example, to direct a call, make an inside call, park a call, or see the availability of an extension.

You must assign the following options to a QCC through system programming:

- QCC operator receiving calls
- QCC Queue Priority
- Call Types
- Elevate Priority
- Hold Return
- Automatic Hold or Automatic Release
- Calls-In-Queue Alert
- Queue Over Threshold
- Extended Call Completion
- Position Busy Backup
- Return Ring Interval
- Message Center Operation

The system can have up to eight operator positions of any combination (QCC and/or DLC).

Keep these facts in mind if your system includes QCCs:

- You must connect a QCC to an extension jack on a 008 MLX or 408 GS/LS-MLX module (Release 2.0 and later only).
- Each 008 or 408 GS/LS-MLX module can carry a maximum of two QCCs.
- You must connect the first QCC to the first extension jack in the system.

- You can connect QCCs only to the first and fifth extension jacks on each module.
- You can connect up to four QCCs for the system.

Direct-Line Consoles

On a Direct-Line Console (DLC), outside lines are assigned to individual buttons. The console can have several calls ringing at the same time.

A DLC operates like other multiline telephones. In all three modes of operation (Key, Hybrid/PBX, and Behind Switch), you assign outside lines to individual buttons on the console. You can assign the lines that have been assigned to a DLC to buttons on other consoles or other telephones. Incoming calls can ring on any of the line buttons, and several calls can ring simultaneously. The system operator directs calls to other users by using the **Transfer** button. **Pool** buttons (Hybrid/PBX mode only) cannot be assigned to a DLC.

A multiline telephone that has been assigned as a DLC through system programming can use system operator features as well as the telephone features available for non-operator multiline telephones to increase call-handling efficiency. The system operator features that you can assign to buttons on the console are Alarm, Night Service, Missed Reminder, and Send/Remove Message.

On a system with 29 or fewer lines, Alarm, Night Service, and Send/Remove Message are assigned by default to analog DLCs on buttons 32-34.

On a system with more than 29 lines:

- Line 30 replaces Alarm
- Line 31 replaces Night Service
- Line 32 replaces Send/Remove Message

Lines 1 through 18 are factory-set as personal lines on MLX-28D telephones assigned as DLCs, regardless of the number of line/trunk connected to the system.

You can use the following telephones as DLCs:

- Digital DLC
  - MLX-20L telephone
  - MLX-28D telephone
- Analog DLC
  - MERLIN II System Display Console with built-in DSS (the only telephone model that is uniquely used as an operator console)

— BIS-34D telephone

— BIS-22D telephone

— BIS-34 (no longer sold)

You can add one or two DSSs to the MLX-20L or MLX-28D telephone to provide 150 (3 pages x 50 buttons) or 300 (3 pages x 50 buttons x 2 DSSs) of additional extension buttons. You cannot attach a DSS to an analog DLC; however, the MERLIN II System Display Console provides a built-in DSS, and Auto Dial buttons can be programmed on BIS phones for rapid access to extensions.

Keep these facts in mind if your system includes DLCs:

- You can connect an analog DLC either to an analog extension jack on a 008 or a 408 analog multiline telephone module, or to a digital extension jack on a 008 MLX or 408 GS/LS-MLX module (Release 2.0 and later only).
- When you assign both DLCs and QCCs in Hybrid/PBX mode, the maximum combined number of system operator positions is eight; no more than four can be QCCs. You can assign a maximum of two DLCs per MLX or analog module.
- Only multiline telephones that are connected to the first and fifth extension jacks on digital or analog modules can be assigned as DLCs. This includes DLCs assigned as calling group supervisors and Call Management System (CMS) supervisors.
- You can use an MLX-20L telephone set up as a DLC for system programming if you connect it to the first or fifth extension jack on the first MLX module and then designate that jack for system programming.

Adapters

Adapters connect adjuncts to the system and to telephones; they provide access to both voice and data signals.

System Adapters

Three system adapters connect directly to the control unit: the Loop-Start Trunk Adapter, the PagePal paging access interface, and the Universal Paging Access Module (UPAM). In addition, several different Channel Service Units (CSUs) connect to the control unit.

ACCULINK 3150 Extended SuperFrame T1 Channel Service Unit

A channel service unit (CSU) is an interface between the 100D module and the Digital Signal 1 TS1 facility provided by the telephone company. (This interface is required for DS1 Primary Rate Interface service on a 100D module.) The DS1 facility supports 24 multiplexed DS0 channels on one 4-pair wire that connects to the back of the CSU. The CSU then connects to the modular jack on the 100D module. One type of channel service unit currently available is the ACCULINK 3150 Extended SuperFrame (ESF) T1 shown in Figure 2–13.

This CSU supports both alternate mark inversion (AMI) and bipolar 8 zero-code substitution (B8ZS) line coding; it includes the following features:

- **Menu-Driven Operation from an LCD Display.** The CSU provides a display screen, six LEDs, and a keypad for operation from the front panel.
- **Security.** Passwords can be set to enhance the security of remote operation.
- **Customizable Performance Reporting.** The CSU gathers and reports information about network performance.
- **Flexible Power Sourcing.** The CSU can operate on either a +24 VDC power source, a -48 VDC single-source battery, or -48 VDC redundant power supply for power backup.
- **PC Software for Local or Remote Operations.** As an alternative to front-panel operation, the CSU can be operated by an on-site PC (connected by the COM ports on the CSU and PC) or by a remote PC communicating with the CSU's 2400-bps modem (connected to the phone system by the CSU's MODEM port). To use this software, you must have Windows 3.1 or later and DOS 3.3 or later, running on a 386 or 486 computer. A mouse is also required.

  In addition, you can use the PC software's Passthrough mode to operate a remote CSU either from the local CSU or the screen of your PC.
- **Testing Capabilities.** Self-testing of the CSU, as well as network testing, is supported. Three network-testing jacks are included on the CSU.

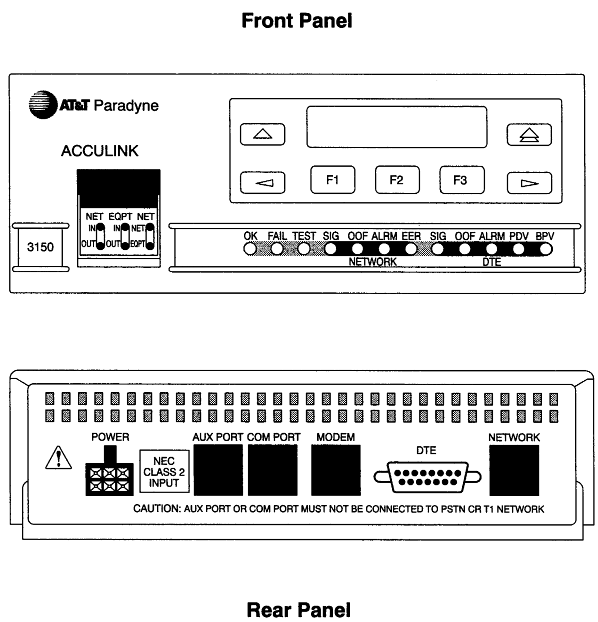


Figure 2–13. ACCULINK 3150 Channel Service Unit Connections

ACCULINK 3160/64 Series Extended SuperFrame T1 Data Service Unit (DSU)/Channel Service Unit

Similar in design to the ACCULINK 3150 CSU, the ACCULINK 3160 and 3164 DSUs/CSUs act as interfaces between the T1 network and the MERLIN LEGEND Communications System 100D module in the control unit. The 3160 and 3164 provides two and four ports respectively, which allow for individual channels on a T1 facility to be directed to data equipment. They also provide the same functionality of the 3150 CSU for connecting a T1 facility to the control unit.

NOTES:

1. The ESF T1 CSU (PEC 2152-ES2), still supported but no longer available for sale or lease, allows the unit to be maintained without interrupting service and provides diagnostic and testing capabilities; it also enables bipolar 8 zero-code substitution (B8ZS) line coding.

2. The 551 T1 LI CSU (PEC 2152-15T), still supported but no longer available for sale or lease, does not provide the B8ZS line coding required for 64-kbps data and for maintenance features. The 551 T1 LI CSU does not provide diagnostic and testing capabilities and is not recommended for video conferencing or other Primary Rate Interface (PRI) applications.

Telephone Adapters

The adapters described below connect adjuncts to telephones.

Multi-Function Module (MFM)

The Multi-Function Module (Figure 2–14) lets you connect tip/ring (T/R) or supplemental alert adjuncts to an MLX telephone. The MFM is a circuit board that mounts inside the telephone. Adjuncts plug into a modular jack on the MFM. The MFM is the only T/R adapter used with MLX telephones. You cannot install an MFM in an MLX-20L telephone that is set up for QCC operation.

WARNING:

Only a qualified technician can install or repair an MFM. To eliminate the risk of electrical shock, do not disassemble the MLX telephone.

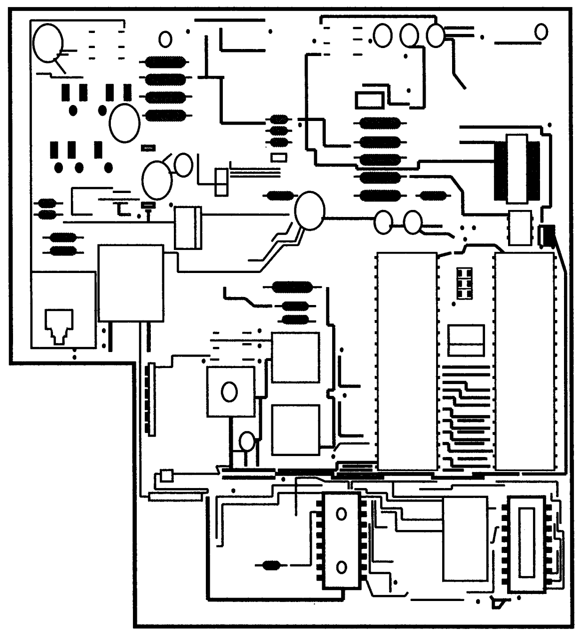


Figure 2–14. Multi-Function Module (MFM)

T/R adjuncts operate independently of the MLX telephone. If the telephone is in use, the adjunct can send and receive voice or data calls. An MFM accommodates the following T/R adjuncts:

- Answering machines
- Fax machines
- Modems
- Credit card verification terminals
- Single-line touch-tone telephones
- Supplemental alerts (bells, chimes, horns, and strobes)

The MFM is shipped with a KS22911-L1 power supply that supports one MFM and one DSS. When you connect two DSSs to a telephone, use a 329A power supply. With either type of power unit, the total cord length cannot exceed 50 ft. (15 m) from the telephone.

The MFM supports only touch-tone dialing and does not detect pulse dialing.

NOTES:

1. The MFM uses one of the two channels when it is active. A channel is used to carry voice or data content of a call between the system and the extension switch. This means you cannot use Voice Announce to Busy and Speakerphone Paging when an adjunct (such as a fax machine) and an MLX telephone are in use at the same time. When Voice Announce to Busy is in use, a person calling an MFM extension gets a busy signal; a person attempting to call out from an extension with an MFM does not get a dial tone.
2. The MFM does not support features activated by a switchhook flash. This means that you cannot transfer or conference calls from an MFM extension.

Two jumper blocks on the MFM let you configure it for either tip/ring (T/R) or supplemental alert operation. The jumpers are factory-set for T/R operation.

CAUTION:

Only a qualified service technician should change the jumper settings.

In T/R mode, the MFM can connect to a 20-Hz AC external alerting device such as a loud external ringer or EICM-type ringer. If you connect several devices to the MFM, only one device can be off-hook at a time, and the total ringer equivalency number (REN) must be less than or equal to 2.0. This is because only one call can ring at an MFM at a time. When a ringing call arrives at an MFM while it is off-hook, the control unit queues the ringing call.

General Purpose Adapter

A General Purpose Adapter (GPA), shown in Figure 2–15, lets you connect a tip/ring device—such as a single-line telephone, modem, or answering machine—to an analog multiline telephone. The device must be touch-tone, not rotary, and you must make calls from the analog multiline telephone, since the GPA has no pulse or touch-tone detectors.

One end of a four-pair cord plugs into the V.T. jack on the back of the GPA; the other end plugs into the OTHER jack on the bottom of the telephone. The one- or two-pair cord from the T/R device plugs into the TEL. EQUIP jack on the GPA.

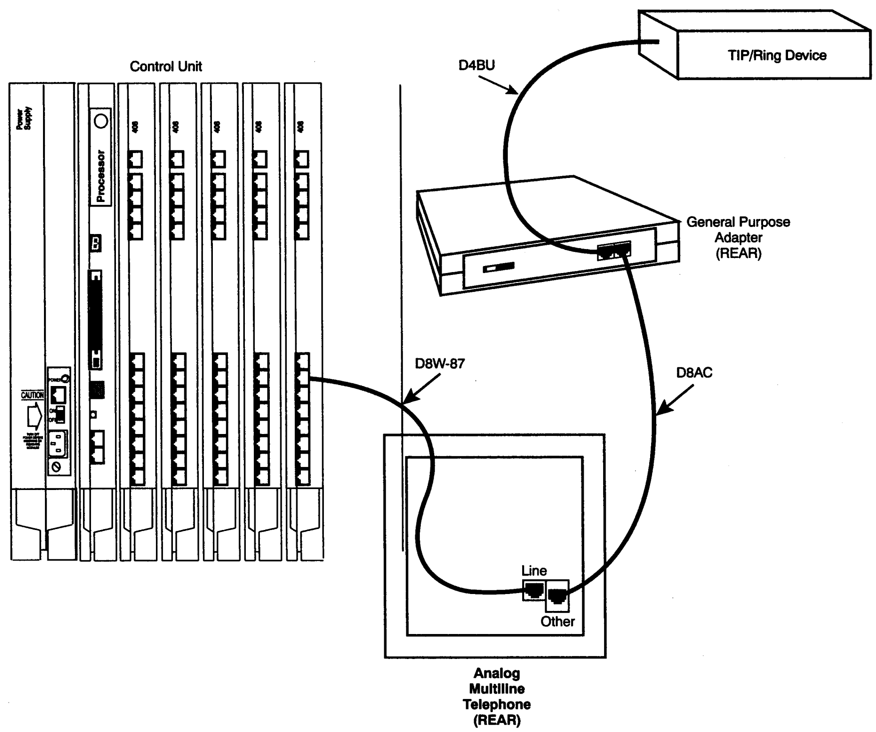


Figure 2–15. GPA Connections

The switch on the back of the GPA lets you choose one of the following services:

- **Basic.** Use this setting to dial and answer calls on an analog multiline telephone or to attach a T/R device such as a single-line telephone or a fax machine. Incoming calls ring only on the analog multiline telephone.
- **Join.** Use this setting to add a recording device or a single-line telephone to a call that is in progress on the analog multiline telephone. You cannot originate or answer calls on this setting.
- **Automatic.** For devices that answer calls, such as answering machines and modems, you need a programmed Auto Answer All button to allow the device to answer calls automatically.

  You also use this setting when you want to make and receive calls on the telephone while the modem attached to the GPA is busy. You cannot make or receive a data call while a voice call is in progress.

NOTE:
When a GPA is used in Auto mode, do not use the telephone. To dial a call using a feature that turns on the speakerphone (such as Authorization Codes, Auto Dial, or Last Number Dial), lift the handset before activating the call.

A GPA is not recommended for use with a fax machine. See "Fax Machines," later in this chapter.

7500B Data Module

The Integrated Services Digital Network (ISDN) 7500B Data Module connects a data terminal to the system on a 008 MLX or 408 GS/LS-MLX module (Release 2.0 and later only), so that a user can make and receive calls at a *7500B data station*. Instead of converting digital data signals to analog signals as a modem does, the data module maintains a digital data format that allows transmission to another 7500B data station or over a PRI facility. On a data terminal, you use the keyboard to dial the number. (For information about connecting analog data stations, see "Modems" later in this chapter. For more information about data communications, see Chapter 5, "Data Communications Support.")

NOTE:
You cannot use a 7500B data module with a QCC.

The 7500B data module provides an RS-232 interface for asynchronous data terminal equipment (DTE) operating at speeds of up to 19.2 kbps. The 7500B data module also provides a CCITT V.35 interface for synchronous DTE operating at speeds of up to 64 kbps. (You must order optional enhancement boards separately. These are described later in this topic.)

You can set up the 7500B data module to handle a variety of data communications equipment (DCE), and it is the only digital adapter approved for use with the system.

The 7500B data module's front panel (see Figure 2–16) has the following features:

- **Power/Test LED.** Lights when the unit is receiving power and flashes when tests are performed.
- **Data LED.** Lights when a call is in progress and flashes to indicate an incoming data call; also flashes when tests are performed.
- **Display.** Displays status information and option settings.
- **Next, Back, and Enter Buttons.** Let you operate the 7500B data module and adjust the screen's contrast.

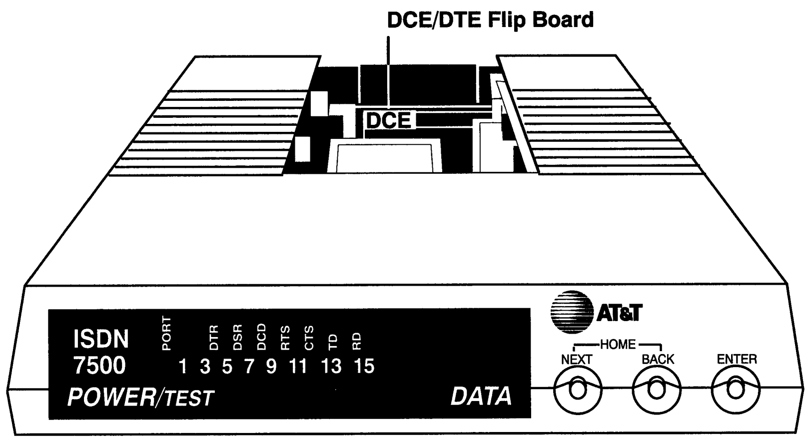


Figure 2–16. 7500B Data Module Front Panel

The 7500B data module's back panel (see Figure 2–17) has the following features:

- **Phone Jack.** Use this jack to connect an MLX telephone to the 7500B data module.
- **Line Jack.** Use this jack to connect the 7500B data module to a line/trunk jack on a line/trunk and extension module.
- **Power Connector.** Connects the 7500B data module to the DC power supply, which connects to an AC outlet.
- **Port 1.** Use this port to connect the 7500B data module to a data terminal, computer, or modem.
- **Port 2.** If an enhancement board is installed for synchronous operation, use this port to connect a second data terminal, an automatic calling device (with an RS-366 interface), or a data terminal (with a V.35 interface).

NOTE:
A modem provides an analog data interface from an MLX telephone with an MFM installed. If an MLX telephone has an MFM, you cannot install the 7500B data module on the same inside line.

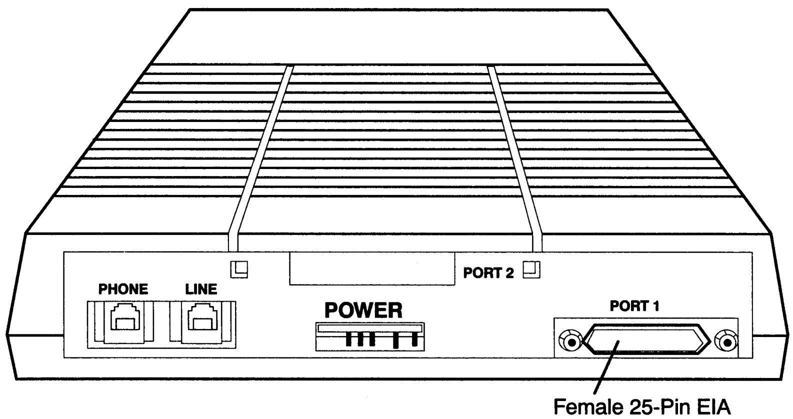


Figure 2–17. 7500B Data Module Back Panel

When you use the 7500B data module with an MLX telephone, one end of the D8W cord plugs into the jack labeled PHONE on the 7500B data module; the other end plugs into the jack labeled LINE on the MLX telephone. The maximum cord length from the 7500B data module to the telephone cannot exceed 80 ft. (24 m). You cannot use the MLX telephone to dial data calls. Similarly, you cannot use the DTE connected to the 7500B data module to dial voice calls. Each device operates independently, and you assign features to each device independently.

NOTE:
Do not connect two 7500B data modules on one inside line.

You can configure the 7500B data module as a stand-alone device by ordering a WP90110,L1 power unit. Alternatively, you can configure the 7500B data module to work in a multiple-mount arrangement by ordering a Z77A data mounting, which provides a common power supply for up to eight 7500B data modules. Both the power unit and the data mounting require a 115-VAC power outlet. Neither the power unit nor the data mounting is provided with the 7500B data module; you must order each separately.

Unlike MLX telephones, the 7500B data module does not have internal 100-ohm line termination. Therefore, when you use a 7500B data module without an MLX telephone, you must install a 100-ohm 440A4 terminating resistor adapter on the line near the 7500B data module.

To provide synchronous operation at speeds up to 64 kbps, you must order the Multipurpose Enhancement Board or the High Speed Synchronous Enhancement Board.

Asynchronous Features Provided with the 7500B Data Module

The 7500B data module offers the following asynchronous features:

- RS-232 interface
- Asynchronous full-duplex operation
- Selected data rates of 300, 1200, 2400, 4800, 9600, and 19,200 bps
- Data options set by the data terminal connected to the RS-232 interface
- Ability to change options without dropping a data call
- Autobaud (also called *data metering* or *speed matching*), which provides the ability to adjust the speed of transmission to match the speed of the data terminal being *called*
- Auto-adjust, which provides the ability to adjust to the speed and parity of the local data terminal connected to the 7500B data module
- Call setup (dialing) from the keyboard of an ASCII data terminal by using local command (CMD) mode or AT modem language mode
- Automatic or manual answering of incoming data calls

Multipurpose Enhancement Board

This board provides an RS-366 automatic calling unit (ACU) interface and converts the RS-232 interface on the main circuit board from asynchronous to synchronous. You must order a V.35 adapter cable separately (from a third-party vendor) for operation at the lower data rates and also at data rates of up to 56 and 64 kbps. Without the adapter cable, data rates are limited to 1200, 2400, 4800, 9600, and 19,200 bps.

The 7500B data module in conjunction with a Multipurpose Enhancement Board offers the following synchronous features:

- RS-232 interface
- Half- or full-duplex operation using the RS-232 interface at data rates of 1200, 2400, 4800, 9600, and 19,200 using data transport Mode 2
- Half- or full-duplex operation at 56 kbps with the V.35 interface adapter cable
- Full-duplex operation at 64 kbps with the V.35 interface adapter cable
- Automatic answering of incoming data calls
- Ability to make outgoing data calls manually and select user-programmable telephone numbers from the 7500B data module display on the front panel

- RS-366 interface to an automatic calling unit

High-Speed Synchronous Enhancement Board

This board provides a V.35 interface at synchronous data rates of 48, 56, or 64 kbps. The board is packaged with a V.35 adapter cable that converts the 25-pin male connector on the board to the industry-standard 34-pin V.35 interface.

The 7500B data module in conjunction with a high-speed enhancement board offers the following synchronous features:

- V.35 interface (includes adapter cable if you order the board using PEC 21624)
- Full-duplex operation at 48, 56, and 64 kbps
- Half-duplex operation at 56 kbps only
- Automatic answering of incoming data calls
- Ability to make data calls manually and select user-programmable telephone numbers from the 7500B data module display on the front panel

Supplemental Alert Adapter

A Supplemental Alert Adapter (SAA) allows the connection of an external alert (such as a bell or chime) to an analog multiline telephone (see Figure 2–17). An external alert notifies people working in noisy areas of incoming calls.

The telephone cord plugs into the TELEPHONE jack; the cord from the extension jack plugs into the CONTROL UNIT jack; and the cord from the external alert device plugs into the ALERTER jack.

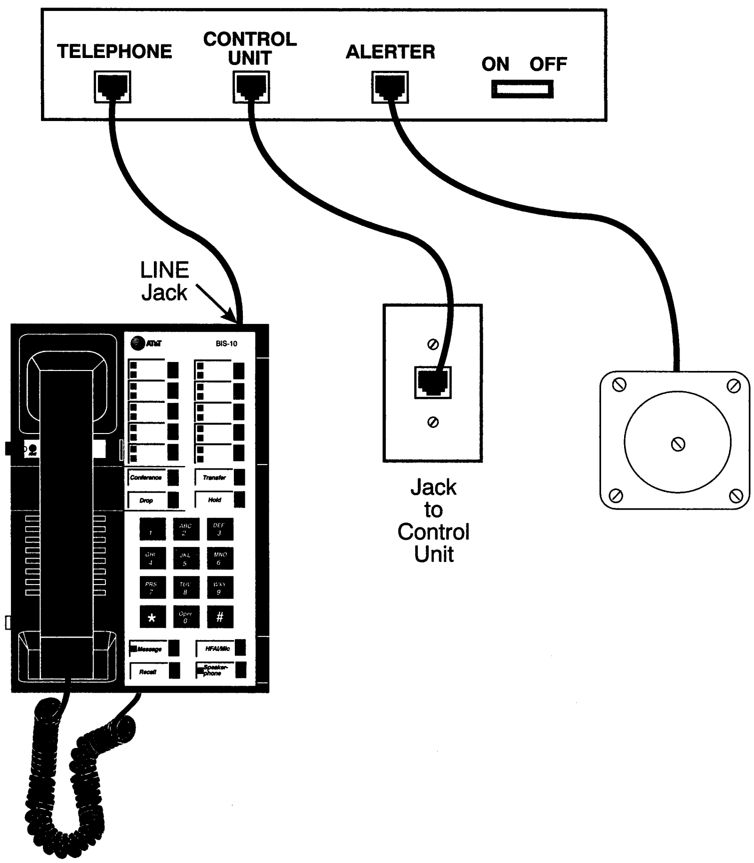


Figure 2–18. SAA Connections

Adjuncts

This section describes the adjuncts that you can use with the system. Table 2–9 summarizes adjuncts that are supported.

Table 2–9. Adjunct Summary

Equipment Type	AT&T Products	Connects to
Alerts (AC) Any audible or visual alert that operates on 20–30 Hz ringing signals. Associated with a specific extension (supplemental alert).	loud external ringer external ringer	Can be connected to: T/R 012/008 OPT extension jack MFM/MLX extension jack GPA/analog extension jack
Alerts (DC) Any audible or visual alert that operates on 48-VDC ringing signals. Associated with a specific extension (supplemental alert) or works on a programmed trunk jack (external alert).	alert bell alert horn alert strobe alert chime alert deluxe horn alert switch	Can be connected to: LS trunk jack *Requires 8301 alerter interface* MFM/MLX extension jack SAA analog extension jack
Answer/Record Machine Industry-standard machine. Low ringer equivalence (less than 0.15 or < 1.0 total REN for T/R port). Ability to recognize 600-ms disconnect signal or other means of automatic disconnect (such as voice reset disconnect timer, fixed recording time). *Cannot be connected to a QCC.*	Model 1300* answering machine Model 1531* remote answering system telephone	Can be connected to: T/R 012/008 OPT extension jack MFM/MLX extension jack GPA/analog extension jack
Credit Card Verification Terminal Must have touch-tone dialing capability when connected by MFM; rotary or touch-tone dialing can be used on T/R port. *Cannot be connected to a QCC.*		Can be connected to: T/R 012/008 OPT extension jack MFM/MLX extension jack *Device originates and receives call independently of associated telephone when used with an MFM.* GPA/analog extension jack
Dial Dictation A device that requires contact closure can be used on LS/GS line jack only with UPAM. *Cannot be connected to a QCC.* *Requires UPAM to provide 48 VDC.*		Can be connected to: LS or GS/LS trunk jack T/R 012/008 OPT extension jack MFM/MLX extension jack GPA/analog extension jack

Continued on next page

Table 2–9, *Continued*

Equipment Type	AT&T Products	Connects to
Direct Station Selector (DSS) A maximum of 2 DSSs can be connected to an operator console. A 329A power supply must be added to an operator console having 2 DSSs. *Connects to DSS jack on operator console.*	DSS	MLX telephones only
Fax Must have touch-tone dialing capability when connected by MFM; rotary or touch-tone dialing can be used on T/R port. Industry-standard analog interface. *Cannot be connected to a QCC.*		Can be connected to: T/R 012/008 OPT extension jack MFM/MLX extension jack *Device originates and receives calls independently of associated telephone when used with an MFM.*
Group Calling Delay Announcement Industry-standard announcement device. Must provide automatic disconnect. Each calling group can have its own announcement (maximum 32). A device can provide delay announcement for more than one group. *Cannot be connected to a QCC.*	Model 1330 answering machine DA-5 Digital Announcer* Digital Announcement Unit	Can be connected to: T/R 012/008 OPT extension jack MFM/MLX extension jack *Device originates and receives calls independently of associated telephone when used with an MFM.* GPA/analog extension jack
Hands-Free Unit *Connects directly to telephone.*	S202A	Analog telephones only
Headpiece	Mirage® StarSet® Supra® Monaural Supra Monaural Noise-Canceling (NC) Supra Binaural Noise-Canceling (NC) Supra Binaural	MLX or analog multiline telephones

Continued on next page

Table 2–9, *Continued*

Equipment Type	AT&T Products	Connects to
Headset Adapter *Need to program Auto Answer All button for use with adapter. Connects directly to telephone OTHER jack.*	502C	Analog telephones only
Loudspeaker Paging External paging system using DTMF signaling connected to LS or GS line jack. CPE paging systems require an interface unit; if CPE has 2–wire input, the BOGEN UPAM-K (58500) can be used.	PagePac Plus with PagePal *Requires no adapter.* PagePac 20* PagePac 20* with Zonemate 9* or Zonemate 39* *2500SM* telephone required.* PagePac 6* *Loop-Start Trunk Adapter (53518) is required when connected to loop-start line jack.*	Can be connected to: LS or GS/LS trunk jack *Bidirectional paging is supported; only one line jack is needed for multizone paging.*
Message-waiting indicator For single-line telephones. Connects directly to telephone.	Z34A	
Modem Industry-standard modem. If the modem supports touch-tone dialing, the keyboard at the associated data terminal can be used for dialing. If the modem does not support touch-tone dialing, an associated single-line telephone can be used for dialing.	DataPort modems	Can be connected to: T/R 012/008 OPT extension jack MFM/MLX extension jack, GPA analog extension jack

Continued on next page

Table 2–9, *Continued*

Equipment Type	AT&T Products	Connects to
Music On Hold: Any FCC-registered 8-ohm music source or recorded announcement device†.	Magic on Hold	Can be connected to: LS or GS/LS trunk jack *Music coupler required.*
Speakerphone	4A* *2500SM telephone required* 203A*	Single-line telephones only *Connects directly to telephone*
SMDR Printer Must be located within 50 ft. (15 m) of control unit or use ADU to extend distance.	serial printer 1200 bps no parity 1 stop bit	Connects to RS-232C jack on processor module.

* Vintage equipment; no longer available for sale or lease.

† If you use equipment that rebroadcasts music or other copyrighted materials, you may be required to obtain a copyright license from and pay license fees to a third party such as American Society of Composers, Artists, and Producers (ASCAP) or Broadcast Music Incorporated (BMI). Or you can purchase a Magic on Hold system, which does not require you to obtain such a license, from AT&T.

NOTE:
You should consider some restrictions and special conditions when you program extensions to serve a certain class of equipment or a special class of service. Modules, telephones, and adjuncts specifically designed for this system provide *equipment type* information to the system software automatically through the port *signature* or terminal/adjunct *classmark*. However, there are types of devices that connect to a T/R interface and provide no classmark or other means of automatic identification to the control unit. This equipment includes Group III (G3) fax terminals, automatic answering machines, modems, automatic dial-announce alarm-sending equipment, external alerting devices, music sources, and paging equipment.

System Adjuncts

System adjuncts connect directly to the control unit. Modems can be connected to an 012 module on the control or to an extension but are described later, under "Telephone Adjuncts."

Station Message Detail Recording Printer

You can connect a Station Message Detail Recording (SMDR) printer to the SMDR jack on the processor module. If the SMDR printer is not within 50 ft. (15 m) of the control unit or is not plugged into the same AC circuit as the control unit, you must use an Asynchronous Data Unit (ADU) to extend the distance.

SMDR captures detailed usage information on incoming and outgoing voice and data calls and sends the information to a printer. Two SMDR report formats are available: the factory-set Basic format and the ISDN format. Use the ISDN format if you subscribe to the AT&T INF02 Automatic Number Identification (ANI) or have an 800 GS/LS-ID module and caller identification service from the CO. If you select the ISDN format during system programming, the number identification information prints in the CALLED NUMBER field of the call report. The remainder of the fields are identical to the basic format.

An SMDR record consists of the following fields:

- **CALL TYPE** (Basic or ISDN)
- **DATE**
- **TIME**
- **CALLED NUMBER**
- **DUR** (duration)
- **LINE** (facility number)
- **STN** (extension)
- **ACCOUNT** (account code)

Also, a Call Accounting Terminal application is available for tracking and printing reports on telephone charges. See "Call Accounting Terminal" in Chapter 4.

AT&T DoorPhone

This adjunct is for use outside an office and connects to a 408 or 800 module line/trunk jack. Through system programming, you specify that the DoorPhone, when its button is pressed, will ring one or more extensions or alerters inside the office. The DoorPhone can be set up so that the door can be opened automatically from an extension where the DoorPhone rings. The person at the extension inside the office may also be able to speak to the person at the DoorPhone.

System Programming and Maintenance PC

You can use a PC with MS-DOS® version 3.3 (or higher) and System Programming and Maintenance (SPM) software to program and maintain the system. The PC connects to the SPM jack on the processor module. See Chapter 4, "Applications," for additional information.

Loudspeaker Paging Systems

Loudspeaker paging systems use a ground-start/loop-start (GS/LS) line jack. You should program the port for loop-start operation and designate it as a paging port. You can program up to three ports as paging ports. If you connect a paging system other than PagePac Plus, you must also install a Universal Paging Access Module (UPAM) or Loop-Start Trunk Adapter.

NOTE:
If you program the loop-start port for paging, you cannot use it for outside calls unless you install a PagePac Port Saver.

PagePac Plus loudspeaker paging system replaces the discontinued (but still supported) PagePac 20 system; PagePac Plus requires no system adapter, nor does it require Zonemate; instead, it comes equipped with 8 built-in zones, expandable to as many as 48 zones with 16-zone zone expansion modules. Use it with the PagePal interface.

PagePac Plus also provides a music source for paging and Music On Hold without a music coupler.

Dial Dictation

You can use a dictation unit as either a system or extension adjunct. Some dictation units connect directly to the control unit via a T/R jack on the 012 module or 008 OPT module, or to a telephone using an MFM or a GPA. Other dictation units connect to a Universal Paging Access Module that is connected to a loop-start port programmed for dial dictation (similar to loudspeaker paging).

Fax Machine

You can connect a fax machine to any T/R jack on the control unit or to an MFM. You should not use a fax machine with a GPA because the fax machine cannot autodial through the GPA. Instructions are packaged with the unit.

A fax machine originates and receives fax calls independently of any associated telephone. You can dial calls from the fax machine's dialpad or from an associated single-line telephone.

If the system does not have Direct Inward Dialing (DID) trunks, you should program fax extensions to personal lines. When the system has DID service, it can direct incoming calls automatically to individual fax extensions or to machines in calling groups.

NOTE:
You can use a fax machine as an MLX telephone adjunct, if you use it in conjunction with an MFM.

Delay Announcements

You can use a delay announcement recording to cover incoming calls that may be delayed before being answered by a calling group. To make announcements, use an industry-standard announcement device, which connects either to an extension jack on a 012 or 008 OPT module or to an MFM.

Telephone Adjuncts

Telephone adjuncts connect to a telephone either directly or through an adapter.

Modems

A modem used at an extension (called a *data station* because a modem is included) may provide the following features, depending upon the type of modem:

- Dialing or ending asynchronous data calls from the keyboard when connected using a basic telephone extension jack on a 012 module or when connected to an MLX telephone using an MFM
- Autobaud (also called *data metering* or *speed matching*), the ability to adjust the speed of transmission to match the speed of the data terminal being called
- Automatic or manual answering of incoming data calls
- Self-test and maintenance procedures
- Ability to set data options at the keyboard for a call and, if necessary, change the options without dropping the call

A modem connected to an MLX telephone requires an MFM (Multi-Function Module); a modem connected to an analog multiline telephone requires a GPA (General-Purpose Adapter). For more information about modem connections and data stations, see Chapter 5, "Data Communications Support."

Modems connected to GPAs, MFMs, and 012 modules have been found to perform at speeds up to 9600 bps. However, various conditions, such as line interference, may lower this rate.

You can connect most industry-standard modems to the system.

Headsets

Headsets are designed for hands-free telephone use and consist of several components, depending upon whether manual or one-touch operation is used. Any AT&T headpiece works in either mode.

Headpieces

Six different headpieces are available as headset components. Each is light, comfortable, and uses a transparent voice tube to eliminate any cumbersome large microphone. Each model comes with a 10-ft. (305-cm) coiled cord and a quick-disconnect latch.

- **Mirage.** This is a small, almost unnoticeable, monaural headset that uses a disk-shaped receiver. It can be worn on either ear, instead of a headband or ear tip. This headset is not useful in noisy environments.
- **StarSet.** This monaural headset is worn without a headband. It uses a one-size-fits-all, soft, pliable ear tip that provides high-quality sound yet allows you to hear other conversations or instructions in the workplace.
- **Supra Monaural.** This monaural headset has an adjustable headband. It offers a soft, comfortable ear cushion that reduces surrounding noise, making it easier to understand the caller.
- **Supra Monaural Noise-Canceling (NC).** Same as above with noise-canceling microphone that reduces background noise by up to 75 percent.
- **Supra Binaural.** Sound in both ears
- **Supra Binaural Noise-Canceling (NC).** Same as above with noise-canceling microphone on flexible boom features windscreen and reduces background noise transmission by up to 75 percent.

Manual Operation (Analog Multiline Telephones Only)

Manual operation is appropriate when a headset is used occasionally. You must pick up the handset to answer a call and replace it to hang up. The manual headset consists of the headpiece, a modular base unit, and the telephone. The headpiece plugs into the modular base unit, and the modular base unit connects to the telephone through the HANDSET jack on the side of the telephone. The telephone handset is plugged into the modular base unit. The modular base unit allows you to adjust the incoming volume, switch between the headset and handset as needed, and temporarily mute the line. (See Figure 2–19.)

One-Touch Operation (All Telephones)

One-touch operation allows you to simply touch a button to answer a call. On analog multiline telephones use one of the headpieces along with a plug prong base unit, and an adapter (502C). The headpiece connects to the plug prong base unit, which in turn connects to the adapter. The adapter plugs into the OTHER jack on the underside of the telephone. The plug-prong unit provides switchhook control for answering calls by pressing a button. You can also adjust the incoming volume. On MLX telephones, the headpiece is attached to a modular base unit, which is attached to the telephone. The handset is also attached to the modular base unit. (See Figure 2–19.)

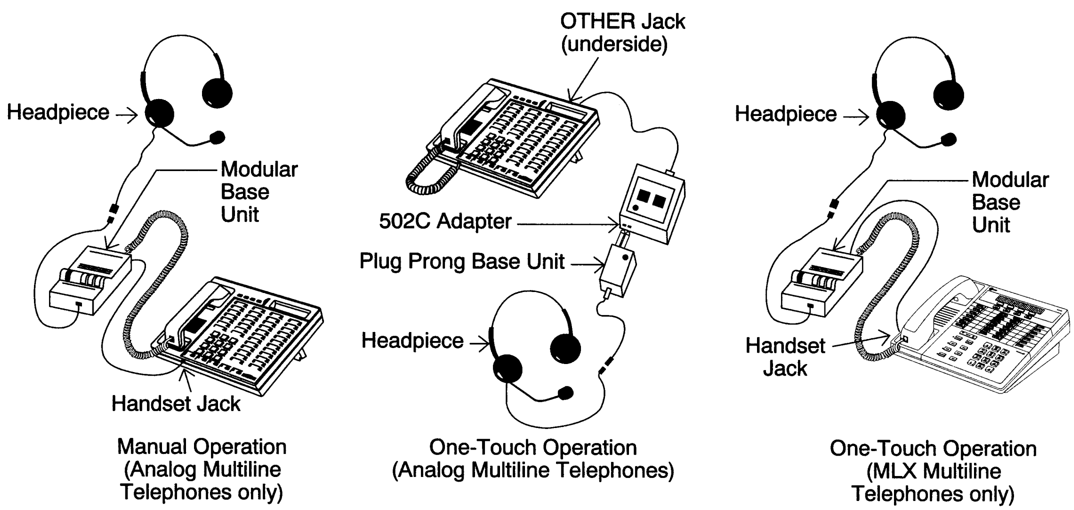


Figure 2–19. Headset Operation

Speakerphones

You can connect the S202A speakerphone to the system. This Hands-Free Unit (HFU) speakerphone for analog multiline telephones allows you to make and receive calls without using the handset. If you program and activate an **Auto Answer Intercom (AAI)** button on the telephone, the HFU will turn on when you receive an inside call.

NOTE:
Equipment in the area of the microphone can lower the quality of speakerphone transmission.

Specialty Handsets

Model K6S handsets for users who are hard of hearing are available for use with MLX telephones.

In addition, beginning with Release 2.1, four specialty handsets are available. They are compatible with earlier releases.

- **Noise-Canceling Handset.** Reduces background noise in an office environment; provides 10dB (nominal) reduction.
- **High Noise-Canceling Handset.** Reduces background noise in a factory- or warehouse-type environment; provides 20-dB (nominal) reduction.
- **Amplified Speech Handset.** Amplifies the voice of the other party; provides 0-dB to 10-dB (nominal) voice gain.
- **Push-to-Talk Handset.** Activates the mouthpiece only when you push the button on he handset.

Message-Waiting Indicator

You can connect the Z34A message-waiting indicator to single-line sets that do not have a Message LED.

Additional Telephone Adjuncts

You can also connect the following adjuncts to telephones:

- Answering machines
- Credit card verification terminals
- PCs (through external modems, internal modems, or 7500B data modules)

Adapters and Adjuncts Not Supported

Do not connect the following analog telephone adjuncts and adapters to the system:

- Basic Telephone and Modem Interface (BTMI)
- Basic Telephone and Modem Interface-2 (BTMI-2)
- ATR Interface (ATRI)
- MTR Interface (MTRI)
- Off-Premises Extension Unit (OPU)
- System 25 Direct Extension Selector (DXS)

CAUTION:
If you connect any adjunct listed above, it may damage the device or the system.

Power-Related Hardware

You can add power-related hardware to the system to provide additional power and protection from power surges. Other accessories apply to specific conditions.

NOTE:
In most cases, you do not need additional power surge protection.

Power Accessories

In a power failure, battery backup units can keep the system running for several hours. When you connect adjuncts and adapters to telephones, the power requirements for the telephones and the system increase. You can add a power accessory to an individual telephone or to the system to accommodate these additional needs.

Battery Backup Power

A 500-VA Uninterruptible Power Supply (UPS) and reserve UPS units can provide battery backup for power to the system. Basic UPS provides power for 15 minutes; however, you can add reserve UPS units to basic UPS. Each reserve unit extends backup power for an additional hour.

The backup durations for normal system operation of one full carrier at maximum system load are as follows:

- **15 Minutes.** Basic 500-VA UPS.
- **1 Hour.** One 500-VA reserve cabinet for each UPS.
- **2 Hours.** Two 500-VA reserve cabinets for each UPS.
- **4 Hours.** Four 500-VA reserve cabinets for each UPS.

Telephone Power Units

The KS22911-L1 and 329A power units provide additional power to telephones that have adjuncts, adapters, and/or two DSSs attached, or to telephones located far from the control unit. These power units are installed between the telephone and the wall jack. Adding local power to a few telephones can reduce system load.

Table 2–10 shows local auxiliary power requirements. You must connect the KS22911-L1 power supply to a 117-VAC outlet not controlled by a switch.

Table 2–10. Local Auxiliary Power Requirements

Number of MLX Telephones	Number of DSSs	Number of MFMs	Number of KS22911-L1 Power Supplies	Number of 329A Power Supplies
1	2	—	1	—
1	1	1	1	—
1	2	1	—	1
3 or more in one carrier	1 per telephone	—	1 for each MLX telephone after the first 2	—
	DSS out of building		1	

CAUTION:

If the system requires additional power and you connect an Uninterruptible Power Supply (UPS), you should also connect the system's auxiliary power unit (Supplemental Power Unit 9040-2) to the UPS.

The total length of wire between the KS22911-L1 or 329A power supply and the MLX telephone must not be more than 50 ft. (15 m).

Do not replace the 2-ft. (61 cm) D8AC cord (packaged with the DSS) with a longer cord. Improper operation may result.

A power supply kit for MLX telephones includes the KS22911-L1 power unit, a D6AP cord, and a 400B or 400B2 adapter. For analog multiline telephones, you need the 329A power unit, a D6AP cord, and a Z400F adapter.

The 329A power unit is not available in a kit, so you must order the D6AP cord and the 400B or 400B2 adapter separately.

The MFM includes the KS22911-L1 power unit, a D6AP cord, and a 400B or 400B2 adapter. DSSs are shipped without power units. Therefore, when DSSs require local power, you must order the KS22911-L1 or 329A power unit, D6AP cord, and 400B or 400B2 adapter separately.

NOTE:
Telephone operation without adjuncts is guaranteed for wiring runs up to 1000 ft. (305 m) from the control unit.

Auxiliary Power Units

The 391A1 power supply provides 54 unit loads to each carrier (the 391A provides 45). If the unit load requirement for a carrier exceeds 54 (or 45) unit loads, an auxiliary power unit is needed to allow that carrier to support an additional 27 unit loads.

NOTE:
You can connect only one auxiliary power unit to the 391A power unit. If the system requires additional 48-VDC power, connect some telephones to KS-22911L1 or 329A telephone power units.

CAUTION:
Running the system with more than 54 or 45 (depending upon the power supply) unit loads per carrier may not appear to do harm. However, this can cause the system to malfunction, creating "No Trouble Found" situations, such as malfunctioning LEDs on multiline telephones, or power unit failure.

To determine the number of unit loads for each power supply module on each carrier, see "Unit Loads" in Chapter 1.

Protection Accessories

Accessories are available for grounding and protecting special telephone connections from power surges, electromagnetic interference, and electrostatic discharge.

In-Range Out-of-Building Protection

An In-Range Out-of-Building (IROB) protection unit is required when equipment is connected to the following jacks and is located in a different building but within 1000 ft. (305 m) of the control unit:

- Analog multiline telephone extension jacks on 008, 408, and 408 GS/LS modules. Analog multiline telephones are supported up to 2000 ft (610m) with local power
- MLX telephone extension jacks (on 008 MLX or 408 GS/LS-MLX (Release 2.0 and later only)

These units protect the equipment and the control unit from lightning strikes and power surges. Each piece of equipment requires two units, one for the control unit end of the wire run, the other for the equipment end.

The system supports the TII Model 343 IROB protector for analog multiline telephones and equipment and the Model 505A IROB protector for MLX telephones and equipment.

NOTE:
Single-line telephones are supported out of building or off premises using an 008 OPT module or an 012 jack with an OPRE.

⚠ CAUTION:
The IROB protectors must be installed by a qualified service technician or installer.

Figures 2–20 and 2–21 illustrate the IROBs. See the documentation packaged with the IROB protector for complete installation instructions.

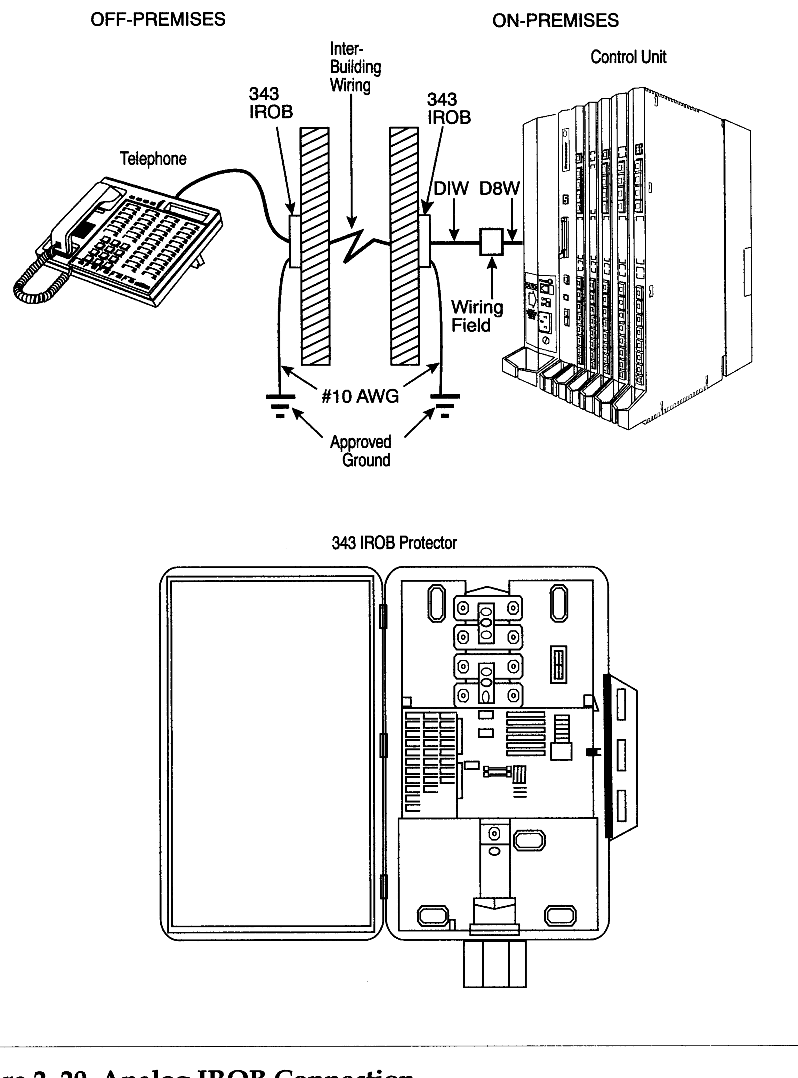


Figure 2–20. Analog IROB Connection

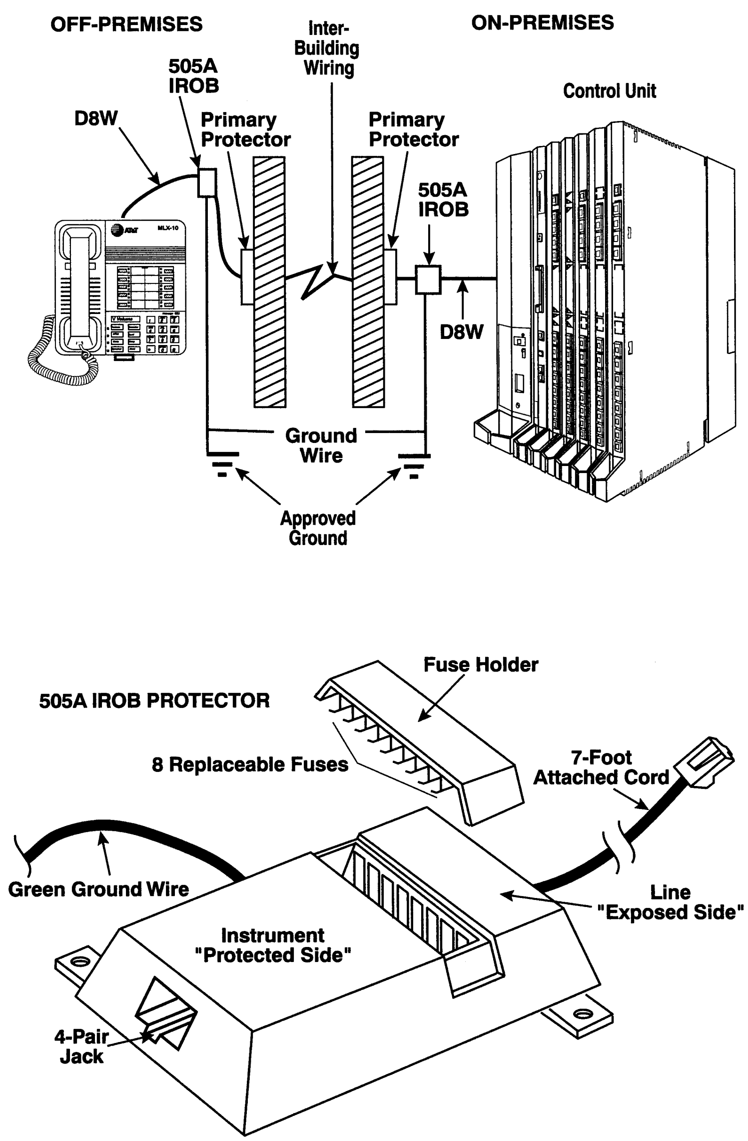


Figure 2–21. MLX IROB Connection

Off-Premises Range Extender

An Off-Premises Range Extender (OPRE) is used for off-premises extensions with the following distance limitations:

- 19 gauge cable: 38K ft. or 7.3 miles

- 22 gauge cable: 27K ft. or 5.2 miles
- 24 gauge cable: 21K ft. or 4.1 miles
- 26 gauge cable: 17K ft. or 3.2 miles

See the documentation packaged with the OPRE for complete installation instructions.

146A and 147A Surge Protectors

To safeguard system operation from lightning and power surges, the control unit may require surge protectors. It is the responsibility of the local telephone company to provide primary protection on the outside lines at the network interface and to ensure that these protectors are properly grounded. If the telephone company line protector is properly grounded and bonded to the AC power ground, most lightning damage is prevented.

The 391A1 power supply has built-in AC line protection. This built-in protection handles almost all situations. Occasionally, the system may require additional AC line protection if a customer is located in a heavy lightning area. You can connect a 147A protector to the system to limit surges from AC lines and outside lines. One 147A protector provides protection for four outside lines. However, you can add up to three 146A protectors to the 147A to provide protection for up to 16 outside lines. For more than 16 lines, additional 147A protectors are required (see Figure 2–22). See the documentation packaged with the surge protectors for complete installation instructions.

NOTE:
The 147A protector is usually not needed with the 391A1 power supply. It may be needed with the older 391A power supply in high-risk lightning areas.

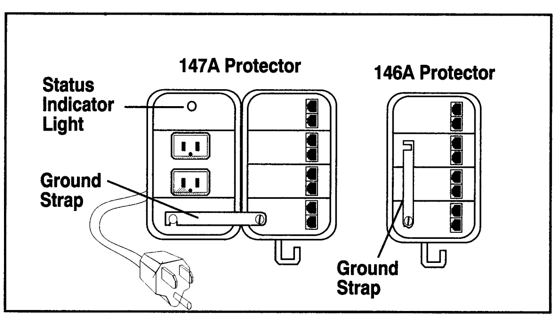


Figure 2–22. Surge Protectors

Electromagnetic Interference Filters

To protect against interference from motors and other devices, you can install a Z200A electromagnetic interference (EMI) filter between the control unit and a telephone. Instead of a D8W cord, the filter cord plugs into the telephone LINE jack. You must install a Z200A filter when you use an SMDR printer. These filters also protect against radio-frequency interference (RFI) from transmitters and other radio frequency generating equipment.

Electrostatic Discharge Suppression Kits

You can install an electrostatic discharge (ESD) suppression kit in older analog multiline telephones—with membranes—to eliminate damage to the telephone that a voltage discharge resulting from electrostatic buildup can cause.

System Alarms

An alarm condition detected by the system can cause the control unit to activate an alarm device on a loop-start port. When the contacts close, a signal is passed on to a Universal Paging Access Module (UPAM) and then to an external alert. Alerting devices can be a strobe, horn, bell, or chime.

A UPAM is needed because 48-VDC alerting devices require two contact closures, and loop-start ports only have one. The UPAM provides the additional closure.

Trouble Alarm

You can use a loop-start jack to activate an alarm by connecting a UPAM (see Figure 2–22) to the jack. When system trouble (software or hardware malfunction) is detected at the operator console, a signal is sent to that jack. The jack's switching contacts close and the signal is passed on to the UPAM. The UPAM then activates the external alert.

Power Failure Alarm

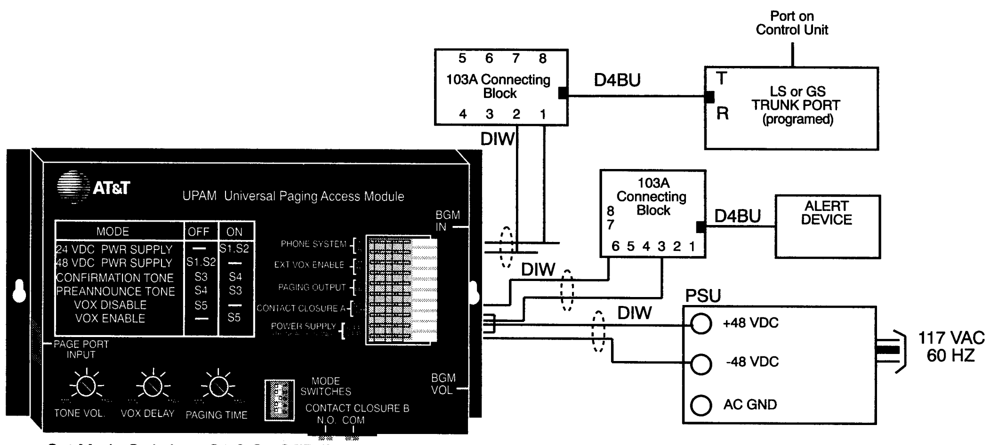
You can use a ground-start or loop-start PFT jack to activate an alerting device during a power failure by connecting the port to a UPAM (see Figure 2–23). When a power failure occurs, the switching contacts on the PFT jack close and a signal is passed on to the UPAM. The UPAM then activates the external alert.

NOTE:
You can connect a PFT telephone to this jack when the jack is connected for a power failure alarm.

Power Failure DID Busy-Out

You can program the PFT jack on a ground-start or loop-start module to automatically short the *busy-out* wire pair associated with a group of DID trunks. Normally, a loop-start line/trunk is used as the busy-out pair. When a power failure occurs, shorting this busy-out pair signals the CO that the DID trunks are out of service. Figure 2–25 shows this connection.

NOTE:
Before you remove the ground-start or loop-start module containing the PFT jack for the DID busy-out, you must short the busy-out pair and then disconnect the modular cord from the PFT jack. Otherwise, a false busy-out occurs. The short is removed after the system is powered up.



Set Mode Switches S1 & S2 OFF (for 48 V operation)

Figure 2–23. Trouble Alarm Connections

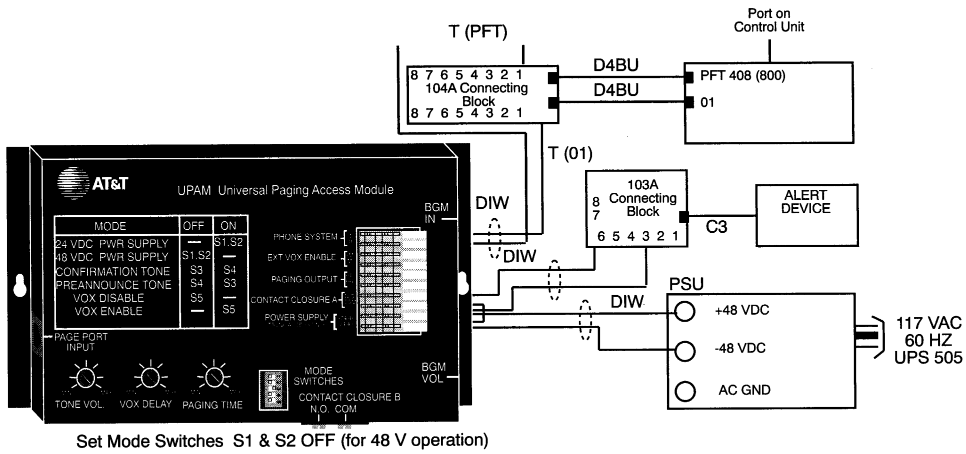


Figure 2–24. Power Failure Alarm Connections

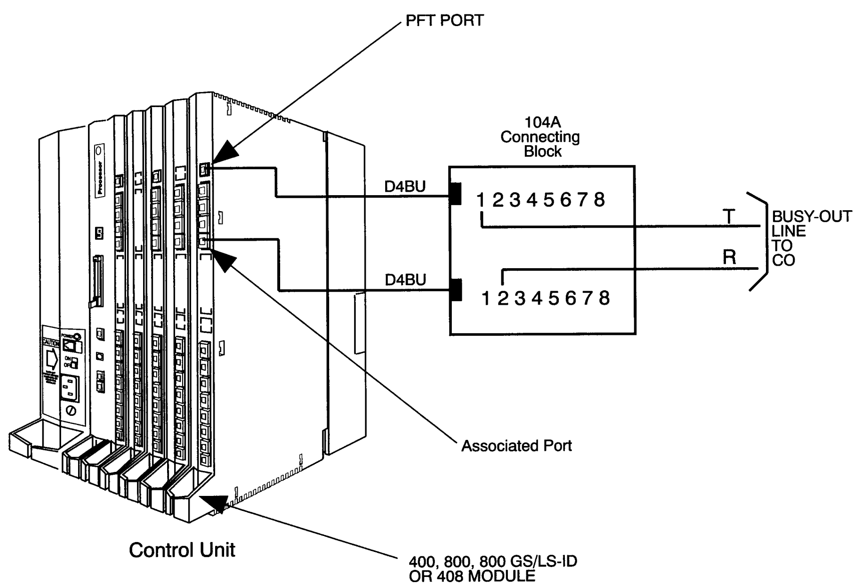


Figure 2–25. Power Failure DID Busy-Out Connections

Power-Failure Transfer Telephone

A power failure transfer (PFT) telephone is any single-line telephone connected to a PFT jack on a 400, 400/GS/LS/TTR, 408, 408 GS/LS, 408 GS/LS-MLX (Release 2.0 and later only), 800, 800 GS/LS, or 800 GS/LS-ID module. In the event of a power failure, the system shuts off and the PFT telephone automatically connects to the associated outside line for making and receiving calls. If a PFT telephone is used on a module with ground-start trunks, you must purchase a GS (ground-start) button for it.

NOTES:

1. The PFT jack does not operate unless a power outage occurs or the power supply units are turned off.
2. A single-line telephone that is connected to an MFM cannot be a PFT telephone.
3. If you plan to connect PFT telephones to ground-start line/trunk, you must add a ground-start button (KS23566,LI) to each PFT telephone. If power fails, the system uses this button when the number is dialed. If the button is used with modular 2500 sets, wire the button from the wall jack.
4. If rotary line/trunk are in the system, you must use rotary telephones (500MMs recommended) as PFT telephones.

Lines and Trunks

3

Contents

Lines and Trunks

3

Telephone lines/trunks are facilities that carry voice or data communications. They are similar in form and function.

Most of the facilities that connect the system to the central office (CO) are technically called *trunks*; connections within the system (for example from the control unit to an extension) are generally called *lines*. (Technically, a loop-start facility is also a line.) As stated in Chapter 1, the terms *lines* and *trunks* are often used interchangeably. However, a system in Hybrid/PBX mode supports *personal lines*. These facilities usually appear on a telephone button and pass transparently through the system to the CO. Selecting a personal line button on a telephone and lifting the handset brings dial tone directly from the CO.

Loop-Start Trunks

Loop-start trunks are the standard for home and small business Key systems. They are often less expensive than other types of facilities and in some areas more expensive, when all costs are considered, than DS1 service. Keep these facts in mind when considering loop-start trunks:

- They are susceptible to *glare*. (Glare occurs when a person tries to make an outside call on a trunk at the same time that an incoming call is being received on that trunk.)
- They can have higher cable losses and, therefore, transmissions of lower quality than ground-start trunks.
- They cannot assure secure toll restriction.
- Loop-start trunks do not provide a reliable disconnect signal to the system, which can cause problems when an automated attendant is used with the system.

- Most Centrex services are only available over loop-start trunks. The system supports Centrex only when using loop-start trunks.
- The system's caller ID function is only available over loop-start trunks connected to an 800 GS/LS-ID module (see Chapter 2, "Line/Trunk and Extension Modules," for details).

Ground-Start Trunks

Ground-start trunks are outside trunks used by businesses where improved signaling is important. Toll restrictions are more secure. They help to prevent glare and provide a reliable disconnect. In addition, they provide cable losses of less than 4.5 dB. Where available, ground-start trunks are recommended for Hybrid/PBX systems.

The following types of outside trunks can be either ground-start or loop-start:

- Basic (for local and long-distance calls)
- OUTWATS
- FX
- 800 service (INWATS)

NOTE:
The system does not support Centrex services that are provided over ground-start trunks.

Tie Trunks

Tie trunks connect two communications systems within the same location or at different locations. A good example of this is a corporate headquarters with several divisions in different buildings. Regardless of the area, calls within the business can operate as though they were at the same site, even if they are in different cities or states.

A tie trunk connection can be either analog or digital:

- **Analog Tie Trunk Connection.** In an analog tie trunk connection, the system's control unit connects to another system's control unit through a 400EM module. If both communication systems are on the same premises and the other system has similar tie trunk facilities, this module can connect directly to the other communications system. For off-premises connection, the 400EM module can connect to another system through the telephone company's facilities.

You can program an analog tie trunk connection for two-way traffic or for one-way traffic (incoming or outgoing). The one-way mode prevents blocked calls caused by glare.

- **Digital Emulated Tie Trunk Connection.** In a digital emulated tie trunk connection, the communication system's control unit connects to another system's control unit through a 100D line/trunk module, which is programmed for T1-type transmission. You can use a back-to-back connection from one DS1 facility to the other when the total cable distance is less than 1300 ft. (396 meters). To reach a remote system, the DS1 facility connects through a channel service unit (CSU) to the telephone company's facilities.

NOTE:
The system does not support digital data transmission through tie trunks.

Tie Trunk Signaling

Tie trunks transmit signals in three different formats; each format is made up of a specific mode and a specific type. The dual in-line packaging (DIP) switches on the 400EM module allow you to select the signaling mode for tie trunk transmission; you select the signaling type through system programming.

Signaling Modes

There are two signaling modes:

- **Simplex.** Two signaling leads superimposed onto the analog transmission leads provide a two-pair wire interface for connecting two local systems at minimal cost.
- **E&M.** This is a standard interface. The E&M signaling leads are isolated from the transmission leads, requiring a three-pair wire interface.

  NOTE:
  Tie trunks use special leads for signaling. These leads are called "E" (for Ear) and "M" (for Mouth). A signal from the local system to the distant system leaves on the "M" lead of the local system and arrives on the "E" lead of the remote system. If E&M leads are not used, the tie trunk is referred to as being "Simplex".

Protective Resistance and Unprotected

Depending on the type of tie trunk installation, protective resistance may be installed for high-voltage transients or fluctuations. In Simplex mode, the circuit always includes protective resistance. E&M mode can be either protected or unprotected. With protection, a resistance is added to the leads to reduce current peaks. Use protection when there is no network interface to protect the circuit from outside interference.

You must use unprotected for an E&M Type 1 Standard interface (see below) in order to meet the specified voltage-drop criteria. Use this mode when there is a network interface.

Signaling Types

Three different signaling types combine with the signaling modes. Together these create the proper signaling format for each system.

Signaling types in each mode are as follows:

- **E&M Mode**

  —**Type 1 Standard** (factory setting). This setting connects two systems through telephone company facilities.

  —**Type 1 Compatible.** This setting connects two systems directly (without using telephone company facilities). One system is set to Type 1 Standard, the other to Type 1 Compatible.

- **Simplex Mode**

  —**Type 5.** This setting connects similar systems or systems with compatible signaling that are located in the same building or on the same business campus.

400EM Module Options

You can add analog tie trunks to the MERLIN LEGEND Communications System using a 400EM module. A 400EM module has four jacks that you must program individually by selecting line/trunk options during system programming. In addition, you must set the DIP switches, located on the front of the 400EM module, for either E&M or Simplex mode and protected or unprotected conditions. The signaling type is set through system programming.

Before you install a 400EM module in the control unit, refer to *System Planning*, Form 3, Incoming Trunks: Tie, to determine the appropriate switch settings. For each 400EM module line/trunk jack, check the form. If the E&M Signal column indicates 1C or 5 for a particular logical ID, set the DIP switches on the front of the 400EM module as shown in Figure 3–1 and Table 3–1. For more details about unprotected modes, as well as signaling, see the topic above, "Tie Trunk Signaling."

Table 3–1. Setting the 400EM Module DIP Switches

Jacks (as numbered in Figure 3–1)		DIP Switch Position #	Signaling Type		
			1S (Default) and 1C Unprotected E&M Mode	1S and1C Protected E&M Mode	5 Simplex Mode
2	4	1	ON	OFF	OFF or ON
		2	ON	OFF	OFF or ON
		3	OFF	OFF	ON
		4	OFF	OFF	ON
		5	OFF	OFF	ON
1	3	6	ON	OFF	OFF or ON
		7	ON	OFF	OFF or ON
		8	OFF	OFF	ON
		9	OFF	OFF	ON
		10	OFF	OFF	ON

NOTE:
DIP switches 1–5 control jack 2 or 4 and switches 6–10 control jacks 1 or 3. Switches 1 and 2 and 6 and 7 determine whether the tie trunks are protected or unprotected. Switches 3, 4, 5, and 8, 9, and 10 determine E&M or Simplex mode. The signaling type (1C, 1S, or 5) is determined through system programming.

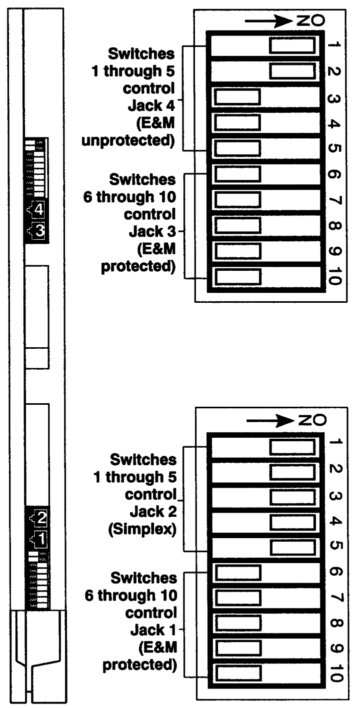


Figure 3–1. Sample settings for the 400EM Module DIP Switches for Signaling Type 1C (Jacks 1 and 3), Type 5 (Jack 2), and 1S (Jack 4)

Programming Tie Trunk Options

You must program the following tie trunk options through system programming.

- **Direction** (`Direction` on System Programming screen)
  - — **Two-way** (factory setting). Calls can be made in either direction.
  - — **Outgoing only.** Calls can be originated but not received (no ringing).
  - — **Incoming only.** Calls can be received but not originated.
- **Trunk Seizure Types** (`Intype` and `Outtype` on System Programming screen)

  Trunk seizure types can be programmed independently for incoming or outgoing directions and all combinations of the following seizure types are possible:

— **Wink** (factory setting). Use this setting if the originating end of the tie trunk transmits an off-hook signal and waits for the remote end to send back a signal (a wink), indicating that it is ready to receive dialing information.

— **Immediate.** Use this setting if no start signal is necessary and dialing can begin immediately after the tie trunk is seized.

— **Delay.** Use this setting if the originating end of the tie trunk transmits an off-hook signal and waits for the remote end to send back an off-hook to on-hook transition as a start dial signal.

— **Automatic.** Use this setting if calls are routed directly to an extension without dialing. When the user picks up the handset, the call rings immediately at the other end. This is also called an *automatic ringdown tie trunk*.

Wink, immediate, and delay types are also called *dial-repeating tie trunks*.

- **E&M Signaling Type** (`E&M Signal` on System Programming screen)

  — **Type 1 Standard** (factory setting). Use this setting if tie trunks connect to the other system through the local telephone company.

  — **Type 1 Compatible.** Use this setting if tie trunks connect directly to a system that uses Type 1 standard signaling and if the trunks are located near this system.

  — **Type 5.** Use this setting if tie trunks connect directly to a system that uses Type 5 signaling and if tie trunks are located near this system.

- **Dial Mode** (`Inmode` and `Outmode` on System Programming screen)

  Inmode and Outmode dialing can be set independently for a tie trunk.

  — Rotary (factory setting)

  — Touch-Tone (cannot be programmed for incoming immediate signaling tie trunks).

  NOTE:
  If you program the 400EM module for touch-tone inward dialing and there are no modules in the system that provide touch-tone receivers (such as 012, 008 OPT, 400, 400 GS/LS/TTR, 800 GS/LS-ID, or 800 DID modules), you must install a 400 GS/LS/TTR module.

- **Dial Tone** (`Dialtone` on System Programming screen)

  This setting determines whether the system provides a dial tone for people calling in on a tie trunk:

  — Remote (factory setting). The system sends a dial tone to the remote end.

  — Local. The system does not send a dial tone to the remote end.

- **Answer Supervision Time** (`AnsSupvr` on System Programming screen)

  Sets a time limit in milliseconds (ms) that an answer supervision signal must be present to be considered valid:

  — 300 ms (factory setting)

  — 20 to 4800 ms (increments of 20 ms)

- **Disconnect Time** (`Disconnect` on System Programming screen)

  Sets a time limit in milliseconds (ms) that a disconnect signal must be present to be considered valid:

  — 300 ms (factory setting)

  — 140 to 2400 ms (increments of 10 ms)

Tie Trunk Compatability

The choice of a tie trunk signaling format to connect two systems depends on the particular application and the systems being connected, including whether or not the tie trunk signals pass through telephone company lines or over customer-owned cable. Table 3–2 shows how to determine tie trunk compatibility between this system and other systems.

Table 3–2. Tie Trunk Compatibility

Installation Situation		Preferred Signaling Format			
From System		System		Remote Extension	
To	**Location**	**Signaling Mode and Type**	**Protected or Unprotected**	**Signaling Mode and Type**	**Protected or Unprotected**
MERLIN II	Same site or interbuilding	Type 5 Simplex	N/A	Type 5 Simplex	N/A
System 25 System 75	Same site or interbuilding	Type 5 Simplex	N/A	Type 5 Simplex	N/A
System 85 or DEFINITY	Same site or interbuilding	Type 5 Simplex	N/A	Type 5 Simplex	N/A
Dimension PBX	Same site	E&M Type 1 Compatible	Unprotected	E&M Type 1 Standard	Unprotected

Continued on next page

Table 3–2, *Continued*

Installation Situation		Preferred Signaling Format			
From System		System		Remote Extension	
To	**Location**	**Signaling Mode and Type**	**Protected or Unprotected**	**Signaling Mode and Type**	**Protected or Unprotected**
Dimension PBX	Interbuilding	E&M Type 1 Compatible	Protected	E&M Type 1 Standard	Protected
Other	Same site	E&M Type 1 Compatible	Unprotected	E&M Type 1	Unprotected
Other	Interbuilding	E&M Type 1 Compatible	Unprotected	E&M Type 1 Standard	Requires a protection unit
LEGEND	Same site or interbuilding	Type 5 Simplex	N/A	Type 5 Simplex	N/A
Network Interface		E&M Type 1 Standard	Unprotected	N/A	N/A

Tie Trunk Networking

The system supports only nontandem tie trunk networking (see Figure 3–2). Use a nontandem tie trunk network to connect telephone lines at both ends; it cannot connect another tie trunk or other facilities.

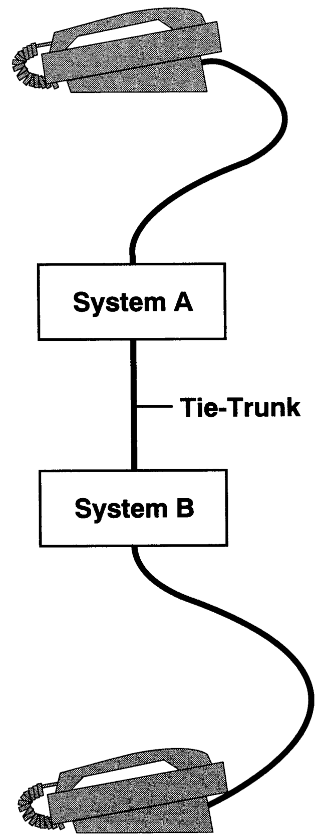


Figure 3–2. Nontandem Tie Trunk Network

Direct Inward Dialing Trunks

Direct Inward Dialing (DID) trunks allow incoming calls to reach specific individuals or groups without the assistance of a system operator. DID trunks are available only in Hybrid/PBX mode. They connect to the system on an 800 DID module or through DID-emulated channels on a 100D DS1 module (Release 2.0 and later).

When DID service is planned for an installation, blocks of telephone numbers are ordered from the central office. The central office determines the quantity of DID numbers for each block. The block is a list of sequential numbers that will route incoming calls to the corresponding DID trunks. Refer to System Programming later in this chapter for setting up DID blocks.

The DID numbers should correspond to the extension numbers for individuals, calling groups, or remote access or pool dial-out codes.

SECURITY ALERT:

DID numbers that correspond to pool dial-out codes (or facility access codes) can be used to evade toll restriction, leading to toll abuse and/or fraud. (See Appendix A, "Customer Support Information," for more information on security.)

Because DID trunks allow calls to come directly to an extension, they cannot be pooled. The telephone company's CO passes the necessary digits to the system, which delivers the call directly to the dialed extension. If the extension numbers in the system are fewer than four digits and the CO sends four, you can program the system to ignore the leading digit(s). For example, if the DID number sent by the CO is 2157, the extension numbers the system can access are 57, 157, or 2157. System programming determines the proper extension numbers to connect.

You assign options to blocks of DID trunks. The system supports a maximum of two blocks of DID trunks, and you can configure each block to match the system numbering plan. For example, the system can have both 3- and 4-digit extension numbers. Trunk block 1 can contain the options needed to reach the 3-digit numbers, and trunk block 2 can contain the options needed to reach the 4-digit numbers.

The system can receive 1- to 4-digit extension numbers over DID trunks. The number of digits received on a specific DID trunk is always the same for that trunk or for the trunks in a particular DID trunk block; however, different DID trunks (or trunk blocks) can receive different numbers of digits.

You can also program them to match more digits than are received from the CO. For example, if you set up the system to match three digits and the CO sends the number 24, the system might insert a 9 in front of the 24 (resulting in the number 924) to complete the match and connect the call.

The system does not route calls until it receives the designated number of digits. Furthermore, the system directs incoming DID numbers that do not match a valid extension to a predesignated extension, such as the operator position, or you can program the system to send back a reorder tone.

System Programming

You must program the following options for each trunk group:

- **DID Trunk Dialing Protocol Type.** Describes the dialing protocol that determines when address digits are sent from the CO to the system.

- **Wink Start** (factory setting). The system signals the CO when it is ready to receive incoming address digits. This is the preferred setting if the local telephone company can support it. Wink-start allows a greater probability of call completion during heavy calling periods.
- **Immediate Start.** The CO sends digits about 65 ms after line seizure (required by the serving CO). You cannot program immediate start for a trunk block that uses dual-tone multi-frequency (DTMF) for passing address signs. Use this setting if the local telephone company can support only immediate start.

■ **DID Trunk Address Signaling Type.** Describes the method by which the system transmits address signals from the serving CO.

- **Dial Pulse** (factory setting). Rotary.
- **DTMF** (not allowed for immediate-start trunks). Touch-tone.

■ **Expected Number of Digits.** Indicates the number of address digits expected from the CO on DID calls in this trunk block. The system blocks a value greater than 4 or less than 1.

- **3** (factory setting)
- **1 to 4**

■ **Number of Digits to Delete** (Delete Digits, 0 to 4, factory setting 0). If the telephone company sends more digits than are supported in the chosen system numbering plan, this is the number of leading digits that the system should delete from the digits sent by the local telephone company.

- **0** (factory setting). Use this setting when the number of digits sent by the telephone company matches the number of digits in the chosen system numbering plan.
- **0 to 4.** The system blocks a value greater than 4.

■ **Digits to Add** (Add Digits, 0 to 9999, factory setting 0). Specifies the digits to prefix before dialed digits in order to determine a routing number. Use this setting when the system must add to the digits sent by the local telephone company because the number of digits sent is fewer than the number of digits in the chosen system numbering plan.

- **0** (factory setting). Use this setting when the number of digits sent by the telephone company matches the number of digits in the chosen system numbering plan.
- 1-to-4-digit number (1 to 9999)

You can program the following options for each DID trunk:

■ **Trunk Disconnect Timing** (10 ms to 2550 ms, factory setting 500 ms). The DID circuit module uses this setting to determine the time needed before the system considers a disconnect from the CO to be valid.

- **Trunk Number.** The line number assigned to every trunk or line in the system on startup serves as the trunk number. You can change the trunk number with the System Numbering feature.
- **Alphanumeric Label** (factory setting Outside). An ASCII string with up to seven characters that you can assign to an individual DID trunk.

Through system programming, you can assign DID trunks as personal lines on any extension in the system, but they are not assigned automatically to any operator position or other extension. A personal line for a DID trunk is not for calls, but for monitoring the facility by observing the light.

Do not assign DID trunks as Music On Hold or paging ports.

Programming Emulated DID Trunks

To program emulated DID trunks on a DS1 circuit module in Release 2.0 and later, note the following options:

- A menu item to program a single channel on the circuit module as DID
- A menu item to program a DS1 circuit module as DID

When a DS1 channel is programmed as an emulated DID trunk, the trunk is automatically added to the first DID block. The following considerations apply:

- The maximum number of DID trunks in a DID trunk block is the system maximum number of trunks (24 or 80, depending on system configuration).
- Only trunks on the 800 DID module and channels programmed as DID on the DS1 facility are allowed in DID trunk blocks.
- A DID trunk cannot appear in more than one DID trunk block.
- If you want to move a DID trunk from one block to another, simply assign the trunk to the new trunk block. The trunk is automatically removed from the previous trunk block and assigned to the new one. The trunk then takes on the programmed attributes of the new trunk block.
- A DID trunk always appears in one of the two DID trunk blocks, even if a physical channel is not present between the circuit module and the CO.

Digital Signal 1 Facilities

A Digital Signal 1 (DS1) facility is a transmission system that transports digital signals. The interface that connects DS1 facilities to the system is the 100D module. Through this module, people can make or receive voice and data calls on a DS1 facility.

Twenty-four Digital Signal 0 (DS0) channels, each operating at 64 kbps, plus framing bits, are multiplexed to form a DS1 signal of 1.544 Mbps. Each DS0 channel within the DS1 signal corresponds to a logical trunk. Even though there is only one physical jack, the 100D module supports up to 24 logical trunks.

In DS1 format, calls to other digital PBXs or COs remain digital, and the system does not need to convert signals to analog for acceptance by the connecting trunk. In addition, you can configure the 100D module to work with T1 or Primary Rate Interface (PRI) service.

To connect the 100D module to an outside DS1 facility, use a channel service unit (CSU). The CSU regulates the transmission into and out of the 100D module so that the module matches the transmission of the outside facility.

The CSU is located on the customer's premises and is used to connect the system to DS1 network facilities. The CSU has three functions:

- To terminate an outside DS1 facility on the 100D module
- To ensure that the signals entering the public network comply with the requirements of the DS1 facility as specified by the FCC
- To include maintenance, diagnostic, and testing capabilities

Both ends of the DS1 facility must be able to communicate. To ensure communication, use system programming to set the following options to match the transmission of the outside DS1 facility:

- Type of service (T1 or PRI)
- Framing format
- Line code
- Line compensation
- Clock synchronization
- Signaling mode (for T1 service only)

The appropriate setting for each option is determined by the transmission facility to which the module is connected. Each option is discussed below.

Type of Service (T1 or PRI)

The system supports two types of service for DS1 facilities: T1 and PRI. You can program the 100D module to operate in either type of service. T1 service transmits and receives analog voice and analog data; PRI transmits and receives digital voice and digital data.

A T1 or PRI line/trunk can provide any combination of the following AT&T Services Network (ASN) services listed below. PRI on the system does not support access to international service.

- **Megacom WATS.** This service supports domestic long-distance outgoing voice calls.
- **Megacom 800.** This service supports domestic toll-free incoming voice calls. T1 and PRI services support Megacom 800 with or without Dialed Number Identification Service (DNIS). This service is provided by the AT&T Switched Network. It routes incoming 800 or 900 calls according to customer-selected parameters, such as area code, state, or time of call. For example, a customer can specify that calls received from a particular area code be routed to a specific individual or group responsible for accounts in the area.
- **Software Defined Network (SDN).** An AT&T Switched Network (ASN) service that supports voice and circuit-switched data calls. SDN lets businesses use portions of the ASN in concert with their dedicated private line networks. However, the system does not support the uniform dialing plan that is necessary for complete integration with SDN.
- **MultiQuest®.** This service supports domestic toll incoming voice calls (900 number). T1 and PRI support MultiQuest with or without DNIS.

PRI supports Shared Access for Switched Services (SASS), which allows both Megacom and Megacom 800 services to be offered over the same line. This eliminates the need to have separate incoming and outgoing trunks. In Release 2.0 and later, PRI also supports Call-by-Call Service Selection. This feature allows maximum utilization of communications lines, providing more than one PRI service for each B-channel.

In addition, in Release 2.0 and later, PRI also supports Station Identification/Automatic Number Identification (SID/ANI) as Calling Party Number. The Calling Party Number (CPN) in Release 1.0 is facility-based, whereas it can be extension-based in Release 2.0 if so programmed. Extension-based CPN is called SID/ANI and can provide the extension number information for outgoing calls when the far-end party supports it.

T1 Service

T1 is the factory setting for DS1 facilities, allowing you to program each of the 24 channels to emulate tie, loop-start, ground-start, and DID trunks in any combination. This means that a single 100D module can take the place of 24 regular outside lines. If you use common-channel signaling, 23 channels are available for emulation and the twenty-fourth channel carries formatting signals.

You can connect the system's control unit to another system's control unit using a digitally emulated tie trunk on a DS1 facility, which is connected to a 100D module and programmed for T1-type transmission. You can use a back-to-back connection from one DS1 facility to the other when the total cable distance is less than 1300 ft. (396 m).

Primary Rate Interface Service

The Primary Rate Interface (PRI) is a standard access arrangement that connects the system to a network providing voice and digital data services through a 4ESS™ Generic 16, a 5ESS Generic 6 (Release 2.0 and later), and a 5ESS serving the FTS2000 (Release 2.0 and later, Federal government only) network.

A PRI facility consists of 24 channels each with a bandwidth of 64 kbps. DS1 refers to the twenty-four 64-kbps channels plus framing and signaling bits multiplexed together to form a 1.544-Mbps signal. You can assign up to 23 B-channels (*bearer channels*); the twenty-fourth is always a D-channel (*data channel*). PRI always uses common channel signaling.

A B-channel carries end-to-end user information, such as the voice or data content of a call, between the system and the far-end switch. Each B-channel provides access to one or more network services. Release 1.0 supports access to only one network service per B-channel. In Release 2.0 and later, Call-by-Call Service Selection allows multiple network services over the same B-channels.

The D-channel conveys signaling information required to set up, control, and clear calls made over the other 23 B-channels. Each PRI must include a D-channel, but may include fewer than 23 B-channels. You cannot use the remaining channels for any other purpose.

You can connect up to three PRIs to the system (no more than 2 per carrier) through separate 100D circuit modules, each of which occupies a slot in the control unit. In terms of system capacity, each PRI line counts as a trunk endpoint, so the maximum number of B-channels supported by the system is 69. Three separate D-channels provide the signaling.

Release 2.1 and later provides direct PRI connection to the following DEFINITY communications systems:

- Generic 1.1
- Generic 2.1
- Generic 3i
- Generic 3r
- Generic 3s

NOTE:
The MERLIN LEGEND Communications System does not support Non-Facility Associated Signaling (NFAS). NFAS supports multiple PRI links over one D-channel and the MERLIN LEGEND Communications System requires a dedicated D-channel for each PRI link.

Framing Format

To identify the DS0 channels, the DS1 signal is segmented into blocks of 193 bits called *frames.* A frame consists of 24 eight-bit words (one for each channel) plus a framing bit at the beginning of each frame: 24 words x 8 bits =192 bits. Thus, the framing bit is the one hundred ninety-third bit of each 1.544-Mbps DS1 signal. Frames repeat at a rate of 8000 per second, with each frame repeating DS0 channels 1 through 24 sequentially.

A 100D module (T1 service) uses two methods of framing: D4 or extended superframe (ESF):

- **D4** (factory setting). This format consists of 24 8-bit time slots and one framing bit. To perform synchronization, the receiving equipment uses the framing information to identify the start of each frame and the frames that contain signaling information. The framing information repeats once every 12 frames; these 12 frames form the D4 frame.
- **ESF.** The ESF format extends the 12-frame D4 frame to a 24-frame. The 24 framing bits include a cyclic redundancy check (CRC) for the entire extended superframe and a facility data link for maintenance. The ESF can detect more errors than D4 framing; however, DS1 equipment does not always support this format.

The framing method you choose must match the framing at the remote extension, and you must program the format selected when service was ordered. To identify the DS0 channels, the DS1 signal is segmented.

Line Coding

The DS1 signal consists of a continuous bit stream of 1s and 0s, encoded into bipolar pulses for transmission. Only the 1s create a pulse; the 0s represent the absence of a pulse. The pulses of the 1s alternate between positive and negative. This type of line coding is called bipolar or alternate mark inversion zero code suppression (AMI-ZCS). The line-coding formats guarantee that the ones-density requirement is met to achieve clock recovery. To meet the ones-density requirement, choose either AMI-ZCS or Bipolar 8 Zero Substitution line coding.

The line coding formats are the following:

- **AMI-ZCS** (factory setting). This line coding monitors each DS0 channel and prevents strings of 8 or more 0s. Upon detecting an all-zero channel octet, AMI-ZCS line coding changes the seventh 0 (second least significant bit) to a 1.

  With AMI-ZCS line coding, an overwritten bit has no noticeable effect on voice and voice-grade data. However, the AMI-ZCS line-coding format can cause errors in digital data transmission.

- **Bipolar 8 Zero Substitution** (B8ZS). B8ZS line coding inserts 8 consecutive 0 bits into a unique binary sequence with a *bipolar violation* in bit positions 4 and 7. Normally for bipolar transmission, 0s are encoded alternately as positive then negative, or negative then positive, pulses. If two positive or two negative pulses are received in succession, a bipolar violation occurs.

  Ordinarily, noise hits to the signal cause bipolar violations. However, B8ZS line coding allows the detection of 8-bit strings at the receiving end and their conversion back into the original sequence.

  B8ZS line coding is better than ZCS because it does not cause errors in data transmission.

  The ACCULINK 3150/3160/3164 channel service units pass B8ZS violations.

Line Compensation

Line compensation adjusts for the amount of cable loss in decibels (dBs), based on the length of cable between the 100D module and the CSU or other far-end connection point. The factory setting is a value of 1, which allows a maximum loss of 0.6 dB. See Table 3–3 for other possible settings.

Table 3–3. Line Compensation Settings

Setting	dB Loss	Cable length (22-Gauge Wire)
1	0.6	0–133 feet
2	1.2	133–266 feet
3	1.8	266–399 feet
4	2.4	399–533 feet
5	3.0	533–655 feet

NOTE:
Cable length in Table 3–3 is the distance between the 100D module and the CSU. If the system does not have a CSU, the distance between 100D modules is doubled.

Clock Synchronization

Clock synchronization is an arrangement where digital facilities operate from a common clock. Whenever digital signals are transmitted over a communications link, the receiving end must be synchronized with the transmitting end to receive the digital signals without errors.

The system synchronizes itself to the network by extracting the timing signal from the incoming digital stream. If the system has more than one 100D module, you must identify the module that provides the primary synchronization for the other 100D modules and for the time-division multiplexing (TDM) bus.

You can provide backup synchronization in the event of a maintenance failure by programming the second and third installed modules as secondary and tertiary synchronization.

In addition, the source of synchronization can be factory set (the factory setting synchronizes the clock to the external station) or reset to Local Clock Reference Source.

Signaling Mode

Signaling is the process of communicating channel-state information (such as dialing) from endpoint to endpoint. The system supports two types of signaling in T1 transmission: robbed-bit signaling (RBS) and common-channel signaling (CCS). You must choose a signaling mode only for T1 service; PRI always uses CCS (23 B-channels and 1 D-channel). Below is a description of the two types of signaling.

- **Robbed-Bit Signaling.** RBS replaces (or robs) the least significant bit of every sixth frame of each DS0 channel with signaling information. RBS is also called *in-band signaling* since signaling information is embedded in the least significant bit of every sixth 8-bit word.

  RBS is appropriate for voice and voice-grade modem data.

- **Common-Channel Signaling.** CCS is a format that places the signaling bits for channels 1 through 23 into the 8-bit word of the twenty-fourth channel. This restricts DS1 from using the twenty-fourth channel for voice or data transmissions. D4 framing does not preclude the use of CCS, but CCS is not compatible with D4 channel banks because the D4 channel banks only recognize RBS. Coupled with B8ZS coding and PRI, CCS can support digital data transmission up to 64 kbps per channel.

Framing Formats and Signaling Modes

When the DS1 facility allows it, you should use ESF framing to take advantage of its improved maintenance, diagnostic, and testing capabilities (the ACCULINK 3150 or 3160/3164 CSU is required to interface with the network). If the transmission between two systems is voice-only, use RBS for all 24 communication paths. For voice transmission, you can use both ZCS and B8ZS line coding to satisfy the ones-density requirement; the preferred line-coding format is B8ZS, which is needed for 64-kbps digital data transmission on PRI.

NOTE:
You must use an ACCULINK 3150 or 3160/3164 CSU for DS1 connections through a central office and between buildings.

Applications

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This chapter provides an *overview* of the applications that you can connect to the system. For complete information about the use of any application discussed here, refer to the documentation for that product.

The system supports the following applications for enhanced call handling and system management capabilities:

- PassageWay Direct Connect Solution
- MERLIN Identifier
- Stand-alone voice messaging applications
  - MERLIN MAIL Voice Messaging System (VMS)
  - AT&T Attendant
- Stand-alone call accounting and management applications
  - Call Accounting System (CAS) for DOS only or Windows
  - Call Accounting Terminal (CAT)
  - Call Management System (CMS)
- Stand-alone system management application: System Programming and Maintenance (SPM) for DOS
- Integrated applications
  - Integrated Solution II (IS II) applications
    - AUDIX Voice Power
    - CAS
    - SPM

- Integrated Solution III (IS III) applications
  - AUDIX Voice Power
  - IS CAS
  - SPM
  - Fax Attendant
  - Integrated Voice Power
- Primary Rate Interface (PRI) applications
  - Group IV (G4) fax
  - PictureTel video conferencing
- Stand-alone Fax and imaging services
  - MERLIN PFC Telephone
  - Automated Document Delivery System (ADDS)
  - Picasso Still-Image Phone
- CONVERSANT
- Centrex operation

Organization of Descriptions

The following sections provide a brief description of each application, service, or system. Most descriptions include the subheadings below. When a subheading is not applicable to a given application, it does not appear.

- **Mode Differences.** Lists any differences or limitations of the application in Key, Hybrid/PBX, or Behind Switch modes of operation.
- **Considerations and Constraints.** Discusses restrictions, capacities, and other information that you should consider before installing or using the application.
- **Feature Interactions.** Provides information about system and telephone features that affect how the application works and notes any features that do not work with the application.
- **System Programming.** Provides an outline of the system programming required to set up the application.
- **Platform Requirements.** Lists the hardware and software required to connect the application to the system.

Also see *System Planning* for planning instructions, *System Programming* for complete system programming instructions, and the documentation provided with the application for connection diagrams and installation instructions.

System Support for Applications

Table 4–1 summarizes the system's capacity to support each application and identifies the modes of operation in which you can use the application.

Table 4–1. Application Capacities and Modes of Operation

Application	Capacity	Key	Hybrid /PBX	Behind Switch
PassageWay Direct Connect Solution	127 (MLX only)	✔	✔	✔
MERLIN MAIL VMS	1 (2 or 4 jacks)*	✔	✔	
Number of mailboxes	40			
AT&T Attendant	4*	✔	✔	
CAS Plus V3/CAS for Windows	1	✔	✔	✔
CAT	1	✔	✔	✔
CMS	2	✔	✔	
Number of lines/trunks (each)	28			
Number of agents (each)	28			
Number of external alerts (each)	4			
SPM (stand-alone)	1	✔	✔	✔
IS II	1	✔	✔	
AUDIX Voice Power	1	✔	✔	
Number of mailboxes	300			
Integrated Voice Power	1	✔	✔	
Automated Attendant	1	✔	✔	
IS CAS	1	✔	✔	
SPM	1			
IS III	1	✔	✔	
AUDIX Voice Power	1	✔	✔	
Number of mailboxes	300			
CAS IS III	1	✔	✔	
SPM	1	✔	✔	
Fax Attendant	1	✔	✔	
PRI				
Group IV (G4) fax		✔	✔	
Video conferencing			✔	

Continued on next page

Table 4–1, *Continued*

Application	Capacity	Key	Hybrid /PBX	Behind Switch
Centrex operation		✔	✔	✔
ADDS	1 voice jack and 1 fax jack	✔	✔	
CONVERSANT	1	✔	✔	✔

* These attendant applications are mutually exclusive.

Supported Printers

The following table shows the printers that are supported with the optional applications discussed in this chapter.

Table 4–2. Applications Printers

Printer	Document No.	Description
AT&T CAS Printer	582-421-105	9-pin dot matrix printer that provides choice of print quality and speed. Uses parallel connection to the computer.
AT&T Applications Printer	582-421-106	9-pin dot matrix printer that provides choice of print quality and speed. Has wide carriage that accommodates pin-feed paper up to 14 7/8 in. (37.8 cm) wide. Uses parallel connection to the computer.
Call Accounting Terminal (CAT) Printer	582-421-100	9-pin dot matrix printer that provides choice of print quality and speed. Uses serial connection to the computer.

PassageWay Direct Connect Solution

IMPORTANT:
This section is intended solely as an overview of the application. For comprehensive information about the use of the application, see the documentation for the product.

NOTE:
This entry describes Passageway Direct Connect Solution, required for MERLIN LEGEND Communications System, Release 3.0. PassageWay Direct Connect Solution (Release 1.0) can be used with Release 2.1 MERLIN LEGEND Communications System.

PassageWay Direct Connect Solution is a collection of software applications and a hardware adapter. It provides an API (applications programming interface) link between a PC with Windows 3.1 or later and the MERLIN LEGEND Communications System via an MLX-28D, MLX-20L, or MLX-10DP telephone. PassageWay Direct Connect Solution includes these applications:

- **AT&T Call.** A cardfile application that enables you to maintain information such as names, addresses, and telephone numbers. You specify the information that you want to store. With AT&T Call, you can place a call directly from the PC and keep a log of all outgoing calls.
- **AT&T Set.** A telephone programming application that enables you to program telephone features for your telephone from your PC. You can also create and save multiple-button programming files for your telephone. You can exchange these files with other AT&T Set users.
- **Log Viewer.** An application that enables you to view the entries stored in the PassageWay Direct Connect Solution call log. The call log stores a record of every call you make using AT&T Call.
- **AT&T Connect.** Management software that provides the basis for other PassageWay Direct Connect Solution software applications and the diagnostic features to troubleshoot these applications. Auto dialing capabilities using the Hayes-compatible command set are also provided.
- **AT&T Buzz.** You can manage incoming calls (answer, hold, or drop) and view the calling party number (Caller ID) for each incoming call at your telephone.

Considerations and Constraints

If there are problems connecting PassageWay Direct Connect Solution to a communications port, see the PassageWay Direct Connect Solution manual for information on PC serial ports.

Feature Interactions

Idle Line Preference

Your MLX telephone should have Idle Line Preference activated. With Idle Line Preference activated, the system automatically selects a line for outgoing calls when you go off-hook.

The system manager should set Automatic Line Selection on your telephone so that your Idle Line Preference is on an **ICOM** button (in Key or Behind Switch mode) or an **SA** button (in Hybrid/PBX mode). Ensuring that Automatic Line Selection is set to an **ICOM** or an **SA** button means you can make both inside and outside calls via AT&T Call. (You make outside calls on an **ICOM** or **SA** button by dialing 9.)

Platform Requirements

To use PassageWay Direct Connect Solution on the system, you must have the following components:

- An AT&T-approved personal computer (PC) with a 286, 386, or more powerful microprocessor and:
  - Available serial port
  - Minimum of 2 MB of RAM (4MB preferred)
  - 3.5-in., 1.44-MB, high-density floppy disk drive or 5.25-in., 1.2-MB, high-density floppy disk drive
  - Hard disk with 2 MB of space available (after Windows is already installed)
  - VGA monitor
  - Windows-compatible pointing device (mouse or trackball recommended)
- Microsoft Windows, version 3.1 or later
- MLX-28D, MLX-20L, or MLX-10DP telephone connected to the system
- PassageWay Direct Connect Solution software
- PassageWay Direct Connect Solution adapter
- 9-pin to 25-pin adapter
- 4-foot, 4-pair keyed modular phone cord (D8AC)

Additional MERLIN LEGEND Communications System requirements are:

- MLX telephones must be wired with 3- or 4-pair extension wiring; otherwise local telephone power is required.
- If you use a console and DSS, local telephone power is required.

Voice Messaging Systems

IMPORTANT:
This section is intended solely as an overview of the applications. For comprehensive information about the use of the applications, see the documentation for the products.

SECURITY ALERT:

Your voice messaging system permits callers to leave verbal messages for system users or gain access to the backup position in an emergency as well as create and distribute voice messages among system users.

The voice messaging system, through proper programming, can help you reduce the risk of unauthorized persons gaining access to the network. However, phone numbers and authorization codes can be compromised when overheard in a public location, are lost through theft of a wallet or purse containing access information, or through carelessness (writing codes on a piece of paper and improperly discarding them). Additionally, hackers may use a computer to dial an access code and then publish the information to other hackers. Substantial charges can accumulate quickly. It is your responsibility to take appropriate steps to implement the features properly, evaluate and program the various restriction levels, protect and carefully distribute access codes.

Under applicable tariffs, you will be responsible for payment of toll charges. AT&T cannot be responsible for such charges and will not make any allowance or give any credit resulting from unauthorized access.

To reduce the risk of unauthorized access through your voice messaging system, please observe the following procedures:

- *Employees who have voice mailboxes should be required to use passwords to protect their mailboxes.*
  - *Have them use random sequence passwords.*
  - *Impress upon them the importance of keeping their passwords a secret.*
  - *Encourage them to change their passwords regularly.*
- *The administrator should remove any unneeded voice mailboxes from the system immediately.*
- *AUDIX Voice Power has the ability to limit transfers to subscribers only. You are strongly urged to limit transfers in this manner.*
- *Use the system programming capability to do the following:*

- — *Block direct access to outgoing lines and force the use of account codes/authorization codes.*
- — *Disallow trunk-to-trunk transfer unless required. If you are using AUDIX Voice Power (AVP), you do not use a MERLIN LEGEND Communications System feature to do this; instead, you use the AVP Transfer to Subscribers Only feature. If you are using MERLIN MAIL, you must use Disallowed Lists rather than restricting trunk-to-trunk transfer. Assign toll restriction levels to all voice messaging ports.*
- — *If you do not need to use the outcalling feature, completely restrict the outward calling capability of the voice messaging ports.*
- — *If you need to use the outcalling feature, use Allowed/Disallowed Lists and calling restrictions (see the* Feature Reference *for details).*

- *Monitor SMDR reports or Call Accounting System reports for outgoing calls that might be originated by voice messaging ports.*

A voice messaging system (VMS) provides call-answering services and may provide voice mail services. Each of the following VMS applications connects to an enhanced T/R port, called a *voice messaging interface (VMI)* port:

- MERLIN MAIL Voice Messaging System
- AT&T Attendant
- AUDIX Voice Power
- Integrated Voice Power (IVP) Automated Attendant

You can program T/R ports on an 012 module either as *generic* VMI ports or *integrated* VMI ports. MERLIN MAIL and AUDIX Voice Power use streams of touch-tone codes, called *mode codes*, to communicate with the control unit. Because these applications use mode codes, you must connect the hardware to integrated VMI ports. You can connect AT&T Attendant and IVP Automated Attendant, which do not use mode codes, to generic VMI ports. See Table 4–6 in the "MERLIN MAIL" section, later in this chapter.

SECURITY ALERT:

Beginning with Release 2.1, an 012 port that is programmed as a generic VMI port can transfer an outside call to an outside number. Previously, only VMI ports programmed as integrated VMI ports could do a trunk-to-trunk transfer. A single-line telephone connected to an integrated VMI port can complete trunk-to-trunk transfers.

Calling restrictions (for example, Disallowed Lists, Toll Restriction, Facility Restriction Levels) should be programmed, as appropriate, to minimize toll fraud abuse, especially if a single-line telephone is connected to an integrated VMI port. Refer to the Feature Reference *for additional information on programming calling restrictions.*

A VMS requires touch-tone receivers (TTRs); the number it requires depends on the number of VMI ports, as shown in Table 4–3. These TTR requirements apply *only* to a VMS; they do *not* include the TTR needs of T/R sets.

Table 4–3. TTRs Required by Voice Messaging Systems

No. of VMI Ports	No. of TTRs Required	No. of 012 Modules*	No. of 400* or 400 GS/LS/TTR* Modules
1	1	1	0
2	1	1	0
3	2	1	0
4	2	1	0
6	3	2	0
8	4	1	1
12	6	3	0
		2	1

* These figures reflect the required number of modules if only these modules are supplying TTRs for the VMSs.

The 012 module supplies two TTRs; The 400 and 400 GS/LS/TTR modules supply four TTRs. The 800 DID and 800 GS/LS-ID modules each supply two TTRs. Table 4–4 identifies the modules and the number of TTRs each supplies. (The 008 OPT module also supplies TTRs, but VMSs cannot be directly connected to it.)

Table 4–4. Modules with TTRs

Module	Number of TTRs
008 OPT	2
012	2
400 GS/LS/TTR	4
400	4
800 DID	2
800 GS/LS-ID	2

NOTE:
A voice messaging system cannot be directly connected to an 008 OPT module. However, the TTRs supplied by the 008 OPT module can be used by the voice messaging system.

The following symptoms indicate that the system needs more TTRs:

- Single-line telephone users do not get dial tone when trying to dial out.
- The voice messaging system fails to transfer calls.
- Calls fail to ring or go to coverage prematurely.

Voice Messaging Interface (VMI) Port Capabilities

VMI ports use switchhook flashes for Hold, Transfer, Conference, and Drop in the same way single-line telephones do. VMI ports also have the ability to perform transfer redirection, respond to far-end disconnect, and, in the case of integrated VMI ports only, send call information and mark a port in or out of service. The following sections describe these capabilities. Beginning with Release 2.1, both integrated and generic VMI ports can perform trunk-to-trunk transfer.

NOTE:
On an 012 module, only four ports can ring simultaneously. If you are using an attendant or voice messaging system that requires eight of the 012 jacks on a single module, you should not use the remaining jacks on the module. If the application uses fewer than 8 jacks, you may use the remaining jacks for T/R devices such as single-line telephones.

Transfer Redirect

If unanswered by the end of the transfer redirect time interval (0–9 rings), a call transferred from a VMI port alerts at the VMS transfer redirect extension, rather than returning to the VMI port that originated the transfer. For example, you might program Extension 15 as a VMI port for a AT&T Attendant and set the transfer redirect time interval to four rings. When a call comes in on Extension 15, the caller listens to a recording and dials a request for Extension 24. The call rings at Extension 24 four times without being answered. The system redirects the call to Extension 10, the system operator; it does not redirect the call back to Extension 15.

On an unsupervised transfer (described in "Automated Attendant," later in this chapter), when the transfer destination is busy or is an invalid extension, the transfer redirect is immediate (no time interval). If the system cannot alert the transfer redirect extension (all buttons are in use), the VMS will keep trying to alert the transfer redirect extension every 20 seconds until the alert is delivered or the caller hangs up.

Far-End Disconnect

When the system detects a far-end disconnect signal on a line/trunk where a VMI extension is receiving a call, the system sends the disconnect signal to the VMI extension, whether or not that extension is the only party left on the call. If another party is still on the call, the VMS decides whether to continue or disconnect the party. (The far-end disconnect signal occurs only if you program the VMI port for Reliable Disconnect.) Loop-start trunks must be programmed for Reliable Disconnect.

Ports In/Out of Service

When a group call to a VMI extension is not answered within 30 seconds, the call is sent to another available VMI extension in the calling group or is queued back to wait for an available extension in the calling group.

For an integrated VMI extension, the control unit sends messages to inform the VMS that the extension is out of service. Both the VMS and the calling group software mark the unavailable port as out of service. If all VMI extensions go out of service, the system generates a hardware error report.

Every 10 minutes, the system tests each out-of-service VMI extension. If the extension responds to the test, the VMS and the calling group software mark it as *in service*. For an integrated VMI extension, the control unit informs the VMS by sending extension-in-service messages.

MERLIN MAIL

IMPORTANT:
This section is intended solely as an overview of the application. For comprehensive information about the use of the application, see the documentation for the product.

The MERLIN MAIL Voice Messaging System (VMS) is a stand-alone application that provides the following integrated call-handling services:

- Automated Attendant service
- Call Answer service
- Voice Mail service

Automated Attendant

Automated Attendant consists of a greeting and one or more menus, providing callers with a number of options that allow them to quickly access an extension, a department, or information by pressing a single dialpad button. This service provides several major benefits, both to the callers and to the company:

- Different greetings, menus, and announcements can be recorded to play during the day and night.

  For example, during the day you may want to tell callers to stay on the line for assistance by an operator. At night, when there may be no operator, you may want to tell callers to stay on the line to leave a message in the General Mailbox.
- Calls are efficiently routed to the correct party.
- Incoming fax calls from machines that product industry-standard fax (CNG) tones, are recognized and automatically routed to the fax extension. If the fax machine is busy or does not answer within 2 rings, the call is disconnected.
- Callers using rotary phones or needing assistance, are automatically transferred to the system operator or General Mailbox, or disconnected, based on your company's preference.
- If the party the caller wants does not answer or the phone is busy, the caller is prompted to leave a message.
- If callers do not know the extension needed, they can access a directory of subscribers or be transferred automatically to an operator.
- Announcements of frequently requested information (such as directions or business hours) can be included as menu options, freeing an employee's time for other tasks.
- You can set up the system to answer calls immediately (immediate call handling) or after a delay (delayed call handling). If the system is set for delayed call handling, calls unanswered by the system operator are answered by Automated Attendant after a specified number of rings.

Call Answer

The system's Call Answer service allows callers to leave messages or transfer to another extension when the extension called is busy or does not answer. When a message is left, Call Answer deposits the message in the subscriber's voice mailbox, then lights the message-waiting indicator on the subscriber's phone. If the subscriber has Outcalling turned on, the system will also place a call to the specified Outcalling number.

In addition to acting as an answering machine, Call Answer enables callers to perform any of the following actions:

- Press *0* for the system operator
- Transfer to another extension by dialing **T* (or **8*) before or after leaving a message
- Review and edit messages before depositing them in the voice mailbox

- Leave messages in the General Mailbox if the subscriber's voice mailbox is full

Voice Mail

Voice Mail service lets subscribers:

- Listen to messages from non-subscribers and other subscribers
- Record their own personal greetings and names
- Forward a received message to one or more subscribers, with additional comments, if desired
- Assign their own passwords, which they can change to ensure that messages are kept confidential
- Create a message and send it to one or more subscribers
- Designate a telephone number or pager/beeper that is notified when a new message arrives in the subscriber's mailbox

Mode Differences

The system must operate in Key or Hybrid/PBX mode. You cannot connect MERLIN MAIL to a system operating in Behind Switch mode.

Considerations and Constraints

The MERLIN MAIL VMS is available in two-port and four-port configurations. Both configurations have four hours of message storage capacity.

The size of a subscriber's mailbox—that is, the total amount of storage for all the messages it can hold—is variable. Available options are 5, 10, or 15 minutes; 60-minute storage is available for special mailboxes.

Callers with rotary telephones whose calls are answered by Automated Attendant service cannot use the features of the MERLIN MAIL VMS. MERLIN MAIL should be set up to direct these calls to the system operator during business hours.

Automated Attendant answers calls immediately (immediate call handling) or after a delay (delayed call handling).

You program the VMS with a touch-tone telephone. To support remote diagnostics, the MERLIN MAIL VMS is equipped with an RS-232 serial port and an external remote maintenance device (modem).

You should assign Disallowed Lists to the VMI ports that connect the MERLIN MAIL VMS to prevent toll calls from being dialed through the VMS and to allow MERLIN MAIL to call out only to the area codes or numbers you specify. Do not use other calling restrictions.

You cannot use MERLIN MAIL VMS with AT&T Attendant.

Feature Interactions

Coverage

Use system programming to assign all extensions that need coverage to a coverage group. The system doesn't do this automatically. It assigns the MERLIN MAIL VMS ports to a calling group and designates the VMS as the coverage receiver for the coverage group.

Subscribers can program their telephones so that only outside calls are sent to coverage.

In Release 2.0 and later, when subscribers activate Coverage VMS Off for their telephones, normally covered outside calls are not covered. No special programming is needed on MERLIN MAIL to activate this feature.

Group Calling

Use system programming to assign the MERLIN MAIL VMI ports to the same calling group.

With integrated VMI ports, the system uses mode codes to identify calls that overflow from one calling group to another calling group as coverage calls. As a result, the overflow calling group's number appears in the called party field of the mode code.

Leave Message

If the target telephone does not have display capabilities, the Leave Message feature sends mode codes to the MERLIN MAIL VMS to deposit a message.

In Release 2.0 and later, when the MERLIN MAIL VMS sends a Leave Message notification to an extension, the system identifies the VMS as the sender of the message. As a result, when a subscriber uses the Return Call feature, the call goes to any available VMS port, not just to the specific port that generated the message. This reduces the chance of getting a busy port.

Night Service

MERLIN MAIL VMS Automated Attendant works with the Night Service feature to provide specialized after-hours service. Automated Attendant can answer calls on lines it does not handle during business hours. A special night announcement can greet after-hours callers.

Privacy

Privacy is automatic for each MERLIN MAIL VMI port.

Ringing Options

If lines set for answering by Automated Attendant appear on telephones other than the system operator console or backup extension, program them for No Ring.

Transfer

If a call received on a line/trunk is transferred to a VMI port, the direct inside access mode code is sent. The call is treated as a transferred call, and the caller hears the greeting assigned for callers within the system.

You can program any calling group, calling group member, or extension as a VMS transfer redirect extension. If the extension is a QCC, the VMS forwards the transfer redirect call to the QCC as a returning call and does not place it in the QCC queue.

If a transferred caller gets no answer and returns to the system operator, the operator has no indication of the origin of the call.

System Programming

Complete the following procedures so that MERLIN MAIL VMS can work on your system. Refer to *System Programming* for complete procedures.

- Assign all MERLIN MAIL VMS ports to a calling group, set the group type to VMI Integrated, and set the hunt type to Linear.
- Program loop-start trunks for reliable disconnect.
- Specify the touch-tone duration and interval between digits in codes sent between the MERLIN MAIL VMS and the system.
- Specify the VMS transfer return interval. This is the number of rings before a call transferred by the MERLIN MAIL VMS is sent to the system operator.
- Set inside (intercom) dial tone to Outside.
- Assign Disallowed Lists to each VMI port.
- When you use Automated Attendant only for Night Service:
  - If the lines/trunks set for answering by Automated Attendant service appear at other extensions, set the No Ring option for the other extensions.
  - Specify Immediate Answer (one ring) for the VMI ports.
  - Specify the VMS calling group as the Night Service operator.

Platform Requirements

To connect the MERLIN MAIL VMS to the system, you need the following equipment:

- MERLIN MAIL VMS unit and power cords
- Remote maintenance device (a modem and power supply)
- Modem cable with a 9-pin connector at one end and a 25-pin connector at the other, for connecting the remote maintenance device to the serial port on the MERLIN MAIL VMS unit
- D4BU modular cords (two for a two-port system or four for a four-port system, plus one for the remote maintenance device)
- 012 module (and ring generator, if the module is an older one that has the apparatus code 517C13, 517D13, 517E13, or 517F13). Current 012 module [apparatus code 517G13 (28) or higher-lettered code] include built-in ring generators and work with all releases of the system. Models 517A13 and 517B13 cannot be used with Release 3.0.

  NOTE:
  The system may require additional TTRs to allow the 012 module to handle a large number of voice connections. Two TTRs are provided on the 012 module. See Table 4–3 earlier in this chapter.

The number of required VMI ports depends on the number of incoming lines/trunks, the number of subscribers programmed for Automated Attendant service, and the number of busy-hour calls. Table 4–5 lists these requirements.

Table 4–5. MERLIN MAIL Voice Messaging System Ports Required

No. of VMI Ports Requested	Incoming Lines/Trunks	No. of Subscribers or Busy-Hour Calls
2	1 to 6	1 to 20
4	7 to 18	21 to 60

MERLIN Identifier

IMPORTANT:
This section is intended solely as an overview of the application. For comprehensive information about the use of the application, see the documentation for the product.

MERLIN Identifier enables customers to receive information about a calling party, as long as the local telephone company provides the Custom Local Area Signaling Service (CLASS) to enable caller identification and the customer subscribes to that service. The caller's jurisdiction must also support the service.

MERLIN Identifier consists of a control unit that supports four central office trunks and up to four individual display units for viewing calling party information, an optional administrative display unit, and an administrative keyboard. MERLIN Identifier can also connect to a personal computer (PC) for administrative or database management or to a serial printer to output caller identification information.

MERLIN Identifier can:

- Display the telephone numbers of incoming calls when the customer has subscribed to caller identification from the local telephone company and the caller's telephone company supports the service.
- If the telephone number matches one in the MERLIN Identifier database, it can display the name associated with the number.
- If the call has been identified as a "priority" call in the MERLIN Identifier database, it can beep to ensure that the caller gets prompt attention.
- Review information for the last 18 calls on each of 4 lines to ensure that callers have received prompt attention.

When combined with a Windows-based contact management software package or a customized contact management software package running on a PC, MERLIN Identifier can also:

- Transfer caller names and numbers between the MERLIN Identifier and PC databases
- Display a business profile as a call comes in from a caller in the database
- During a call, add notes or a memo to the profile
- Maintain a log or history of answered and unanswered calls

MERLIN Identifier uses existing MERLIN LEGEND Communications System wiring and is compatible with all releases of the system.

AT&T Attendant

IMPORTANT:
This section is intended solely as an overview of the application. For comprehensive information about the use of the application, see the documentation for the product.

AT&T Attendant answers incoming calls and plays a menu of recorded prompts. A caller can respond to the prompts by dialing touch tones. AT&T Attendant then routes the call to an inside extension. AT&T Attendant transfers callers with rotary telephones to a designated extension (Route 0) for further call handling and routing.

You can program AT&T Attendant to transfer calls in either of two ways:

- **Unsupervised Transfer.** AT&T Attendant dials the extension or department requested by the caller and disconnects. If the call is not answered or the extension is busy, communications system routes the call to the redirect extension.
- **Supervised Transfer.** AT&T Attendant transfers the call and retrieves it if the transfer is unsuccessful. AT&T Attendant then directs the call to another telephone, allows the caller a second route choice, or plays a failed-transfer announcement, depending on how you program it.

AT&T Attendant can answer calls immediately *(primary call handling)* or after a delay *(secondary call handling).*

Mode Differences

The system must operate in Key or Hybrid/PBX mode. You cannot use AT&T Attendant on a system operating in Behind Switch mode.

Considerations and Constraints

You cannot connect AT&T Attendant to the system if AUDIX Voice Power or MERLIN MAIL is installed.

You can connect a maximum of four AT&T Attendants to the system.

You can program AT&T Attendant to answer every incoming call or only calls on certain lines/trunks.

You can route calls to an answering machine to allow callers to leave messages if a called extension is busy, if a call is unanswered, or if it is after business hours.

AT&T Attendant can transfer calls to fax machines, if the fax extension number is specified and the caller dials it. (AT&T Attendant does not automatically detect fax tones.)

AT&T Attendant provides 64 seconds for recording up to 5 standard messages, including the caller greetings used during and after business hours, a hold announcement for a caller who is being transferred, a connect announcement for the department or extension receiving a transferred call, and an announcement explaining that a call cannot be completed.

Feature Interactions

Coverage

An inside call on a VMI port that transfers to an inside extension will not go to coverage but will continue to ring at the inside extension until the transfer redirect feature is configured.

In Release 2.0 and later, outside calls that would normally proceed to AT&T Attendant as coverage calls do not do so if the telephone that sends the call to Group Coverage has activated Coverage VMS Off. No special action is needed in AT&T Attendant programming to activate this feature.

Forwarding

Remote Call Forwarding is supported on ports programmed as generic VMI.

Group Calling

You must assign all AT&T Attendants that you connect to the system to the same calling group.

Night Service

AT&T Attendant works with the system's Night Service feature to provide specialized after-hours service. AT&T Attendant can answer calls on lines it does not handle during business hours or can direct calls to ring at a specific night extension or department, such as Building Security. A special night announcement can greet after-hours callers.

Privacy

You must program Privacy for each AT&T Attendant VMI port.

Transfer

If a caller incorrectly specifies the answering VMI port as the desired transfer destination extension, the VMI port may park the call.

System Programming

The following procedures must be completed for AT&T Attendant to function on your system. Refer to *System Programming* for complete procedures.

- Assign all AT&T Attendant ports to a calling group and set the group type to VMI Generic.
- Set inside dial tone to Outside.

- Designate a transfer redirect extension, such as the system operator, to receive calls that were originally transferred to unanswered or busy extensions, or to receive calls when a caller fails to respond to the announcement.
- Program all calling groups as auto-logout factory setting.
- Assign Privacy to each AT&T Attendant VMI port.

Platform Requirements

To connect AT&T Attendant to the system, you need the following equipment:

- AT&T Attendant unit
- 6-wire modular telephone cord
- 012 module (and ring generator, if the module has the apparatus code 517F13). Current 012 modules [apparatus code 517G13 (28) or higher-letter or lower letter] include a built-in ring generator and work with all releases of the system.

  NOTE:
  The system may require additional TTRs to allow the 012 module to handle a large number of voice connections. Two TTRs are provided on the 012 module. See Table 4–3 earlier in this chapter.

The number of AT&T Attendants that the system requires depends on the number of incoming lines/trunks and the number of busy-hour calls. One is normally sufficient for handling after-hours calls only and for delayed call handling. Table 4–6 shows the requirements when you program AT&T Attendant for primary call handling.

Table 4–6. AT&T Attendants Required

No. of Attendants Required	Incoming Lines/Trunks	Busy-Hour Calls
2	1 to 6	1 to 25
3	7 to 9	25 to 50
4	10 to 12	50 to 100

Call Accounting System

IMPORTANT:
This section is intended solely as an overview of the application. For comprehensive information about the use of the application, see the documentation for the product.

A Call Accounting System (CAS) is a software application for businesses that need to manage telephone usage and control costs by tracking, sorting, and recovering telephone charges. CAS provides a menu-driven user interface and on-line help.

There are three versions of CAS:

- **CAS Integrated with IS III** (IS CAS). Starting with Release 1.2 of IS III, this application differs from two versions of CAS described below. Please also see "Integrated Solution III" in this chapter for more information.
- **CAS Plus V3.** This version, for general business use, is a stand-alone application that runs on an AT&T-approved DOS PC.
- **CAS for Windows.** This version takes advantage of the easy-to-use graphical user interface of Microsoft Windows. It is also a stand-alone application that runs on an AT&T-approved DOS PC. It allows a single CAS system to be used for both local and remote business sites.

All three versions allow businesses to calculate the cost of calls using the rates charged by long-distance carriers in one of 11 major metropolitan areas. In addition, you can customize CAS for Windows, CAS Plus V3, and IS CAS by programming additional rate tables.

All three versions provide the following services and features:

- **Call Record Processing.** This feature collects, stores, and produces records of calls and calculates costs using the selected rate table. You can program the system to process all calls or only calls that exceed a specified cost threshold. It can also add a service charge to calls before billing them to clients, departments, or projects.

  In addition, IS CAS collects and processes Automatic Number Identification (ANI) information as well as Caller Identification information provided by the 800 GS/LS-ID module and/or a DS1 module with PRI and Station Identification automatic Number Identification (SID/ANI) service. However, the availability of this information may be limited, depending on the legal jurisdiction and the equipment at the CO serving the caller. IS CAS also includes custom rate tables, required for the application.
- **Report Generation.** This feature organizes and prints call record information in the following formats:

- — Summary reports provide consolidated information on call activity. A wide variety of summary reports is available, based on all the types of data available about the application: for example by department, by extension, by area code, by cost, by time of day or by trunk facility used.
- — Detail reports provide detailed, call-by-call information for each extension.
- — Selection reports organize information on the basis of user-specified criteria, allowing trends and problems to be highlighted.
- — Account Code Detail Report lists every call associated with each account code entered by users.
- — Facility and Cost Center Reports show the distribution of line/trunk usage over organizations or cost centers.
- — Preselected Reports provide a choice of up to five reports from any of the other report categories and can be set to print on demand or at a specified time and date.

- **System Management.** The system manager can customize and maintain CAS activities, by editing tables, setting up reports, and updating call rate information.
- **Directory Lookup and Message Center.** Callers can look up anyone in the organization by name or extension, leave a message, and print or display messages.
- **HackerTracker System for CAS Plus V3 and IS CAS.** Telephone systems with auto attendant, voice mail, or remote access lines are common targets for toll theft. HackerTracker is designed to help detect fraudulent use of the system by detecting abnormal calling activity and tracking authorization code usage.

CAS Plus V3

The following steps are necessary for implementation of Caller ID information and CAS Plus V3 on the MERLIN LEGEND Communications System, Release 3.0:

NOTE:
If the customer has PRI and ICLID trunks, the system administrator will have to create a customized facility name in the telephone system configuration to distinguish between the two types of calls. If this is not administered, both types of calls will be costed and reported as ANI/ANIAB calls.

1. Administer the Release 3.0 options, SMDR format for ISDN and SMDR call report for In/Out collection.

2. Install a new PBX/KTS interface, MERLIN LEGEND Communications System - ISDN interface for the CAS Plus V3. You will need to have the PBX/KTS interface disk to complete this task.
3. Update facility tables, if necessary, in CAS under telephone system configuration for the new ICLID trunk numbers. If the customer has ISDN facilities, see NOTE.
4. Administer the dialed digit processing (DDP) in CAS Plus V3 to identify both the ICLID calls that are completed (ANI) and ICLID calls that are abandoned (ANIAB). Based upon how the customer wants the calls to be displayed on CAS reports would determine how to adminster the DDP records.
5. A report should be run to verify that ICLID information is being processed and reported in CAs according to the customer's needs and requirements.

Considerations and Constraints

You can connect only one CAS device to the system.

The system does not provide Station Message Detail Recording (SMDR) for calls within the system.

The number of calls about which CAS can store information depends on the amount of available disk space. In its largest configuration, CAS records data for up to 5000 extensions and 15,000 account codes.

Feature Interactions

Account Code Entry CAS uses the account codes entered by users before or during calls to provide reports by account code.

SMDR CAS collects call information from the SMDR output of the system. To collect Caller ID or ANI information, administer SMDR format to ISDN format.

Platform Requirements

To install CAS Plus V3 with the system, you need the following components:

- An AT&T-approved 386 PC, with:
  - MS-DOS 5.0 or higher (for new installations)
  - 640KB RAM, 80 MB hard disk, and 3.5- or 5.25-in. floppy disk drive
  - 1 parallel port and 2 serial ports
  - VGA or Super VGA monitor

— Real-time clock card

- 132- or 80-column IBM-compatible graphics parallel printer
- D8W modular cord and 355AF adapter connecting the SMDR jack on the system to the COM1 serial port on the PC (CAS Plus V3 only; CAS IS III connects to the COM2 port).

To use CAS for Windows with the system, the following components are recommended:

- For a single-site system, an NCR 3315 PC (20-MHz 386) with 6 MB of RAM and a 120-MB hard disk
- For a multi-site system, an NCR 3332 PC (66-MHz 486) with 16 MB of RAM and a 340-MB hard disk
  - — MS-DOS 5.0 or higher
  - — Windows 3.1
- VGA color monitor
- Bus mouse
- For a single-site system, a 120-MB tape drive
- For a multi-site system, a 525-MB tape drive
- AT&T Applications printer

For communications using CAS for Windows, the following components are recommended:

- For a single-site system, one parallel port and two built-in serial ports (DB9 for direct switch connection and DB25 for other connections)
- For a multi-site system, one parallel port and a four-port Equinox Mark-IV board with four RJ45 connections for direct switch hookup
- For remote diagnostics, a Remote Maintenance Board
- If a modem is used, an AT&T COMSPHERE 3830 or compatible
- If you are using the 9-pin port on your PC for the direct switch connection, you need a DB9-to-modular adapter
- An RJ45 modular cable to connect the PC's COM1 port with the control unit's SMDR port

For information on IS CAS, see "Integrated Solution III."

Call Accounting Terminal

IMPORTANT:
This section is intended solely as an overview of the application. For comprehensive information about the use of the application, see the documentation for the product.

A Call Accounting Terminal (CAT) is a dedicated terminal and printer designed to track, sort, and print reports on telephone charges. See Figures 4–1 and 4–2.

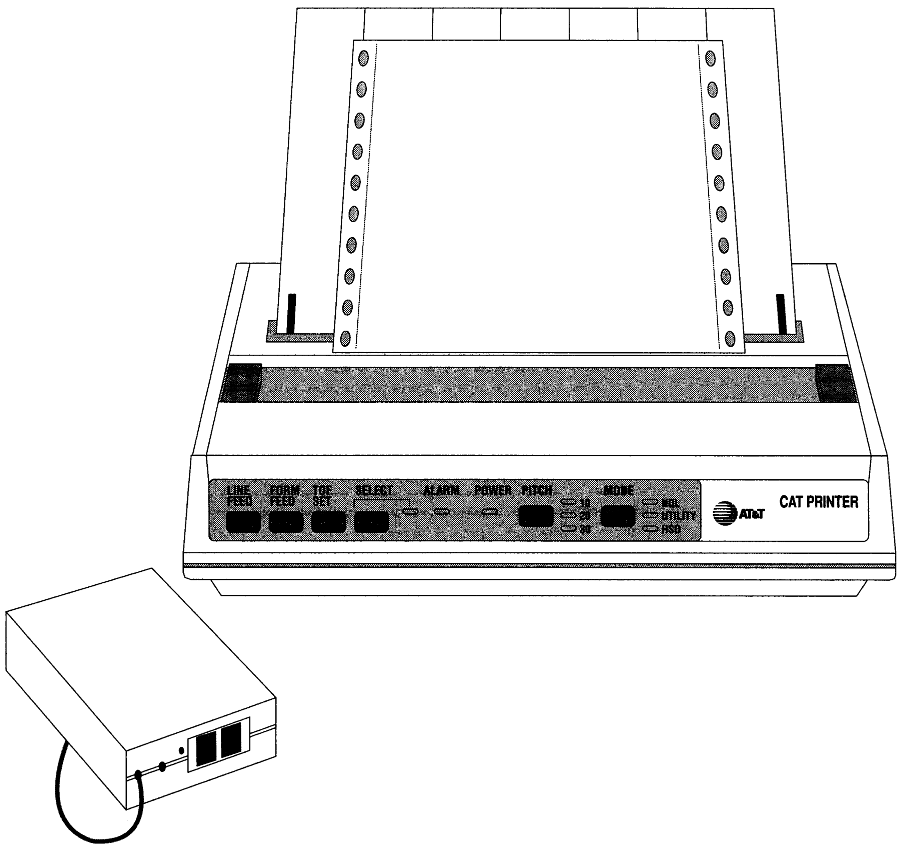

Figure 4–1. Call Accounting Terminal Basic

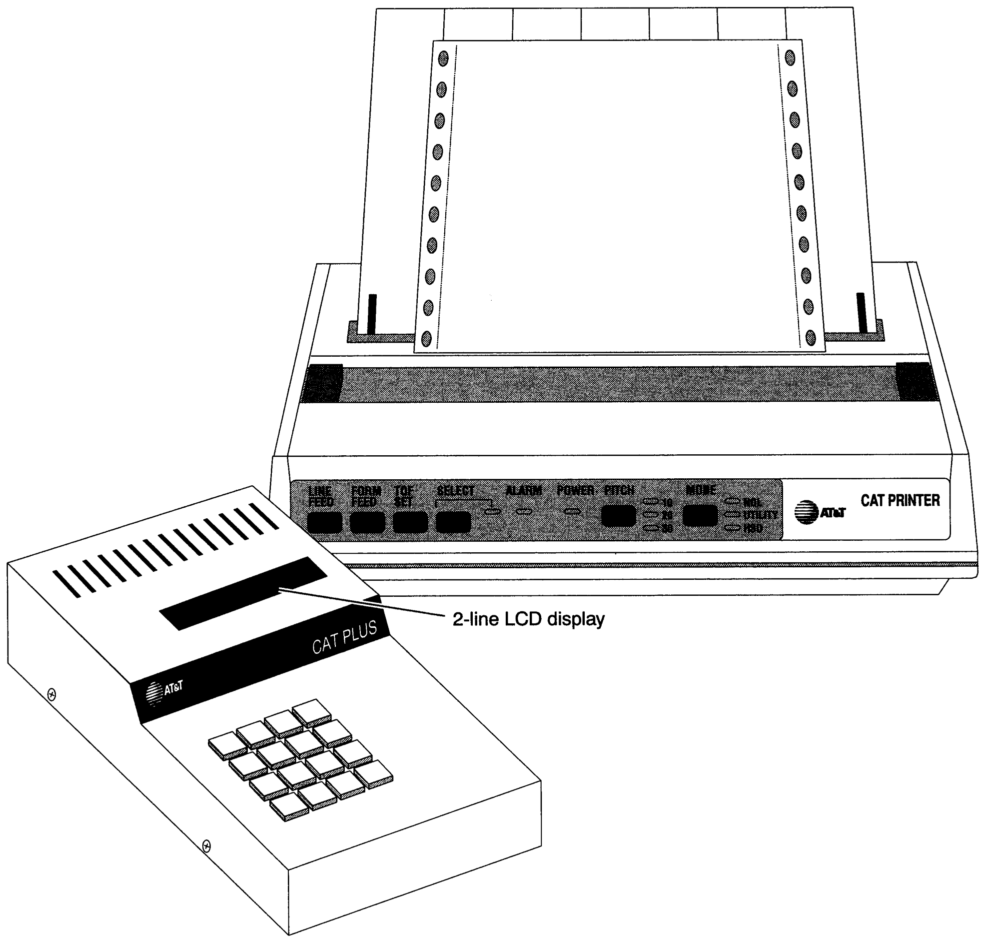

Figure 4–2. Call Accounting Terminal Plus

Three versions of CAT are available:

- **CAT Basic.** This version is an entry-level system for small businesses.
- **CAT Plus.** This version is for larger businesses and includes a two-line display.
- **CAT Plus/Hospitality.** This version, for hotels and health care facilities, also includes a two-line display.

You can set up a CAT to calculate the cost of calls using toll rates or by-the-minute charges. The CAT can apply service charges and discounts to calls made to local and long-distance numbers and to directory assistance. It can also identify calls to specified area codes (such as 900) for special treatment.

You customize CAT with current local and long-distance rates for your company's location. As rates change or a new area code or exchange is added, you can update the rate information by exchanging a chip inside the terminal. When you add a new telephone line or account code to the system, the CAT adds the information to its memory automatically the first time the new line or code is used.

The CAT provides a variety of reports that it can print on a regular schedule or automatically when call information reaches 90 percent of the terminal's storage capacity. The available reports include the following, depending on the version of CAT you have:

- A variety of summary and detail reports. For example, CAT can print reports on all extensions or rooms, a single extension or room, account codes, time of day, duration, and trunk facility.
- Management analyses organize call information by time of day, cost and duration of calls, area codes and exchanges called, and trunk facilities.

CAT can receive and process ANI information as well as Caller ID information provided by the incoming line identification from SMDR. The system gets such information from the AT&T Megacom 800 service or caller identification and puts it into the SMDR.

CAT Plus features an LCD display.

NOTE:
The availability of caller identification information may be limited by local-serving (caller's) jurisdiction, availability, or telephone company equipment.

Considerations and Constraints

You can connect only one CAT to the system.

CAT Basic can store information on up to 1200 calls for 100 extensions and 49 lines.

CAT Plus/Business can store information on 6500 calls made from up to 200 telephones that share up to 49 lines. When 90 percent of this capacity is reached and 5850 of these calls have been processed, reports are printed and memory is cleared. Any calls that come in during this process are held until reports are printed again.

System Programming

Set SMDR options:

- Select Basic or ISDN call report format. Select ISDN if you want ANI or Caller ID information reported.
- Specify the minimum call length to be recorded (10 seconds is recommended).
- Specify whether information is to be recorded for both incoming and outgoing calls or only for outgoing calls.

Feature Interactions

Account Code Entry CAT uses the 9-digit account codes that callers enter before or during calls to associate calls with accounts and individuals; these codes appear on CAT reports.

SMDR CAT collects call information from the SMDR output of the system.

Platform Requirements

To connect CAT to the system, you need the following equipment:

- CAT Basic, CAT Plus/Business, or CAT Plus/Hospitality
- CAT Printer
- D8W modular cord to connect CAT to SMDR port on the CU and 355AF adapter to connect the CAT to the printer.

Call Management System

IMPORTANT:
This section is intended solely as an overview of the application. For comprehensive information about the use of the application, see the documentation for the product.

Call Management System (CMS) is a DOS-based application that simulates the actions of a system operator by answering calls and distributing them to individual agent extensions. If no agents are available, CMS puts calls on hold and, if programmed, plays a recorded announcement to the callers. When agents become available, CMS searches the system for the appropriate agent—usually the one who has been idle the longest—and transfers the call to that agent's extension.

CMS is designed for businesses with large groups of personnel who perform a common function, such as airline ticketing, filling catalog orders, or providing customer service. You can divide agents within these groups into *splits*, or subgroups, to handle different kinds of calls or customers. For example, you might divide the agents in a travel agency into three splits: one that handles personal vacations, one that handles business trips, and one that handles group charters. You can designate another split to provide support when call traffic is particularly heavy in the other splits. Calls come in to each split on a group of lines designated to ring into that split.

Agents make themselves available and unavailable to take calls by logging in and out. In addition, agents can enter the after-call work (ACW) state, which allows them to complete work on their last call without being interrupted by new CMS calls. You can set up the system so that agents are automatically in the ACW state whenever they complete a CMS call or so that agents must dial a feature code or press a programmed button to enter ACW.

CMS provides the following additional features:

- Management reports that analyze call volume and patterns and agent activity. Summary reports can span from 1 to 93 days.
- The Answer Delay option to specify how long a call rings before it is designated as unanswered and connected to the recorded announcement
- The Forced Delay option to connect all calls to the recorded delay announcement regardless of whether all agents are busy
- Priority lines to ensure that calls coming in on those lines are answered first
- Display of current agent activity on system status screens to allow monitoring, tracking, and analyzing of short- and long-term performance
- Music on Hold for callers waiting for available agents
- Allows connection of up to four external alerts to indicate an exception, for example, an LED that lights when the oldest call has waited longer than 30 seconds. Exception thresholds are programmed.
- Real-time dynamic reconfiguration, allowing you to modify the call flow on-line

Mode Differences

The system must operate in Hybrid/PBX or Key mode. You cannot connect CMS equipment to a system operating in Behind Switch mode.

Considerations and Constraints

You can connect a maximum of two CMSs to the system.

You must install CMS on an AT&T-approved DOS PC, which is dedicated to the application. You must connect the two CMS interface card ports on the PC to two analog multiline extension jacks on the same module in the control unit (008 or 408). These jacks must be system operator positions. If two system operator position jacks are not available on the same module, you must install another of these modules in the control unit to provide them.

Each CMS can handle calls for up to 28 agents on up to 28 lines, and it can answer calls on two lines at the same time with the same announcement.

You can designate up to six agent splits for each CMS, with 28 agents per split.

The CMS supervisor's console is a Direct-Line Console (DLC). CMS agents can have any MLX telephone or any analog multiline telephone. You must connect agent telephones to the first 40 extension jacks on the control unit.

Lines/trunks ringing in to CMS can be loop-start, ground-start, T1-emulated ground-start, or PRI.

You can use up to four external alerts to alert agents and supervisors when the number of calls waiting in the queue reaches the programmed threshold.

You can use AT&T Attendant to direct callers to the appropriate CMS group by use of a *loopback* or *loop-around* arrangement, where AT&T Attendant transfers group calls to a tip/ring (T/R) port on an 012 or 008 OPT module that in turn rings a loop-start line assigned to CMS. Calls transferred in this way look like new outside calls to CMS. This also applies for DS1 applications using DNIS to route calls to CMS.

Connect a Music On Hold audio source and coupler to play music for callers waiting in the queue.

NOTE:
If such equipment is used to rebroadcast music or other copyrighted materials, it may be necessary to obtain a copyright license from or pay license fees to a third party, such as ASCAP or BMI. Or you can purchase a Magic on Hold system, which does not require such a license, from AT&T.

Feature Interactions

Extension Status

A CMS supervisor uses the Extension Status feature to control and monitor when agents are in the available, unavailable, or ACW state. A CMS agent does not have to be a member of a calling group to be available or unavailable. The system can be programmed for CMS or for Hotel/Motel Extension Status, but not for both.

Group Calling

CMS agents log in and out by using the same buttons or codes as calling group members.

System Renumbering

CMS uses 2-digit extension numbers only.

System Programming

Set basic system operating conditions:

- Select the 2-digit system renumbering plan (2-Digit is the factory setting) or Set-Up Space with CMS agents numbered for 2 digits. 3- and 4-digit numbering are not supported.
- Set transfer return time for 3 to 5 rings.
- Set transfer audible to Ringback.
- Select the Group Calling/CMS option for the Extension Status feature.

Remove CMS lines from all telephones (Key mode only) or from trunk pools (Hybrid/PBX mode only).

Set up three DLC system operator positions, two for CMS PC positions and one for the CMS supervisor position (if a CMS supervisor telephone is required):

- Assign the positions.
- Assign CMS lines and external alerts to the CMS supervisor's console and copy the assignments to the CMS PC ports.

Set up a CMS fallback plan:

- Designate the CMS supervisor console as a group coverage sender.
- Assign the agent telephones to a calling group and assign group coverage to the calling group.

Set up optional equipment and features, including headsets and paging groups.

Set the Ringing options for lines assigned to CMS ports to No Ring.

Platform Requirements

To connect CMS to the system, you need the following equipment:

- An approved AT&T DOS PC, with:
  - 640 KB of RAM
  - 3.5-in. floppy disk drive
  - 20-MB hard disk drive
  - Monochrome or color monitor
- CMS interface card with two 14-ft. (430-cm), 4-pair modular extension cords
- CMS software
- Digital Announcer Unit with one 14-ft. (430-cm) DIN connector cord
- Parallel printer with cable to connect to the PC parallel port
- Supervisor console, any DLC position
- Agent telephones, any MLX or analog multiline telephones
- One analog multiline module (008 or 408) to connect the two PC ports to the extension jacks assigned as DLC ports

System Programming and Maintenance

IMPORTANT:
This section is intended solely as an overview of the application. For comprehensive information about the use of the application, see the documentation for the product.

System Programming and Maintenance (SPM) software lets you program and maintain the system. It performs the same functions as an MLX-20L telephone set up as a system programming console, providing a display that emulates the console display. SPM also has additional features, such as the ability to back up and restore system programming and to print reports.

Two versions of SPM are available:

- **SPM Stand-Alone.** This version runs on an AT&T-approved DOS PC.
- **SPM Integrated with IS II or III** (SPM IS II or III). This version runs under the UNIX system. See "Integrated Solution II" and "Integrated Solution III" for more information.

You can connect a PC with DOS-based SPM directly to the control unit or you can access the system remotely in one of the following ways:

- The system programmer dials the system directly. You can set up a password to prevent unauthorized access.
- The system programmer dials the system operator and asks to be transferred to the system's built-in modem (dial code *10).

You can use SPM IS III only through a direct local connection.

You can program SPM to operate in English, French, or Spanish. Independently of the overall language setting, you can select one of these languages for the console-simulation window for the duration of the current session.

Considerations and Constraints

You *must* upgrade SPM to version 2.16 to function with Release 2.0 and later of the system. You *must* upgrade SPM to version 3.xx to function with Release 3.0 and later of the system.

Unless the system is being backed up or restored, a remote SPM connection takes priority over a local user. If the local user is programming when a remote user connects to the system, the system sends a warning message to the local user and disconnects that user.

The SPM PC connects to the lower RS-232 jack on the processor module in the control unit. This connection runs at 1200 or 2400 bps with autobaud switching.

You can print SPM reports and/or save them on the PC's hard or floppy disk drive. At the same time, the report is displayed on the screen together with prompts for browsing.

You should not print SPM reports when the system is handling more than 100 calls per hour.

You can use a printer connected to the SPM PC to print system reports. Alternatively, you can send reports to a printer connected to the SMDR port on the control unit. However, SMDR information may be lost while system programming reports are being printed through the SMDR port.

Platform Requirements

To support stand-alone SPM, you need:

- AT&T-approved DOS PC
- MS-DOS 3.3 or higher
- At least 128 KB of RAM
- A double-sided floppy diskette drive, either 5.25-in. or 3.5-in. (hard disk is optional but recommended)
- A serial port assigned to COM1 or COM2. The serial port can use either a DB-9 or DB-25 connector. If a DB-9 connector is used, a 9-pin to 25-pin adapter is also required. The 9-pin side must be female.
- A monochrome or color monitor
- A D8W modular cord and a 355AF modular adapter if the PC is less than 50 ft. (15 m) from the control unit. Distances of greater than 50 ft. (15 m) require back-to-back Asynchronous Data Units (ADUS).

Integrated Solution II

IMPORTANT:
This section is intended solely as an overview of the application. For comprehensive information about the use of the application, see the documentation for the product.

Integrated Solution II (IS II) is a complete package of UNIX System-based voice processing and call analysis software applications. IS II offers a single interface to any of the following applications:

- **Integrated Voice Power Automated Attendant** (IVP AA). Answers telephones automatically and transfers callers to the appropriate departments or extensions. Also provides callers with a menu of recorded prompts that they can respond to by dialing numbers on a touch-tone telephone.

  The system transfers callers without touch-tone telephones to the system operator. You can set up separate menus for day and night service, as well as multilevel menus and corresponding announcements to ensure that callers reach the right person or department as quickly as possible.

IVP AA can operate in touch-tone gate mode or in no-gate mode. To speed handling of calls from touch-tone telephones, gate mode prompts callers to dial 1 to continue to the main menu. If a 1 is not dialed within a programmed interval, calls are automatically transferred to the system operator. In the no-gate mode, callers hear the main menu immediately and, if no response is received after the main menu is played, calls are transferred to the system operator.

IVP AA is a low-cost alternative for businesses that need enhanced call handling without the added voice messaging capabilities of AUDIX Voice Power IS II.

- **AUDIX Voice Power IS II** (AVP IS II). Offers the features of the IVP AA plus the following services:
  - **Call Answer Service.** Allows callers who reach a busy or unanswered extension to leave a message, transfer to another extension, or transfer to a system operator. Individual extension users can program a personal greeting or select a standard greeting; users can also program a password to prevent others from retrieving their messages.
  - **Voice Mail Service.** Allows users to send messages to other extensions in the system, forward messages received with comments, and reply to messages received. The system manager can send general messages to everyone in the system.
  - **Information Service.** Provides a customer-oriented calling information service that plays a recorded message and then disconnects the caller.
  - **Message Drop.** Offers an answering service, similar to an answering machine, that plays a message to the caller and then allows the caller to "drop off" a message, such as a request for service or an order. Callers cannot direct their messages to specific extensions.
- **Call Accounting System IS II** (CAS IS II). Collects and analyzes call information, calculates the prices of calls by using rates selected by the business, organizes calls by client or project, and prints reports on a daily or as-needed basis. For more information on the features of CAS, see "Call Accounting System," earlier in this chapter.
- **System Programming and Maintenance IS II** (SPM IS II). This programming package, built into IS II, allows a system manager or technician to upgrade and maintain the system and its features and to add, change, or rearrange telephones. The system manager or technician can program on site or remotely.

Additional IS II features include the following:

- **Dial by Name.** Permits AVP users to call subscribers by dialing the last name of the subscriber instead of dialing the extension number.
- **Alternate Personal Greetings.** Allows a user to record a second personal greeting in addition to the primary call-answer greeting.
- **Fax Transfer.** Directs incoming fax calls to a designated fax machine.
- **Class of Service.** Allows the system manager to assign one of 16 predefined parameters to a subscriber. These parameters define the size of the mailbox, the type of coverage service, and the activation of the outcalling feature.
- **General Mailbox Options.** Provides two special mailboxes that have reserve extensions associated with them. Callers using rotary telephones or needing assistance can be transferred to leave messages in a general mailbox. Subscribers having problems with the system can report problems to the trouble mailbox.

The number of incoming lines and subscribers programmed for AVP or IVP AA and the number of busy-hour calls determine how many voice channels are required for the system (see Table 4–7).

Table 4–7. Voice Channels Required: IS II

Channels Required	Lines	Subscribers	Busy-Hour Calls
2	1 to 6	1 to 20	1 to 20
4	7 to 18	21 to 60	21 to 60
6	19 to 24	61 to 80	61 to 80
8	25 to 42	81 to 200	81 to 200
12	Over 42	201 to 300	201 to 300

Mode Differences

Of the available IS II applications, you can connect only CAS IS II and SPM IS II applications to a system that operates in Behind Switch mode.

Considerations and Constraints

IS II uses UNIX System V Release 3.2.2.

IS II stores up to 12 hours of voice-mail messages when IS II includes AVP and over 200,000 call accounting records when IS II includes CAS.

You can install either IVP AA or AVP on the system, but not both.

The system supports up to 12 IVP AA ports (on three circuit boards).

If IS II includes AVP, when users receive voice-mail messages, the Message LEDs on their telephones light, provided that a mailbox has been assigned to each of those telephones.

For AVP or IVP AA, the following symptoms indicate that the system needs more TTRs:

- Single-line telephone users do not get dial tone when trying to dial out.
- AVP or IVP AA fails to transfer calls.
- Calls fail to ring or go to coverage prematurely.

You can print SPM IS II reports or save them on disk (floppy or hard disk). At the same time, the report is displayed on the screen together with prompts for browsing.

You should not print SPM IS II reports when the system is handling more than 100 calls per hour.

Feature Interactions

Account Code Entry

CAS IS II associates account codes entered by users before or during calls with accounts and individuals; they appear on CAS IS II reports.

Coverage

An inside call on a VMI port that transfers to an inside extension does not go to coverage. It continues to ring at the inside extension.

If a sender programs the telephone so that only outside calls are sent to coverage, calls received on **ICOM** or **SA** buttons will not be sent to voice mail.

In Release 2.0 and later, outside calls that would normally proceed to AUDIX Voice Power as coverage calls do not do so if the telephone that sends the call to Group Coverage has activated Coverage VMS Off. No special action is needed in AUDIX Voice Power programming to activate this feature.

Group Calling

Calls answered by an overflow calling group get coverage mode codes; the overflow calling group's number appears in the Called Party field of the mode code.

Leave Message

If a Leave Message notification is left in a mailbox in a system with heavy VMI traffic, the user may have to dial out manually for messages.

In Release 2.0 and later, when AUDIX Voice Power sends a Leave Message notification to an extension, the system identifies the voice mail system as the sender of the message. As a result, when the voice mail subscriber uses the Return Call feature, the call goes to any available voice mail port, not just to the specific port that generated the message. This improves access by reducing the chance of getting a busy port.

Night Service

If the AVP Automated Attendanthandles only after-hours calls, a phantom extension (an unused extension or vacant port) must be programmed as a member of a Night Service group associated with the system operator. In turn, this phantom extension is covered by a calling group with integrated VMI ports as members. If an incoming call is not answered in the programmed number of rings, the control unit sends the call to the calling group with the VMI ports. Because of prior programming, AVP recognizes the call to be from the phantom extension and provides Automated Attendant service rather than the usual Call Answer service.

SMDR

CAS IS II uses the call information provided by the system's built-in SMDR feature to process calls. There are two system formats for SMDR: Basic and ISDN.

Transfer

Beginning with Release 2.1, any VMI ports can transfer an incoming call to an outgoing line/trunk. In earlier versions, only integrated VMI ports could transfer to an outgoing line/trunk.

If a caller incorrectly specifies the answering VMI port as the desired transfer destination telephone, the VMI port may inadvertently park the call.

You can program any calling group, calling group member, or telephone to be a VMS transfer redirect extension. If you program a QCC as such, the transfer redirect call is delivered to the QCC as a returning call and is not placed in the QCC queue.

If a transferred caller gets no answer and returns to the system operator, the system operator has no indication of the origin of the call.

System Programming

You must program the following options when IS II includes IVP AA:

- Designate Inside Dial Tone to be the same as the outside line/trunk dial tone.
- Assign all Automated Attendants to the same calling group and set the group type to VMI Generic.
- Program each VMI loop-start port for reliable far-end disconnect.
- Designate a backup position, such as the system operator, to receive calls that originally transferred to unanswered or busy extensions or to receive calls when a caller fails to respond to a message.
- Specify the number of rings before a call transferred by the voice messaging system is sent to the backup position.

You must program the following options when IS II includes AVP:

- Assign AVP ports to a calling group and specify the group type as VMI Integrated.
- Program each VMI loop-start port for Reliable far-end disconnect.
- Specify the touch-tone duration and interval between digits in codes sent between the AVP and the system.
- Specify the number of rings before a call transferred by AVP is sent to the backup position (usually the system operator).
- When you use AVP Automated Attendant for Night Service only, do the following:
  - If the lines/trunks set for answering by Automated Attendant appear at other extensions, set the No Ring option for the other telephones.
  - Assign the phantom extension to a Night Service group for each system operator position.
  - Assign the phantom extension to a coverage group, and assign the VMI calling group to cover that coverage group.
  - Specify the VMI ports that provide Automated Attendant to be Automated Attendant ports.
  - Specify the business schedule for AVP.

Platform Requirements

You need the following equipment:

- A 200-MB fixed disk if IS II includes either IVP AA or AVP
- IS II uses an AT&T Master Controller, based on a 6386/SX WGS processor with UNIX System V/386 Release 3.2.2. It includes:
  - Master Controller II processor (with a 40-MB, 80-MB, or 200-MB fixed disk and a 3.5-in. floppy disk drive)

- Monitor (monochrome or color)
- Keyboard
- Optional tape drive (required for systems with a 200-MB fixed disk for saving UNIX files, application program files, programming files, and voice system files during backup)

■ A 355AF adapter for connecting the Master Controller to the serial port on the control unit if they are within 50 ft. (15 m) of each other and are on the same AC branch circuit

■ Asynchronous Data Units for connecting the Master Controller to the serial port on the control unit if they are not within 50 ft. (15 m) of each other and are not on the same AC branch unit

■ Any additional hardware required by each application included in IS II, including the cables and adapters for connecting the applications to the system. See the instruction booklet that comes with each application.

■ IVP4 boards

■ 012 basic telephone module to provide the tip/ring interface for IVP AA or AVP

Integrated Solution III

IMPORTANT:
This section is intended solely as an overview of the application. For comprehensive information about the use of the application, see the documentation for the product.

Integrated Solution III (IS III) Release 1.2 is an interface to a complete package of UNIX System-based voice processing and call management software applications. It provides a single integrated interface to any of the following applications:

■ **AUDIX Voice Power 2.1.1** (AVP). Combines the following voice messaging services and features:

- **Call Answer Service.** Allows callers who reach a busy or unanswered extension to leave a message, transfer to another extension, or transfer to a system operator. Individual subscribers can program a personal greeting or select a standard greeting and can program a password to prevent others from retrieving their messages.
- **Voice Mail Service.** Allows subscribers to send messages to other system extensions, forward messages with comments, and reply to messages. The system manager can broadcast messages to all subscribers.

- — **Information Service.** Provides a call-in information service that plays a recorded message and then disconnects the caller.
- — **Message Drop.** Provides an answering service, similar to an answering machine, that plays a message to callers and then allows a caller to "drop off" a message, such as a request for service or an order. Callers cannot direct messages to specific extensions.
- — **Automated Attendant Service.** Answers incoming calls and plays a menu of recorded prompts. A caller can respond to the prompts by dialing touch tones, and Automated Attendant routes the call to an inside extension accordingly. If there is no answer or the extension is busy, the caller can be given the option to leave a message or try another extension.

  A caller with a rotary telephone is transferred to the system operator for further call handling and routing.

  The system manager can record multiple levels of menus and announcements, including separate menus for day and night service.
- — **Outcalling.** When a user or subscriber receives a new message, the system can automatically call a programmed number, for example, a beeper or a home telephone number. The subscriber can then login to the VMS.

  NOTE:
  To restrict outcalling, use AVP's Transfer to Subscribers Only feature. Do not use MERLIN LEGEND Communications System calling restrictions.

- **Fax Attendant 2.1.1.** Provides an integrated voice/fax mailbox, fax broadcasting, fax bulletin board, and coverage for busy or off-line fax machines. Fax Attendant only works with AUDIX Voice Power and not by itself. Fax Attendant includes the following services and features:
  - — **Fax Call Answer.** Allows Fax Attendant to receive fax messages for subscribers whose fax machines are busy or out of paper. This feature also lets subscribers (who have personal fax mailboxes) without fax machines receive fax messages. In such a case, Fax Call Coverage gives the appearance of a personal fax machine by automatically answering and receiving fax messages for the specified telephone number. Faxes are temporarily stored for printing later at a convenient time.
  - — **Personal Fax Messaging.** Because subscribers have fax mailboxes similar to voice mail mailboxes, inbound faxes can be stored until the subscriber asks that they be printed, at any fax machine he or she specifies, on company premises or off-site (when the subscriber retrieves fax messages remotely). This feature protects the confidentiality of sensitive documents. Fax Attendant can even inform subscribers of waiting faxes by calling to an outside number.

- — **Fax Mail.** Allows subscribers to send fax messages, get fax messages, record personal greetings, program outcalling (stand-alone configuration with AVP only: create fax distribution lists, change account passwords, deliver report settings, and autoprint setting).
- — **Fax Response.** Allows users to dedicate a phone number from which callers can retrieve information. This feature directs callers through a series of prompts that asks information about their fax machines. Callers are greeted with spoken prompts that guide them in pressing touch-tone buttons to access the information and to receive their information within minutes by fax transmission.
- — **Fax Broadcast.** Provides a simple way to send one fax to as many as 1000 fax numbers. Because it uses multiple ports, faxes are sent in the fraction of the time required for a broadcast fax machine and fax machines are free to receive incoming calls and send other faxes. Features such as Economy Delivery and Intelligent Auto Retry save time and money. AVP voice prompts make operation easy.

■ **Integrated Solution Call Accounting System** (IS CAS). Collects and analyzes call record information, calculates costs using rate tables selected by the customer, organizes calls by client or project, and prints reports daily or as needed. With IS III Release 1.2, IS CAS is distinct from the stand-alone CAS Plus V3 application. Here are ways that it differs:

- — **ANI and Caller ID Support.** IS CAS collects and processes Automatic Number Identification (ANI) information as well as Caller ID information provided through the 800 GS/LS-ID module. However, the availability of this information may be limited, depending on the legal jurisdiction and the equipment at the CO serving the caller.
- — **Custom Rate Tables.** Custom rate table software is required for IS CAS and is included in the package.

■ **System Programming and Maintenance.** Provides a maintenance and programming interface to the system. SPM IS III provides the same functionality as the stand-alone SPM application, except for remote connection to the control unit.

Integrated Administration

For more complete information on this application, refer to *Feature Reference*. Integrated Administration is the integration of AUDIX Voice Power and Fax Attendant programming with the system parameters that the two applications use. Integrated Administration consists of four parts, accessed by menu selection:

- Extension Directory
- Extension Directory Setup
- Reconciliation

- Integrated AUDIX Voice Power and Fax Attendant Administration (System Programming/Switch Administration, Fax Channel Administration)

Integrated Administration is intended for system technicians who are responsible for programming the applications and the system through IS III. Users can also use Integrated Administration to make changes to their system. All three parts of Integrated Administration are available to system technicians. Only the Extension Directory and System Programming/Switch Administration are available to users through the IS III menu.

Initial Installation

The system technician performing initial installation logs in to IS III and selects **System Programming and Maintenance (SPM)** from the menu. The SPM screen appears. The technician performs basic system programming, such as dial plan, mode, attendant, phantom extensions, and lines in pools, and exits from SPM.

Through the Technician Maintenance menu item on the Integrated Solution III Maintenance menu, the technician selects **AUDIX Voice Power Switch Defaults** and changes the defaults for calling groups if necessary. Through the same menu item, the technician performs an Extension Directory Setup, which downloads the system dial plan and directory labels into the Extension Directory database.

Through the Integrated Solution III Maintenance menu, the technician performs an Extension Directory update, that is, steps through each extension and attaches a name label and other information.

Assigning the AVP application to a user through the Integrated Administration Extension Directory automatically covers the user by AUDIX Voice Power, if specified, and produces the AVP Subscriber screen.

The AVP Subscriber screen uses the name, extension, and coverage information contained in the Integrated Administration Extension Directory. This information cannot be changed on the AVP Subscriber Screen.

The technician then selects **System Programming/Switch Admin** from the AVP or AVP/FA main menu and performs Integrated AVP or AVP/FA programming. (The technician is guided through a series of choices and forms that direct the programming.) The technician then completes the remaining AVP, FA, and system programming.

Installation on an Existing System

The system technician installing Integrated Administration on an existing system selects **AUDIX Voice Power Switch Defaults** from the Technician Maintenance item on the Integrated Solution III Maintenance menu, and changes the defaults for calling groups if necessary. Through the same menu item, the technician performs an Extension Directory setup. This downloads the system dial plan and directory labels into the Extension Directory database.

Through the Integrated Solution III Maintenance menu, the technician performs an Extension Directory update, that is, steps through each extension and adds them as AVP subscribers if necessary.

The technician then selects **System Programming/Switch Admin** from the AVP or AVP/FA main menu and performs integrated AVP or AVP/FA programming. (The technician is guided through a series of choices and forms that direct the programming.) During the programming, the technician presses the Save key (without entering any information) whenever prompted to enter lines/pools. This allows the flow of information to proceed without sending any outside calls directly to the AVP ports.

The technician completes the remaining AVP and FA programming, including programming of greetings and other voice prompts, and through the System Programming/Switch Administration menu, goes back to each installed service and adds the appropriate lines.

Branch Operation (Technicians only)

The system technician can perform initial installation if the branch location is equipped with a surrogate system and IS III. Using the remote location's system and Multi-Application Processor/5 (MAP/5) the technician programs the customer's configuration, as specified earlier in "Initial Installation."

Through SPM, the technician backs up the system configuration and, through the Technician Maintenance menu, backs up the Extension Directory database files. The technician then dials up the customer location and accesses the internal modem, and, through remote SPM, restores the customer's system from the translations made at the remote location.

After requesting pass-through to the customer's MAP/5, the technician, through the Technician Maintenance menu, restores the customer's database files from the database files backed up at the remote location.

NOTE:
Technicians can access the Extension Directory and Integrated Administration screens remotely, but the information is stored in a file and executed after the remote caller hangs up. Also, a change made to System Renumbering is not reflected in the Extension Directory; the reconciliation program that ran automatically in earlier releases is disabled in Release 1.2 and later.

Mode Differences

The system must operate in Key or Hybrid/PBX mode for all IS III applications except IS CAS and SPM. Those are the only two applications that you can connect to a system operating in Behind Switch mode.

Considerations and Constraints

The MAP/5 is a 32-bit, i486SX computer that comes in a range of hard disk/tape sizes with a VGA monitor. The following considerations and constraints should be reviewerd when configuring a system:

On a 500-MB fixed disk, IS III can store up to 36 hours of voice mail messages for AUDIX Voice Power, as many as 3000 pages of faxes for Fax Attendant, and 332,000 call records for IS CAS.

If both IS CAS and Fax Attendant System are part of IS III, they share the same disk area for record storage and so share the same maximum based on disk size.

On a 200-MB fixed disk, IS III can store over 200,000 call records for IS CAS, 12 hours of messages for AUDIX Voice Power, and as many as 1000 pages of faxes for Fax Attendant.

Fax Attendant is not supported on a 100-MB fixed disk system or MAP/5.

You cannot install Fax Attendant without AUDIX Voice Power.

Fax Attendant versions prior to Release 2.1.1 (optional with IS III Release 1.2) used IFP2 two-port (two-channel) boards. These boards are compatible with Release 2.1.1, but IFP4 four-port (four-channel) boards are also available. The IFP2 and IFP4 boards cannot be mixed in the same IS III configuration.

You cannot install Automated Attendant as a stand-alone application but only in conjunction with AUDIX Voice Power.

When an AUDIX Voice Power subscriber receives a voice mail message, the Message LED on the telephone lights. To properly update Message LEDs on system extensions, link each AVP mailbox with telephones connected to the control unit.

If an AUDIX Voice Power mailbox is needed for a person with no telephone, you must assign a phantom extension to the control unit.

You should synchronize AUDIX Voice Power time with system time.

For Integrated Administration, the MAP/5 provides a separate backup and restore capability that saves the directory information on the hard disk. This is available through the Backup and Restore menu options under Maintenance.

The subscriber must have **AVP** in the Applications field of the Integrated Administration Extension Directory in order to be added to or deleted from AUDIX Voice Power.

All extensions and lines that you program through Integrated Administration must be idle.

Feature Interactions

Account Code Entry

IS CAS uses the account codes people enter before or during calls to associate calls with accounts and individuals; these codes appear on IS CAS reports.

Coverage

An inside call on a VMI port that transfers to an inside extension does not go to coverage, but continues to ring at the inside extension.

If a sender programs the telephone so that only outside calls are sent to coverage, calls received on **ICOM** or **SA** buttons are not sent to voice mail.

In Release 2.0 and later, outside calls that would normally proceed to AUDIX Voice Power as coverage do not do so if the telephone that sends the call to group coverage has activated Coverage VMS Off. No special action is needed in AUDIX Voice Power programming to activate this feature.

Group Calling

Mode codes identify calls that overflow from one calling group to another calling group with integrated VMI ports coverage calls. As a result, the overflow calling group's number appears in the Called Party field of the mode code.

Labeling

Names entered through the Integrated Administration Extension Directory are sent to the control unit and are available through Switch Labeling screens.

Leave Message

If a Leave Message notification is left in a mailbox in a system with heavy VMI traffic, the subscriber may have to dial out manually to retrieve the message.

In Release 2.0 and later, when AUDIX Voice Power sends a Leave Message notification to an extension, the system identifies the voice mail system as the sender of the message. As a result, when the voice mail subscriber uses the Return Call feature, the call goes to any available voice mail port, not just to the specific port that generated the message. This improves access by reducing the chance of getting a busy port.

Night Service

If Automated Attendant handles only after-hours calls, you must program a phantom extension as a member of a Night Service group associated with a system operator. This phantom extension is covered by a calling group with integrated VMI ports as members. If an incoming call is not answered within the programmed number of rings, the control unit sends it to the calling group with the VMI ports. You must program AUDIX Voice Power to recognize the call from the phantom extension, and you must provide Automated Attendant service rather than the usual Call Answer service.

SMDR

IS CAS collects call information from the SMDR output of the system.

System Renumbering

System Renumbering can be done only through SPM or MLX-20L system programming. Integrated Administration uses System Renumbering to read extension numbers and adjuncts.

Transfer

VMI ports can transfer an incoming call to an outgoing line/trunk.

If a caller incorrectly specifies the answering VMI port as the desired transfer destination extension, the VMI port may park the call.

You can program any calling group, calling group member, or extension as a VMS transfer redirect extension. If you program a QCC as such, the transfer redirect call is delivered to the QCC as a returning call and is not placed in the QCC queue.

If a transferred caller gets no answer and returns through voice mail to the system operator, the system operator has no indication of the origin of the call.

System Programming

AUDIX Voice Power requires the following system programming:

- Assign AUDIX Voice Power ports to a calling group and specify the group type as VMI integrated.

- Specify the touch-tone duration and interval between digits in codes sent between AUDIX Voice Power and the system.
- Specify the number of rings before a call transferred by AUDIX Voice Power is sent to the backup position (usually the system operator).

AUDIX Voice Power with Automated Attendant requires the following system programming:

- Set inside dial tone to Outside.
- Assign Automated Attendants to a calling group and specify the group type as VMI Integrated.
- Designate a backup position, such as the system operator, to receive calls that originally transferred to unanswered or busy extensions or to receive calls when a caller fails to respond to a message.
- Specify the number of rings before a call transferred by the VMS is sent to the backup position.

When you use AUDIX Voice Power Automated Attendant only for Night Service:

- If the lines/trunks set for answering by Automated Attendant appear at other extensions, set the No Ring option for the other extensions.
- Specify the VMI ports that provide Automated Attendant service as Automated Attendant ports.

Integrated Administration with Fax Attendant requires the following system programming:

- **IVP 4/6 Board Jacks.** For fax response service, you must program the following items:
  - Assign the tip/ring extensions dedicated to fax response into a calling group for Integrated Administration.
  - Set the calling group type to VMI Integrated-Automatic.
  - Assign outside lines to the calling group.
  - Assign appropriate labels to the lines.
- **Fax Board Ports.** Each fax board connects to a T/R extension jack on the control unit. These extension jacks are regular T/R extension jacks and do not have to be identified as fax extensions on the system. You must program the following options:
  - Assign appropriate labels to the T/R extensions.
  - Assign the T/R extensions to the Night Service Exclusion List; this enables off-site fax delivery to function at night.
  - If users want a waiting fax to light Message LEDs, use SPM to identify the ports as fax extension jacks on the control unit.

- **Private Fax Extensions.** A private fax extension either connects to an actual fax machine used by an individual or is a phantom extension associated with an individual's voice extension. Programming for a private fax extension depends on whether or not the system's configuration supports DID lines. In systems with DID, unique DID extension numbers are sufficient for private fax extensions because outside calls placed to a DID number ring the fax machine or phantom extension. Systems without DID must rely on personal lines.

For private fax extensions in DID configurations, you assign the DID number of a phantom extension or actual fax machine as a private fax extension.

You must program the following items for private fax extensions in non-DID configurations:

- Assign a personal line to a phantom extension or to the extension connected to an actual fax machine.
- Using SPM, assign the phantom extension or fax machine as the owner of that line.

You must program the following remaining items for private fax extensions in both configurations:

- Assign the individual as a subscriber to Fax Attendant service. Specifying AUDIX Voice Power as an application automatically subscribes the user to both AUDIX Voice Power and Fax Attendant.
- Assign the phantom extension or fax machine as a private fax extension.
- Assign the private fax extension to a coverage group.
- Set the label of the private fax extension appropriately.
- Assign the specified coverage group previously to be covered by the calling group of Automated Attendant, Call Answer, or Voice Mail-Automatic.

One private fax extension can be used by a group of individuals through parameters that you set with Fax Attendant setup screens on the MAP/5. To facilitate these configurations, select a group fax administrator. You must program the following items:

- Assign a non-valid extension number as a special-purpose extension for the group fax administrator.
- Assign a private fax extension to the special-purpose extension.
- Set group members as Fax Attendant subscribers. Do not program any of these users for private fax extensions.

Platform Requirements

When IS III is delivered, it is installed and configured according to the applications you ordered. The system consists of a MAP/5 running UNIX System V Release 3.2.2. Various hardware configurations are available; see the *AT&T Integrated Solution III Installation and Maintenance Guide* for details.

If AUDIX Voice Power is installed, a 012 module is required.

Personal Fax Messaging and most group mailbox applications require that DID numbers or personal lines be assigned to each personal mailbox subscriber or group administrator. A separate DID extension number is needed for each subscriber using the Personal Fax Messaging application.

The number of voice channels required for AUDIX Voice Power depends on the number of incoming lines/trunks, the number of subscribers programmed for the system, and the number of busy-hour calls. Table 4–8 shows these requirements.

Table 4–8. Voice Channels Required: IS III

No. of Channels Required	Lines	Subscribers	Busy-Hour Calls
2	1 to 6	1 to 20	1 to 20
4	7 to 18	21 to 60	21 to 60
6	19 to 24	61 to 80	61 to 80
8	25 to 42	81 to 200	81 to 200
12	Over 42	201 to 300	201 to 300

Fax Attendant's Fax Mail and Fax Call Answering services share voice channels with AUDIX Voice Power's Voice Mail, Voice Call Answering, and Automated Attendant services; one port can handle all these services. The number of voice channels needed for Fax Attendant depends upon the services used and the traffic on each service, but Fax Response, Message Drop, and Information Service all require dedicated voice ports.

The number of data channels required for Fax Attendant depends on the number of faxes sent and received per hour, assuming three pages per fax. Table 4–9 estimates these requirements.

Table 4–9. Data channels Required

Faxes Sent/Received Per Hr.	Channels Required
1 to 20	4
21 to 80	8
81 to 130	12 (maximum)

Primary Rate Interface

Primary Rate Interface (PRI) is a standard access arrangement that can be used to connect the system to a network providing voice and digital data services.

Group IV Fax

Group IV (G4) fax is an application that enables the system to use the advanced Group IV fax equipment—one of the new services accessible with Primary Rate Interface (PRI) trunks. Group IV fax equipment provides several advantages:

- High-speed transmission
- High-quality laser reproductions
- High-speed, high-capacity printing
- Virtually error-free transmission
- Office copying on a fax machine

Documents received from Group IV fax equipment are virtually perfect reproductions of the original document. Therefore, any company involved in graphic media (such as detailed engineering or architectural drawings or advertising graphic layouts) is an ideal candidate for this application.

Depending on the fax machine's interface, you can connect the Group IV equipment in the following ways:

- Direct RS-232 (the recommended method)
- V.35 interface connecting to a 7500B data module RS-232D interface
- V.35 interface connecting to a 7500B data module

Each configuration requires additional equipment. See Chapter 5, "Data Communications Support," for additional information about Group IV fax.

Group Video Conferencing

Group video conferencing, available with PRI service, enables groups of people in different geographical locations to meet face to face. Conferees can exchange information, documents, ideas, and data while employing a variety of visual aids. Visual aids can include interactive writing and drawing, prepared text and graphic materials, and prerecorded audio and video material. Improved technology and superior camera optics and digital audio signals result in video pictures that are equal to commercial broadcast quality.

Users start video conferences from an easy-to-use control console and conduct the conference as easily as they operate a telephone. No special technical expertise is required.

The system supports PictureTel 4000 video conferencing. The video conference is made up of the following subsystems:

- **Conference Control.** Establishes and terminates connections and allows camera control (including pan, tilt, and zoom), as well as audio control for volume and privacy.
- **Video.** Includes a full-motion video camera, a video monitor, and various video switching circuits. May also include an auxiliary room camera, a document camera, and a video cassette player/recorder.
- **Audio.** Allows video participants to hear and speak at the same time. Includes microphones, a microphone mixer, and an echo canceler. Microphones may include tabletop microphones, wired lapel microphones, and/or wireless hand-held or lapel microphones.
- **Video Codec.** A signal processing computer that digitizes, merges, and compresses audio and video signal input from the camera and microphone mixer for transmission to the far-end conference unit.

You can integrate these basic components in a mobile roll-about console that can be wheeled easily into a conference room or executive office prior to a scheduled video conference call. Alternatively, the components can be built into a video conference room.

The system may require additional equipment, for example, an interface converter for the video codec, and you can add applications to the basic components to enhance the information being sent back and forth during an interactive broadcast. See Chapter 5, “Data Communications Support,” for additional information about video conferencing.

Centrex Operation

Centrex is an optional telephone service for business customers. Available in various packages from local telephone companies, it provides telephone features that formerly were available only from a PBX located on the customer's premises.

Basic Centrex features generally include three-way conference, drop, hold, recall, call forwarding, call waiting, call pickup, group pickup, and automatic callback. You may also be able to add additional features such as speed dialing and night service.

You can use the system with either full or limited Centrex service. Full Centrex service requires that each telephone has a Centrex line (called a *prime line*) and that users depend primarily on Centrex features for their communications needs. Full Centrex can also be used when only some telephones have direct lines, while others share lines or have no direct line assigned. Limited Centrex service is for customers who use the MERLIN LEGEND Communications System features for most of their communications needs.

IMPORTANT:
Normally, Centrex is only available on loop-start trunks. However, in some areas, this type of service is becoming available on ground-start trunks. The MERLIN LEGEND Communications System is not compatible with Centrex services provided on ground-start trunks.

Timed Switchhook Flash

In Releases 1.0 and 1.1, you can generate a timed switchhook flash on a prime line button or a **Pool** button only, as long as the call is not a conference call and the facility is a loop-start line. You cannot generate a timed switchhook flash for an outside call that terminates on an **SA** button.

In Release 2.0 and later, you can generate a timed switchhook flash on a call that terminates on an **SA** button, as well as on a prime line or **Pool** button. This includes, among others, transferred calls, calling group calls, and forwarded calls as long as the call has arrived on a loop-start line. A timed switchhook flash allows access to Centrex features and not the features of the communications system.

The following factors apply to timed switchhook flash (Release 2.0 and later):

- **Dial Access to Pools.** If you press the **Recall** or **Flash** button during dialing while connected to a trunk or when end-of-dial is reached, a timed switchhook flash is generated, the accessed trunk is kept, and restrictions are applied.

- **Automatic Route Selection (ARS).** While an ARS call is being dialed, a timed switchhook flash cannot be generated. When dialing is complete, pressing the **Recall** or **Flash** button generates a timed switchhook flash, the accessed trunk is kept, and restrictions are applied.
- **Rotary Trunks.** A timed switchhook flash cannot be generated during dialing. When dialing is completed, pressing the **Recall** or **Flash** button generates a timed switchhook flash, the accessed trunk is kept, and restrictions are applied.

Full Centrex Service

With full Centrex service, each telephone has a prime line to the Centrex provider's CO. The prime line allows users to dial outside numbers directly after dialing an access code (usually a 9). The prime line is also used to call other four-digit Centrex extension numbers (these may or may not be on the local system as well). System intercom lines are used to dial other extensions on the system.

Users with full Centrex service can send a switchhook flash using the **Recall** or **Flash** button without the on-site system intercepting or responding to the signals. Full Centrex service requires that the planning form for each MLX telephone using a direct line be marked for programming of a **Recall** or **Flash** button (code *775). In addition, **Conference**, **Drop**, and **Transfer** buttons are programmed to access host features, not system features.

Limited Centrex Service

Limited Centrex service is for customers who primarily use the system features but wish to retain access to the network or other Centrex locations by use of a limited number of lines.

With limited Centrex service, some telephones have direct Centrex lines, while others do not. Some telephones may be assigned ground-start, tie, or DID lines. Others may use **SA** buttons to access pooled facilities. Generally, users rely heavily on the features of the system.

In a limited Centrex configuration, the system provides the primary connection to the CO, serving as a *local* switch between the telephones and the CO. A switchhook flash, feature access code, or **Feature** button signal is interpreted by the system to be a system command, not a Centrex command.

Mode Differences

For full Centrex operation, configure the system for Behind Switch mode. For limited Centrex operation, in cases where Centrex features are dominant, also configure the system for Behind Switch mode; if system features are dominant, you should configure the system for Key or Hybrid/PBX mode.

You can use the system as a PBX behind a host system by combining the features of a Behind Switch system with the ground-start capabilities of a PBX system. If a ground-start line is connected directly to the control unit, the FCC considers the system a hybrid (MF or PF classification), regardless of the system mode. See "FCC Registration" in Chapter 1 for more information.

Considerations and Constraints

The following considerations apply to full Centrex service:

- During periods of high traffic, users may experience delays in obtaining dial tone from the host. If a user begins to dial too rapidly, the first and second digits can be lost and the call misdialed. This situation is more common when the host is another PBX, not a CO Centrex. With full Centrex service, the delay in dial tone can cause misdialing when using System Speed Dial or Personal Speed Dial.
- Dependence on loop-start lines during a high-traffic period can cause a glare condition when calls grab the same line simultaneously. The loop-start lines normally used in Centrex service do not protect against glare. This is not usually a problem in full Centrex configurations where every telephone has its own prime line.
- Some Centrex features require 2- or 3-digit codes for access. These must be obtained from the telephone company and provided to the customer at installation.
- Loop-start lines have higher cable losses than ground-start lines and cannot assure secure toll restrictions.
- Single-line telephones have limited functionality when connected directly to the CO Centrex host. They cannot access system features or make inside calls. They can, however, use all of the Centrex features.

  If a single-line telephone's Idle Line Preference is set to an intercom line, the telephone cannot use Conference, Transfer, or Drop, even on inside calls.

The following considerations apply to limited Centrex service:

- If you program the limited Centrex configuration for Hybrid/PBX mode, you can use calling groups, **Shared SA** buttons, pools, other features, and applications such as MERLIN MAIL, CAS, and CMS. If the mode is Behind Switch, you cannot use the applications.

- Extension numbers should be the last four digits of the Centrex prime line telephone number. A brief ring delay occurs when calling a Centrex or PBX host extension number because the call is being processed through two systems. No delay occurs when making a system intercom call.

MERLIN PFC Telephone

IMPORTANT:
This section is intended solely as an overview of the product. For comprehensive information about its use, see the documentation for the product.

The MERLIN PFC (Phone-Fax-Copier) telephone is a BIS-34D (34–button) display telephone with a built-in fax machine and personal copier that provides a fax machine and personal copier in one compact unit. See Figure 4–3 for an illustration of the PFC telephone.

The MERLIN PFC telephone allows users to:

- Make and receive inside and outside calls with the built-in speakerphone and use the BIS-34D telephone features provided by the system.
- Send and receive fax transmissions while using the telephone.
- Make quick photocopies while using the telephone.

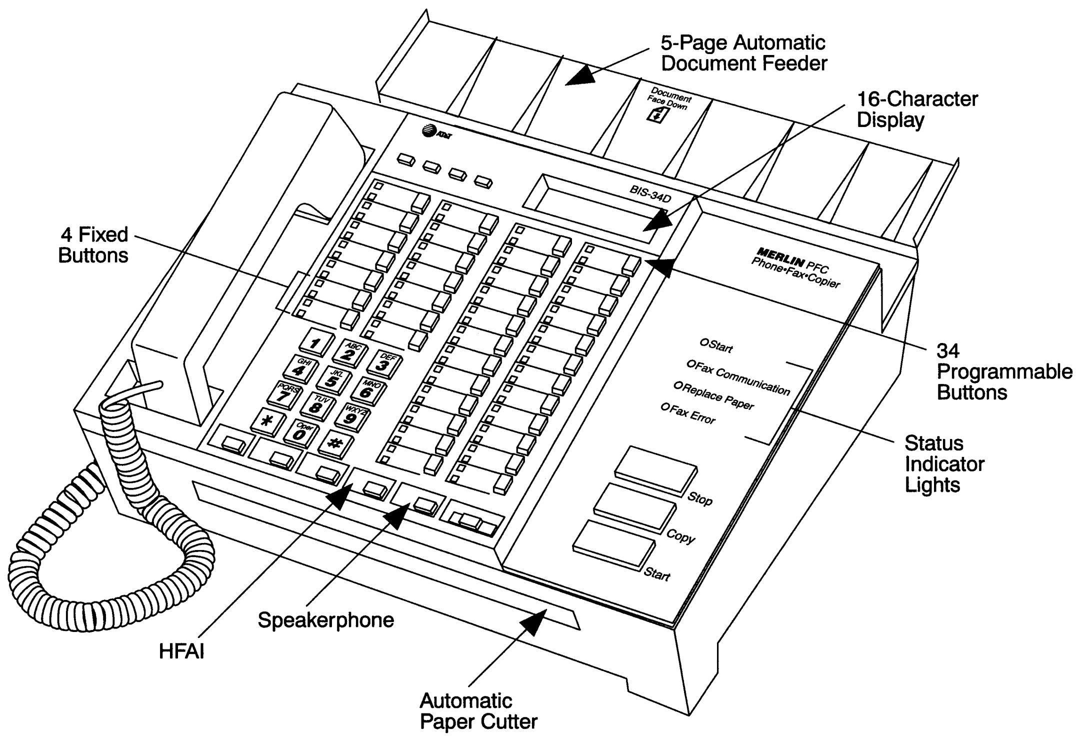

Figure 4–3. MERLIN PFC Telephone

The system must have two analog extension jacks available on the control unit. In Behind Switch mode, the system also requires a dedicated fax line for incoming fax calls; in Hybrid/PBX or Key mode, the system can have either a dedicated fax line or a Direct Inward Dial (DID) trunk.

NOTE:
The MERLIN PFC telephone's built-in fax does not transmit the date, time, and fax number.

Mode Differences

Hybrid/PBX and Key Modes

You must connect the dedicated fax line for incoming fax calls from the CO to a line/trunk jack in the control unit. You cannot assign the fax line to any pool.

If you use DID, you must assign a DID number to the fax extension.

If you use a dedicated private line, you must assign the fax line to the voice extension.

You cannot assign lines and line pools to the fax extension.

At the fax extension you should program the dedicated fax line to Immediate Ring and any other lines to No Ring.

Behind Switch Mode

You can assign the dedicated fax line only to the MERLIN PFC telephone fax extension.

You cannot assign the dedicated fax line to a pool.

You should assign the dedicated fax line as the secondary line on the MERLIN PFC telephone.

Considerations and Constraints

The MERLIN PFC telephone requires two analog extension jacks on the control unit: one for the voice line and one for the fax line.

You must install the telephone wiring between the control unit and the MERLIN PFC telephone in the same building.

You cannot install a MERLIN PFC telephone outside of the building.

You must remove all button assignments, except the one for the fax line, from the fax extension.

You should remove the Voice Announce feature from the fax extension.

Feature Interactions

If the dedicated fax line is shared for outgoing calls only, you must program the Ringing Option to No Ring at any extension except the MERLIN PFC Telephone fax extension.

Automated Document Delivery System

IMPORTANT:
This section is intended solely as an overview of the application. For comprehensive information about the use of the application, see the documentation for the product.

The Automated Document Delivery System (ADDS) is a computer-based system that stores documents in a database and automatically faxes them on request 24 hours a day. This type of fax application is called *fax response* or *fax-on-demand*, although it is different from the Fax Response feature of Fax Attendant with IS III and AUDIX Voice Power. ADDS has one voice port for handling incoming calls and one fax port for delivering fax documents. Using a touch-tone telephone, a caller accesses the system and is guided by prompts through the process of selecting a document and indicating the fax number to which the information is to be sent. The caller then receives the requested information in minutes by fax transmission.

Callers may be required to enter a password to gain access to ADDS. Access to system programming also requires the use of a password. The application can be configured to allow callers to request more than one document per call. Also, callers can leave a message after requesting a document.

For each transmission, a record is maintained by ADDS, including the filenames of documents the system has transmitted or attempted to transmit, the phone number of the destination fax machine, the time and date of the transmission attempt, and whether the transmission succeeded or failed.

Considerations and Constraints

Using one line for fax transmission limits ADDS to approximately 100 calls per day. Businesses anticipating more than 100 calls per day should consider IS III with AUDIX Voice Power and Fax Attendant.

You can use ADDS with an Automated Attendant.

ADDS must be connected to an 012 module, not to a telephone using an adapter.

Platform Requirements

To set up ADDS, a business must have:

- ADDS software
- A touch-tone telephone
- A Group III (G3) fax machine with an integrated handset

To request and receive information, a caller must have a touch-tone telephone and a Group III fax machine.

CONVERSANT

IMPORTANT:
This section is intended solely as an overview of the application. For comprehensive information about the use of the application, see the documentation for the product.

CONVERSANT is a voice response system that enables a user to run Integrated Voice Response (IVR) applications. CONVERSANT can automatically answer and route calls and execute telephone transactions. It is particularly useful for order-taking, for example.

You can configure the CONVERSANT software in one of two ways:

- As an application development environment in which all the tools to create an application are available
- As a platform to run applications that are already developed

CONVERSANT consists of the hardware and software that supports transaction processing, data retrieval, and data entry using a touch-tone telephone connected to a public telephone network. When a telephone connection is made to CONVERSANT, the application running on CONVERSANT prompts the caller with a synthesized voice in an application-dependent dialogue. The caller enters the appropriate responses by using the touch-tone keys on the telephone. This interaction continues until the caller ends the call.

You can develop applications that allow calls to be transferred to an attendant telephone during some part of the dialogue. Calls also can be transferred to an attendant telephone automatically if the application determines that an attendant is required. CONVERSANT Intro also supports scripts that allow callers to record and play back information.

CONVERSANT offers the following capabilities:

- Customized inbound call management or call routing
- Functions that are performed by choosing options in windows displayed on the screen
- Multiple script configuration possibilities that allow for different paths within the same script for handling calls during normal business hours, after hours, and on holidays
- Simple prompt recording using a telephone

- Optional seasonal greetings to be played during set time intervals
- Interaction of applications with voice mailboxes, with the ability to leave and retrieve messages, execute voice mail scripts, or get subscriber information
- Creation of tables and retrieval and updating of data using database tables
- Logging and displaying error messages
- Management reports and a system monitor for monitoring daily and ongoing system progress

Considerations and Constraints

CONVERSANT supports a maximum of 24 channels of analog ports, or up to 6 IVP4 boards. In a co-resident environment, such as CONVERSANT and AUDIX Voice Power, the system supports a maximum of 16 channels. The number of channels assigned to AUDIX Voice Power can *never* exceed 12.

Platform Requirements

The platform for CONVERSANT is the Master Controller III, a high-performance 32-bit computer built around an 486SX microprocessor. It has 8 MB of random-access memory (RAM) and a 500-MB fixed disk drive.

The system unit has a 250-MB tape drive and a 3.5-in. floppy disk drive. Two serial ports and one parallel port are integrated on the main board with connectors on the back panel of the system unit. A diskette drive controller and fixed disk drive interface are also integrated on the main board. A Video Graphics Array (VGA) video display controller and a tape drive controller are provided on separate add-in boards. Six additional Extended Industry Standard Architecture (EISA) slots are available for other Input/Output (I/O) cards.

The Master Controller III uses AT&T UNIX System V version 3.2.2. It includes a system unit, a monitor, and a keyboard.

Picasso Still-Image Phone

IMPORTANT:
This section is intended solely as an overview of the application. For comprehensive information about the use of the application, see the documentation for the product.

This single-line telephone allows users to transmit full-color, still images and discuss them simultaneously. It allows rapid, accurate communication of visual images for remote conferences, project reviews, and remote presentations. It provides the following features:

- Storage of up to 32 images in the phone itself
- Hardcopy output from a video printer supplied by the customer
- Optional wireless remote control
- Optional real-time annotation using a Windows-compatible annotation device such as a mouse
- Windows interface software
- Support for required customer-supplied input device
- Support for required customer-supplied monitor

Considerations and Constraints

Image transmission requires 5 to 40 seconds.

The customer supplies a required input device and monitor.

Both parties on a call must have a Picasso Still-Image Phone.

Platform Requirements

The Picasso Still-Image Phone connects to a port on an 012 module and uses any analog central office line/trunk. Depending upon the customer's configuration and the optional features the customer requires, the following components may be needed:

- Windows version 3.1 or higher
- Replacement handset
- Annotation device
- TV monitor or LCD monitor (required)
- Customer-supplied camcorder, electronic camera, document camera, VCR, or photo CD player as an input device (required)
- AC power supply with power supply cord (required)
- RCA video cable
- 4-pin mini DIN video cable
- BNC-RCA adapter
- RF modulator
- F connector cable
- 6-outlet power strip

Data Communications Support

5

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Data Communications Support

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The system uses circuit-switched connections to establish a dedicated communications path between two data terminals for the transfer of data. This connectivity enables you to share the system's resources and to manage connections between computers and other data input and output devices. You can also use a variety of voice features, such as Automatic Route Selection (ARS), calling restrictions, and Idle Line Preference to enhance data facilities.

This chapter describes the system's data communication capabilities, the configurations and features that support those capabilities, and typical data communication applications. For instructions on making and answering data calls, see *Data User's Guide*. For information about installation, see *Installation*.

Overview

The system's control unit, in conjunction with other external devices, provides data connectivity for the following:

- Internal data stations connected to analog extension jacks and equipped with modems, called *modem data stations*
- Internal data stations connected to MLX extension jacks and equipped with modems, also called *modem data stations*
- Internal digital data stations connected to MLX extension jacks and equipped with ISDN 7500B Data Modules, called *7500B data stations*
- Connection to external modem data stations using analog ground-start, loop-start, tie, or Direct Inward Dial line/trunks; dedicated analog facilities; DS1 T1 service emulating ground-start, loop-start, tie, or Direct Inward Dialing functionality or Primary Rate Interface (PRI)
- Connection to external digital data stations using a PRI facility

- Circuit-switched connections between two system data stations that are connected to the same types of extension jacks (analog or MLX)
- Data hunt groups (DHGs)
- Simultaneous voice and data on analog and MLX extension jacks
- On-premises host computer access
- Local area network (LAN) access through a modem or 7500B data module connected to an RS-232 port on a workstation on the LAN

The following sections describe the various equipment configurations and connectivity arrangements noted above. The system also supports the use of *modem/7500B data module pools*, which are described in application note, *MERLIN LEGEND Communications System Modem Pooling*, and allow communications between modem data stations that communicate over analog facilities and 7500B data stations that communicate over digital facilities.

Data Stations

A *data station* consists of *data terminal equipment* (DTE), such as a personal computer (PC), terminal, printer, or fax machine, and *data communications equipment* (DCE), such as an internal or external modem or a 7500B data module. The DTE connects to the control unit through the DCE, which has capabilities similar to a telephone. The DCE places the data call, maintains its connection, and terminates the data call.

Data stations (PC-based or workstation-based) require communications or terminal emulation software to transfer and receive data. The communications setup for each data station depends on specific configuration requirements and equipment limitations. (See the communications or terminal emulation software documentation for instructions.)

The DTE and DCE may have hardware and/or software options, such as parity and bit rate, that you can set for transferring and receiving data. Other options may differ, depending on whether the transfer of data is synchronous or asynchronous. See the documentation packaged with the data terminal equipment (DTE), communications software, and data communications equipment (DCE) for configuration compatibility requirements, options for transferring and receiving data, and guidelines for changing options.

Data stations may include a telephone for users who also need voice capabilities. The user may only be able to use one device at a time, or, with the right setup, he or she can use both the data terminal and the telephone simultaneously. When a data station uses an MLX extension jack, only one jack is required to provide simultaneous voice and data transmission; either device can dial. When a data station uses an analog extension jack, two jacks and an analog multiline telephone are needed for simultaneous voice and data; only the telephone can dial. If a telephone is used at a data station served by a single analog extension jack, the telephone dials; it cannot be used for voice communications during data communications sessions.

Modem Data Stations

A modem data station uses a modem as its DCE and communicates over analog line/trunks, rather than the digital line/trunks offered by the Primary Rate Interface (PRI) arrangement. The modem converts the signals from the DTE into signals usable over telephone lines. It transmits these signals as continuously varying electrical voltages in the voice-frequency band. It converts incoming analog line signals into the correct format for the DTE. You can connect most types of Hayes-compatible modems to the control unit (see "Modems" in Chapter 2 for information about recommended modems).

From a physical viewpoint, the modem provides a T/R analog signal interface to the telephone network or communications system and an EIA RS-232 digital interface (or other type) to the data equipment.

On this system, the tip/ring (T/R) interface for an modem data station is provided by one of the following:

- An extension jack on a T/R module (012 T/R or 008 OPT) for a modem directly connected
- A General-Purpose Adapter (GPA) for analog multiline telephones
- A Multi-Function Module (MFM) for MLX telephones

Modem Data Station Configurations

T/R interfaces support the following modem data stations:

- **Analog Voice and Modem Data.** This data station includes a data terminal (with keyboard and display) or a computer connected to a modem through an EIA-type RS-232 interface. The GPA connects the modem to the analog multiline telephone to provide the T/R interface for the modem. The telephone provides the dialing capability for the data station. Providing simultaneous voice and data requires two adjacent odd/even extension jacks on a 408, 408 GS/LS, or 008 module. The even jack is for voice, and the odd jack is for data. The bridging adapter joins the odd/even jack pair for connection to the analog multiline telephone.

NOTE:
If the distance between the extension jack and the modem exceeds 5000 ft. (1524 m), this configuration may require *data grade* facilities with 4.5 dB maximum loss.

- **Modem Data-Only.** This data station includes a data terminal (with keyboard and display) or a computer connected to a modem through an EIA RS-232 interface.

  NOTE:
  A modem data-only station may be off-site, in which case the modem connects to a jack on a 008 OPT module.

  The modem connects to an extension jack on an MFM, 012, or 008 OPT module. A telephone may or may not be connected, depending on modem capabilities. The terminal keyboard provides the dialing capability for the data station. This data station does not allow simultaneous voice and data transmission. The maximum data transmission speed is 9600 bps for an 012 module and 4800 bps for an MFM & 008 OPT module.

- **MLX Voice and Modem Data.** This data station includes a data terminal (with keyboard and display) or a computer connected to a modem through an RS-232 interface. The modem connects to an MLX telephone equipped with an MFM. The MFM provides the T/R interface for the modem (see "Multi-Function Module" in Chapter 2 for a description of the MFM). You use the data terminal keyboard to dial data calls and the telephone dialpad to dial voice calls. You cannot use the MLX telephone to dial data calls, or the data terminal keyboard to dial voice calls. Each device operates independently of the other, and you assign features to each device independently. This data station allows a maximum speed of 4800 bps.

Other Supported Modem Data Endpoints

Other data equipment that you can connect to a modem data station includes the following:

- A local host computer, as described later in this chapter
- Group III (G3) fax terminal
- A device that shows what is received, such as a printer or display
- A personal computer equipped with an internal modem card

7500B Data Stations

The 7500B data module adapts the PC or data terminal to the MLX environment. It is configured between the EIA type RS-232 interface to the data equipment and the MLX extension jack in the control unit. The 7500B data module does not convert signals from the DTE to analog line signals but sends them as a sequence of separate electrical impulses (see "7500B Data Module" in Chapter 2 for hardware specifications and requirements). Along with the features described in Chapter 2, the system also supports the 7500B data module's capability for circuit-switched data connections on a B-channel.

To communicate with the system, the digital data endpoint uses D-channel messages during call setup and termination. When a data call is set up, the system establishes a connection between the calling and called endpoints on a B-channel. The system sends the appropriate messages to drive the endpoints into data mode. In this mode, the 7500B data module transmits and receives data over a B-channel using transport modes as defined in the Digital Multiplexed Interface (DMI).

This capability uses data transport mode 3/2 adaptive or mode 2 only for asynchronous transmission, and modes 0, 1, and 2 for synchronous transmission. The transport modes are defined in the DMI, and the system does not interact with modes of data transport. (The 7500B data module can share the MLX port with an MLX telephone; however, the two units operate independently of each other. See "7500B Data Station Configurations" for details.)

The 7500B data module and the terminal keyboard can both provide dialing and answering capabilities to the data station.

NOTE:
Although the 7500B data module supports packet-switched data on the D-channel, the system does not support this data mode, so you cannot use it.

MLX Port Connection Requirements

When you connect digital data equipment to an MLX port, the following requirements and/or restrictions apply:

- You should connect only *one* 7500B data module to an MLX port. When two 7500B data modules are connected, the system cannot address a specific 7500B data module for incoming calls. Although outgoing calls can be made, incoming calls may not be answered by the intended party.
- If no MLX telephone is connected, you must install a 440A4 terminating resistor adapter to provide 100-ohm termination for each transmission pair. The 7500B data module does not provide termination.

- An MLX telephone works independently from the 7500B data module; however, the telephone may cause B-channel conflict between the telephone and the 7500B data module when it is voice-signaled while active on a call. If a slight chance of data call blocking is unacceptable, either do not connect an MLX telephone to a 7500B data module or disable the Voice Announce feature at the extension.
- The maximum cord length from an MLX telephone to a 7500B data module is 80 ft. (24 m). Keep this in mind if you plan to use a port's voice capability and the MLX telephone is located some distance away from the 7500B data module.

7500B Data Station Configurations

- **MLX Voice and 7500B Data.** This data station includes a data terminal (with keyboard and display) or a computer connected to a 7500B data module through an RS-232 or V.35 interface. The 7500B data module connects to a jack on a 008 MLX, 408 MLX, or 408 GS/LS-MLX (Release 2.0 and later only) module. The telephone and the 7500B data module share the MLX port, but they operate independently of each other. The terminal keyboard provides the dialing capability for the data station.

  NOTES:
  1. For MLX voice and 7500B data stations, the MLX telephone cannot contain an MFM, because the MFM interferes with communication to the system.
  2. The Voice Announce feature should be turned off at this type of station, because the system requires both MLX B-channels for voice signaling.

- **7500B Data-Only.** This data station includes a data terminal (with keyboard and display) or a computer connected to a 7500B data module through an RS-232 or V.35 interface. There is no MLX telephone, so the control unit requires a 440A4 terminating resistor. The 7500B data module connects to a 008 MLX, 408 MLX, or 408 GS/LS-MLX (Release 2.0 and later only) module. The keyboard provides the dialing capability.

Other Supported 7500B Data Stations

Other data equipment that you can connect to a 7500B data station through a 7500B data module includes the following:

- Video conferencing system
- Group IV (G4) fax machine

See "PRI Applications" later in this chapter.

Data Station Configurations Summary

Table 5–1 summarizes the hardware requirements and port assignments needed to support the various data station configurations.

Table 5–1. Configuration of Data Stations

	Equipment Configuration					
Type of Data Station	**Computer or Terminal**	**Modem**	**7500B Data Module**	**Analog Telephone**	**MLX Telephone**	**Type of Module**
Analog voice and modem data	✔	✔		✔ Requires GPA		008 408 408 GS/LS
Modem data only	✔	✔		May connect phone		012 008 OPT
MLX voice and modem data*	✔	✔			✔ Requires MFM	008 MLX 408 MLX 408 GS/LS-MLX
MLX voice and 7500B data*	✔		✔		✔ No MFM allowed	008 MLX 408 MLX 408 GS/LS-MLX
7500B data only	✔		✔			008 MLX 408 MLX 408 GS/LS-MLX

* Disable the Voice Announce feature for this type of data station.

Data Hunt Groups

A data hunt group (DHG) is a calling group to which are assigned data stations connected to the same type of extension jack, either MLX or analog. When a call is placed to a DHG's number, the system performs a circular search (starting with the extension listed after the one that received the last call) to find the first idle extension. The system then alerts the first idle extension it finds. If the extension answers the call, the system connects the extension to the originator. If all the extensions in the DHG are busy, the originator hears ringback. The system accommodates up to 32 calling groups. Each calling group can have up to 20 members. A data station can be a member of only one calling group.

The following list shows examples of data hunt groups:

- 7500B data-only and MLX voice and 7500B data stations that communicate with a local host computer's 7500B data module.
- Modem data-only and analog voice and modem data stations that communicate with a local host computer's modem.
- 7500B data-only and MLX voice and 7500B data stations that communicate with a workstation (gateway) on a LAN that is equipped with a 7500B data module.
- Modem data-only and analog voice and modem data stations that communicate with a workstation (gateway) on a LAN that is equipped with a modem.

NOTE:
MLX voice and modem data stations cannot be grouped with analog voice and modem data stations or modem-only data stations, because they connect to different types of extension jacks (see Table 5–1). These data stations require a modem/7500B pool in order to communicate with other types of data stations.

DHGs support the following:

- Modem/7500B pools
- Dedicated lines for data service (DID trunks can be used for inbound services only)
- A host computer with multiple ports
- A workstation (gateway) on a LAN

Local Area Network Connectivity

A Local Area Network (LAN) is an interconnected chain of PCs or terminals that pass data to and from a file/client server computer or among themselves. The control unit connects to the LAN through a workstation that operates as a *gateway.* The gateway provides the ports for the modem and 7500B data module connection to the control unit and to the LAN. It also provides the protocol required for the transfer of data between a data endpoint on the system and the workstation (or data terminal) on the LAN.

The modems connect to the gateway through EIA-type RS-232 (or other type) interfaces and provide the T/R interface to the 012 module. The 7500B data modules also connect to the gateway through EIA-type RS-232 interfaces (or other type); on the system end, they connect to the 008 MLX, 408 MLX, or 408 GS/LS-MLX module (Releases 2.0 and later only). The system requires terminating resistors for the 7500B data module connection.

Once a connection is established between a data station on the system and a workstation within the LAN, all of the features and capabilities of the LAN environment are available to the data station on the system. However, limitations or hardware requirements may restrict the usage of some LAN capabilities.

Outside Trunks

The types of outside trunks that you can use to make and receive data calls to and from data stations outside the system are described in the following sections.

Ground-Start and Loop-Start

A ground-start trunk is preferred for communication with outside modem data stations. Ground-start trunks provide improved signaling and reliable far-end disconnect for secure toll restriction. However, a loop-start trunk is standard for home and small businesses and is the least expensive trunk in some areas. Keep in mind the following disadvantages of loop-start trunks:

- They do not protect against glare, a condition that occurs when an outside call is made at the same time that an incoming call arrives on the same trunk. Thus, an outside caller may accidentally connect to a modem.
- They cannot provide reliable far-end disconnect for toll restriction.

You can use these types of ground-start/loop-start trunks for data communications: basic, WATS, FX, and 800 service (IN-WATS).

You can connect ground-start and loop-start trunks to ground-start or loop-start jacks on these modules: 800 GS/LS, 400 GS/LS/TTR, 408 (LS trunks only), 408 GS/LS, 400 (LS trunks only), 800 (LS trunks only), 408 GS/LS-MLX (Releases 2.0 and later only). You can connect loop-start trunks to 800 LS-ID modules (Release 3.0 and later only) as well, but since this module provides calling number identification information to MLX display telephones, it is not the best choice for data stations.

Tie

A tie trunk provides communication between two telephone communication systems. A tie trunk "ties" the two systems together, providing access to all telephones or data stations on each system. Tie trunks are often used for data communication with modem data stations that are connected to a system at a different location (for example, a different floor of a building, a different building, or a different city or state). Tie trunks connect to jacks on the 400EM module.

NOTE:
The system only supports analog data transmission through dedicated analog tie trunks that connect two MERLIN LEGEND Communications Systems.

Direct Inward Dial

DID trunks allow incoming calls to reach specific extensions in the system without operator assistance. DID trunks are available only in Hybrid/PBX mode. Use a DID trunk to receive incoming calls from outside modem data stations; it cannot be used for outgoing calls. DID trunks connect to jacks on 800 DID modules.

Digital Signal 1 Facility

This facility carries digital signals in DS1 format. DS1 format multiplexes 24 Digital Signal 0 (DSO) channels of 64 kbps each and one 8-kbps framing signal, for a total of 1.544 MBps. You can use a DS1 trunk for communication with outside digital or modem data stations.

DS1 trunks connect to jacks on 100D modules. Even though the 100D module only has one jack, it supports up to 24 logical trunks for voice and data calls. Each channel in the DS1 signal corresponds to a logical trunk.

A DS1 trunk provides either T1-only or PRI access:

- Use a T1 facility for communication with outside modem data stations. You can program the 24 channels on a T1 facility individually in any combination to emulate a loop-start, ground-start, E&M tie, or DID trunk, so a single 100D module replaces 24 trunks. 7500B data calls cannot be placed through this facility unless PRI is used.
- PRI is the standard format provided by connection to a 5ESS CO switch or a 4ESS toll switch or in Release 2.0 and later, direct connection to DEFINITY systems. You only send 7500B data over PRI.

PRI service offers the same benefits for data calls as for other types of calls (see Chapter 3, "Digital Signal 1 Facilities," for more information about DS1 and PRI). In particular, however, speed of data communication is much improved over DS1 facilities. Digital data transmission can take place at speeds up to 19.2 kbps for asynchronous communications and 64 kbps for synchronous communications. PRI also provides Accunet®-switched digital service for 56-kbps, 64-kbps restricted, and 64-kbps clear circuit-switched data calls.

Data Communication Features

You can use the following system features provided for voice service to enhance data communications:

- **Account Code Entry.** Allows tracking of outgoing data calls for billing, forecasting, or budget reports.

- **Auto Answer All.** Allows a modem with auto-answer capability to answer data calls when the user is away from the station. Use this feature for analog voice and modem data stations only.
- **ARS** (Hybrid/PBX mode only). Routes calls over outside trunks according to dialed number and trunk availability. Therefore, you can program the system to select the least expensive route for each data call.
- **Calling Restrictions** (such as Allowed Lists and Disallowed Lists). These features allow companies to control and manage communications costs for outgoing data calls.
- **Data Status Button.** Monitors call activity of data stations. The green LED on the button lights to indicate busy status. Unlike programmed Auto Dial and Signaling buttons, which dial specific numbers, the programmed Data Status button does not *dial* numbers. Pressing it, therefore, has no adverse effect on data calls in progress. The Data Status button is the only button you should use to monitor data station activity.
- **Personal Line.** Provides access from a data station to outside lines.
- **Idle Line Preference.** Automatically selects the first available line for data calls.
- **Last Number Dial.** Automatically places a call to the last number dialed from that extension. When programming this feature, remember to include a dial-out code for outside calls, if your system requires it.
- **Personal Speed Dial and System Speed Dial.** Allows quick dialing of frequently used numbers on 10-button phones or at data terminals. When programming these features, remember to include a dial-out code for outside calls if your system requires it.

  NOTE:
  Many data communications software packages provide a directory features that allows speed dialing of frequently-used telephone numbers from a PC.

- **Pool Access to External Transmission Facillties** (Hybrid/PBX Mode only). Allows data endpoint dialing to seize access to pool numbers for outside lines and trunks.
- **Privacy.** Prevents loss of data by ensuring that data transmission is not interrupted accidentally. The Privacy feature is automatic for data calls on 7500B data stations and on modem data stations with analog multiline telephones. You must activate it manually on other modem data stations.

The following system features may interfere with data connections and should be disabled:

- Voice Announce at stations where there is an MLX telephone
- Call Waiting (factory setting: off)

- Automatic Callback (factory setting: off)

Station Communication Features

The system supports the use of data equipment or data software. These features are provided by the data station hardware/software in the PC, data terminal, or DCE device (modem or 7500B data module):

- Data transport mode selection (for example, 7500B data modules support DMI modes 0, 1, 2, and 3)
- Data metering, or speed matching, of bit rates between digital data endpoints
- Data terminal dialing
- Automatic answering of data calls by the DCE (used with the system Auto Answer All feature at analog voice and modem data stations)

PRI Applications

The system provides an interface between PRI services and businesses. It also supports some advanced digital applications. These applications include high-speed fax transmissions and video conferencing.

This following sections provide the configurations supported for fax transmissions on Group IV (G4) fax machines and a general description of video conferencing connections.

Fax Transmissions Application

Group IV fax machine is a fax unit, offering 400 by 100 dots per inch (dpi) in fine mode. It can operate at any speed for communications with a Group III (G3) fax machine or another Group IV fax machine. When speed is essential, it can transmit data at 64 kbps and achieve speeds as fast as 3 seconds per page.

Supported Configurations

There are three ways to connect a Group IV fax machine to the system for transmitting and receiving data:

- Direct RS-232 interface (requires 8-pin connector/adapter)
- RS-232 to V.35 interface conversion (requires 8-pin connector/adapter)
- Direct V.35 interface (requires 8-pin connector/adapter)

These configurations are a guide, not an assurance that different fax machines with other proprietary interface connections will operate properly. The configurations are based on compatibility testing of Canon and Ricoh fax models on this system. The communications device is the 7500B data module (see "7500B Data Module" in Chapter 2 for details).

Group IV fax machines are available from a number of manufacturers, each of whom uses proprietary interface connections. One manufacturer may even use different interfaces as standard from one model to another. Some require the buyer to specify various CCITT versions for the standard EIA-RS-232 interface, for example V.11, V.28, or V.35. In most cases, these interfaces are simple plug-in connections and are off-the-shelf items. It is important to know what interfaces are required *before* buying the fax machine. This also means knowing the transmission mode in which the machine normally operates—asynchronous or synchronous. (When operating behind this system, the fax transmission is synchronous.)

Direct RS-232 Interface

This is the recommended configuration. The other two methods, described later in this section, are shown as alternatives when the EIA interface on the fax machine requires that they be used. The system does support the three methods, but there is virtually no advantage in using one method rather than another; it is only a matter of which interface the fax machine uses. Figure 5–1 shows the configuration for a direct RS-232 interface, a synchronous DTE-to-synchronous DCE configuration that includes communication between a 7500B data module and a Group IV fax machine.

To use this configuration, it is important that you order an EIA-RS-232 connection, with V.28 or V.35 interface, with the fax machine. In this configuration, there is a direct connection between the Group IV fax machine and a 7500B data module. The standard RS-232 interface on the back panel of the Group IV fax machine connects an EIA-232D cable to the RS-232D interface jack (Port 1) on the 7500B data module. This connection is a simple jack-to-jack plug-in operation. It provides the imagery transmission path.

The 7500B data module connects an unattended DTE or another DCE to a 008 MLX or 408 GS/LS-MLX (Release 2.0 and later only) module. This application shows the Group IV fax machine as synchronous DTE, sending and receiving image transmissions through the digital network.

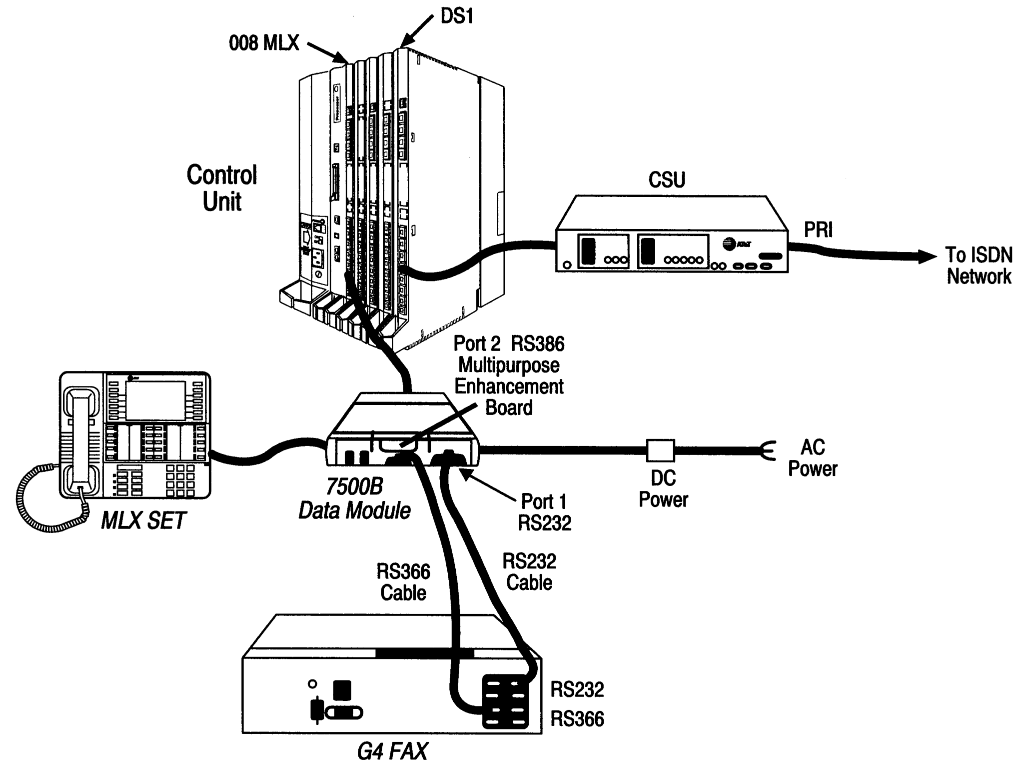


Figure 5–1. Direct RS-232 Interface

The dialing path in this configuration requires a Multipurpose Enhancement Board that is installed in Port 2 of the 7500B data module. The dialing path is then established by connecting the RS-366 jack on the Group IV (G4) fax machine to the RS-366 jack in Port 2 on the back panel of the 7500B data module. An RS-366 cable is required for the connection.

The LINE jack on the back panel of the 7500B data module connects to an MLX port on an 008 MLX or 408 GS/LS-MLX module (Release 2.0 and later only). Completing the link is the PRI facility connection that plugs into the jack on the Channel Service Unit (CSU) shown in Figure 5–2. The CSU is then plugged into the 100 DS1 Module in the control unit.

Use this connection when the fax transmission telephone numbers are dialed manually from the fax dialpad. You should also place the 7500B data module in a safe location, to avoid accidental tampering with configuration settings that could result in misdialed calls.

For the 7500B data module to operate properly, the DCE/DTE Flipboard circuit card in the unit must be in the proper position before the Group IV fax machine can operate as a DCE unit.

The MLX telephone in the configuration shares the same MLX port as the 7500B data module and can be used to send and receive voice calls. It is not essential to the fax operation. However, if an MLX set is not connected to the 7500B data module, a 100-ohm terminating resistor adapter must be installed close to the 7500B data module on the line to the control unit.

RS-232 to V.35 Interface Conversion

Use this configuration when users need to operate other adjunct equipment from the 7500B data module, Other equipment may include modems, automatic calling equipment (RS-366 interface), or DTE with a V.35 interface. Figure 5–2 shows the connections required for a Group IV fax V.35 interface connecting to a data module RS-232D interface.

You must use a V.35-to-RS-232 converter between the fax machine and the 7500B data module. This configuration also requires the use of a Multipurpose Enhancement Board in Port 2 of the 7500B data module for setting up the RS-366 dialing interface. This configuration allows the Group IV fax machine to be connected to Port 1 for image transmission. The fax machine is connected by a V.35 cable to the converter's V.35 port, and then from the converter's RS-232 port to the 7500B data module's RS-232 port. The dialing is accomplished from the RS-366 interface on the fax machine to the RS-366 interface in Port 2 of the 7500B data module.

The Multipurpose Enhancement Board provides an RS-366 auto dial interface on Port 2. It converts the RS-232 interface on Port 1 on the main circuit board from asynchronous to synchronous mode. You must order the V.35 adapter cable separately from the board in order to operate at data rates of 56 and 64 Kbps. Without the cable, data rates are limited to 1200, 2400, 4800, 9600, and 19,200 bps.

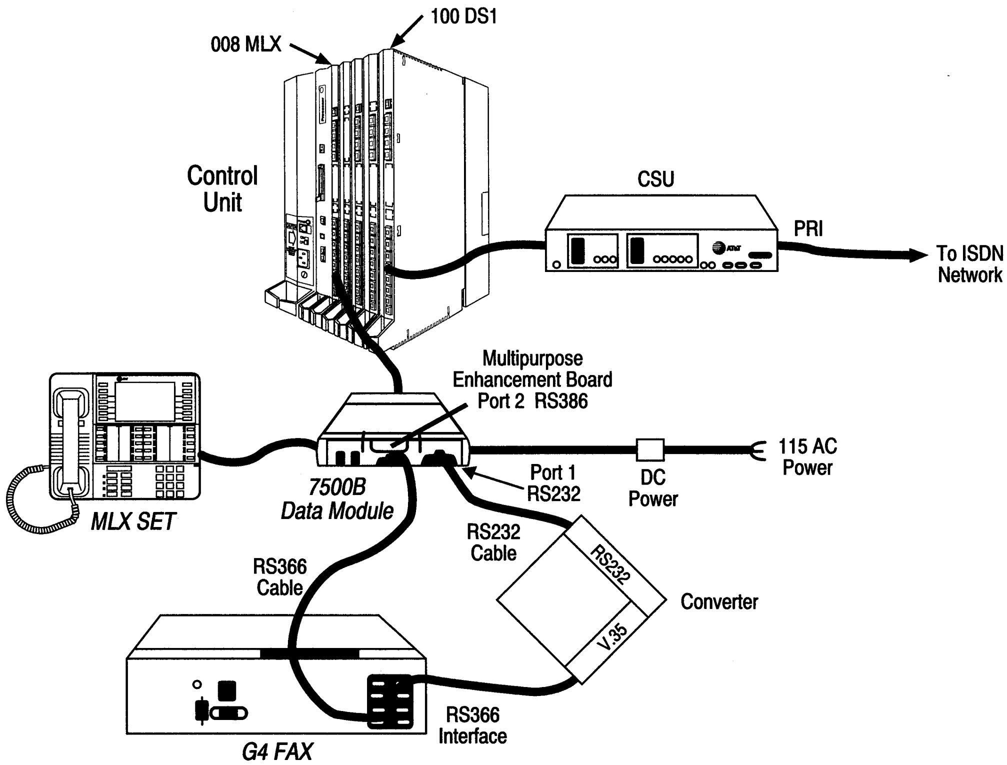


Figure 5–2. RS-232 to V.35 Interface Conversion

A D8W cord connects the 7500B data module to the MLX port on the 008 MLX or 408 GS/LS-MLX module (Release 2.0 and later only). The 100 DS1 module establishes the PRI interface to the CSU and then to the ESS switch.

For the purposes of the compatibility tests, a Shore Microsystems Model SM-100 RS-232/V.35 converter was used to test this configuration. The converter is customer-supplied equipment that can be purchased from a data equipment vendor.

Direct V.35 Interface

If the Group IV (G4) fax machine is equipped with a CCITT-V.35 interface, use this configuration method. If the fax machine is equipped with a V.35 interface, you must use an optional connection board with the 7500B data module (see below). This configuration is required if the customer does not wish to purchase a converter or if the dialing is to be done on the front face of the 7500B data module.

This connection requires dialing from the 7500B data module. That is, you must locate the 7500B data module in a work area where it cannot be accidentally reconfigured. Accidental reconfiguration can cause fax transmissions to malfunction. Also, dialing for fax transmissions from the 7500B data module is subject to a high error rate, due to misdials.

Figure 5–3 shows the connection between a Group IV fax machine and a 7500B data module through an EIA-V.35 interface.

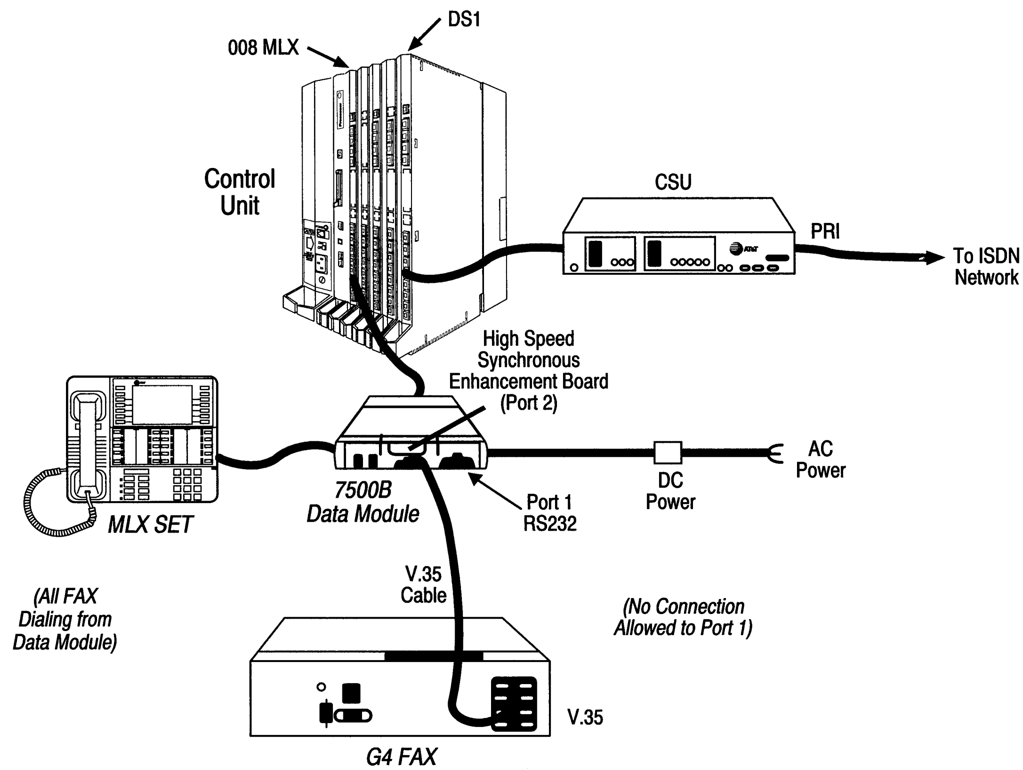


Figure 5–3. Direct V.35 Interface

This connection requires you to install a High Speed Synchronous Enhancement Board in the 7500B data module. This optional board provides a V.35 interface at synchronous data rates of 48, 56, or 64 Kbps on Port 2. The connection is through an V.35 external adapter cable that converts the 25-pin male connector on Port 2 to the industry-standard 34-pin V.35 interface. The cable is packaged with the board along with an adhesive V.35 label that you must affix to the back panel of the 7500B data module (so that Port 2 is not mistaken for a second EIA-232D interface). When you use a high-speed board, no connection to Port 1 is allowed.

In this configuration, no dialing connection is made between the Group IV fax machine and the 7500B data module, and all dialing must be done from the front panel of the data module.

A D8W cord connects the LINE jack on the 7500B data module to the 008 MLX or GS/LS-MLX module. The PRI interface is made from the 100 DS1 module.

Video Conferencing

This section describes, in general terms, the hardware requirements for connecting video conferencing capabilities to the MERLIN LEGEND Communications System. This information is intended as a guideline only. For additional information, see the documentation packaged with the video conferencing equipment.

Video equipment (cameras, monitors, microphones, speakers) must be connected to the MERLIN LEGEND Communications System control unit through a coder/decoder called a codec. The video codecs available in today's market are not equipped for direct connection to the control unit because the codec's do not have an ISDN interface port built into them. The codecs do provide a V.35 communication interface and an RS-366 dialing interface.

The codec must be connected to the control unit through two 7500B Data Modules, equipped with Multipurpose Enhancement Boards. The Multipurpose Enhancement Boards can support the RS-366 dialing interface, but does not support the V.35 communication interface required for the video codec—it supports V.24 communications interface. Converters must be installed between the video codec and the 7500B Data Modules.

The control unit must have 008 MLX or 408 GS/LS MLX modules installed for connecting the 7500B Data Modules with the Multipurpose Enhancement Boards in them. Also, a 100D module must be installed for connecting the trunk from the network to the control unit with PRI services be available from a 4ESS or 5ESS switch.

A T1 Channel Service Unit must be connected between the MERLIN LEGEND Communications System and the PRI interface at the customer location.

Hardware Requirements

- 008 MLX or 408 GS/LS-MLX module and 100D module in control unit
- ACCULINK 3150 Extended SuperFrame T1 CSU or ESF T1 CSU
- Two Shore Microsystems SM-100 EIA-232/V.35 converters (or equivalent)
- Two 7500B data modules

- Two 7500B data module feature package 2 upgrades
- Two Multipurpose Enhancement Boards (to provide synchronous communication and RS-366 ACU interface)
- Two WP-90110-L1 power supplies (one per stand-alone 7500B data module) Two 440A4 terminating resisting adapters
- Z77A mounting for multiple 7500B data modules
- Cables:
  - Two male/male EIA-232 cables, 8 ft. (24 m), to connect the PORT 1 jacks on the data modules to the EIA-232/V.35 converters
  - Two male/male V.35 Communication cables (DB-37), 8 ft. (24 m), to connect the V.35 communication ports on the video codec to the EIA-232-/V.35 converters.
  - Two male/male EIA-232 cables (DB-25), 8 ft. (24 m), to connect the RS-366 dialing port of the video codec to PORT 2 on the 7500B data module

Video Conferencing Connections

Figure 5–4 shows an example of video conferencing connections.

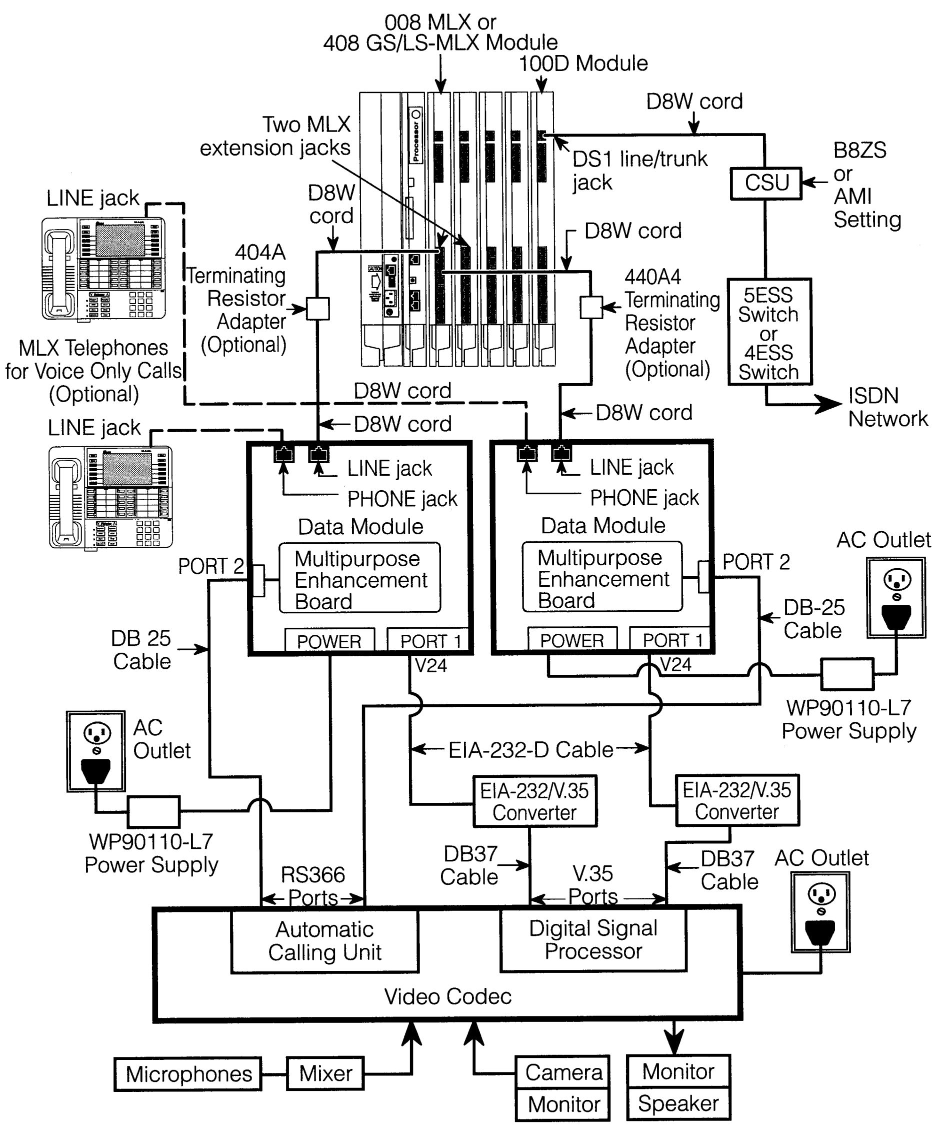


Figure 5–4. Video Conferencing Connections

NOTE:
If you connect MLX telephones, omit both 440A4 terminating resistors, and note that the maximum cord length from the 7500B data module to the telephone is 80 ft. (24 m).

Customer Support Information

A

Support Telephone Number

In the U.S.A. only, AT&T provides a toll-tree customer Helpline (1-800-628-2888) 24 hours a day. If you need assistance when installing, programming, or using your system, call the Helpline, or your AT&T representative.

Outside the U.S.A., if you need assistance when installing, programming, or using your system, contact your AT&T representative.

Federal Communications Commission (FCC) Electromagnetic Interference Information

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at his or her own expense.

Canadian Department of Communications (DOC) Interference Information

This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the radio interference regulations of the Canadian Department of Communications.

Le Présent Appareil Numérique n'émet pas de bruits radioelectriques dépassant les limites applicables aux appareils numériques de la class A préscrites dans le reglement sur le brouillage radioélectrique edicté par le ministère des Communications du Canada.

FCC Notification and Repair Information

This equipment is registered with the FCC in accordance with Part 68 of its rules. In compliance with those rules, you are advised of the following:

- **Means of Connection.** Connection of this equipment to the telephone network shall be through a standard network interface jack, USOC RJ11C, RJ14C, RJ21X. Connection to E&M tie trunks requires a USOC RJ2GX. Connection to off-premises extensions requires a USOC RJ11C or RJ14C. Connection to 1.544-Mbps digital facilities must be through a USOC RJ48C or RJ48X. Connection to DID requires a USOC RJ11C, RJ14C, or RJ21X. These USOCs must be ordered from your telephone company.
- **Party Lines and Coin Telephones.** This equipment may not be used with party lines or coin telephone lines.
- **Notification to the Telephone Companies.** Before connecting this equipment, you or your equipment supplier must notify your local telephone company's business office of the following:
  - — The telephone number(s) you will be using with this equipment.
  - — The appropriate registration number and ringer equivalence number (REN), which can be found on the back or bottom of the control unit, as follows:
    - If this equipment is to be used as a Key system, report the number AS593M-72914-KF-E.
    - If the system provides both manual and automatic selection of incoming/outgoing access to the network, report the number AS593M-72682-MF-E.
    - If there are no directly terminated trunks, or if the only directly terminated facilities are personal lines, report the number AS5USA-65646-PF-E.

- The REN (Ringer Equivalency Number) for all three systems is 1.5A.

— For tie line connection, the facility interface code (FIC) is TL31M and the service order code (SOC) is 9.0F.

— For connection to off-premises stations, the FIC is OL13C and the SOC is 9.0F.

— For equipment to be connected to 1.544-Mbps digital service, the FIC is 04DU9-B for D4 framing format or 04DU9-C for extended framing format, and the SOC is 6.0P.

— For equipment to be connected to DID facilities, the FIC is 02RV2-T and the SOC is 9.0F.

— The quantities and USOC numbers of the jacks required.

— For each jack, the sequence in which lines are to be connected, the line types, the FIC, and the REN by position when applicable.

- **Disconnection.** You must also notify your local telephone company if and when this equipment is permanently disconnected from the line(s).
- **REN.** The REN is used to determine the number of devices that may be connected to the telephone line. Excessive RENs on the line may result in the devices not ringing in response to an incoming call. In most, but not all, areas the sum of the RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to the line, as determined by the total RENS, contact the telephone company to determine the maximum REN for the calling area.

Installation and Operational Procedures

The manuals for your system contain information about installation and operational procedures.

- **Repair Instructions.** If you experience trouble because your equipment is malfunctioning, the FCC requires that the equipment not be used and that it be disconnected from the network until the problem has been corrected. Repairs to this equipment can be made only by the manufacturers, their authorized agents, or others who may be authorized by the FCC. In the event repairs are needed on this equipment, contact your authorized AT&T dealer or, **in the U.S.A. only**, contact the National Service Assistance Center (NSAC) at 1-800-628-2888.
- **Rights of the Local Telephone Company.** If this equipment causes harm to the telephone network, the local telephone company may discontinue your service temporarily. If possible, they will notify you in advance. But if advance notice is not practical, you will be notified as soon as possible. You will also be informed of your right to file a complaint with the FCC.

- **Changes at Local Telephone Company.** Your local telephone company may make changes in its facilities, equipment, operations, or procedures that affect the proper functioning of this equipment. If they do, you will be notified in advance to give you an opportunity to maintain uninterrupted telephone service.
- **Hearing Aid Compatibility.** The custom telephone sets for this system are compatible with inductively coupled hearing aids as prescribed by the FCC.
- **Automatic Dialers.** WHEN PROGRAMMING EMERGENCY NUMBERS AND/OR MAKING TEST CALLS TO EMERGENCY NUMBERS:
  - — Remain on the line and briefly explain to the dispatcher the reason for the call.
  - — Perform such activities in off-peak hours, such as early morning or late evening.
- **Direct Inward Dialing (DID).** This equipment returns answer supervision signals to the Public Switched Telephone Network when:
  - — Answered by the called station
  - — Answered by the attendant
  - — Routed to a recorded announcement that can be administered by the customer premises equipment user
  - — Routed to a dial prompt

  This equipment returns answer supervision on all DID calls forwarded back to the Public Switched Telephone Network. Permissible exceptions are when:
  - — A call is unanswered
  - — A busy tone is received
  - — A reorder tone is received

  Allowing this equipment to be operated in such a manner as not to provide proper answer supervision signaling is in violation of Part 68 rules.

DOC Notification and Repair Information

NOTICE: The Canadian Department of Communications (DOC) label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational, and safety requirements. The DOC does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to connect it to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. In some cases, the company's inside wiring for single-line individual service may be extended by means of a certified connector assembly (telephone extension cord). The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or any equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines, and internal metallic water pipe system, if present, are connected. This precaution may be particularly important in rural areas.

CAUTION:

Users should not attempt to make such connections themselves, but should contact the appropriate electrical inspection authority or electrician, as appropriate.

To prevent overloading, the Load Number (LN) assigned to each terminal device denotes the percentage of the total load to be connected to a telephone loop used by the device. The termination on a loop may consist of any combination of devices subject only to the requirement that the total of the Load Numbers of all the devices does not exceed 100.

DOC Certification No.: 230 4095A
CSA Certification No.: LR 56260
Load No.: 6

Renseignements sur la notification du ministère des Communications du Canada et la réparation

AVIS: L'étiquette du ministère des Communications du Canada identifie le matériel homologué. Cette étiquette certifie que le matériel est conforme à certaines normes de protection, d'exploitation et de sécurité des réseaux de télécommunications. Le ministère n'assure toutefois pas que le matériel fonctionnera à la satisfaction de l'utilisateur.

Avant d'installer ce matériel, l'utilisateur doit s'assurer qu'il est permis de le raccorder aux installations de l'entreprise locale de télécommunication. Le matériel doit également être installé en suivant une méthode acceptée de raccordement. Dans certains cas, les fils intérieurs de l'enterprise utilisés pour un service individuel à ligne unique peuvent être prolongés au moyen d'un dispositif homologué de raccordement (cordon prolongateur téléphonique interne). L'abonné ne doit pas oublier qu'il est possible que la conformité aux conditions énoncées ci-dessus n'empêchent pas la dégradation du service dans certaines situations. Actuellement, les entreprises de télécommunication ne permettent pas que l'on raccorde leur matériel aux jacks d'abonné, sauf dans les cas précis prévus pas les tarifs particuliers de ces entreprises.

Les réparations de matériel homologué doivent être effectuées par un centre d'entretien canadien autorisé désigné par le fournisseur. La compagnie de télécommunications peut demander à l'utilisateur de débrancher un appareil à la suite de réparations ou de modifications effectuées par l'utilisateur ou à cause de mauvais fonctionnement.

Pour sa propre protection, l'utilisateur doit s'assurer que tous les fils de mise à la terre de la source d'énergie électrique, des lignes téléphoniques et des canalisations d'eau métalliques, s'il y en a, sont raccordés ensemble. Cette précaution est particulièrement importante dans les régions rurales.

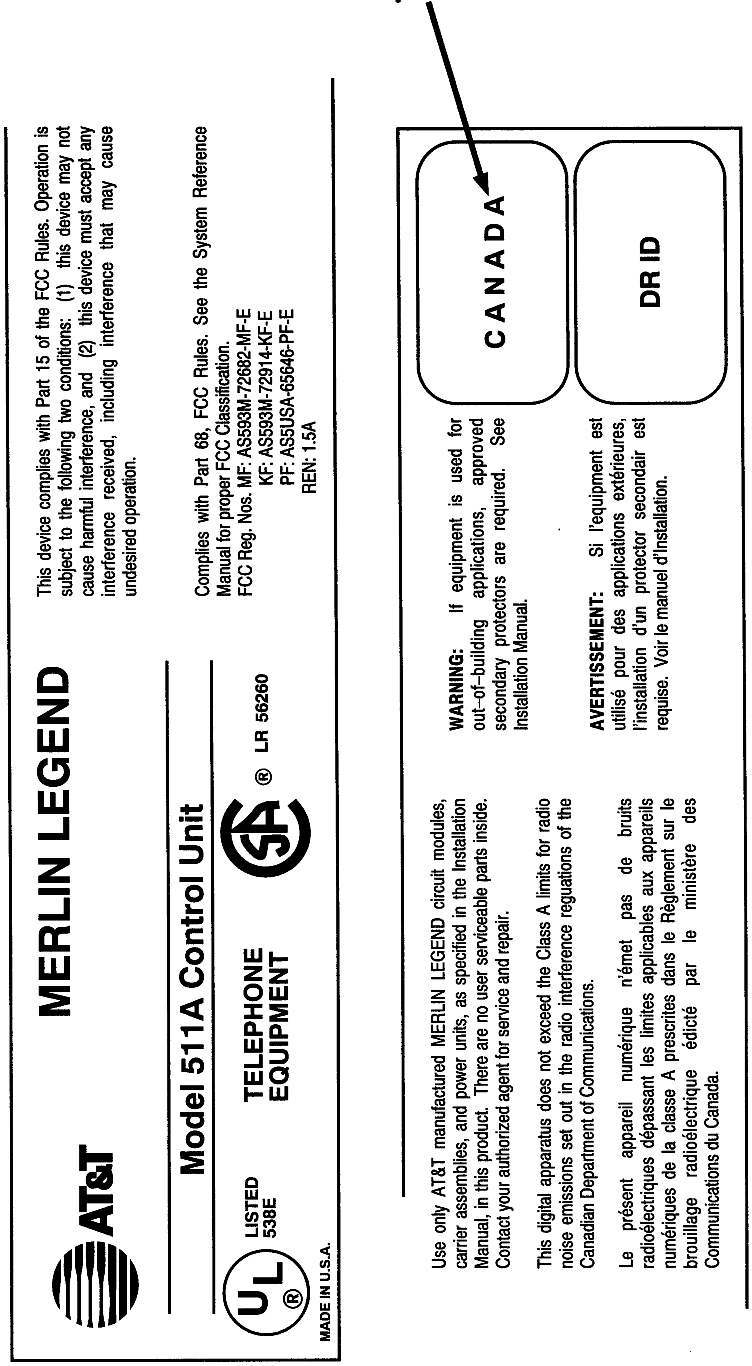
AVERTISSEMENT: L'utilisateur ne doit pas tenter de faire ces raccordements lui-même; il doit avoir recours à un service d'inspection des installations électriques, ou à un electricien, selon le cas.

L'indice de charge (IC) assigné à chaque dispositif terminal indique, pour éviter toute surcharge, le pourcentage de la charge totale qui peut être raccordée à un circuit téléphonique bouclé utilisé par ce dispositif. La terminaison du circuit bouclé peut être constituée de n'importe quelle combinaison de dispositifs, pourvu que la somme des indices de charge de l'ensemble des dispositits ne dépasse pas 100.

No d'homologation: 230 4095A
No de certification: CSA LR 56260
L'indice de charge: 6

MERLIN LEGEND D.O.C. Location Label Placement

Ministère des Communications du Canada emplacement de l'étiquette



Security of Your System: Preventing Toll Fraud

As a customer of a new telephone system, you should be aware that there is an increasing problem of telephone toll fraud. Telephone toll fraud can occur in many forms, despite the numerous efforts of telephone companies and telephone equipment manufacturers to control it. Some individuals use electronic devices to prevent or falsify records of these calls. Others charge calls to someone else's number by illegally using lost or stolen calling cards, billing innocent parties, clipping on to someone else's line, and breaking into someone else's telephone equipment physically or electronically. In certain instances, unauthorized individuals make connections to the telephone network through the use of remote access features.

The Remote Access feature of your system, if you choose to use it, permits off-premises callers to access the system from a remote telephone by using a telephone number with or without a barrier code. The system returns an acknowledgment signaling the user to key in his or her barrier code, which is selected and administered by the system manager. After the barrier code is accepted, the system returns dial tone to the user. If you do not program specific outward calling restrictions, the user will be able to place any call normally dialed from a telephone associated with the system. Such an off premises network call is originated at, and will be billed from, the system location.

The Remote Access feature, as designed, helps the customer, through proper administration, to minimize the ability of unauthorized persons to gain access to the network. Most commonly, phone numbers and codes are compromised when overheard in a public location, through theft of a wallet or purse containing access information, or through carelessness (for example, writing codes on a piece of paper and improperly discarding it). Additionally, hackers may use a computer to dial an access code and then publish the information to other hackers. Enormous charges can be run up quickly. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes, and distribute access codes only to individuals who have been fully advised of the sensitive nature of the access information.

Common carriers are required by law to collect their tariffed charges. While these charges are fraudulent charges made by persons with criminal intent, applicable tariffs state that the customer of record is responsible for payment of all long-distance or other network charges. AT&T cannot be responsible for such charges and will not make any allowance or give any credit for charges that result from unauthorized access.

To minimize the risk of unauthorized access to your communications system:

- Use a nonpublished Remote Access number.

- Assign access codes randomly to users on a need-to-have basis, keeping a log of *all* authorized users and assigning one code to one person.
- Use random-sequence access codes, which are less likely to be easily broken.
- Deactivate all unassigned codes promptly.
- Ensure that Remote Access users are aware of their responsibility to keep the telephone number and any access codes secure.
- When possible, restrict the off-network capability of off-premises callers, using calling restrictions, Facility Restriction Levels, and Disallowed List capabilities.
- When possible, block out-of-hours calling.
- Frequently monitor system call detail reports for quicker detection of any unauthorized or abnormal calling patterns.
- Limit Remote Call Forwarding to persons on a need-to-have basis.

Limited Warranty and Limitation of Liability

AT&T warrants to you, the customer, that your MERLIN LEGEND Communications System will be in good working order on the date AT&T or its authorized reseller delivers or installs the system, whichever is later ("Warranty Date"). If you notify AT&T or its authorized reseller within one year of the Warranty Date that your system is not in good working order, AT&T will without charge to you repair or replace, at its option, the system components that are not in good working order. Repair or replacement parts may be new or refurbished and will be provided on an exchange basis. If AT&T determines that your system cannot be repaired or replaced, AT&T will remove the system and, at your option, refund the purchase price of your system, or apply the purchase price towards the purchase of another AT&T system.

If you purchased your system directly from AT&T, AT&T will perform warranty repair in accordance with the terms and conditions of the specific type of AT&T maintenance coverage you selected. If you purchased your system from an AT&T-authorized reseller, contact your reseller for the details of the maintenance plan applicable to your system.

This AT&T limited warranty covers damage to the system caused by power surges, including power surges due to lightning.

The following will not be deemed to impair the good working order of the system, and AT&T will not be responsible under the limited warranty for damages resulting from:

- Failure to follow AT&T's installation, operation, or maintenance instructions
- Unauthorized system modification, movement, or alteration
- Unauthorized use of common carrier communication services accessed through the system
- Abuse, misuse, or negligent acts or omissions of the customer and persons under the customer's control
- Acts of third parties and acts of God

AT&T'S OBLIGATION TO REPAIR, REPLACE, OR REFUND AS SET FORTH ABOVE IS YOUR EXCLUSIVE REMEDY.

EXCEPT AS SPECIFICALLY SET FORTH ABOVE, AT&T, ITS AFFILIATES, SUPPLIERS, AND AUTHORIZED RESELLERS MAKE NO WARRANTIES, EXPRESS OR IMPLIED, AND SPECIFICALLY DISCLAIM ANY WARRANTIES OF MERCHANTABILITY OR FITNESS FOR A PARTICULAR PURPOSE.

Limitation of Liability

EXCEPT FOR PERSONAL INJURY, DIRECT DAMAGES TO TANGIBLE PERSONAL PROPERTY PROXIMATELY CAUSED BY AT&T, AND LIABILITY OTHERWISE EXPRESSLY ASSUMED IN A WRITTEN AGREEMENT SIGNED BY AT&T, THE LIABILITY OF AT&T, ITS AFFILIATES, SUPPLIERS, AND AUTHORIZED RESELLERS FOR ANY CLAIMS, LOSSES, DAMAGES, OR EXPENSES FROM ANY CAUSE WHATSOEVER (INCLUDING ACTS OR OMISSIONS OF THIRD PARTIES), REGARDLESS OF THE FORM OF ACTION, WHETHER IN CONTRACT, TORT OR OTHERWISE, SHALL NOT EXCEED AN AMOUNT EQUAL TO THE LESSER OF THE DIRECT DAMAGES PROVEN OR THE PURCHASE PRICE OF THE SYSTEM. IN NO EVENT SHALL AT&T OR ITS AFFILIATES, SUPPLIERS, OR AUTHORIZED RESELLERS BE LIABLE FOR INCIDENTAL, RELIANCE, CONSEQUENTLY, OR ANY OTHER INDIRECT LOSS OR DAMAGE (INCLUDING LOST PROFITS OR REVENUES) INCURRED IN CONNECTION WITH THE SYSTEM. THIS LIMITATION OF LIABILITY SHALL SURVIVE FAILURE OF THE EXCLUSIVE REMEDY SET FORTH IN THE LIMITED WARRANTY ABOVE.

Voice Mail Systems

Your voice mail system permits callers to leave verbal messages for system users or gain access to the backup position in an emergency as well as create and distribute voice messages among system users.

The voice mail system, through proper administration, can help you reduce the risk of unauthorized persons gaining access to the network. However, phone numbers and authorization codes can be compromised when overheard in a public location, are lost through theft of a wallet or purse containing access information, or through carelessness (writing codes on a piece of paper and improperly discarding them). Additionally, hackers may use a computer to dial an access code and then publish the information to other hackers. Substantial charges can accumulate quickly. It is your responsibility to take appropriate steps to implement the features properly, evaluate and administer the various restriction levels, protect and carefully distribute access codes.

Under applicable tariffs, you will be responsible for payment of toll charges. AT&T cannot be responsible for such charges and will not make any allowance or give any credit resulting from unauthorized access.

To reduce the risk of unauthorized access through your voice mail system, please observe the following procedures:

- Employees who have voice mailboxes should be required to use the passwords to protect their mailboxes.
  - Have them use random sequence passwords.
  - Impress upon them the importance of keeping their passwords a secret.
  - Encourage them to change their passwords regularly.
- The administrator should remove any unneeded voice mailboxes from the system immediately.
- AUDIX Voice Power has the ability to limit transfers to subscribers only. You are strongly urged to limit transfers in this manner.
- Use the Hybrid/PBX or Key system administration capability to do the following:
  - Block direct access to outgoing lines and force the use of account codes/authorization codes.
  - Disallow trunk-to-trunk transfer unless required.
  - Assign toll restriction levels to all AUDIX Voice Power ports or other voice mail ports.
  - If you do not need to use the Outcalling feature, completely restrict the outward calling capability of the AUDIX Voice Power ports. Use voice mail application features to do this.
- Monitor SMDR reports or Call Accounting System reports for outgoing calls that might be originated by AUDIX Voice Power ports or other voice mail ports.

Remote Administration and Maintenance

The Remote Administration and Maintenance feature of your telecommunications system, if you choose to use it, permits users to change the system features and capabilities from a remote location.

The Remote Administration and Maintenance feature, through proper administration, can help you reduce the risk of unauthorized persons gaining access to the network. However, telephone numbers and access codes can be compromised when overheard in a public location, are lost through theft of a wallet or purse containing access information, or through carelessness (for example, writing codes on a piece of paper and improperly discarding them). Additionally, hackers may use a computer to dial an access code and then publish the information to other hackers. Substantial charges can accumulate quickly. It is your responsibility to take appropriate steps to implement the features properly, evaluate and administer the various restriction levels, and protect and carefully distribute access codes.

Under applicable tariffs, you will be responsible for payment of toll charges. AT&T cannot be responsible for such charges and will not make any allowance or give any credit resulting from unauthorized access.

To reduce the risk of unauthorized access through Remote Administration and Maintenance, please observe the following procedures:

- The System Administration and Maintenance capability of a Hybrid/PBX or Key system is protected by a password.
  - Change the default password immediately.
  - Continue to change the password regularly.
  - Only give the password to people who need it and impress upon them the need to keep it secret.
  - If anyone who knows the password leaves the company, change the password immediately.
- If you have a special telephone line connected to your Hybrid/PBX or Key system for Remote Administration and Maintenance, you should do one of the following:
  - Unplug the line when it is not being used.
  - Install a switch in the line to turn it off when it is not being used.
  - Keep the Remote Administration and Maintenance telephone number secret. Only give it to people who need to know it, and impress upon them the need to keep it a secret. Do not write the telephone number on the Hybrid/PBX or Key system, the connecting equipment, or anywhere else in the system room.
- If your Remote Administration and Maintenance feature requires that someone in your office transfer the caller to the Remote Administration and Maintenance extension, you should impress upon your employees the importance of only transferring authorized individuals to that extension.

Ordering Codes

B

Component	PEC	Comcode	App. Code
Control Unit			
MERLIN LEGEND Control Unit	6140-CU3		
Power Supply Module		106257199	391A2
R3 Processor		107040438	517A33
Translation Card		107245243	10A1
Backplane/Basic Housing and Carrier		107007114	
R3 Customer Reference Manuals*		107259921	403G
MERLIN LEGEND Upgrade—R1/R2 to R3	6141-103A		
R3 Processor		107040438	517A33
Translation Card		107245243	10A1
R3 Customer Reference Manuals*		107259921	
SPM—UNIX (Attribute: QTY 11)		107259913	
SPM—DOS (Attribute: QTY 11)		107259905	
No Material (Attribute: QTY 99)		011111111	
MERLIN LEGEND Upgrade—M II to R3	6141-U3LA		
R3 Processor		107040438	517A33
Translation Card		107245243	10A1
R3 Customer Reference Manuals*		107259921	
Kit of Parts (Cover Labels and Ferrite Cores)		107005027	D182764

* The *R3 Customer Reference Manuals* package contains *Feature Reference*, *System Programming*, and *System Manager's Guide*.

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
MERLIN LEGEND R3 Translation Card	61501	107245243	10A1
Expansion Unit	61490		
Expansion Wall Mount with Top/Front Cover		107007122	403H
Power Supply Module		106257199	391A2
Top and Front Cover (Choose One)		106905953	18A
Cov99 (No Covers)			
Cov01* (One Top/One Front Cover)			
Cov02 (Two Top/Two Front Covers)			
Empty Module (Choose One)		107005720	19A
MOD90 (No Module)			
MOD01* (One Module)			
Kit of Parts (Cover Labels and Ferrite Cores)		107005027	D182764
Plastic Backboard Hardware			
Template		847009206	
Backboard (31.5" x 27")		847007523	
Shipping Container		847087376	
Shipping Tray		847087392	
Shipping Insert (pair)		847087384	
Network X-Conn: RJ-21X		403613003	
Station X-Conn: BR2580-66 Block		405464777	

* Default

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
Trunk and Station Modules			
008 (ATL)	61485	105351092	517B3
008 MLX	61486	105628010	517A21
008 OPT	61489	106995269	517D28
012(T/R)	61487	106767379	517F13
012 (T/R) + Ring Generator	61494	106933773	517G13
100D(DS1)	61491	105461560	517B15
400EM (tie trunk)	61492	105311401	517A14
400 GS/LS/TTR	61483	107044869	517C18
408 GS/LS	61481	107044877	517C26
408 GS/LS-MLX	61493	107044851	517B29
800 DID	61488	106995251	517D20
800 GS/LS	61484	105628069	517B19
800 GS/LS ID	61502	106975584	517A31
Vintage Trunk and Station Modules			
400 (with TTRs)	61379	105408892	517B12
408 LS	61482	105512495	517C1
800 LS	61384	105351100	517B4
Telephones			
MLX Telephones			
MLX-10			
English (black)	3156-02B	106902497	7712D01C-003
English (white)	3156-02W	106902505	7712D01C-264
French (black)	3156-F2I	106633886	7712D01A(29)-003
French (white)	3156-F2I	106633894	7712D01A(29)-264
Spanish (black)	3156-S2I	106613508	7712D01A(22)-003
Spanish (white)	3156-S2I	106613516	7712D01A(22)-264
Heat/Humidity (black)		107108722	7712D01D-003
Heat/Humidity (white)		107108748	7712D01D-264

Ordering Codes, *continued*

Component	PEC	Comcode	App. Code
MLX-10D			
English (black)	3156-03B	106902513	7712D02C-003
English (white)	3156-03W	106902521	7712D02C-264
French (black)	3156-F3I	106633928	7712D02A(29)-003
French (white)	3156-F3I	106633936	7712D02A(29)-264
Spanish (black)	3156-S3I	106613524	7712D02A(22)-003
Spanish (white)	3156-S3I	106613532	7712D02A(22)-264
Heat/Humidity (black)		107108979	713D01D(22)-003
Heat/Humidity (white)		107108987	713D010(22)-264
MLX-10DP			
English (black)	3156-06B	106943269	7712D04A-003
English (white)	3156-06W	106943277	7712D04A-264
Heat/Humidity (black)		107108946	712D04D-003
Heat/Humidity (white)		107108953	7712D04D-264
MLX-20L			
English (black)	3156-05B	106902539	7713D01C-003
English (while)	3156-05W	106902547	7713D01C-264
French (black)	3156-F5I	106634421	7713D01A(29)-003
French (white)	3156-F5I	106634439	7713D01A(29)-264
Spanish (black)	3156-S5I	106613557	7713D01A(22)-003
Spanish (white)	3156-S5I	106613573	7713D01A(22)-264
Heat/Humidity (black)		107108979	713D01D-003
Heat/Humidity (white)		107108987	713D01D-264
MLX-28D			
English (black)	3156-04B	106841166	7713D02C-003
English (white)	3156-04W	106902554	7713D02C-264
French (black)	3156-F4I	106634470	7713D02A(29)-003
French (white)	3156-F4I	106634488	7713D02A(29)-264
Spanish (black)	3156-S4I	106613599	7713D02A(22)-003
Spanish (white)	3156-S4I	106613607	7713D02A(22)-264
Heat/Humidity (black)		107115800	713D02D-003
Heat/Humidity (white)		107115818	713D02D-264

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
MLX Secure Telephones			
MLX-10DS			
English (black)	3156-03S	107019804	7712D02A-003
MLX-28DS			
English (black)	3156-04S	107019820	7713D01A-003
MLX-20LS			
English (black)	3156-05S	107019846	7713D02A-003
800 LS/E&M Card	61395	406981209	93030.2 2/4 WIRE PRN
Fiber Interface Card with Ring Generator	61393	406981217	93030.2 FIB INT PRN
Chassis with Power Supply Blank Cover	6139-FS	406981225	93030.8C MINI
800 LS Card	61394	406981241	903030.3 2 WIRE PRN
Analog Multiline Telephones (black)			
MLC-5	3168-MLC	105515332	7312HO1C-003
BIS-10	3165-10B	105161061	7313HO1C-003
BIS-22	3166-22B	105188809	7314HO1C-003
BIS-22D	3166-DSB	105630420	7315HO1D-003
BIS-34D	3167-DSB	105630529	7317HO1D-003
PFC paper	31690	106673361	
5-Button	3160-111	105217426	Z7302H01D-003
10-Button	3161-172	105217509	Z7303H01D-003
10-Button HFAI	3161-161	105371942	Z7309H01C-003
34-Button	3162-412	103842050	Z7305H01B-003
34-Button Deluxe	3162-417	105217715	Z7305H02D-003
34-Button BIS	3162-BIS	103981965	Z7305H03D-003
34-Button BIS/DIS	3162-DIS	103981981	Z7305H04C-003
MERLIN PFC (ATL)	3169-PF2	106681562	
Single-Line Telephones			
8110 Analog Voice	3139-001		
Black		106745715	8110A01C-003 811
White		106745730	8110A01C-264 811

Ordering Codes, *continued*

Component	PEC	Comcode	App. Code
Single Line Telephones			
8102 Analog Voice	3192-001		
Black		106745698	8102A01C-003 810
White		106745706	8102A01C-264 810
8101 Analog Voice	3192-101		
Black		106272289	8101A01-003
White		106272297	8101A01-264
2500 YMGL	3101-KFD		
Black		107005043	2500YMGL-003
Misty Cream		107005050	2500YMGL-215
2500 YMGK (message waiting, recall,touch-tone, desk)	3178-NHL		
Black		105480578	2500YMGK-003
Misty cream		105480560	2500YMGK-215
2500 MMGK (recall, touch-tone, desk)	3101-ETR		
Black		105414130	2500MMGK-003
Misty cream		105414122	2500MMGK-215
2500 MMGJ (touch-tone, desk)	3101-EBD		
Black		105414155	2500MMGJ-003
Misty cream		105414148	2500MMGJ-215
2554 MMGJ (touch-tone, wall)	3101-EBW		
Black		105480081	2554MMGJ-003
Misty cream		105480032	2554MMGJ-215
500 MM (rotary, desk)	3100-ORD		
Black		103870234	500MM-03
Ivory		103870226	500MM-50
Beige		103870267	500MM-60

Ordering Codes, *continued*

Component	PEC	Comcode	App. Code
554 BMPA (rotary, wall)	3100-ORW		
Black		103823498	554BMPA-3
Ivory		103823506	554BMPA-50
Beige		103823555	554BMPA-60
Cordless/Wireless Telephones			
Model 5405	3103-405	106440472	
Model 5455	3103-455	106440464	
MDC 9000			
TransTalk Cordless Tel Set (white)	3203-03W	106739089	7311H11A-264
TransTalk Cordless Tel Set (black)	3203-03B	106738073	7311H11A-003
MDW 9000 (standalone product shipped w/power pack)			
TransTalk Wireless Tel Set (white)	3204-01W	107017030	7815H01A-264
TransTalk Wireless Tel Set (black)	3204-01B	107017022	7815H01A-003
MDW 9000 (ship with wireless carrier assembly)			
TransTalk Wireless Tel Set (white)	3204-W1W	107077463	7815H02A-264
TransTalk Wireless Tel Set (black)	3204-W1B	107077455	7815H02A-003
TransTalk Wireless Carrier Assembly	3204-CR1	107073330	117A1
Headset	3122-041	407156892	
Battery Pack			
White	32034	106760812	
Black	32036	106760804	
Charging Cradle			
White	32040	107076762	
Black	32040	107076754	

Ordering Codes, *continued*

Component	PEC	Comcode	App. Code
Special Purpose Telephones			
Explosive Atmosphere Telephones			
520B			
Rotary, Desk	3129-ERW	103873048	520B-3
2520B			
Touch-tone, Wall	3129-ETW	103873030	2520B-3
Special-Purpose Outdoor Telephones			
Rotary Outdoor WL	3129-WRW	105727444	526
Touch-tone Outdoor WL	3129-WTW	105725386	2526
Non-Dial Outdoor WL	3129-WNW	105725402	526
Auto-Dial Outdoor WL	3129-WAW	105725394	526 AMACADL
Consoles			
MERLIN II System Display			
Console	61392	105229744	7318H01A-003
DSS			
English (black)	3156-DCB	105685481	604B1-003
English (white)	3156-DCW	105685499	604B1-264
Spanish (black)	3156-SDI	106613672	604A1(22)-003
Spanish (white)	3156-SDI	106613680	604A1(22)-264
Heat/Humidity (white)		106902463	604B1-003
Heat/Humidity (black)		106902489	604B1-264
Applications			
SPM Version 2.0—DOS	61495	106906092	
SPM Version 2.0—UNIX System	61496		
SPM Version 3.15—DOS			
SPM Version 3.15—UNIX System			
Call Accounting System (CAS)			
CAS Plus V3 Bundle w/80-col. Parallel Printer	1201-NP1		
CAS Plus V3 Bundle w/132-col. Parallel Printer	1201-WP1		

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
CAS Plus V3 Software	1201-DR1†	406362244	
Rate Table*	12010	406158444	3302KA51
CAS Plus upgrade	12009	406477679	3801E
CAS V3 Hacker Tracker			
(MS-DOS)	1202-10D	406774513	3399EA
Hacker/Tracker	12014	406806166	PCCB6201
Fax/Modem	1201-U14	406898262	UN/CAS 250
UNIX CAS (All Sets)			
UNIX CAS Rate Tables	12003		
UNIX CAS Upgrade (250-500)	1201-U15	406898254	UN/CAS UPGR
UNIX Hacker Tracker	1201-U13	406898270	SFTW-ISIII
Call Accounting Terminal (CAT)			
CAT BASIC/B (LEGEND)	3600-010		
CAT + LEGEND/H	3600-024		
CAT + LEGEND/B	3600-023		
CAT Basic Rate Table*			
(Update Chip)	36014	406669739	
CAT/B Rate Table* (update)	36023	406478792	
CAT/H Rate Table* (update)	36024	406478784	
Call Management System (CMS)	1207-100		
5 ¼" floppy disk		106496540	
3 ½" floppy disk		106496532	
Board	8301-100	106198815	
MII/ML CMS Alerter	83010		
Block Connector		105164859	104A-246
Power Supply		405331711	KS22911L2 120VAC

† Includes Hacker Tracker
* Consult AT&T or an authorized dealer for other area-specific information.
* Consult an AT&T representative for other area-specific information.

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
Applications (continued)			
CONVERSANT INTRO Application			
Casual Dev. & Data	4201-102		
MERLIN LEGEND Integrated			
Solution III Controllers 100 MB			
MC-II + Processor	4200-503		
4 x 100 MB MC-II + Processor		406506329	
4MB Memory Upgrade		106219553	
Color Monitor		406504571	
Keyboard		406504563	
9 to 25 Pin Adapter		406139394	
Cartridge Tapes (qty. 2)		106220866	
MERLIN LEGEND v1.2 IS III			
AVP8hr	6146-210/A	107013062	
DOC-MERLIN LEGEND v1.2			
IS II		07135485	
MERLIN LEGEND v1.2 AVP 2.1		107013013	
125 MB Cartridge Tape		106220866	
IVP4 Bd		106248651	
4x100 MB MC2+ CPU		406506329	
2 MB MEM Upgrade		406504555	
Color Monitor		406504571	
Keyboard		406504563	
9 to 25 Pin Adapter		406708503	
Surge Protector		106282551	
A/B Switch Kit		106368814	
RMD		407002278	
Cartridge Tape Utilities		106632938	

Ordering Codes, *continued*

Component	PEC	Comcode	App. Code
Applications (continued)			
MERLIN LEGEND v1.2 IS III AVP8hrCAS	6146-211A	107013062	
DOC-MERLIN LEGEND v1.2 IS II		1070135485	
MERLIN LEGEND v1.2 AVP 2.1		107013013	
125 MB Cartridge Tape		106220866	
IVP4 Bd		106248651	
4x100 MB MC2+ CPU		406506329	
2 MB MEM Upgrade		406504555	
Color Monitor		406504571	
Keyboard		406504563	
9 to 25 Pin Adapter		406708503	
Surge Protector		106282551	
IS CAS 250		406898262	
A/B Switch Kit		106368814	
RMD		407002278	
Cartridge Tape Utilities		106632938	
MERLIN LEGEND v1.2 IS III AVP12hr	6146-220A	107013062	
DOC-MERLIN LEGEND v1.2 IS II		107135485	
MERLIN LEGEND v1.2 AVP 2.1		107013013	
125 MB Cartridge Tape		106220866	
IVP4 Bd		106248651	
4x200 MB MC2+ CPU		406506337	
2 MB MEM Upgrade		406504555	
Color Monitor		406504571	
Keyboard		406504563	
9 to 25 Pin Adapter		406708503	
Surge Protector		106282551	
A/B Switch Kit		106368814	
RMD		407002278	
Cartridge Tape Utilities		106632938	

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
Applications (continued)			
MERLIN LEGEND v1.2 IS III AVP12hrCAS	6146-221A	107013062	
DOC-MERLIN LEGEND v1.2 IS II		107135485	
MERLIN LEGEND v1.2 AVP 2.1		107013013	
125 MB Cartridge Tape		106220866	
IVP4 Bd		106248651	
4x200 MB MC2+ CPU		406506337	
2 MB MEM Upgrade		406504555	
Color Monitor		406504571	
Keyboard		406504563	
9 to 25 Pin Adapter		406708503	
Surge Protector		106282551	
IS CAS 250		406898262	
A/B Switch Kit		106368814	
RMD		407002278	
Cartridge Tape Utilities		106632938	
MERLIN LEGEND v1.2 IS III AVP12hr	6146-320A	107013062	
DOC-MERLIN LEGEND v1.2 IS II		107135485	
MERLIN LEGEND v1.2 AVP 2.1		107013013	
250 MB Cartridge Tape		406506329	
IVP4 Bd		106248651	
8x200 MB MC2+ CPU		406700930	
2 MB MEM Upgrade		406504555	
Color Monitor		406504571	
Keyboard		406504563	
9 to 25 Pin Adapter		406708503	
Surge Protector		106282551	
A/B Switch Kit		106368814	
RMD-HS		407002278	
Cartridge Tape Utilities		106632938	

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
Applications (continued)			
MERLIN LEGEND v1.2 IS III AVP12hrCAS	6146-321A	107013062	
DOC-MERLIN LEGEND v1.2 IS II		107135485	
MERLIN LEGEND v1.2 AVP 2.1		107013013	
250 MB Cartridge Tape		406760009	
IVP4 Bd		106220866	
8x200 MB MC2+ CPU		406700930	
Color Monitor		406504555	
Keyboard		406504571	
9 to 25 Pin Adapter		406504563	
MERLIN LEGEND v1.2 IS III FAX		07013039	
IFP4 Bd		406794966	
Surge Protector		106282551	
A/B Switch Kit		106368814	
RMD-HS		406701052	
Cartridge Tape Utilities		106632938	
MERLIN LEGEND v1.2 IS III AVP12hrFA	6146-322A	107013062	
DOC-MERLIN LEGEND v1.2 IS II		107135485	
MERLIN LEGEND v1.2 AVP 2.1		107013013	
250 MB Cartridge Tape		406760009	
IVP4 Bd		106248651	
8x200 MB MC3 CPU		406700930	
Color Monitor		406504571	
Keyboard		406504563	
9 to 25 Pin Adapter		406708503	
Surge Protector		106282551	
MERLIN LEGEND v1.2 IS III FAX		107013039	
IFP4 Bd		406794966	
A/B Switch Kit		106368814	
RMD-HS		406701052	
Cartridge Tape Utilities		106632938	

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
Applications (continued)			
MERLIN LEGEND v1.2 IS III AVP12hrFACAS	6146-323A	107013062	
DOC-MERLIN LEGEND v1.2 IS II		107135485	
MERLIN LEGEND v1.2 AVP 2.1		107013013	
250 MB Cartridge Tape		406760009	
IVP4 Bd		106248651	
8x200 MB MC3 CPU		406700930	
Color Monitor		406504571	
Keyboard		406504563	
9 to 25 Pin Adapter		406708503	
Surge Protector		106282551	
MERLIN LEGEND v1.2 IS III FAX		107013039	
IFP4 Bd		406794966	
IS CAS 250		406898262	
A/B Switch Kit		106368814	
RMD-HS		406701052	
Cartridge Tape Utilities		106632938	
MERLIN LEGEND v1.2 IS III AVP36hr	6146-350A	107013062	
DOC-MERLIN LEGEND v1.2 IS II		107135485	
MERLIN LEGEND v1.2 AVP 2.1		107013013	
250 MB Cartridge Tape		406760009	
IVP4 Bd		106248651	
8x500 MB MC3 CPU		406700914	
Color Monitor		406504571	
Keyboard		406504563	
9 to 25 Pin Adapter		406708503	
Surge Protector		106282551	
A/B Switch Kit		106368814	
RMD-HS		407002278	
Cartridge Tape Utilities		106632938	

Ordering Codes, *continued*

Component	PEC	Comcode	App. Code
Applications (continued)			
MERLIN LEGEND v1.2 IS III			
AVP36hrCAS	6146-351A	107013062	
DOC-MERLIN LEGEND v1.2			
IS II		107135485	
MERLIN LEGEND v1.2 AVP 2.1		107013013	
250 MB Cartridge Tape		406760009	
IVP4 Bd		406700914	
8x500 MB MC3 CPU		406506329	
Color Monitor		406504571	
Keyboard		406504563	
9 to 25 Pin Adapter		406708503	
Surge Protector		106282551	
IS CAS 250		406898262	
A/B Switch Kit		106368814	
RMD-HS		406701052	
Cartridge Tape Utilities		106632938	
MERLIN LEGEND v1.2 IS III			
AVP36hrFA	6146-352A	107013062	
DOC-MERLIN LEGEND v1.2			
IS II		107135485	
MERLIN LEGEND v1.2 AVP 2.1		107013013	
250 MB Cartridge Tape		406760009	
IVP4 Bd		406700914	
8x500 MB MC3 CPU		406506329	
Color Monitor		406504571	
Keyboard		406504563	
9 to 25 Pin Adapter		406708503	
Surge Protector		106282551	
MERLIN LEGEND		107013039	
IFP4 Bd		406794966	
A/B Switch Kit		106368814	
RMD-HS		406701052	
Cartridge Tape Utilities		106632938	

Ordering Codes, *continued*

Component	PEC	Comcode	App. Code
Applications (continued)			
MERLIN LEGEND v1.2 IS III AVP36hrFACAS	6146-353A	107013062	
DOC-MERLIN LEGEND v1.2 IS II		107135485	
MERLIN LEGEND v1.2 AVP 2.1		107013013	
250 MB Cartridge Tape		406760009	
IVP4 Bd		406700914	
8x500 MB MC3 CPU		406506329	
Color Monitor		406504571	
Keyboard		406504563	
9 to 25 Pin Adapter		406708503	
Surge Protector		106282551	
MERLIN LEGEND v1.2 IS3 FAX		107013039	
IFP4 Bd		406794966	
IS CAS 250		406898262	
A/B Switch Kit		106368814	
RMD-HS		406701052	
Cartridge Tape Utilities		106632938	
MERLIN LEGEND v1.2 IS II to IS III Upgrade	6146-IS3U		
MERLIN LEGEND v.1.2 IS III		107013062	
DOC MERLIN LEGEND v1.2 IS III		107135485	
MERLIN LEGEND v1.2 AVP 2.1		107013013	
250 MB Cartridge Tape		406760009	
PassageWay Direct Connect Solution	8302-500		
Fast Call Software	8330-101A	407160019	
Commence 2.1	8330-201A	407160027	
Commence Startup	8330-202A	407160043	
OnTime 1.54	8330-301A	407127349	

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
Applications (continued)			
MERLIN Attendant	6125-ATT		
Hardware		406406090	
Documentation		106431265	
MERLIN MAIL Voice			
Messaging System for the			
MERLIN LEGEND			
Communications			
System	6107-005		
Two-port		406824532	
MERLIN MAIL unit		406466193	
Remote maintenance	6107-006		
device		406824532	
Four-port		406466193	
MERLIN MAIL unit			
Remote maintenance	6107-007	406824524	
device			
Two-port line card (R2)			
(upgrade from two to four)			

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
Applications (continued)			
MERLIN Identifier (for MERLIN LEGEND R2.x)			
Controller Assembly with Display Keyboard Administration	6128-KBD	406891556	
Controller Assembly with PC Administration	6128-PCA	406891564	
Display Assembly with Wall-Mounting	6128-DIS	406891572	
Call Alert Software	6128-SFW	406891721	
Bracket Assembly, ATL Telephone Mounting	6128-BKT	406891937	
Fixture, Display Wallmount		406891929	
PC Administration Adapter Kit		406960930	
Printer Adapter Kit		406960948	
Printer Port to PC Adapter Kit		406960955	
Installation and System Administration Manual		406891713	
Quick Reference Card for MERLIN Identifier Users		406891705	
Display Unit			
Keyboard 101		406891663	
Controller with Mounting		406891655	
Panel		406891648	
Cable, Serial RS-232, Controller to PC		406891903	
System Adjuncts and Adapters			
Auxiliary Power Unit 9024	61416	406467142	9024
Channel service units (CSUs)			
T1 CSU (3150 CSU)	21581	107087546	
T1 ESF CSU Standalone		107063828	21581-00001
115VAC in line Transformer		406942284	
Converter Cable		107083711	3100-F1-560
RJ48M to RJ48M Unshielded Twisted Pair Cable (T1)		406941559	3110-F1-500

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
Applications (continued)			
System Adjuncts and Adapters (continued)			
Channel Service Units (continued)			
To order parts separetely use:			
T1 ESF CSU Standalone		107063828	21581-00001
115VAC in line Transformer	21573	406942284	
Converter Cable	21583	107063711	3100-F1-560
RJ48M to RJ48M Unshielded Twisted Pair Cable (T1)	21587	406941559	3110-F1-500
Optional Equipment:			
Unshielded TW Pair Cable (T1) Canada	21582	107063703	3100-F1-510
Straight-Thru Cable PC Serial Port	21584	406941542	3100-F1-550
Straight-Thru Cable Terminal/Printer	21585	406941534	3100-F1-540
Modular DC Voltage Adapter	21586	406941492	3100-F1-250
Wall Mount Kit	21590	406941674	3100-F1-400
Cables for Mounting			
15' M25M Cable	31500	106473317	ASSY-3230-903F
25' M25M Cable	31501	106473325	ASSY-3230-904F
50' M25M Cable	31502	106473333	ASSY-3230-905F
2' D4BU-29 Cord	31503	106472905	ASSY-4400-F1-530
25' D4BU-29 Cord	31504	106472921	ASSY-4400-F1-533
Dial Back Modem FLD	31505	106842271	ASSY-3400-F2-201
Dial Back Modem FAC	31506	106842289	ASSY-3400-G2-201
Dial Back Modem NFLD	31507	106842305	ASSY-4000-F2-201
Dial Back Modem NFAC	31508	106842297	ASSY-4000-G2-201
Prism MUX Field	31509	106842313	ASSY-3400-F2-200

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
System Adjuncts and Adapters (continued)			
Optional Equipment			
Peripheral Interface	62515	105179303	KIT PRTS-D181558
Async. Data Unit, Receptacle	2169-004	103964185	Z3A4
Async, Data Unit, Plug	2169-001	103963963	Z3A1
Aux Power (2 required)			
Transformer	21691	102600517	
Adapter (248B)		102802113	
Cord		102937620	
Adapter (400B)		103848859	
Electrostatic discharge/ (ESD) suppression kits			
D-181574		105179329	D181574
D-181589		105201891	D181589
D-181590		105201909	D181590
D-181591		105201917	D181591
D-181593		105201933	D181593
EMI filter		103965208	Z200A
In-Range Out-of-Building -343B (IROB) unit —analog multiline*	32918	406721738	343B
IROB unit—MLX*	32919	106417447	505A ASSY 0A WD
Fuse block 505A for IROB (8 fuse blocks per box)		406610337	
7500B Data Module	2164-BDM	105657654	Z750B-L1
Stand-alone power supply	21625	405509852	WP90110L7
Multiple mounting	21626	105441166	Z77A
7500A upgrade kit	21627	105688501	D182208
Off-Premises Range Unit	2302-OPT	106460405	122A-215

* Any multiline off-premises telephone must have an appropriate IROB protector at the control unit location and at the off-premises location.

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
System Adjuncts and Adapters (continued)			
Digital Magic on Hold			
Basic Prerecorded Package	3128-020		
Digital Deck		406659326	DMOH1DIGITAL
Cassette		406876672	DMOH-02 GENERIC
Personalized Recording Package	3128-030		
Digital Deck		406659326	DMOH1 DIGITAL
Cassette		406876664	DMOH-01 PERSONALIZE
Custom Production Package	3128-040		
Digital Deck		406659326	DMOH1 DIGITAL
Cassette		406876680	DMOH-05 SIN F/CUST
Standalone Single Custom Production Package	31284	405135344	INDIV
Standalone Package of 3 Custom Productions	31283	406876649	DMOH-03
Standalone Package of 4 Custom Productions	31280	405126632	M4 FOUR
Duplicate of a Custom Production (for Second Location)	31289	405127945	D-IP/EM DUB IND MSTR
Re-License of Music	31288	405127879	D24 24 DUB
Digital Announcer Unit (one minute)	3119-001		
Announcer		406887828	
Recorder		406659342	RCDR-DMOH2
Cassette		406769455	CSTT-DMOH5

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
System Adjuncts and Adapters (continued)			
Digital Announcer Unit (three minute)	3119-003		
Announcer		406941948	
Recorder		406659342	
Cassette		406659359	
Modem 2224G	2224-CEO	105659965	2224C-L1 D/2
Music Coupler	61398	406143925	ASSY-K23395 L3
PagePac Plus			
PagePac Plus Controller	5323-100	406914598	UNIT-22051-000
PagePac Plus 16 Zone	5335-100	406914614	UNIT-22051-016
D20 PagePac Plus Amplicenter	5328-020	406915280	UNIT-22051-020
D100 PagePac Plus Amplicenter	5328-100	406915264	UNIT-22051-100
D300 PagePac Plus Amplicenter	5328-300	406915330	UNIT-22051-300
AT&T Door Phone Speaker	53240	406269860	
200 WATT Amplicenter	52120	403305493	52120 AT
50 WATT Amplicenter	52150	403305501	52150 AT 50W
Ring generator unit	61388	105213201	129B RING GEN
SMDR Printers			
AP Printer (80-column)	4200-570	406637314	ML182
AT&T 571 Parallel Printer		406516989	571-MCII 6FT
(132-column)	4200-571	406712067	ML321P
AP CAT Printer (serial)	4200-572	406516989	571-MCII 6FT

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
System Adjuncts and Adapters (continued)			
Uninterruptible Power Supply (UPS)		406716464	ML182-R2
500 VA (15 min)	2403-050	105610141	515005C111
Reserve (1 hr)	24035	105610174	0053150
Universal Paging Access Module (UPAM)	58500	405891698	KIT-UPAM
TAM-B		405899972	D181900
PRS-48		405742735	D181900
WMT-1A		405891680	D181900
Zonemate™ 9	53505		
Dialer unit		404057911	DIAL UNIT-9ZONE
Control unit		405024134	CNTL 22050-020
Zonemate 39	53506		
Dialer unit		404057929	39 ZONE SELECT
Control unit		405024134	CNTL-22050-020C
External Alerts			
Loud external ringer	31016A	103117016	RINGER-L1AMP-49
E1CM-type	31019A		
Gray		102872934	RINGER-E1CM-49
Ivory		102917952	RINGER-E1CM-50
E1CM ringer and parts	61211		D-181233
290A adapter		102992252	290A ADPTR
Ringer		102872934	E1CM-49
Mounting plate		102988466	1049A
Cord		103938494	CORD-D4CH-87-25
Supplemental Alerts			
Alert Bell	5580-002	406293720	TB591-1
Network Interface Alert Bell	61211	102872934	RINGER-E1CM-49
Alert Horn	5580-021	406207217	THET4-1
Alert Strobe	5580-041	403319197	AT-WHL LK
Alert Chime	5580-030	405136060	CHBT2-1

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
Telephone Adjuncts and Adapters			
General Purpose Adapter			
(GPA) (analog)	2301-GPA	103977997	Z1C
Multi-Function Module			
(digital)	3156-MFM	105746474	540A1
Supplemental Alert Adapter			
(SAA)	2301-SSA	105031199	ADPTR-856A
MLX-10 and MLX-10D			
protective cover		406648469	
MLX Telephone Power	31757		
48V Power Supply		405331711	KS22911L2
Modular Power Cord		102937620	D6AP-87
400B2 Adapter		104152558	400B2
Analog Multiline Telephone			
Power	62510	105105514	D181522
48V Power Supply		405331711	KS22911L2
Modular Power Cord		102937620	D6AP-87
Z400F Adapter		103942857	Z400F
Single-line telephones			
Program, Pause, and			
Auto Dial button			
conceal kit for 8100			Kit-D 182363
series telephones	31931	106248370	Analog
4A Speakerphone	3120-02W		4A
Power unit		102139938	PWR UNIT-85B1
Block connector		102434925	BLK CON-82B-49
Adapter for single-line			ADPTR-223C
telephone		102813888	
Adapter for multiline			
telephone		102949013	ADPTR-223D IP
Transmitter (black)		103971891	TRMR-680AF-03
Transmitter (ivory)		103971909	TRMR-680AF-50
Loudspeakers			
Black		103873873	LSPK-108AA-03
Ivory		103873881	LSPK-108AA-50
Green		103873899	LSPK-108AA-51
Beige		103873907	LSPK-108AA-60
White		103873964	LSPK-108AA-58

Ordering Codes. *continued*

Component	PEC	Comcode	App. Code
Supplemental Alerts			
Alert Bell	5580-002	406293720	TB591-1
Network Interface Alert Bell	61211	102872934	RINGER-E1CM-49
Alert Horn	5580-021	406207217	THET4-1
Alert Strobe	5580-041	403319197	AT-WHL LK
Alert Chime	5580-030	405136060	CHBT2-1
S201 Speakerphone	3152-007A	103786786	D8W-87 7FT
Black		106192651	MOD-S201AP-003
Misty cream		106192693	MOD-S201AP-215
CS201 Conference			
Speakerphone	3131-004A	103786786	D8W-87 7FT
Black		106270325	MOD-CS201A-003
Misty cream		106270333	MOD-CS201A-215
S202A Speakerphone	3152-008		
Black		105721088	TEL-S202A-003
Misty cream		105721096	TEL-S202A-215
S203A Speakerphone	3131-008		
Black		106058340	MOD-S203A-003
Misty cream		106508365	MOD-S203A-215
Message Waiting Indicator	31032	103966396	Z34A
Hands Free Unit (HFU)	3163-HFU	103814356	MOD-S102A
Headsets and Adapters			
StarSet Headset	3122-030	406445627	KS23822L3
Mirage Headset	3122-050	406445783	KS23822L4
Supra Monaural Headset	3122-040	406445791	
Supra Monaural Headset			
w/Noise Cancelling	3122-055	406445809	
Supra Binaural Headset	3122-045	406976076	
Supra Binaural Headset	3122-060	406445817	
w/Noise Cancelling			
Headset Adapter	3164-HFA	105752042	ADPTR-502C-003
500A Headset Adapter	3152-001	106690043	Adapter EL-500A-265
		405331711	Pwr Sup-KS2291 1L2
		102479904	Cord-D4BU-29 Std 7FT
		104152558	Adapter-40082
Modular Amplifier	122-020	406741900	KS23822L2
Plug Prong Amplifier	3122-010	406445601	KS23822L1

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
MLX Telephones—Miscellaneous Add-Ons/Replacement Parts			
Handsets and Cords			
Handset (black)		106050065	K2S1-003
Handset (white)		106053408	K2S1-264
Handset, amplified hearing	31052		
Black		105581896	K6S2-003
White		106248248	K6S2-264
Misty cream		105581904	K6S2-215
Noise Cancelling Handset	31056		
Black		406712463	KS23843L7
White		406712471	KS23843L8
Misty Cream*		406712489	KS23843L9
High-Noise Cancelling Handset	31057		
Black		406712497	KS23843L10
White		406712505	KS23843L11
Misty Cream*		406712513	KS23843L12
Amplified Speech Handset	31054		
Black		406712406	KS23843L1
White		406712414	KS23843L2
Misty Cream*		406712422	KS23843L3
Push-to-Talk Handset	31055		
Black		406712430	KS23843L4
White		406712448	KS23843L5
Misty Cream*		406712455	KS23843L6
Push-to-Listen Handset	31053		
Black		406382344	K8S2-003
White		406382369	K8S2-264
Misty Cream*		406382351	K8S2-215
Handset cord, 9' (2.74 m), black		105635429	H4DU-003 9 FT
Handset cord, 9' (2.74 m), white		105701809	H4DU-264 9'BULK
Handset cord, 12' (3.66 m), black		102401445	H4DU-3 12FT IP
Handset cord, 12' (3.66 m), white		102402609	H4DU-26412'IP
Handset cord, 25' (7.62 m), black		105523866	H4DU-3 25'
DSS line cord, 2' (61 cm)		106187545	CORD D8AC-87

* Default

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
MLX Telephones—Miscellaneous Add-Ons/Replacement Parts (continued)			
Designation (Button Assignment) Cards and Covers			
Desk Stands and User Trays			
Stand (large, black)		846320851	STAND-LARGE BL
Stand (large, white)		846320844	STAND-LARGE WH
Stand (small, black)		846320810	STAND-SMALL BL
Stand (small, white)		846320802	STAND-SMALL WH
User tray (black)		846320240	USER TRAY DWR B
User tray (white)		846320232	USER TRAY DWR W
Card*—MLX-10, MLX-10D, MLX-10DP, MLX-20L, MLX-28D		847193273	
Card set-DSS†		106448756	KIT-D182464
Card covers-DSS (black)†		106448731	KIT-D182462 PRT
Card covers-DSS (white)†		106448749	KIT-D182463 PRT
Card set-QCC‡		106561673	KIT-D182562 PRT
Card covers§—MLX-10, MLX-10D, MLX-20L		106448681	KIT-D182457 PRT
Card covers§-MLX-28D		106448699	KIT-D182458 PRT
Analog Multiline Telephones—Miscellaneous Add-Ons/Replacement Parts			
Desk Stands and Wall Mounts			
Adjustable desk stand, 10-button	32002	103746855	11A
Adjustable desk stand, 34-button	32003	103746863	11C
Fixed desk stand, 5- & 10-button	32004	103746848	10A
Desk stand/wall mount 14A,		103804290	14A-003
BIS-10		103964458	Z14B-003
Desk stand/wall mount 14B,		103979837	14C-003
BIS-22		103804290	14A
Desk stand/wall mount 14C,	32000	103995882	D-181230
BIS-34		103747846	201A
Fixed desk stand and wall mount, 5-button	32001	103995882	D-181230
Kit of parts	32006	103747853	203A

* 10 sheets per package

† Includes both top and bottom cards or covers

‡ 8 cards per kit (four sets)

§ 4 per package

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
Analog Multiline Telephones—Miscellaneous Add-Ons/Replacement Parts (continued)			
Wall mount, 10-button Kit of parts		103995882	D-181230
Wall mount, 34-button Kit of parts			
Faceplates			
BIS-10		105203186	KIT PRTS-D-181582
BIS-22		105336986	KIT PRTS-D-181786
BIS-22D		105690762	KIT PRTS-D-182210
BIS-34 and BIS-34D		105203194	KIT PRTS-D-181583
Button Label Sheets			
BIS-10		105336978	KIT PRTS-D-181785
BIS-22		105336960	KIT PRTS-D-181784
BIS-22D		105690770	KIT PRTS-D-182211
BIS-34 and BIS-34D		105336956	KIT PRTS-D-181783
Display console (FM1) (includes one faceplate)		105299754	KIT PRTS-D-181727
Display console (FM2 & R3) (includes one faceplate)		105486252	KIT PRTS-D-182041
Single-Line Telephones—Miscellaneous Add-Ons			
Ground-Start Button	31021	405792839	Key-LS23566L1
Wiring Kits			
Interconnect Wiring Kit			
110AB1-100JP12		104409396	
110A1 trough		104407960	
D-Rings		842139248	
D8W cords		103786802	
Parts list			

Ordering Codes, ***continued***

Component	PEC	Comcode	App. Code
Single-Line Telephones—Miscellaneous Add-Ons			
SYSTIMAX® MERLIN Wiring Kit	3103-MER	106393671	
110A1 trough (5)		104407960	
110AB1-100JP12 modular block (2)		104409960	
110AB1-100 FT punch down block (1)		103823845	
D-Rings (6)		842139248	
Patch cords 12 cords, 4-pair, 5' (1.5 m)		846619989	
D8W cords 24 cords, 14' (4.3 m)		103786802	
Template		846613933	
Instruction sheet		846613941	
Parts List		846623924	

Glossary

GL

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7500B data module — See *ISDN 7500B Data Module.*

7500B data station — A type of data station that includes an ISDN 7500B Data Module as its DCE. It may also include an MLX telephone for simultaneous voice and data (7500B data-only station). These data stations connect to MLX extension jack modules for digital transmission of data over a DS1 facility.

A

account code — Code used to associate incoming and outgoing calls with corresponding accounts, employees, projects, and clients.

ACCUNET — AT&T's switched digital service for 56-kbps, 64-kbps restricted, and 64-kbps clear circuit-switched data calls.

address — A coded representation of the destination of data or of the data's originating terminal, such as the dialed extension number assigned to the data terminal. Multiple terminals on one communication line must each have a unique address.

ADDS — (Automated Document Delivery System) Computer-based application that stores documents in a database and automatically faxes them on request.

adjunct Optional equipment used with the communications system, such as an alerting device or *modem* that connects to a multiline telephone or to an extension jack.

ALS (Automatic Line Selection) Programmed order in which the system makes outside lines available to a user.

AMI (alternate mark inversion) Line coding format in which a binary one is represented by a positive or negative pulse, a binary zero is represented by no line signal, and subsequent binary ones must alternate in polarity; otherwise, a *bipolar violation* occurs. AMI is used in the *DS1* interface.

analog transmission Mode of transmission in which information is represented in continuously variable physical quantities such as amplitude, frequency, phase, or resistance. See also *digital transmission.*

Analog data station See *modem data station*

ANI (automatic number identification) Process of automatically identifying a caller's billing number and transmitting that number from the caller's local central office to another point on or off the public network.

application Software and/or hardware that adds functional capabilities to the system. For example, MERLIN Identifier is an application that provides caller identification information (if available in the local area or jurisdiction).

ARS (Automatic Route Selection) System feature that routes calls on outside trunks according to the number dialed and trunk availability.

ASCAP (American Society of Composers, Artists, and Producers)

ASN (AT&T Switched Network) AT&T telecommunications services provided through an Integrated Digital Services Network Primary Rate Interface (ISDN-PRI) trunk, *Accunet* switched digital service, *Megacom*, *Megacom 800*, Software Defined Network (*SDN*), Multiquest, and Shared Access for Switch Services (SASS).

asynchronous data transmission A method of transmitting a short bitstream of digital data, such as printable characters represented by a 7- or 8-*bit* ASCII code. Each string of data bits is preceded by a start bit and followed by a stop bit, thus permitting data to be transmitted at irregular intervals. See also *synchronous data transmission.*

AT&T Attendant — Application with equipment that connects to one or more *tip/ring* extension jacks and automatically answers incoming calls with a recorded announcement; directs calls in response to touch tones.

AT&T Switched Network — See *ASN*.

AUDIX Voice Power — A voice-processing application, part of *IS II/III*, that provides Automated Attendant, Call Answer, Information Service, Message Drop, Voice Mail, and, optionally, *Fax Attendant System* for use with the system.

Automated Attendant — *IS II/III*, *MERLIN MAIL*, and MERLIN *Attendant* application that automatically answers incoming calls with a recorded announcement and directs callers to a department, an extension, or the system operator.

Automatic Line Selection — See *ALS*.

Automatic Number Identification — See *ANI*.

automatic ringdown tie-trunk — See *automatic-start tie trunk.*

Automatic Route Selection — See *ARS*.

automatic-start tie trunk — *Tie trunk* on which incoming calls are routed to an operator or other designated destination without a start signal, as soon as the trunk is seized; the destination is specified during programming. Also called "automatic ringdown" or "auto-in" tie trunk.

auxiliary power unit — Device that provides additional power to the system.

B

B8ZS — (bipolar 8 zero substitution) Line-coding format that encodes a string of eight zeros in a unique binary sequence to detect *bipolar violation*. See also *bipolar signal.*

backup — Procedure for saving a copy of system programming onto a floppy disk or *memory card*. See also *restore.*

bandwidth Difference, expressed in hertz, between the highest and lowest frequencies in a range that determines channel capacity.

barrier code Password used to limit access to the *Remote Access* feature of the system.

basic carrier Hardware that holds and connects the *processor*, *power supply*, and up to five modules in the system. See also *expansion carrier.*

baud rate Strictly speaking, a measurement of transmission speed equal to the number of signal level changes per second. In practice, often used synonymously with *bit rate* and *bps*.

B-channel (Bearer-channel) 64-kbps channel that carries a variety of digital information streams, such as voice at 64 kbps, data at up to 64 kbps, wideband voice encoded at 64 kbps, and voice at less than 64 kbps, alone or combined.

Bearer-channel See *B-channel.*

Behind Switch mode One of three modes of system operation, in which the control unit is connected to (behind) another telephone switching system, such as *Centrex* or Definity, which provides features and services to telephone users. See also *Hybrid/PBX mode* and *Key mode.*

binary code Electrical representation of quantities or symbols expressed in the base-2 number system, which includes zeros and ones.

bipolar 8 zero substitution See *B8ZS.*

bipolar signal Digital signal in which pulses (ones) alternate between positive and negative. See also *AMI*, *B8ZS*, and *bipolar violation.*

bipolar violation Condition occurring when two positive or two negative pulses are received in succession. See also *AMI* and *B8ZS.*

BIS (Built-In Speakerphone) Part of the model name of some analog multiline telephones.

bit (binary digit) One unit of information in binary notation; it can have one of two values, zero or one.

bit rate Speed at which bits are transmitted, usually expressed in *bps*. Also called "data rate." See also *baud rate.*

blocking Condition in which end-to-end connections cannot be made on calls because of a full load on all possible services and facilities. See also *glare.*

BMI (Broadcast Music Incorporated)

board A *module*, for example, 100D or 408 MLX GS/LS, that allows you to connect lines/trunks and extensions to the communications system or holds the processor or power supply.

board assignment *SPM* procedure for assigning *line/trunk and extension modules* to slots on the *control unit*.

board renumbering System programming procedure for renumbering *line/trunk and extension* modules that have already been assigned to specific *slots* on the *control unit*.

BRI (Basic Rate Interface) Standard digital *framing format* that specifies the protocol between the communication system and a terminal. BRI runs at 19.2-kbps and provides two 64-kbps voice or B-channels and on 16-kbps signaling or D-channel per port. The remaining 48-kbps are used for framing and D-channel contention.

broadband Transmission path having a bandwidth greater than a voice-grade channel.

BTMI (basic telephone modem interface)

bus Multiconductor electrical path used to transfer information over a common connection from any of several sources to any of several destinations.

button Key on the face of a telephone that is used to access a line, activate a feature, or enter a code on a communications system.

byte Sequence of *bit*s (usually eight) processed together. Also called "octet."

C

Call Accounting System See *CAS*.

Call Accounting Terminal See *CAT*.

Caller ID A service provided by some local telephone companies (if local regulations allow) that supplies the calling party telephone number. In Release 3.0 and later, an 800 GS/LS-ID module on the system can capture this information and display it on the screeens of MLX telephones. See also *ANI*.

Calling group Team of individuals who answer the same types of calls.

Call Management System	See *CMS*.
campus cable	Cable that runs between buildings connected to the same communications system.
CAS	(Call Accounting System) DOS- or UNIX System-based application that monitors and manages telecommunications costs.
CAT	(Call Accounting Terminal) Stand-alone unit with a built-in microprocessor and data buffer that provides simple call accounting at a low cost.
CCITT	(International Telegraph and Telephone Consultative Committee)
CCS	(common-channel signaling) Signaling in which one channel of a group of *channels* carries signaling information for each of the remaining channels, permitting each of the remaining channels to be used to nearly full capacity. In the system's 100D module, channel 24 can be designated as the signaling channel for channels 1–23.
centralized telephone programming	Programming of features on individual telephones; performed at a central location by the *system manager*. See also *system programming* and *extension programming.*
central office	See *CO*.
Centrex	Set of system features to which a user can subscribe on telephone lines/trunks from the local telephone company.
channel	Telecommunications transmission path for voice and/or data.
channel service unit	See *CSU*.
checksum	Sum of ones in a sequence of ones and zeros used to detect or correct errors in data transmission.
circuit-switched data call	Data call made through an exclusively established and maintained connection between *data stations*.
class of restriction	See *COR*.
clock synchronization	Operation of digital facilities from a common clock.
CMS	(Call Management System) DOS-based application that simulates the actions of a system operator by answering and distributing calls. Also produces reports for call analysis.

CO	(central office) Location of telephone switching equipment that provides local telephone service and access to toll facilities for long-distance calling.
coaxial cable	Cable consisting of one conductor, usually a small copper tube or wire within and insulated from another conductor of larger diameter, usually copper tubing or copper braid.
codec	(coder-decoder) Device used to convert analog signals such as speech, music, or television to digital form for transmission over a digital medium and back to the original analog form.
common channel signaling	See *CCS*.
communications system	Software-controlled processor complex that interprets dialing pulses, tones, and or keyboard characters and makes the proper interconnections both inside and outside. Consists of a computer, software, a storage device, and *carriers* with special hardware to perform the actual connections. Provides voice and/or data communications services, including access to public and private networks, for telephones and other equipment. Also referred to in this guide as "system," short for MERLIN LEGEND Communications System.
control unit	*Processor module*, *power supply modules*, *line/trunk and extension modules*, *carrier*s, and housing of the system.
console	Refers to telephone and adjuncts (if any) at operator or system programmer extension.
CONVERSANT Intro	Entry-level voice response application that automatically answers and routes calls and executes telephone transactions.
conversion resource	See *modem pool*.
COR	(class of restriction) Various types of restrictions that can be assigned to *remote access* trunks or barrier codes. These restrictions consist of calling restrictions, *ARS* Facility Restriction Levels (*FRL*s), Allowed Lists, Disallowed Lists, and Automatic Callback queuing.
Coverage	Set of system features that can determine how extensions' calls are covered when the person at the extension is busy or not available.
CRC	(cyclic redundancy check) An error-detection code used on *DS1* facilities with the extended superframe format (*ESF*).

CSU (channel service unit) Equipment used on customer premises to provide *DS1* facility terminations and signaling compatibility.

cyclic redundancy check See *CRC*.

D

D4 framing format *Framing format* consisting of a sequence of individual frames of 24 eight-*bit* slots and one signal bit (193 bits) in a 12-frame superframe. See also *ESF*.

Data-channel See *D-channel*.

data communications equipment See *DCE*.

data hunt group See *DHG*.

data module See *ISDN 7500B Data Module*.

data rate See *bps*.

data station Special type of extension where data communications take place; includes *DTE* and *DCE*; sometimes a telephone is also part of a data station.

data terminal An input/output (*I/O*) device (often a personal computer) that can be connected to the control unit via an interface.

data terminal equipment See *DTE* and *data terminal*.

DCE (data communications equipment) Equipment such as *modem*s or data modules used to establish, maintain, and terminate a connection between the system and data terminal equipment (*DTE*), such as printers, personal computers, host computers, or network workstations.

D-channel (Data-channel) 64-kbps channel that carries signaling information or data on a *PRI* or 16-kpbs channel to carry signaling information on *BRI*.

DCP (Digital Communications Protocol) AT&T proprietary protocol to transmit digitized voice and data over the same communications link.

dedicated feature buttons The imprinted feature buttons on a multiline telephone: **Conf** or **Conference**, **Drop**, **Feature**, **HFAI** (Hands Free Answer on Intercom), **Hold**, **Message**, **Mute** or **Microphone**, **Recall**, **Speaker** or **Spkrphone**, and **Transfer**.

delay-dial-start tie trunk See *dial-repeating tie trunk*

DFT (direct facility termination) See *personal line*.

DHG (data hunt group) Group of analog or digital *data stations* that share a common access code. Calls are connected in a round-robin fashion to the first available data station in the group.

dial access See *feature code*.

Dialed Number identification Service See *DNIS*.

dial-out code Digit (usually a 9) or digits dialed by telephone users to get an outside line.

dial plan Numbering scheme for system extensions, lines, and trunks.

dial-repeating tie trunk *Tie trunk* on which the originating end of the tie trunk transmits an off-hook signal to the receiving end and waits for the receiving end to send an off-hook signal followed by an on-hook signal. Also called "dial-repeating tie trunk."

DID (Direct Inward Dialing) Service that transmits from the telephone company central office and routes incoming calls directly to the called extension, *calling group*, or outgoing trunk *pool*, bypassing the system operator.

DID trunk Incoming trunk that receives dialed digits from the local exchange, allowing the system to connect directly to an extension without assistance from the system operator.

digital Representation of information in discrete elements such as off and on or zero and one. See also *analog transmission*.

Digital Communications Protocol See *DCP*.

digital data station See *7500B data Station*

Digital Signal 0 See *DS0*.

Digital Signal 1 See *DS1*.

digital switch element See *DSE*.

digital transmission	Mode of transmission in which the information to be transmitted is first converted to digital form and then transmitted as a serial stream of pulses. See also *analog transmission*.
DIP switch	(dual in-line package) Switch on a 400EM module used to select the signaling format for tie-line transmission. Also used on other equipment for setting hardware options.
direct facility termination	(DFT) See *personal line*.
Direct Inward Dialing	See *DID*.
Direct-Line Console	See *DLC*.
Direct Station Selector	See *DSS*.
display buttons	Buttons on an MLX display telephone used to access the telephone's display.
DLC	(Direct-Line Console) Telephone used by a system operator to answer outside calls (not directed to an individual or a group) and inside calls, transfer calls, make outside calls for users with outward calling restrictions, set up conference calls, and monitor system operation.
DNIS	(Dialed Number Identification Service) Service provided by the AT&T Switched Network (*ASN*); it routes incoming 800 or 900 calls according to customer-selected parameters, such as area code, state, or time of call.
door answering unit	Device connected to a basic telephone jack and used at an unattended extension or front desk.
DOS	(disk operating system)
DS0	(Digital Signal 0) Single 64-kbps voice or data *channel*.
DS1	(Digital Signal 1) *Bit*-oriented signaling interface that multiplexes twenty-four 64-kbps channels into a single 1.544-Mbps stream.
DSS	(Direct Station Selector) 60-button *adjunct* that enhances the call-handling capabilities of an *MLX-20L or MLX-28D telephone* used as an operator console.
DTE	(data terminal equipment) Equipment that makes the endpoints in a connection over a data connection, for example, a data terminal, personal computer, host computer, or printer.

DTMF signaling	(dual-tone multifrequency signaling) Touch-tone signaling from telephones using the voice transmission path. DTMF signaling provides 12 distinct signals, each representing a dialed digit or character, and each composed of two voiceband frequencies.

E

E&M signaling	Trunk supervisory signaling, used between two communications systems, in which signaling information is transferred through two-state voltage conditions (on the Ear and Mouth leads) for analog applications and through two *bit*s for digital applications. See also *tie trunk*.
EIA	(Electronic Industries Association)
EIA-232-D	Physical interface, specified by the *EIA*, that transmits and receives asynchronous data at speeds of up to 19.2-kbps over cable distances of 50 ft. (15 m.)
Electronic Switching System	See *ESS*.
endpoint	Final destination in the path of an electrical or telecommunications signal.
ESF	(extended superframe format) *Framing format* consisting of individual frames of 24 eight-*bit* slots and one signal bit (193 bits) in a 24-frame extended superframe. See also *D4 framing format.*
ESS	(Electronic Switching System) Class of central office (*CO*) switching systems developed by AT&T in which the control functions are performed principally by electronic data processors operating under the direction of a stored program.
expansion carrier	*Carrier* added to the control unit when the basic carrier cannot house all of the required modules. Houses a power supply and up to six additional modules.
extension	An endpoint on the internal side of the communications system. An extension can be a telephone with or without an adjunct. Also called "station." See also *data station.*
extension jack	An analog, digital, or *tip/ring* physical interface on a module in the control unit for connecting a telephone or other device to the system. Also called "station jack."

extension programming — Programming performed at an extension to customize telephones for personal needs; users can program features on buttons, set the telephone ringing pattern, and so on. See also *centralized telephone programming* and *system programming*.

extended superframe format — See *ESF*.

F

facility — Equipment (often a *trunk*) constituting a telecommunications path between the system and the telephone company central office (*CO*).

Facility Restriction Level — See *FRL*.

factory setting — Default state of a device or feature when an optional setting is not programmed by the user or system manager.

fax — (facsimile) Scanning and transmission of a graphic image over a telecommunications facility, or the resulting reproduced image, or the machine that does the scanning and transmitting.

Fax Attendant System — Fax handling and processing application available with *AUDIX Voice Power*.

FCC — (Federal Communications Commission)

feature — Function or service provided by the system.

feature code — Code entered on a dialpad to activate a feature.

feature module — Prior to Release 3.0, a circuit pack inserted into the *processor* module, used to provide system features and replaced when the system is upgraded.

Feature screen — Display screen on MLX display telephones; provides quick access to commonly used features.

ferrite core — Attachment to the AC power cord and ground wire of the carrier power supply for compliance with FCC, part 15 requirements.

Flash ROM — Beginning with Release 3.0, a type of read-only memory provided on the *processor module*, used to supply system features.

foil shield — Copper foil sheet (for power units) used to prevent excessive noise on the module.

forced idle	Condition of the system during certain programming or maintenance procedure; system prevents initiation of new calls.
foreign exchange	See *FX*.
frame	One of several segments of an analog or digital signal that has a repetitive characteristic. For example, a *DS1* frame consists of a framing *bit* and 24 bytes, which equals 193 bits.
framing format	Pattern of *frames* used in transmissions.
frequency generator	See *ring generator*.
FRL	(Facility Restriction Level) *ARS* calling restriction type that restricts outgoing calls to certain specified routes.
FX	(Foreign exchange) Central office (*CO*) other than the one that is providing local access to the public telephone network.

G

General-Purpose Adapter	See *GPA*.
glare	Condition that occurs when a user tries to call out on a *loop-start* trunk at the same time that another call arrives on the same trunk.
GPA	(General-Purpose Adapter) Device that connects an analog multiline telephone to optional equipment such as an answering machine or a fax machine.
ground-start trunk	Trunk on which the communications system, after verifying that the trunk is idle (no ground on tip lead), transmits a request for service (puts ground on ring lead) to the telephone company central office (*CO*).
Group IV (G4) fax machine	A fax unit, offering 400 by 100 dots per inch (DPI) in fine mode, that can operate at any speed for communication with a Group III (G3) fax machine or another Group IV (G4) fax machine.

H

Hands Free Answer on Intercom See *HFAI*.

hands-free unit See *HFU*.

headset Lightweight earpiece, microphone, and adapter used for hands-free telephone operation.

HFAI (Hands Free Answer on Intercom) Feature that allows a user to answer a voice-announced call.

HFU (Hands-Free Unit) Unit for older analog multiline telephones that allows users to make and receive calls on the speakerphone without using the handset.

Home screen Display normally shown on an MLX display telephone; shows time, date, and call information, and shows when some features are in use.

host Telephone company or other switch providing features and services to the system users, usually when the system is operating in *Behind Switch mode*.

Hybrid/PBX mode One of three modes of system operation, in which the system uses trunk *pools* and *ARS* in addition to *personal lines*. Provides a single interface (***SA** buttons*) to users for both internal and external calling. See also *Behind Switch mode* and *Key mode*.

I

ICLID (Incoming Call Line Identification) See *Caller ID*.

ICOM buttons (intercom buttons) Telephone buttons that provide access to inside system lines for calling other extensions or receiving calls from them.

immediate-start tie trunk *Tie trunk* on which no start signal is necessary; dialing can begin immediately after the trunk is seized.

in-band signaling See *robbed-bit signaling*.

inside dial tone A tone users hear when they are off-hook on an ***SA*** or ***ICOM** button*.

Inspect screen Display screen on an MLX display telephone that allows the user to preview incoming calls and see a list of the features programmed on line buttons.

Integrated Administration Capability of *IS III* that simplifies the programming of common information for the system, *AUDIX Voice Power*, and, if it is also installed, *Fax Attendant System*.

Integrated Services Digital Network See *ISDN*.

Integrated Solution II/III See *IS II/III*.

Integrated Voice Power Automated Attendant *IS II* application that automatically answers incoming calls with a recorded announcement and directs callers to a department, an extension, or the system operator.

intercom buttons See *ICOM buttons*.

interface Hardware and/or software that links systems, programs, or devices.

I/O device (input/output device) Equipment that can be attached to a computer internally or externally for managing a computer system's input and output of information.

IROB protector (In-Range Out-of-Building protector) Surge-protection device for off-premises telephones at a location within 1000 feet (305 m) of cable distance from the control unit.

IS II/III (Integrated Solution II or Integrated Solution III) Set of UNIX System-based applications that augments and provides additional services using the system.

ISDN (Integrated Services Digital Network) Public or private network that provides end-to-end digital connectivity for all services to which users have access by a limited set of standard multipurpose user and *network interfaces*; provides digital circuit-switched or packet-switched connections within the network and to other networks for national and international digital connectivity.

ISDN 7500B Data Module Data communications device that allows connection between an RS-232 *DTE* device and the control unit via MLX extension jacks on the 008 MLX or 408 GS/LS-MLX module.

J

jack Physical connection point to the system for a telephone, trunk, or other device. Also called "port."

K

kbps	Kilobits per second.
Key mode	One of three modes of system operation, in which the system uses *personal lines* on line buttons for outside calls, with a separate interface (***ICOM*** *buttons*) for internal calling. See also *Behind Switch mode* and *Hybrid/PBX mode.*

L

LAN	(local area network) Arrangement of interconnected personal computers or terminals, sometimes accessing a host computer, sometimes sharing resources like files and printers.
LDN	(listed directory number)
LED	(light-emitting diode) Semiconductor device that produces light when voltage is applied; light on a telephone.
line	Connection between extensions within the communications system or loop-start communications path with *CO*; often, however, used synonymously with *trunk*.
line and trunk assignment	Assignment of lines and trunks connected to the system control unit to specific buttons on each telephone.
line coding	Pattern that data assumes as it is transmitted over a communications channel.
line compensation	Adjustment for the amount of cable loss in decibels (dB), based on the length of cable between a 100D module and a channel service unit (*CSU*) or other far-end connection point.
line/trunk	Refers to inside system lines and outside trunks in general terms. See also *line* and *trunk*.
line/trunk jack	Physical interface on a module in the control unit for connecting an outside trunk to the communications system. Also called “trunk jack.”
line/trunk and extension module	Module on which the jacks for connecting central office lines/trunks and/or the jacks for connecting the extensions are located.

local host computer access A method for connecting an extension jack to an on-site computer for data-only calls through a *modem* or data module.

local loop See *access line.*

logical ID Unique numeric identifier for each *extension* and *line/trunk jack* in the system control unit.

loop-start trunk Trunk on which a closure between the tip and ring leads is used to originate or answer a call. High-voltage 20-Hz AC ringing current from the central office signals an incoming call.

M

Magic on Hold An AT&T Music On Hold enhancement that promotes a company's products or services or provides custom music selection.

Mbps megabits per second

Megacom AT&T's tariffed digital *WATS* offering for outward calling.

Megacom 800 AT&T's tariffed digital 800 offering for inward calling.

memory card Storage medium, similar in function to a floppy disk, that allows information to be added to or obtained from the communication system through the PCMCIA interface slot on the processor module.

MERLIN Identifier Adjunct that allows users to receive, store, and use information provided by caller identification services provided by the local telephone company.

MERLIN Mail Voice Messaging System Application that provides automated attendant, call answering, and voice-mail services on the system.

MFM (Multi-Function Module) Adapter that has a *tip/ring* mode for answering machines, modems, fax machines, and tip/ring alerts, and an *SAA* mode for -48 VDC alerts. Installed inside an MLX telephone, and is used to connect optional equipment to the telephone. The optional equipment and the telephone operate simultaneously and independently.

MLX-10 or MLX-10D telephone 10-line button digital telephone offered with (MLX-10D) or without (MLX-10) a 2-line by 24-character display.

MLX-10DP Same as an MLX-10D except it has an adjunct in the back for connecting the *Passageway Direct Connect Solution* application.

MLX-20L telephone 20-line button digital telephone with a 7-line by 24-character display.

MLX-28D telephone 28-line button digital telephone with a 2-line by 24-character display.

mode codes Streams of touch-tone codes used by voice messaging applications to communicate with the system's control unit.

modem (modulator-demodulator) device that converts digital data signals to analog signals for transmission over a telephone line, and analog signals received on a telephone line to digital signals.

modem data station A type of data station that includes a modem as its DCE. It may also include an MLX telephone for simultaneous voice and data (MLX voice and modem data station), an analog multiline telephone (analog voice and modem data station), or a single-line telephone for dialing only (modem data-only station). These data stations connect respectively to MLX, analog, or tip/ring extension jack modules. They provide analog transmission of data.

modem pool Pair, or group of pairs, of *modems* and data modules with interconnected RS-232 interfaces that converts digital signals to analog, or analog signals to digital, thereby allowing users with *7500B data stations* to communicate with users who have analog *modem data stations*.

module Circuit pack in the control unit that provides the physical jacks for connection of telephones and/or outside lines/trunks to the communications system. In the name of a module, the first digit indicates the number of *line/trunk* jacks it contains; the last digit indicates the number of *extension jacks* it contains. If no letters appear after the number, a line/trunk module provides *loop-start trunks* or an extension jack module provides analog or *tip/ring* jacks. For example, a 408 GS/LS MLX module contains four line/trunk jacks and eight digital (MLX) extension jacks, provides either loop-start (LS) or *ground-start* (GS)*trunks*. There are also modules for the processor and power supply.

Multi-Function Module See *MFM*.

multiline telephone An analog or digital (MLX) telephone that provides multiple line buttons for making or receiving calls or programming features.

multiplexing The division of a transmission channel into two or more independent channels, either by splitting the frequency band into a number of narrower bands or by dividing the channel into successive time slots.

Music On Hold Customer-provided music source or *Magic on Hold* connected to the system through a *loop-start* jack.

N

network Configuration of communications devices and software connected for information interchange.

network interface Hardware, software, or both that links two systems in an interconnected group of systems, for example, between the local telephone company and a PBX.

O

off-hook Telephone is said to be off-hook when the user has lifted the handset, pressed the **Speaker** button to turn on the speakerphone, or used a headset to connect to the communications system or the telephone network.

off-premises telephone See *OPT*.

ones density Requirement for channelized *DS1* service to the public network that eight consecutive zeros cannot occur in a digital data stream.

on-hook Telephone is said to be on-hook when the handset is hung up, the speakerphone is turned off, and the user is not using a headset to connect to the communications system or the telephone network.

OPT (off-premises telephone) *Single-line telephone* or other *tip/ring* device connected to the system by an 008 OPT module in the control unit. Appears as an inside extension to the system, but may be physically located away from the system.

OPX (off-premises extension)

out-of-band signaling Signaling that uses the same path as voice-frequency transmission and in which the signaling is outside the band used for voice frequencies.

P

parity The addition of a *bit* to a bit string so that the total number of ones is odd or even, used to detect and correct transmission errors.

PassageWay Direct Connect Solution Set of software applications to provide an interface between a personal computer and the system: cardfile, telephone programming application, call log and viewer, incoming call management and identification and applications manager.

pass-through Connection from the internal *modem* to an attached IS II/IS III application on the system.

PBX (private branch exchange) Local electronic telephone switch that serves local stations (for example, extensions within a business) and provides them with access to the public network.

PC (personal computer)

PCMCIA memory card Personal Computer Memory Card International Association memory card) See *memory card.*

personal line Central office trunk that terminates directly on one or more telephones. In *Hybrid/PBX mode*, a personal line cannot be part of a trunk *pool.* Also called "DFT" (direct facility termination). Also refers to lines represented on line buttons in *Key Mode.*

PFT (Power Failure Transfer) Feature that provides continuity of telephone service during a commercial power failure by switching some of the system's trunk connections to telephones connected to specially designated extension jacks.

pool In *Hybrid/PBX mode*, a group of outside trunks that users can access with a **Pool** button or by dialing an access code on an ***SA*** *button.* Also used by the *ARS* feature when choosing the least expensive route for a call.

port See *jack.* Also, refers to *extension* or *line jacks* before these are numbered according to the *dial plan* during programming. The lowest jack on a module is always Port 1.

Power Failure Transfer See *PFT.*

power supply module Device that directs electricity to modules and telephones on the system. One power supply module is needed for each carrier, and an *auxiliary power unit* is added if the module exceeds capacity.

PRI (Primary Rate Interface) Standard interface that specifies the protocol used between two or more communications systems. As used in North America, provides twenty-three 64-kbps *B-channels* for voice and/or data and one 64-kbps *D-channel*, which carries multiplexed signaling information for the other 23 channels.

primary system operator position First jack on the first MLX or analog multiline extension module in the control unit, that is, the extension jack with the lowest *logical ID* in the system.

prime line Individual extension number assigned to a telephone in a system operating in *Behind Switch mode*. Each telephone user has his or her own prime line and is automatically connected to that line when he or she lifts the handset.

processor module Module in the second slot of the control unit (Slot 0, to the right of the *power supply module*). Includes the software and memory that runs the system.

programming port reassignment Reassignment of the system programming jack position to any of the first five extension jacks on the first MLX module in the control unit.

protocol Set of conventions governing the format and timing of message exchanges between devices, such as an MLX telephone and the control unit.

public network Network that is commonly accessible for local or long-distance calling. Also called "public switched telephone network."

Q

QCC (Queued Call Console) MLX-20L telephone used by a system operator in *Hybrid/PBX mode* only. Used to answer outside calls (directed to a system operator position) and inside calls, direct inside and outside calls to an extension or an outside telephone number, serve as a message center, make outside calls for users with outward calling restrictions, set up conference calls, and monitor system operation.

R

RAM (random-access memory) Computer memory in which an individual *byte* or range of bytes can be addressed and read or changed without affecting other parts of memory.

read-only memory	See *ROM*.
Remote Access	System feature that allows an outside caller to gain access to the system, almost as if at a system extension.
restore	Procedure whereby saved and archived system programming is reinstated on the system, from a floppy disk or *memory card*. See also *backup*.
ring generator	Circuit pack added to the power supply that generates a high-voltage, 20–30 Hz signal to ring a telephone.
riser cable	Cable that runs between floors in a multistory building and connects wiring closets.
RS-232	Physical interface, specified by the Electronics Industries Association (*EIA*), that transmits and receives *asynchronous* data at distances of up to 50 feet (15 m).
robbed-bit signaling	Signaling in which the least significant *bit* of every sixth frame per channel is used for signaling in that channel.
ROM	(read-only memory) Computer memory that can be read but cannot be changed.

S

SAA	(Supplemental Alert Adapter) Device that permits -48-VDC alerting equipment to be connected to an analog multiline telephone jack so that people working in noisy or remote areas of a building can be alerted to incoming calls.
SA buttons	Telephone buttons that provide a single interface to users for both inside and outside calling.
SDN	(Software Defined Network) AT&T private networking service created by specialized software within the public network.
SID	(station identification)
simplex signaling	Transmission of signals in one direction only across a telecommunications channel.
signaling	Sending of control and status information between devices to set up, maintain, or cease a connection such as a telephone call.
single-line telephone	Industry-standard touch-tone or rotary-dial telephone that handles only one call at a time and is connected to the system via an *extension jack* on a basic 012 or 008 OPT module.

slot	Position in a *carrier* for a module; numbered from 0 (*processor module*).
SMDR	(Station Message Detail Recording) Feature that captures detailed usage information on incoming and outgoing voice and data calls.
SMDR printer	Printer used to produce SMDR reports. Connected to the system via an RS-232 jack on the *processor* module.
Software Defined Network	See *SDN*.
special character	Pause, Stop, or End-of-Dialing signal in a programmed dialing sequence such as an Auto Dial or Personal Speed Dial number.
SPM	(System Programming and Maintenance) *DOS*- or UNIX system-based application for programming and maintaining the system.
square key	Configuration in *Key mode* operation in which all outside lines appear on all telephones.
station	See *extension*.
station jack	See *extension jack*.
Station Message Detail Recording	See *SMDR*.
Supplemental Alert Adapter	See *SAA*.
switchhook flash	Momentary (320 ms to 1 second) on-hook signal used as a control signal. May be directed either to the control unit or to a *host* switch outside the system. Also called "Recall" or "timed flash."
synchronous data transmission	Method of transmitting a continuous digital data stream in which the transmission of each binary *bit* is synchronized with a master clock. See also *asynchronous data transmission*.
system acceptance test	Test of all trunks, telephones, data terminals, and features after installation to ensure that they are working correctly.
System Access buttons	See *SA buttons*.
system date and time	Date and time that appear on MLX display telephones and *SMDR* reports.
system programming	Programming of system functions and features that affect most users, performed from an *MLX-20L telephone* or a *PC* using *SPM*. See also *extension programming* and *centralized telephone programming.*

System Programming and Maintenance	See *SPM.*
system renumbering	Procedure used to change the numbers assigned to telephones, adjuncts, *calling group*s, paging groups, park zones. *remote access*, and lines/trunks.

T

T1	Type of digital transmission facility that in North America transmits at the *DS1* rate of 1.544 Mbps.
TDM	(time-division multiplex) Process where the transmission channel is divided into time slots.
telephone power supply unit	Equipment that provides power to an individual telephone.
tie trunk	Private trunk directly connecting two telephone switches. See also *automatic-start tie trunk*, *delay-dial-start tie trunk*, *immediate-start tie trunk*, and *wink-start tie trunk.*
timed flash	See *switchhook flash.*
tip/ring	Contacts and associated conductors of a *single-line telephone* plug or jack.
touch-tone receiver	See *TTR.*
T/R	See *tip/ring.*
trunk	A telecommunications path between the communications system and the telephone company central office (*CO*) or another switch. Often used synonymously with *line.*
trunk jack	See *line/trunk jack.*
trunk pool	See *pool.*
TTR	(touch-tone receiver) Device used to decode *DTMF* touch-tones dialed from *single-line telephones* or *Remote Access* telephones.

U

uninterruptible power supply	See *UPS.*

unit load Measure of the power load drain of a module, telephone, or *adjunct*.

UPS (uninterruptible power supply) Device that connects to the system to provide 117 VAC to the equipment when the commercial power source fails.

V

VAC Alternating-current voltage.

VDC Direct-current voltage.

VMI (voice messaging interface) An enhanced *tip/ring* port.

voice-band channel A transmission channel. generally in the 300–3400-Hz frequency band.

voice mail Application that allows users to send messages to other extensions in the system, forward messages received with comments, and reply to messages.

voice messaging interface See *VMI*.

W

WATS (Wide Area Telecommunications Service) Service that allows calls to certain areas for a flat-rate charge based on expected usage.

wink-start tie trunk *Tie trunk* on which the originating end transmits an off-hook signal and waits for the remote end to send back a signal (a wink) that it is ready for transmission.

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