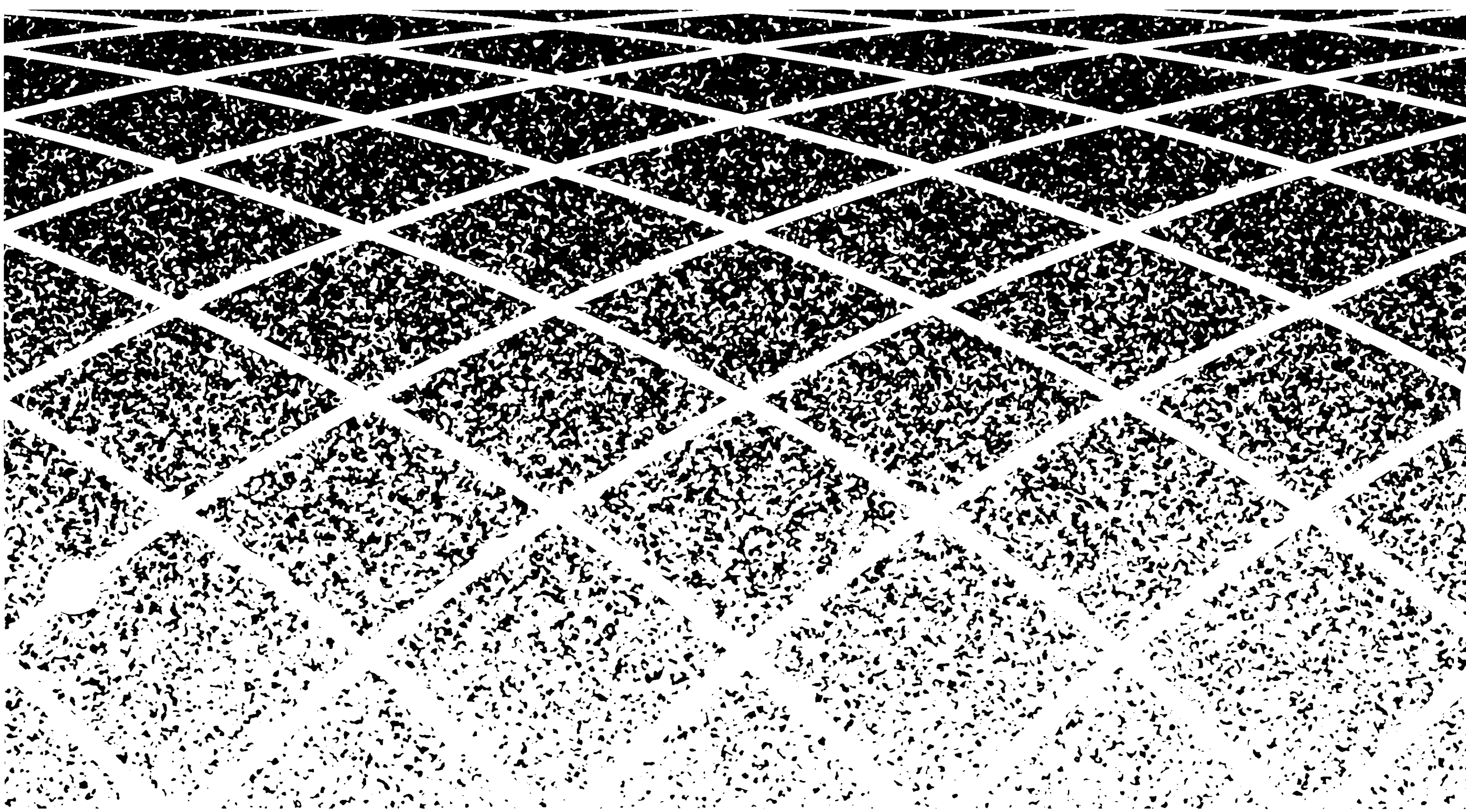
AT&T 555-620-115
Issue 1
October 1992

MERLIN LEGEND™ Communications System
Release 2.0

Equipment and Operations Reference



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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.

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 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 radioélectriques dépassant Ies Iimites applicable aux appareils numériques de la class A prescribes clans Ie Réglement sur Ie brouillage radioélectrique édicté par Ie ministère des Communications du Canada.

Trademarks

5ESS, ACCUNET, CONVERSANT, Dimension, Horizon, Magic On Hold, MEGACOM, MERLIN, SYSTIMAX, and MultiQuest are registered trademarks and 4ESS, AUDIX Voice Power, FAX Attendant System, InnManager, MERLIN LEGEND, MERLIN MAIL, MLX-10, MLX-10D, MLX-20L, MLX-28D, and MERLIN PFC are trademarks of AT&T in the U.S. and other countries.

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Support Telephone Number

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

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he 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 (511A) control unit.
- Use only AT&T-recommended/approved MERLIN LEGEND Communications System accessories.
- If equipment connected to the analog station 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 three-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).**

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

Customer Support Information

Support Telephone Number

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.

Outside the U. S. A., if you need assistance when installing, programming, or using your system, contact your authorized AT&T dealer.

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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,

Le Présent Appareil Numérique n'émet pas de bruits radioelectriques depassant Ies Iimites applicable aux appareils numériques de la class A prescribes clans Ie reglement sur Ie brouillage radioelectrique edicté par Ie 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 stations 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.

  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 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 for all three systems is 1.5A.
  — For tie line connection, the facility interface code (FIC) is TL31 M and the service order code (SOC) is 9.0F.
  — For connection to off-premises stations, the FIC is OLI3C 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.

  You must also notify your local telephone company if and when this equipment is permanently disconnected from the line(s).

  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 T&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.

  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).**

  a. This equipment returns answer supervision signals to the Public Switched Telephone Network when:
  (1) answered by the called station
  (2) answered by the attendant
  (3) routed to a recorded announcement that can be administered by the customer premises equipment user
  (4) routed to a dial prompt

  b. This equipment returns answer supervision on all DID calls forwarded back to the Public Switched Telephone Network, Permissible exceptions are when:
  (1) a call is unanswered
  (2) a busy tone is received
  (3) 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, Ies fils intérieurs de l'enterprise utilisés pour un service individual à Iigne 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 degradation du service clans certaines situations. Actuellement, Ies entreprises de télécommunication ne permettent pas que I'on raccorde Ieur matériel à des jacks d'abonné, sauf clans Ies cas précis prévus pas Ies tarifs particuliers de ces entreprises.

Les réparations de matériel homologué doivent être effectuées par un centre d'entretien canadien autorisé désigné par Ie fournisseur. La compagnie de télécommunications peut demander à I'utilisateur de débrancher un appareil à la suite de reparations ou de modifications effectuées par I'utilisateur ou à cause de mauvais fonctionnement.

Pour sa propre protection, I'utilisateur doit s'assurer que tous Ies fils de mise à la terre de la source d'énergie électrique, des Iignes téléphoniques et des canalisations d'eau métalliques, s'il y en a, sent raccordés ensemble. Cette precaution est particulièrement importance clans Ies régions rurales.

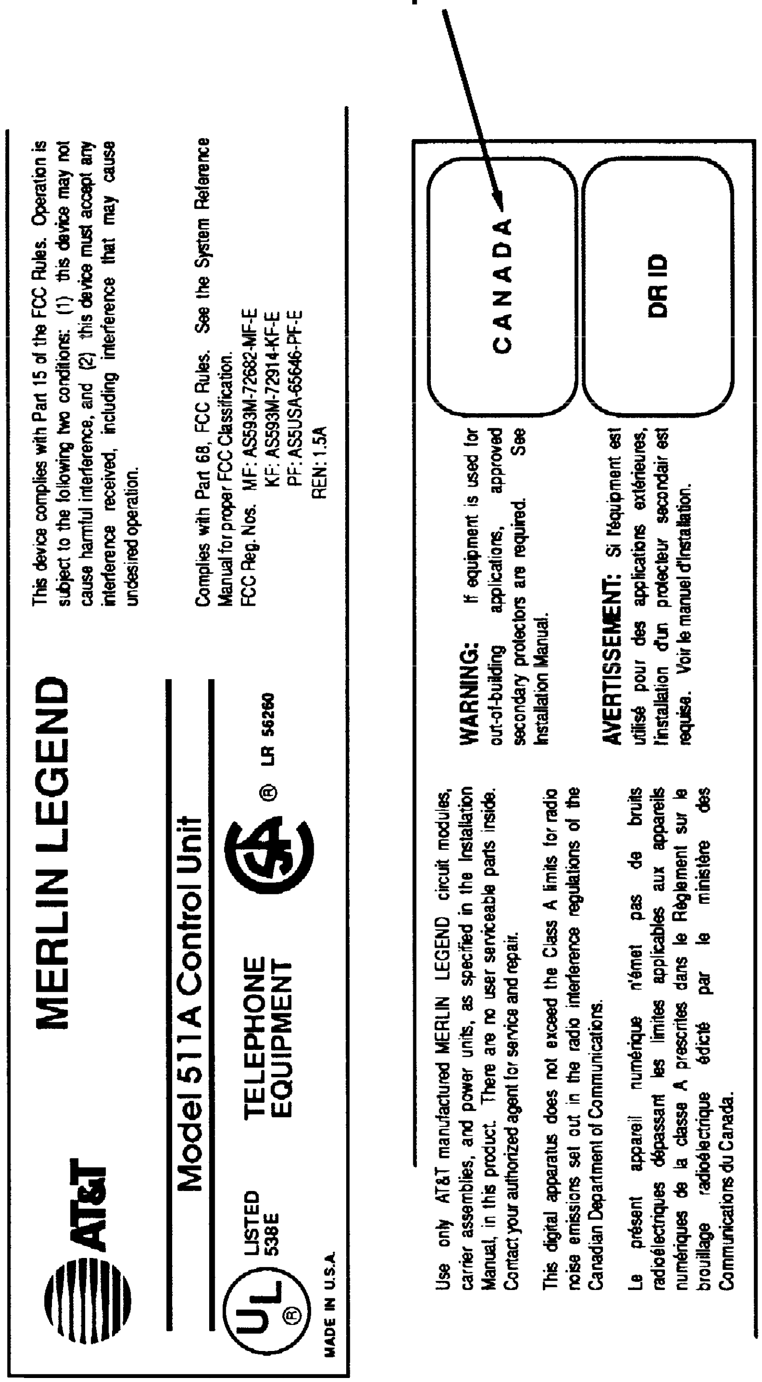
AVERTISSEMENT: L'utilisateur ne doit pas tenter de faire ces raccordements lui-même; il doit avoir recours à un service d'inspection des installations électriques, ou à un electrician, selon Ie cas.

L'indite de charge (IC) assigné à chaque dispositif terminal indique, pour éviter toute surcharge, Ie 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 constitute de n'importe quelle combinaison de dispositifs, pourvu que la somme des indices de charge de I'ensemble des dispositifs ne dépasse pas 100.

No d'homologation: 230 4095A
Node certification: CSA LR 56260
L'indite 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 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. 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 an 800 number or a 7- or 10-digit telephone number. The system returns an acknowledgement signaling the user to key in his or her authorization code, which is selected and administered by the system manager. After the authorization code is accepted, the system returns dial tone to the user. If you do not program specific egress 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 (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 authorization 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 authorization 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 authorization codes secure.
- When possible, restrict the off-network capability of off-premises callers, via use of Call Restrictions 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 Forward 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 back-up 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 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.
  - If you do not need to use the Outcalling feature, completely restrict the outward calling capability of the AUDIX Voice Power ports.
- Monitor SMDR reports or Call Accounting System reports for outgoing calls that might be originated by AUDIX Voice Power 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 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, 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 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 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 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.

About This Book

This document covers all aspects of the MERLIN LEGEND™ Communications System Release 2.0, a state-of-the-art telephone switching system that provides both voice and data communication features.

The document is intended for use by anyone who needs detailed information about the hardware and software that apply to the communications system, including support personnel, technicians, sales representatives, and account executives. It describes system components and capabilities, modes of operation, lines and trunks, applications, and data communications support.

The following documents may be referenced for additional information:

- *Feature Reference*
- *System Planning*
- System Programming

See "Related Documents" later in this section.

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Terms and Conventions Used

In this document, the following terms are used to describe components of the communications system:

- telephone (synonymous with voice terminal)
- extension (synonymous with station)
- control unit (synonymous with switch)

Although the terms **line** and **trunk** technically refer to different facilities, they are often used interchangeably. In this document, **trunk** refers to either facility.

Typographical conventions are used in this document to distinguish certain kinds of information. The conventions are as follows:

- **Bold type** is used for emphasis and for telephone buttons.

  Press **Drop** to delete the current entry.

- `Constant width` type is used for information on telephone display screens or on a PC screen,

  Select `Sys Program`,

- **`Bold constant width`** type indicates information that you enter exactly as shown.

  Type **`install;`** dial #55.

Product Safety Labels

An exclamation point inside a triangle and the word "caution" or "warning" indicate hazardous situations. These product safety labels appear as follows.

⚠ **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

The use of passwords prevents unauthorized users from abusing the communications system. It is strongly recommended that passwords be assigned wherever possible and that the passwords are provided only to those persons directly responsible for system administration and maintenance.

Non-displaying access codes and telephone numbers provide another layer of security. The following cautionary note pertains to security:

⚠ **CAUTION:**
For more information about the security of your communications system to prevent toll fraud, see the "Customer Support Information" section at the front of this document.

Related Documents

A number of related documents are available, providing additional information about the communications system. Whenever a reference to a related document is given within a document, the reference uses a shortened version of the document's title. For example, MERLIN LEGEND *Communications System Release 2.0 System Programming* is referred to as *System Programming.*

Within the continental United States, these documents can be ordered from the AT&T Customer Information Center (CIC) by calling 1-800-432-8600 or by contacting your local sales representative or authorized dealer.

Document No.	Title
	System Documents
555-620-114	*System Overview*
555-620-110	*Feature Reference*
555-620-115	*Equipment and Operations Reference*
555-620-116	*Pocket Reference*
555-620-111	*System Programming*
555-620-112	*System Planning*
555-62~113	*System Planning Forms*
	Telephone User Support
555-620-122	*MLX-10D, MLX-28D, and MLX-20L Display Telephones User's Guide*
555-620-123	*MLX-10D, MLX-28D, and MLX-20L Display Telephones Quick Reference*
555-620-150	MLX-10D (Display) Telephone Tray Cards (6 cards)
555-620-152	MLX-28D and MLX-20L Telephone Tray Cards (5 cards)
555-620-124	*MLX-10 Non-Display Telephone User's Guide*
555-620-125	*MLX-10 Non-Display Telephone Quick Reference*
555-620-151	MLX-10 (Non-Display) Telephone Tray Cards (6 cards)
555-620-120	*Analog Multiline Telephones User's Guide*
555-620-121	*Analog Multiline Telephones Quick Reference*
555-620-128	*ML C-5 Cordless Telephone Quick Reference*
555-620-126	*Single-Line Telephones User's Guide*
555-620-127	*Single-Line Telephones Quick Reference*
	System Operator Support
555-620-134	*MLX Direct-Line Consoles Operator's Guide*
555-620-135	*MLX Direct-Line Consoles Quick Reference*
555-620-132	*Analog Direct-Line Consoles Operator's Guide*
555-620-133	*Analog Direct-Line Consoles Quick Reference*
555-620-136	*MLX Queued Call Console Operator's Guide*
555-620-137	*MLX Queued Call Console Quick Reference*
	Miscellaneous User Support
555-620-130	*Calling Group Supervisor's Guide*
555-620-131	*Calling Group Supervisor's Quick Reference*
555-620-129	*Data User's Guide*
	Documentation for Qualified Technicians
555-620-140	*Installation, Programming, & Maintenance (lP&M) Binder* (consists of 555-620-141,555-620-142, 555-620-143, and 555-620-1 44)
555-620-141	*Installation*
555-620-142	*System Programming & Maintenance (SPM)*
555-620-143	*Maintenance and Troubleshooting*
555-620-144	*Programming Summary*

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Room 2A25
Parsippany, NJ 07054.

Introduction

1

The MERLIN LEGEND Communications System is an advanced digital switching system that integrates voice and data communications features. Voice features combine traditional telephone features, such as Transfer and Hold, with advanced 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—the required and optional equipment that makes up the system
- Functional Description-the functional units that make up the control unit, their relationships, and the process of signaling
- Modes of Operation—the three modes for which the system can be configured: Key mode, Hybrid/PBX mode, and Behind Switch mode
- Programming-general information about programming the system and telephones
- System Capacities and Requirements—the technical requirements and capacities of the system, for example, hardware and software capacities and environmental, power, and grounding requirements

Release Differences-enhancements provided by Release 1.1 and Release 2.0 of the system

Components

The system consists of the following basic components and optional auxiliary components:

- Basic components
  - Control unit
  - Telephones
- Auxiliary components
  - Adjuncts
  - Adapters
  - 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 station modules. Each expansion carrier contains a power supply module and line/trunk and station modules.

Telephones

The telephones that can be used with the system are MLX (digital) telephones, analog multiline telephones (including cordless telephones), and single-line telephones.

MLX Telephones

The following MLX telephones can be used:

- MLX-10D™ (10 buttons with display)
- MLX-20L™ (20 buttons with display)
- MLX-28D™ (28 buttons with display)
- MLX-10™ (10 buttons, no display)

Analog Multiline Telephones

The following analog multiline telephones can be used:

- 5-button* (5 buttons, membrane, no adjuncts supported)

- 10-button* (10 buttons, membrane)
- 34-button* (34 buttons, membrane)
- 34-button Deluxe* (34 buttons, membrane)
- 10-button HFAI* (10 buttons, hands-free-answer, no adjuncts supported)
- 34-button BIS* (34 buttons, built-in speakerphone)
- 34-button BIS/DIS* (34 buttons, built-in speakerphone, 16-character display)
- 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 (34 buttons, built-in speakerphone)
- BIS-34D (34 buttons, built-in speakerphone, 16-character display)
- MLC-5 Cordless (5 buttons, cordless)
- MERLIN PFC™ Telephone (telephone, fax machine, and copier)

* Vintage telephone; no longer available for sale or lease

† Requires two analog multiline ports

Single-Line Telephones

- 2500MMGB (desk telephone)
- 2554MMGJ (wall telephone)
- 2500YMGK* (desk telephone, message light, **Recall** button)
- 2500SM (desk telephone used with 4A speakerphone)
- 2514 BMW (desk telephone with built-in headset jack)
- 2526BMG (outdoor telephone used with waterproof enclosure)
- 7101A* (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)
- CS6402U0IA* (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)
- 500MM, 554BMPA, 500SM (rotary dial)

* Vintage telephone; no longer available for sale or lease

Adjuncts

Adjuncts are pieces of equipment that connect directly to the control unit or to a telephone through an adapter (see Adapters). Answering machines, credit card verification terminals, and alerts are examples of adjuncts.

Adapters

Adapters enable the connection of equipment or, in the case of a channel service unit (CSU), of Digital Signal 1 (DS1) facilities to the control unit. Some adapters connect directly to the control unit (system adapters) while others connect to telephones (telephone adapters).

- System adapters
  - — ESF T1 CSU
  - — 551 T1 L1 CSU
  - — Universal Paging Access Module (UPAM)
  - — Loop trunk adapter for paging
- Telephone adapters
  - — Multi-Function Module (MFM) for MLX telephones

    WARNING:
    The MFM can be installed or repaired only by a qualified technician or an authorized dealer representative. To eliminate the risk of electrical shock, the MLX telephone should not be disassembled.

  - — General Purpose Adapter (GPA) for analog multiline telephones

    7500B Data Module for connecting digital data equipment either directly to the control unit or to an MLX telephone (for simultaneous voice and data transmission)

  - — 7500B Data Module for connecting digital data equipment modem for connecting digital equipment (such as a personal computer) to a tip/ring (T/R) interface

— Supplemental Alert Adapter (SAA) for connecting an alert (such as a horn or strobe) to an analog multiline telephone

Applications

The following applications for the system consist of software and/or hardware that add functions to the system. See the Applications chapter for details.

- MERLIN MAIL™ Voice Messaging System
- MERLIN® Attendant
- Call Accounting System (CAS)
- Call Accounting Terminal (CAT)
- Call Management System (CMS)
- InnManager™ Guest Management System
- System Programming and Maintenance (SPM)
- Integrated Solution II (IS II)
  - — 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)
  - — Integrated Voice Power Automated Attendant (IVP AA—IS III)
  - — AUDIX Voice Power—IS III (AVP—IS III)
  - – Call Accounting System–IS III (CAS-IS III)
  - — System Programming and Maintenance—IS III (SPM–IS III)
  - — FAX Attendant System™
- Primary Rate Interface (PRI) Applications
  - — Group IV (G4) Fax
  - — Video Conferencing
- Centrex service
- MERLIN PFC
- Automated Document Delivery System (ADDS)
- CONVERSANT® Intro

Figure 1-1 shows the components of the system. The numbered paragraphs following the figure correspond to the numbered items in the figure.

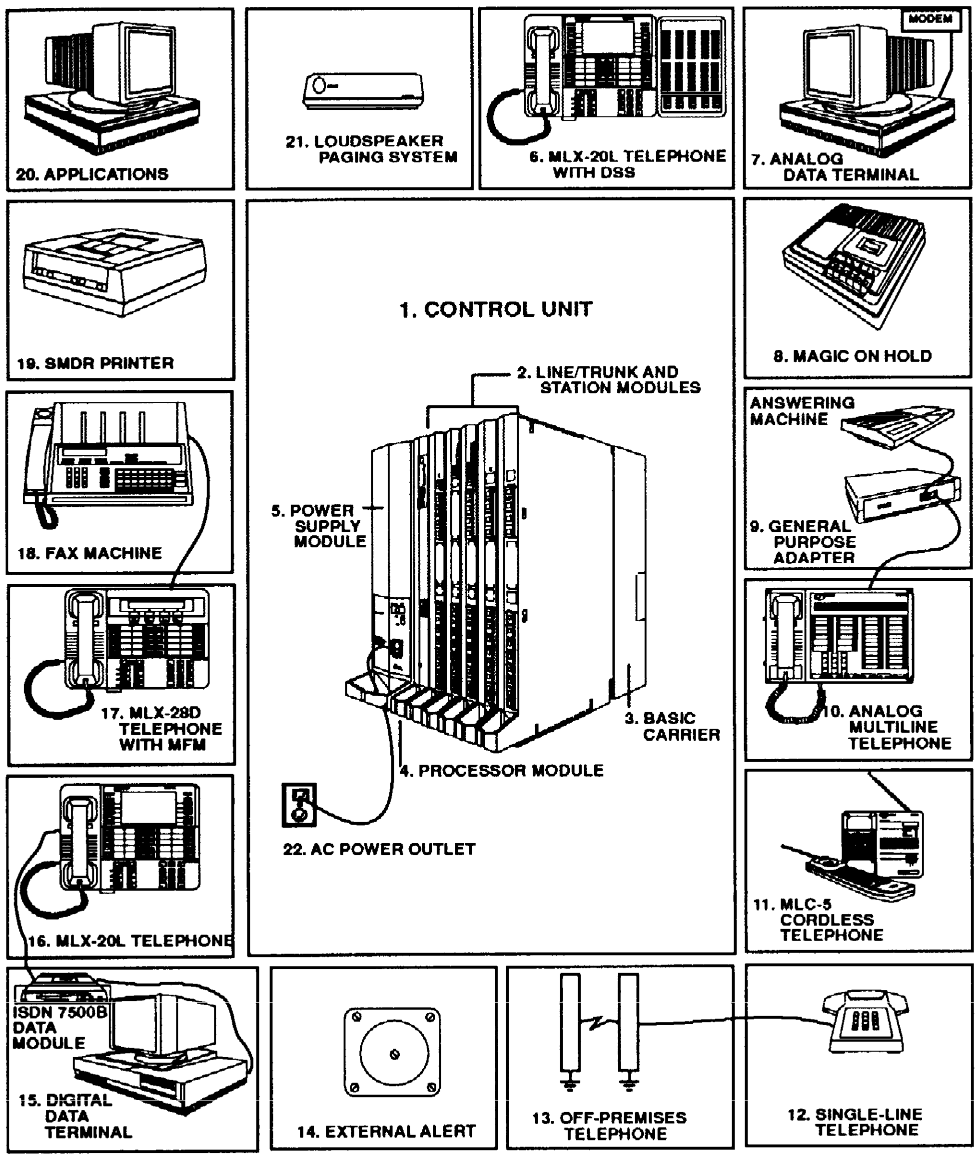


Figure 1-1. System Components

1. **Control Unit.** The backbone of the system, consisting of the basic and expansion carriers, power supply module, processor module, and line/trunk and station modules. The control unit connects telephone company lines/trunks with stations such as telephones and adjuncts.
2. **Line/Trunk and Station Modules.** The components that connect telephone company lines/trunks and terminal equipment such as telephones, external alerts, and fax machines via jacks to the control unit.
3. **Carrier (Basic).** The component attached to the backboard used to hold the modules needed for system operation. The basic carrier houses the processor module, power supply module, and up to five line/trunk and station modules. Each expansion carrier houses its own power supply and up to six additional line/trunk and/or station modules. One or two expansion carriers can be added.
4. **Processor Module.** A miniature computer that controls most of the system's features, and supplies the system's diagnostics. The processor provides two jacks, one for Station Message Detail Recording (SMDR) and the other for system programming and maintenance via a personal computer (PC).
5. **Power Supply Module.** The component that supplies DC power for the modules and telephones (one power supply unit is needed per carrier). If the system's power requirements exceed the capacity of the power supply, an auxiliary power supply unit can be added.
6. **Direct Station Selector (DSS).** A console that adds 50 buttons for one-touch extension dialing to the MLX-20L or MLX-28D telephone and speeds call handling.
7. **Analog Data Station.** A data terminal such as a PC, printer, or optical reader that connects, via a modem (for transmitting and receiving analog signals), to a 012 basic telephone module or a 008 off-premises telephone (OPT) module. A data terminal can also be connected to an MLX telephone using an MFM or to an analog multiline telephone using a GPA.
8. **Magic On Hold®.** Optional equipment that connects to the system through a ground-start/loop-start (GS/LS) jack programmed for Music-on-Hold. (A customer-provided music source can be connected instead of Magic On Hold.)
9. **General Purpose Adapter** (GPA). An adapter used to connect a variety of tip/ring (T/R) adjuncts to an analog multiline telephone (shown here with an answering machine).
10. **Analog Multiline Telephone.** A 34-button telephone with built-in speakerphone that connects to the system via an analog station jack. Other analog multiline telephones compatible with the system include the 22- and 34-button with built-in speakerphone and a l-line, 16-character display, and the 10- and 22-button with built-in speakerphone, without display. In addition, the following vintage membrane telephones (no

longer available for sale or lease) are compatible with the system: 5-button, 10-button, 34-button, and 34-button Deluxe.

11. **MLC-5 Cordless Telephone.** A cordless multiline telephone that connects to the control unit via an analog station jack.
12. **Industry-Standard Single-Line Telephone.** A touch-tone or rotary industry-standard telephone connected to the system via a 012 basic telephone module, a 008 OPT module, or an MLX telephone via an MFM.
13. **Off-Premises Telephone** (OPT). A single-line, touch-tone or rotary, industry-standard telephone located in a different building from the control unit.
14. **External Alert.** Alerting devices such as balls, chimes, and strobe lights that connect to a jack on a 012 basic telephone module or a 008 OPT module, or to an MFM or SAA.
15. **Digital Data Station.** A data terminal such as a PC, printer, or optical reader that connects via a 7500B Data Module to a 008 MLX or 408 GS/LS-MLX module (Release 2.0 only) and that can also include an MLX telephone.
16. **MLX-20L Telephone.** An MLX telephone with 20 line buttons and a display with seven lines of 24 characters each. The MLX-20L telephone can be used as a system programming console. Other MLX telephones are as follows:

    — MLX-10 Telephone: a 10-button MLX telephone without a two-line, 24-character display

    — MLX-10D Telephone: a 10-button MLX telephone with a two-line 24-character display

17. **MLX-28D Telephone with Multi-Function Module.** An MLX telephone with 28 line buttons and a two-line, 24-character display.

    A Multi-Function Module (MFM) is a circuit board mounted inside an MLX telephone that provides a jack to connect optional equipment such as answering machines, fax machines, external alerts, and modems to the telephone.

18. **Fax.** Industry-standard fax machines connected to the control unit via a jack on a 012 basic telephone module or a 008 OPT module, an MFM, or a GPA.
19. **SMDR Printer.** A printer for SMDR call records, connected via a RS-232 jack on the processor.
20. **Applications.** Software and hardware for the system that can be connected to the control unit to provide more functions.
21. **Loudspeaker Paging.** A single-zone or multizone system such as PagePac® with Zonemate™ 9 or 39 that connects via an administered jack on a GS/LS module.

22. **AC Power Outlet.** A dedicated 115-VAC wall outlet (not controlled by an on/off switch) that supplies power to the control unit.

Functional Description

This section describes the functional units that make up the control unit, their relationships, and how signals are processed.

The control unit contains the following functional units:

- Processor module
- Power supply module
- Carrier
- Line/trunk and station modules

Functional Units

The functional units are the processor module, which controls the operation of the system and its features; the power supply module, which supplies power to the control unit; the carrier with its backplane assembly, which contains the input/output (I/0) bus and the time-division multiplex (TDM) bus; and the line/trunk and station modules, which connect the outside lines/trunks to the stations. All the modules are electrically connected to the backplane, which provides common circuitry for the I/O bus, the TDM bus, and power distribution,

The processor is connected to intelligent ports on the line/trunk and station modules, through the I/0 bus, by the digital switch element (DSE) on each line/trunk and station module. The TDM bus is also connected to the DSE of each line/trunk and station module. The two buses are illustrated in Figure 1-2.

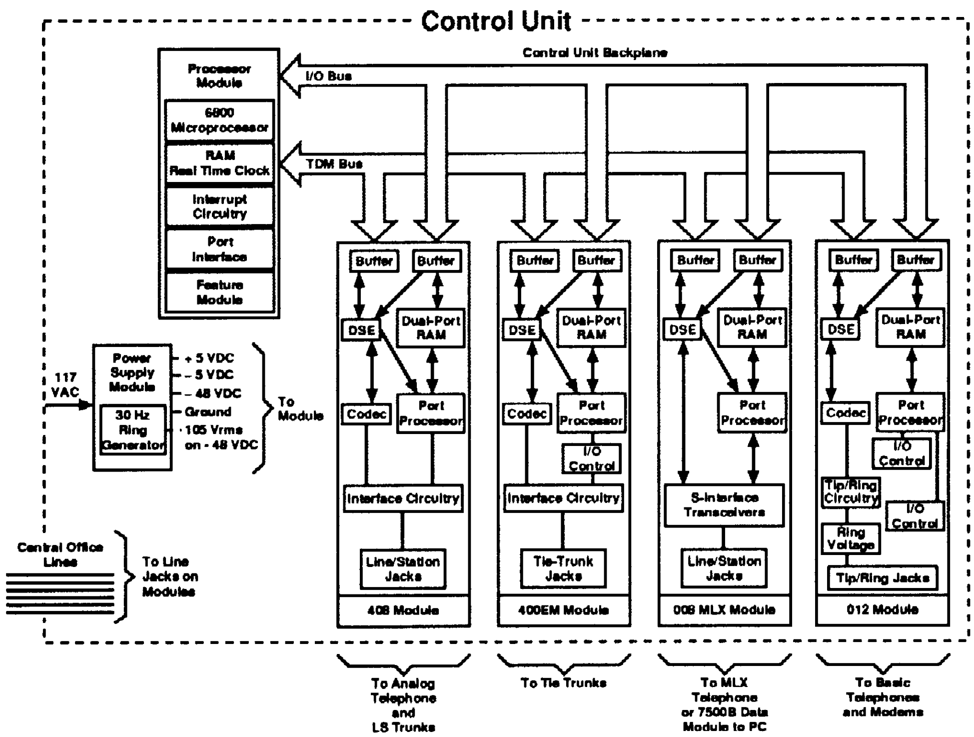


Figure 1-2. Functional Units

- **Input/Output Bus.** The I/O bus contains a 16-bit address bus and an 8-bit data bus. The address bus selects the module that receives instructions from the 68000 microprocessor in the processor module. The microprocessor provides instructions to the port processors and DSEs through the 8-bit data bus.
- **Time-Division Multiplex Bus.** 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, tones, and clocks. 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-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 up to 64 kbps.

Digital Switching

Because the system is internally a digital system in a world of both analog and digital devices, it must accurately translate analog signals, Doing this involves signal conversion and switching. Codecs provide analog-to-digital and digital-to-analog conversion. The 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, signals can be transmitted to one or several destinations and reconstructed at the original amplitude. The result is no signal loss during switching and transmission from one point 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 station modules.

The processor uses the DSE to specify time slots for various functions, For example, during a conversation between station A and station B, a time slot is reserved for station A to transmit on and for station B to receive on. For example, station A talks on time slot 150 and listens on time slot 160. Station B talks on time slot 160 and listens on time slot 150.

A digital tone plant in the processor module provides touch-tone and call-progress signals to stations via time slots 0 to 39. 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, the DSE ignores the data.

Each module has a DSE to interface codecs or digital transceivers to the TDM bus. The actual digital switching occurs when the DSE is programmed by the system I/O bus 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.

Modes of Operation

NOTE:
Although the terms “line” and “trunk” technically refer to different facilities, the differentiation is not as clear as it once was, and the terms are usually treated as if they are interchangeable. A “line” traditionally connects a piece of equipment to a switching system; for example, it connects your home telephone to the local telephone company. A “trunk” connects one switching system to another switching system; for example, it connects a communications system like this (except for facilities line personal lines that pass transparently through the system) to the local telephone company’s central office (CO). Since the industry trend seems to be toward using “trunk” to refer to either facility, this standard is used in this section.

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

- The types of outside trunks that can be connected to the system
- How telephone users access outside trunks
- The types of system operator consoles allowed
- The features available and how they work

The following sections describe each mode of operation, and Table 1-1 summarizes the modes.

Table 1-1. Modes of Operation Summary

	Key	Hybrid/PBX	Behind Switch
Trunks connected directly to control unit:			
■ Ground-start			
KF registration (FCC)	No	No	No
MF registration (FCC)	Yes	Yes	Yes
PF registration (FCC)	No	Yes	No
■ Loop-start	Yes	Yes	Yes
■ PRI	Yes	Yes	Yes
■ DSI	Yes	Yes	Yes

Continued on next page

Table 1-1. - *Continued*

	Key	Hybrid/PBX	Behind Switch
■ 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, that is, outside trunks assigned to buttons on telephone	Yes	Yes	Yes
Shared trunks	Yes (outside trunks only)	Yes (outside trunks and SA buttons)	Yes (outside trunks only)
Prime Lines	No	No	Yes
Queued Call Console (QCC)	No	Yes	No
Number of extensions:			
■ <5(3	Good	Good	Good
■ >50	Not recommended	Good	Good up to 80
FCC registration	KF or MF	MF or PF	KF or MF

Key Mode

A Key system is the simplest way to provide users 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 communications system operates in Key mode, telephones are programmed with two kinds of buttons:

- Line buttons (or keys) are associated with specific outside (telephone company) trunks. Line buttons allow users to see activity on other telephones, join conversations, and make and receive calls.
- **Intercom** buttons are used to make and receive internal calls.

The Key mode of operation accommodates the following kinds of outside trunks:

- Loop-start trunks, including basic lines, WATS, and foreign exchange (FX)
- Ground-start trunks (only if registered as MF and if not strapped for Key mode, as described below) and emulated ground-start trunks on T1 facilities
- DS1 facilities
- Tie trunks and emulated tie trunks 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 CO recognizes a call attempt and the time the call is processed. This delay is minimal and virtually unnoticeable. Most residence and small business telephones have loop-start trunks.

The communications system is configured for Key mode operation by system programming, depending on how the system is registered with the Federal Communications Commission (FCC), as described later in this section. If the system is modified for Key-only operation by the hardware strap in the processor module, no ground-start trunks can be connected to it. However, ground-start emulation on a T1 facility is allowed.

NOTE:

- The default programmed mode is Key.
- On initialization of a Release 1.0 system, all loop-start and ground-start trunk programming reverts to loop-start. In Releases 1.1 and 2.0, if the system is programmed for Key mode, the strap is checked on initialization. If the strap is in (Key-only operation), all trunks revert to loop-start. If the strap is not in, any programmed designation of ground-start trunks is retained.

The following features are not available in Key mode:

- Direct inward dialing (DID) trunks
- Trunk pools
- Automatic Route Selection (ARS)
- QCCs and associated features
- **System Access** buttons

Line Access

In Key mode (whether strapped and/or programmed), each outside line must be assigned to a line button on at least one telephone. As a result, the telephones most commonly used in Key mode are multibutton telephones. See Figure 1-3.

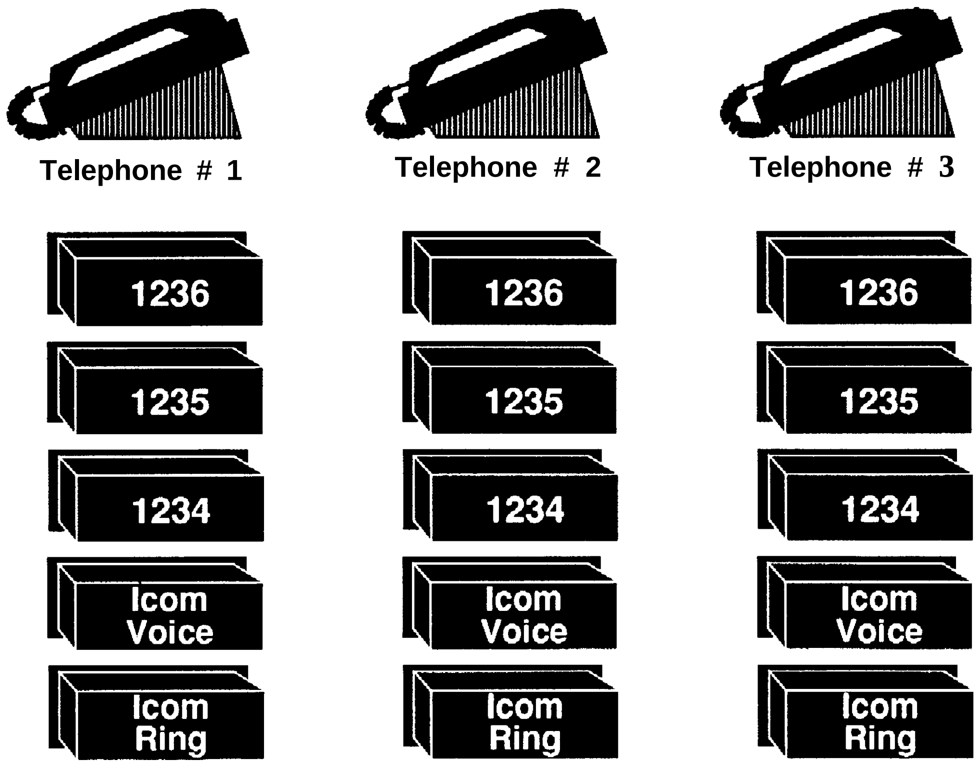


Figure 1-3. Lines Labeled for Key System Telephones

A user selects an outside line by pressing a personal line button-a button labeled with a telephone number. Upon hearing dial tone, the user can dial out. When a line is in use, the green LED goes on next to the corresponding personal line button on any telephones that share that personal line.

The telephones in a Key system also have Intercom buttons, labeled **ICOM,** that allow users to make and receive calls to and from other extensions within the system. An inside talk path is provided when an **ICOM** button is pressed. The user hears a system dial tone, which can be programmed to sound different from outside dial tone. The factory setting is to provide a different internal dial tone.

The following types of **ICOM** buttons can be used to make and receive inside calls in Key mode:

- An **ICOM Ring** button is used to make inside calls and to receive inside and outside calls transferred from another extension. When an ICOM Ring button is used to make an inside call, the telephone at the destination extension rings once per ring cycle to indicate an inside call.
- An **ICOM Voice** button is used to make inside calls and to receive inside and outside calls transferred from another extension. When an **ICOM Voice** button is used to make an inside call, the person at the destination extension hears the caller's voice on the speakerphone after a beep, rather than ringing. (If he or she has a single-line telephone, does not have a speakerphone, or has disabled voice announcements, the telephone rings the same as if the call had been made on an **ICOM Ring** button.)
- An **ICOM Originate Only** button is used only to make inside calls. Neither inside nor outside calls are received on an **ICOM Originate Only** button. This type of button ensures that the user always has a button available to make or transfer a call, establish a conference call, answer a Call Waiting call, or pickup parked calls. The button can be programmed for either voice or ring operation.

A combination of up to 10 **ICOM Voice, ICOM Ring,** and **ICOM Originate Only** buttons can be assigned to each telephone on buttons 1 through 10. The number of personal 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. See *System Planning* for button diagrams.

Key System Configurations

In Key mode, the system can be configured as a square, modified, or hybrid Key system, as described in the following sections. (The communications system does not distinguish among these configurations.)

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 trunk in the system terminates on a personal line button on every telephone in the system.

The system illustrated in Figure 1-3 is a square Key system in an office with three outside trunks and three telephones. Each of the three trunks is assigned to the same button on each telephone. When a trunk is assigned to a personal line button on more than one telephone, it is considered a shared line.

Modified Key System

A Key system can be modified through system programming to provide trunk access for special business needs. For example, some business do not require every user to have access to a tie trunk, so the system can be programmed so that some telephones do not have access to all outside trunks.

Figure 1-4 shows an example of personal line button assignments in a modified Key system with two outside trunks and one tie trunk. Each trunk is not assigned to a button on every telephone.

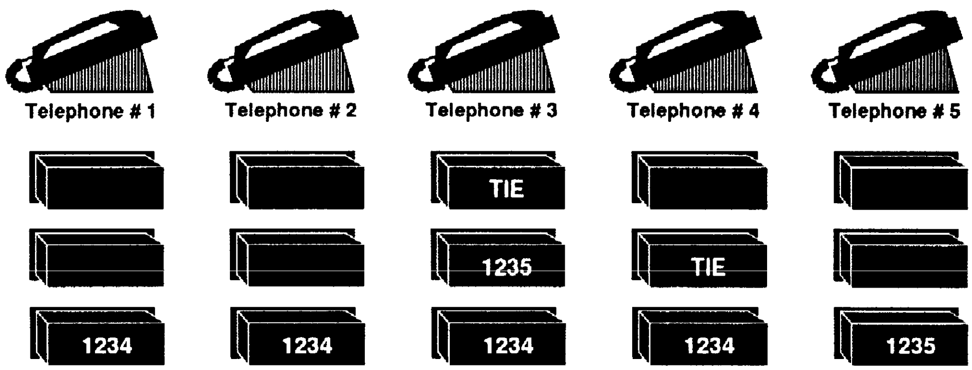


Figure 1-4. Lines Labeled for Modified Key System Telephones

Hybrid Key System

A hybrid Key system allows ground-start trunks to be connected directly to the system's control unit. To program ground-start trunks in Key mode, the system may **not** be strapped for Key-only operation and must be registered with the FCC under an MF classification, as described later in this chapter, in the section FCC Registration. In this configuration, all outside trunks, including ground-start trunks, are assigned to personal line buttons on each telephone.

Key Mode Considerations

- The multibutton telephones most commonly used in a Key system provide easy access to outside trunks. To get a dial tone, the user simply lifts the handset and an outside line is automatically selected.
- The intercom path helps ensure that outside trunks are available when needed.

- Key mode has the flexibility to provide trunk access according to user needs. For example, tie trunks can be terminated on the telephones of only those users who need them.
- The loop-start trunks traditionally associated with Key mode operation can cost less than the trunks used in the other modes.
- Key systems are best suited to smaller businesses.
- To take advantage of the features and functionality of the system, all users should have multibutton telephones when the system is operating in Key mode.
- If the number of trunks connected to the system is larger than the number of buttons available on the DLC, Hybrid/PBX mode, which offers the QCC, may be more functional.
- To make more efficient use of outside trunks by grouping them into pools for shared use, or to use ARS, the system must be programmed to operate for Hybrid/PBX operation.

Hybrid/PBX Mode

A private branch exchange (PBX) originally was a large switchboard installed at a customer's office that functioned like a small, self-contained telephone company. The switchboard was manually operated, and the system operators physically connected calls by plugging cords into the board's jacks. Today's PBX is a processor in the communication system control unit programmed to connect both inside and outside calls on a single 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, the communications system operating as a PBX still requires the user to request an outside trunk. A user simply dials a dial-out code (usually a 9) and the telephone number on an **SA** button, and the system routes the call to an available outside trunk.

Thus, the major distinction of Hybrid/PBX mode, is that both inside and outside calls can be made on the same button.

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

- Loop-start trunks, including basic lines, WATS, and FX
- Ground-start trunks, including basic lines, WATS, and FX
- DS1 facilities
- Tie trunks and emulated tie trunks on T1 facilities
- DID trunks

Programming the system for Hybrid/PBX mode automatically arranges the outside trunks in functional groups, or pools, within the control unit. (See Figure 1-5.) 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 a pool is assigned to a line button during system programming, it is called a *pool button.* Users request specific trunk pools by dialing the trunk pool number (870-879) for the pool or by pressing a pool button, which gives one-touch access to a group of trunks.

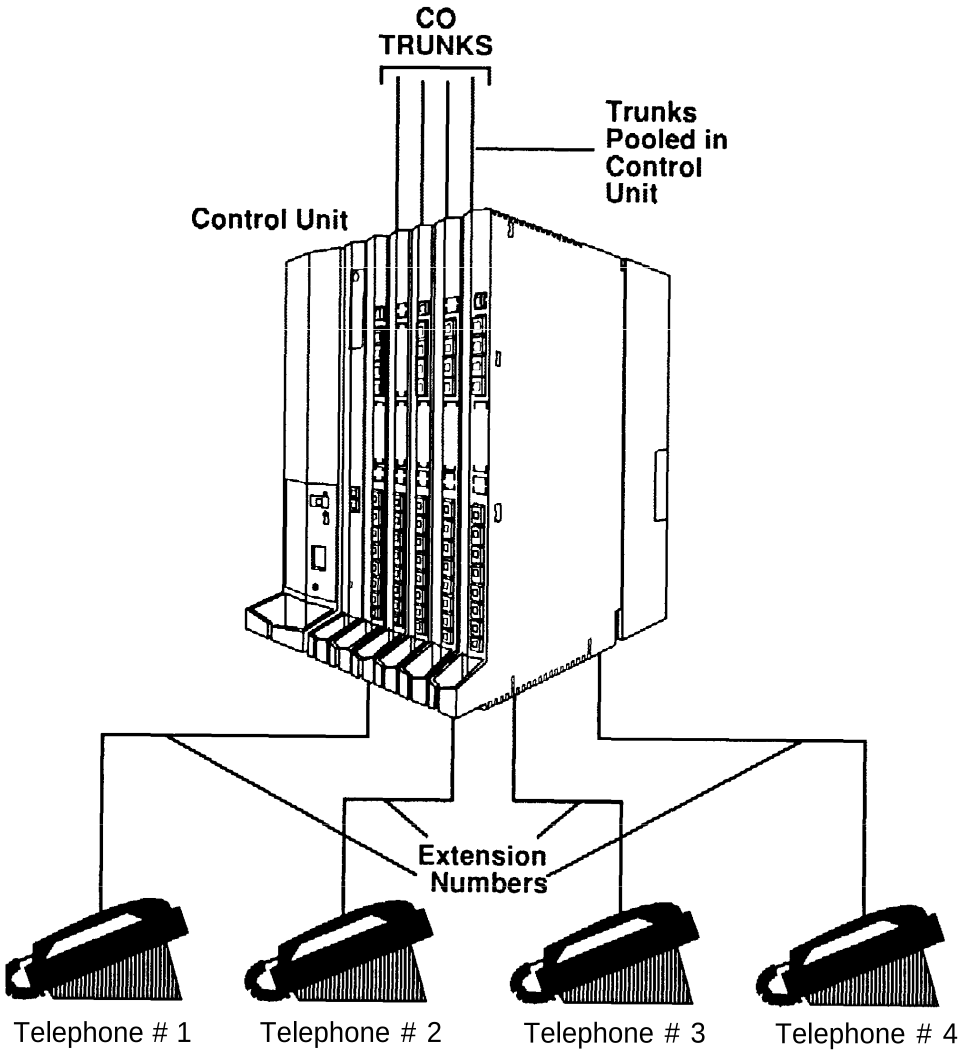


Figure 1-5. Hybrid/PBX Mode

Outside calls are normally answered by a system operator and transferred to individual system users.

A feature commonly used in Hybrid/PBX mode is ARS. When an **SA** button is used to make an outside call and the ARS dial-out code is entered, the system selects the next available trunk from the type of pool that is most cost-effective for the call and gives the user access to that trunk.

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

- Loop-start trunks, by default, are assigned to pool number 70. This pool is called the *loop-start pool* or *main pool.*
- Ground-start trunks, by default, are assigned to pool number 890. This pool is called the *ground-start pool.*

  NOTE:
  On initialization of a Release 1.0 system, all Imp-start and ground-start trunk programming reverts to Imp-start. The ground-start pool never has trunks assigned to it automatically, but must be programmed after the ground-start ports are designated. In Releases 1.1 and 2.0, ground-start trunks are assigned to the ground-start pool on initialization, except in a system strapped for Key mode operation.

- Tie trunks, by default, are assigned to pool number 891. This pool is called the *tie pool.*

Through system programming, the three automatically assigned pools can be rearranged and special-function or special-user pools can be created. For example, the main pool can be divided and smaller pools of loop-start trunks can be assigned 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 different kinds of calls to be made from the same buttom--outside calls on basic loop-start or ground-start trunks or on tie trunks or special service facilities such as WATS, and inside calls to other extensions in the system.

The following types of buttons can be assigned to multiline telephone users:

- An **SA Ring** button is used to make and receive inside and outside calls. When an **SA Ring** button is used to make an inside call, the telephone at the destination extension rings once per cycle to indicate an inside call.
- An **SA Voice** button is used to make and receive inside and outside calls. When an **SA Voice** button is used to make an inside call, the person at the destination extension hears the caller’s voice on the speakerphone after a single beep, rather than ringing. (If the person at the destination extension has a single-line telephone, does not have a speakerphone, or has disabled voice announcements, the telephone rings the same as if the call had been made on an **SA Ring** button.)

- An **SA Originate Only** button is used only to make inside and outside calls. Neither inside nor outside calls are received on an **SA Originate Only** button. The purpose of this type of button is to ensure that the user always has 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, the button can be programmed for either voice or ring operation.
- A **Shared SA** button is used 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 designated as the principal (or primary) extension. This extension is the telephone from which **SA Ring, SA Voice,** and/or **SA Originate Only** buttons are assigned as **Shared SA** buttons on one or more telephones in the Shared System Access arrangement.

  Shared SA buttons are often provided to secretaries and their bosses, as well as to others who work closely together, such as a customer service department. For inside calls, the button can be programmed for either voice or ring operation.
- A pool button is used to make outside calls using a specific trunk pool, To make an outside call, the user presses the appropriate pool button—no dial-out code is necessary.
- A personal line button is used to dedicate an outside trunk for use by one or more telephones in the system. The personal line button is used to make and receive only outside calls. To make a call, the user presses the appropriate personal line button—no dial-out code is necessary.

A combination of up to 10 **SA Voice, SA Ring, SA Originate Only,** and **Shared SA** buttons can be assigned to each telephone (except for the QCC) on buttons 1 through 10. See System Planning for button diagrams, The number of personal 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.

Queued Call Console

The type of system operator position typically used in Hybrid/PBX mode is the 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 system control unit exceeds the number of buttons available on a DLC.

A call is held in queue until the system operator is available, After the call is delivered to a Call button on the QCC, the operator transfers it to the desired internal destination. The operator can switch between the caller and the called person to screen calls.

Hybrid/PBX Mode Considerations

- Hybrid/PBX mode provides the most efficient use of outside trunks since they can be pooled and are more readily available to users. The ARS feature can be programmed for more cost-effective use of pools.
- Hybrid/PBX mode provides greater functionality for single-line telephones than other modes of operation. The telephone user can make both inside and outside calls by accessing a pool of trunks.
- 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

The system operates in the Behind Switch mode when the control unit is connected to (is “behind”) another system. The other system is referred to as the host, and can be either a PBX or Centrex service (a telephone company service that provides PBX-like capabilities but is housed at the CO).

Figure 1-6 illustrates a very simple Behind Switch configuration in which the outside trunks are connected to the host system. The lines connecting the two control units are like extensions that provide users with access to the host system’s trunks, as if via a trunk pool.

Each extension number from the host system is assigned to individual telephones as a Prime Line. This line rings when the user receives an outside call and the user is connected to this line (even when the line is in use) unless the user manually selects a different line.

Each Prime Line can be assigned to additional telephones as secondary lines so that users can see activity on other telephones sharing the button and join their co-worker’s conversations.

All Prime Lines are also automatically assigned 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 allows the system operator to answer calls received by other users on their Prime Lines.

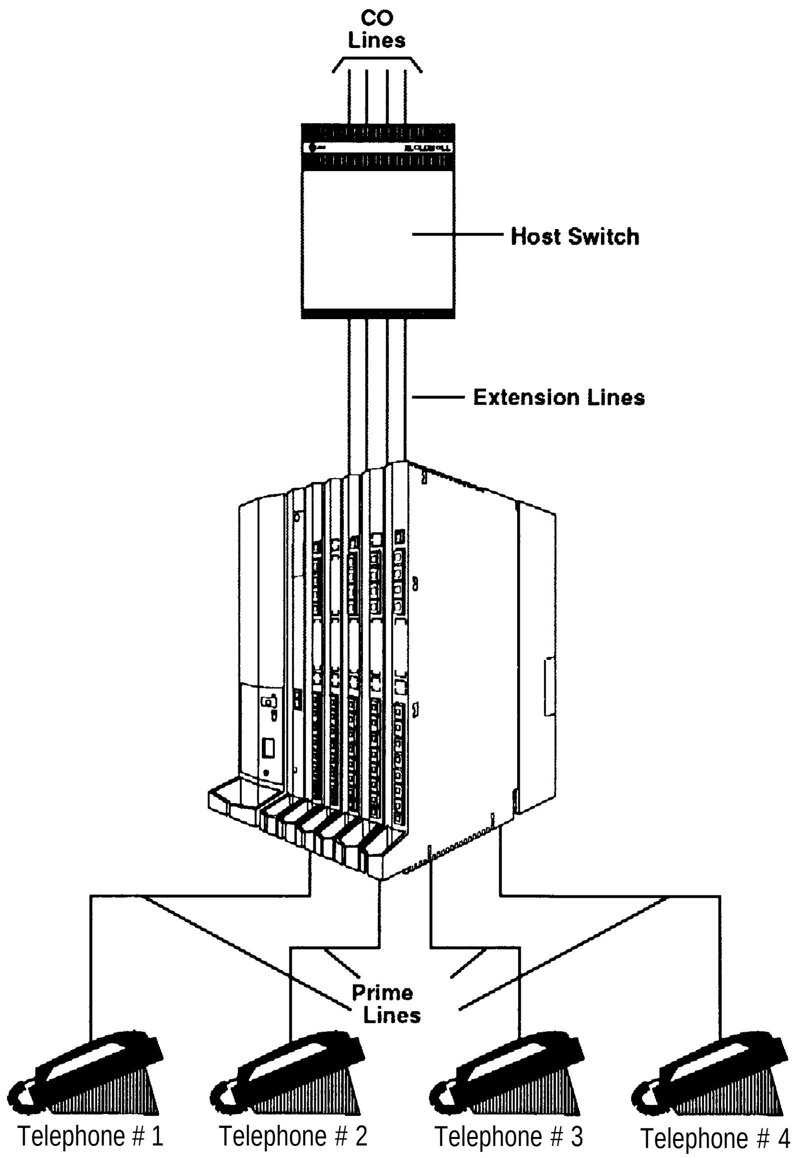


Figure 1-6. Behind Switch Mode

Modified Configurations

In addition to accessing the host's outside trunks, the Behind Switch system can be modified 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 this system's users. Figure 1-7 illustrates this configuration,

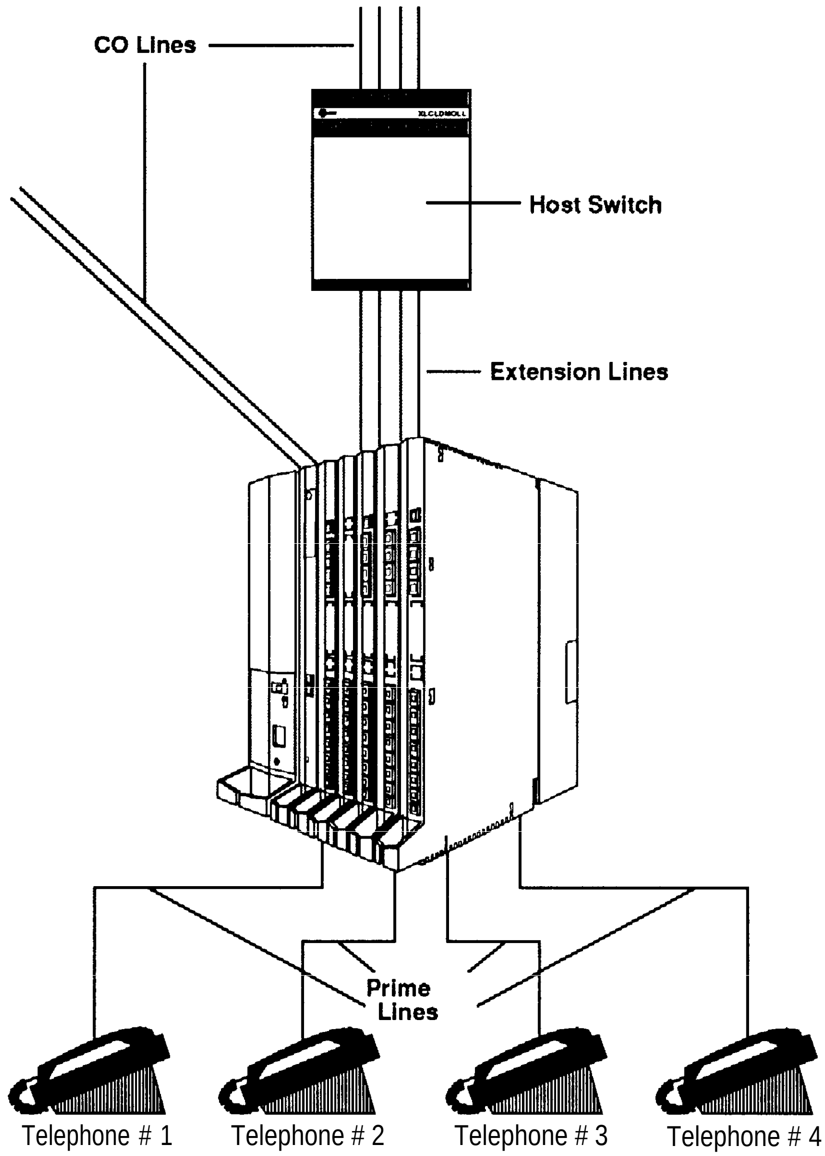


Figure 1-7. Behind Switch Mode with Direct Outside Trunks

Depending on business needs, the following kinds of direct outside trunks can be added:

- Loop-start trunks, including basic lines, WATS, and FX
- DS1 facilities

- Tie trunks
- Ground-start trunks (only if *not* registered as KF and *not* strapped for Key mode)

The direct outside trunks must be terminated on individual telephones and must appear on the telephones' line buttons. For example, if tie trunks are assigned to buttons on telephones 1 and 2, the buttons on those telephones appear as shown in Figure 1-8.

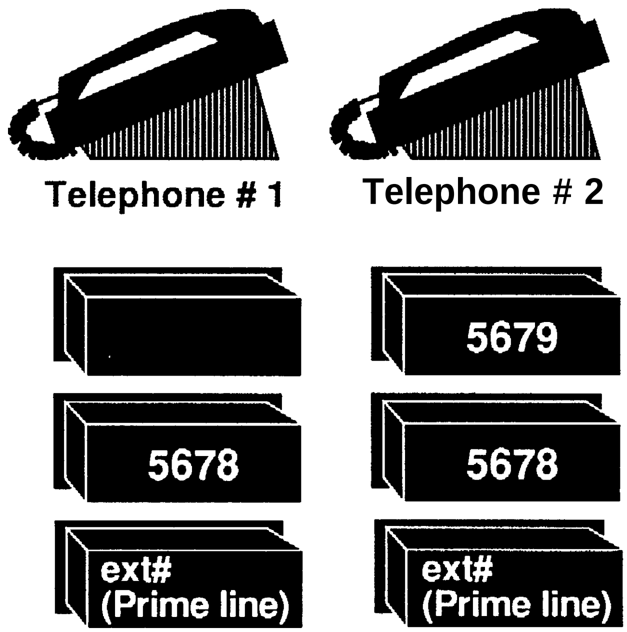


Figure 1-8. Labeled Line Buttons for Behind Switch Telephones

The bottom 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 1-8, 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
- ARS
- QCC
- **SA** buttons

Line Access

In Behind Switch mode, the telephones used most commonly are multibutton telephones. Single-line telephones can also be used and can be set up in two ways. In one configuration, the user has constant access to the Prime Line buttons, but cannot make intercom calls or use system features. In the second configuration, the user can make and receive intercom and Prime Line calls and can use system features, but must dial a dial-out code to make an outside call.

When the system is programmed for Behind Switch mode, the system assigns a single Prime Line, an **ICOM Ring** button, and an **ICOM Voice** button to each multiline telephone. Two **ICOM Ring** buttons are assigned to each single-line telephone; no Prime Lines are assigned. When the telephone handset is lifted, the Prime Line is selected automatically (even when it is busy) unless the user has first selected a different button. The Prime Line connects only with the host system, not directly with an outside trunk.

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

In a Behind Switch system, **ICOM** buttons allow users to call other people connected to the system. When a user presses an **ICOM** button, the system provides an inside talk path and dial tone from the Behind Switch system (not from the host system). The user can then reach co-workers without tying up a Prime Line.

The following types of **ICOM** buttons can be used to make and receive inside calls in Behind Switch mode:

- An **ICOM Ring** button is used to make inside calls and to receive inside calls and outside calls transferred from another extension. When an **ICOM Ring** button is used to make an inside call, the telephone at the destination extension rings with a one-burst ring to indicate an inside call.
- An **ICOM Voice** button is used to make inside calls and to receive inside calls and outside calls transferred from another extension. When an **ICOM Voice** button is used to make an inside call, the person at the destination extension hears the caller's voice on the speakerphone after a single beep, rather than ringing. (If he or she has a single-line telephone, does not have a speakerphone, or has disabled voice announcements, the telephone rings the same as if the call had been made on an **ICOM Ring** button.)
- An **ICOM Originate Only** button is used only to make inside calls. Neither inside nor outside calls are received on an **ICOM Originate Only** button. The purpose of this type of button is to ensure that the user always has a button available to make or transfer a call, establish a conference call, answer a Call Waiting call, or pick up parked calls. The button can be programmed for either voice or ring operation.

A combination of up to 10 **ICOM Voice, ICOM Ring** , and **ICOM Originate Only** buttons can be assigned to each telephone, except for single-line telephones, 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.

In Behind Switch mode, users have access to the special features of both the Behind Switch system and the host system. When both systems have common features, the customer must decide which system will be used for those features through system programming, A *fixed* Conference, Drop, or Transfer button automatically activates that feature on the host system. A user can program an unlabeled button for any of those features to access that feature on the Behind Switch system. Each system must be programmed accordingly, and the users must be given the appropriate access instructions.

When users press a fixed Conference, Drop, or **Transfer** button, the respective host features are activated: a timed switchhook flash is sent to the host, followed by a programmed feature access code. The person programming the system must obtain the feature access code for the host system and program it to the appropriate button.

NOTE:
A single-line telephone user has access only to the host system's Conference, Drop, and Transfer features.

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.

FCC Registration

The account representative or authorized dealer who planned the system's mode of operation provides the FCC registration number that the customer reports to the local telephone company. Depending on mode of operation and the hardware strap in the processor module, this number includes the letters KF, MF, or PF, loosely corresponding to "key function," "multi-function," or "PBX function, " respectively. (The FCC has no Behind Switch classification.)

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

Table 1-2 lists the registration number(s) used for each mode of operation.

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

The following guidelines are used to determine which classification is used.

KF Classification

The system's KF classification number is AS593M-72914-KF-E. This classification is applicable only to the Key and Behind Switch modes of operation. The system is registered under the KF classification if any of the following conditions are met.

Key Mode of Operation

- The system is strapped for Key-only operation.
- All outside trunks terminate on one or more telephones.
- All outside trunks are loop-start, tie, DS1 facilities, or ground-start emulation on DID (or a combination of these).
- No trunks are pooled.

Behind Switch Mode of Operation

- The system is strapped for Key-only operation.
- No outside trunks are connected directly to the control unit. The communications system accesses only trunks connected to the host switch.
- No ground-start trunks are connected directly to the control unit, except for ground-start emulation on DID.
- No trunks are pooled.

MF Classification

The system's MF classification number is AS593M-72682-MF-E. This classification is applicable to all three modes of operation—Key, Hybrid/PBX, and Behind Switch. The system is registered under the MF classification if any of the following conditions are met.

Key Mode of Operation

- One or more ground-start trunks are connected directly to the control unit. These trunks connect to a 400 GS/LS/TTR, 408 GS/LS, 800 GS/LS, or 408 GS/LS-MLX module (Release 2.0 only).

Hybrid/PBX Mode of Operation

- All outside trunks are pooled; no trunks are terminated directly on a telephone.
- The only directly terminated trunks are personal lines, not shared lines—only one telephone accesses the trunk.

Behind Switch Mode of Operation

- One or more ground-start trunks are connected directly to the control unit. These trunks connect to a 400 GS/LS/TTR, 408 GS/LS, 800 GS/LS, or 408 GS/LS-MLX module (Release 2.0 only), The processor module must not be strapped for Key-only operation.

PF Classification

The system's PF classification number is AS5USA-65646-PF-E. This classification is applicable only to the Hybrid/PBX mode of operation. The system is registered under the PF classification if either of the following conditions is met.

Hybrid/PBX Mode of Operation

- All outside trunks are pooled; no trunks are terminated directly on a telephone.
- The only directly terminated trunks are personal lines, not shared lines—only one telephone accesses the trunk.

Programming

System Programming

The system can be programmed with options for the following:

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

The system can be programmed by using one of the following:

- An MLX-20L telephone connected to one of the first five ports on the first MLX module in the control unit
- The built-in modem in the processor, which permits remote programming via the public network. For example, support personnel can access the system by using a PC with a modem and with SPM software; support personnel call the system and enter a password to gain access. The system must be programmed for Remote Access.
- A PC with SPM software connected to the lower RS-232 port on the processor

The programming options are accessed from display screen menus. For more information, see *System Programming.* To use SPM to program your system on a personal computer, you need the SPM diskette and an PC with version 3.3 (or a later version) of MS-DOS® .

Your PC should include the following:

- At least 640 kbytes of random access memory (RAM)
- A floppy disk drive that will accommodate the SPM diskette
- A serial port that can use either a DB-9 or DB-25 connector

  NOTE:
  For a DB-9 connector, use a 9-pin to 25-pin adapter to convert the 25-pin connector to a modular connector.

- Either a 355AF modular adapter (if a male connector is on the interface cable) or 355A modular adapter (if the connector is female)

- A 4-pair modular cord (D8W)

The monitor can be either monochrome or color.

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

NOTE:
SPM uses Interrupt 4 and I/O address 3F8 for COM1. It uses Interrupt 3 and I/O address 2F8 for COM2.

When you use a PC with SPM to program the system, the maximum number of MLX-20L telephones that can be connected to the system is reduced by one.

Telephone Programming

There are two kinds of telephone programming: Centralized Telephone Programming and Extension Programming.

Centralized Telephone Programming

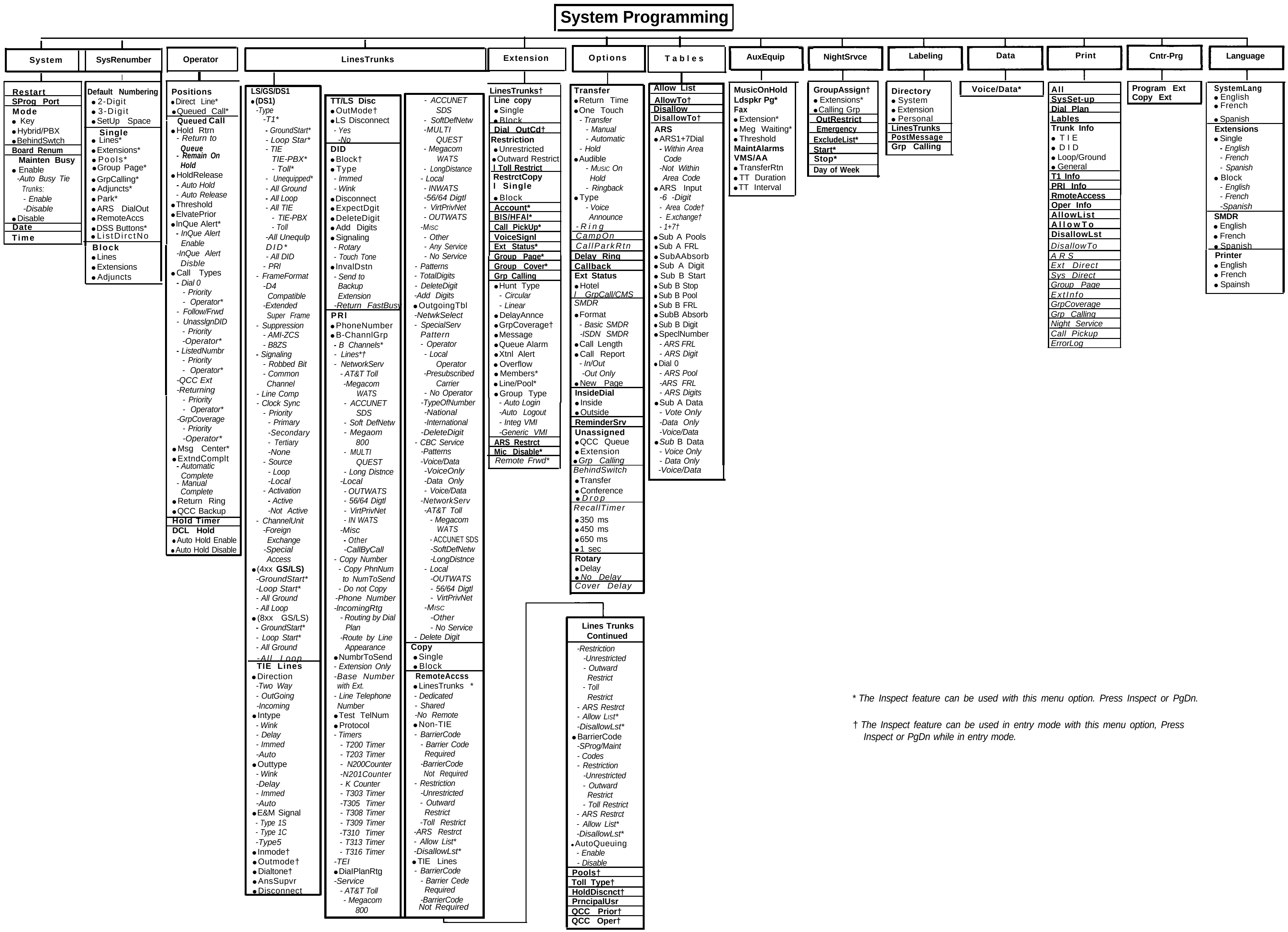
Centralized Telephone Programming is an option you can choose from the System Programming menu to program any feature onto a telephone. Although many features can also be programmed by individual telephone users Or system operators, the following features can be programmed *only* by centralized telephone programming (and not by individual users):

- Barge-In
- Headset Hang Up
- **ICOM** buttons—all types (Key and Behind Switch only)
- System Access buttons—all types (Hybrid/PBX 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 buttons on the telephone. In addition, many other features can be programmed on both multiline telephones and single-line telephones that do not require button assignment, such as Call Waiting.

Users can program theier telephones by dialing programming codes or, on MLX display telephones, selecting features from the display. When a telephone is in the program mode, the system considers it busy; therefore, no incoming calls ring at the telephone until it is back in the call-handling mode.



* The Inspect feature can be used with this menu option. Press Inspect or PgDn.

† The Inspect feature can be used in entry mode with this menu option, Press Inspect or PgDn while in entry mode.

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

Capacities

The system can be arranged as a stand-alone system 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 station endpoints that support a combination of the following:
  - — Up to 144 physical station jacks for telephones and adjuncts
  - — Up to 127 logical digital data ports (via 7500B Data Modules connected to jacks on the MLX module) providing RS-232 connections to data terminals and personal or multiport computers
- System call-handling capability of 3888 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 stations); however, each MLX module station jack supports two logical endpoints (station devices that can operate simultaneously and independently). For example, an MLX telephone with a Multi-Function Module (MFM) plugs into only one station jack, but the jack supports the telephone and the equipment connected to the MFM (such as a fax machine or an answering machine).

In a similar way, although the 100D module has only one jack, it can serve up to 24 endpoints (emulated lines/trunks or PRI lines/trunks).

Thus, the entire system can be configured to connect up to 80 lines/trunks and 255 station endpoints, a total of 335 endpoints.

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

The following table lists the hardware and software capacities of the system.

Table 1-3, Hardware and Software Capacities

	Limit	Constraining Factor
Allowed Lists		
■ Number of lists	8	
■ Entries per list	10	
■ Digits per entry	7	
Automatic Route Selection		
■ Number of ARS patterns	18	
■ Subpatterns per pattern	2	
■ Routes per subpattern	6	
■ Number of fully programmable ARS tables	16	
■ 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	QCCs cannot be members
■ 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	First slot of basic earner used for processor module
■ Line/trunk and station module slots per basic earner	5	
■ Line/trunk and station module slots per expansion carrier	6	
■ Maximum slots available for line/trunk and station modules	17	
Coverage groups		
■ Number of groups	30	
■ Senders per group	144	QCCs cannot be senders
■ Groups per sender	1	
■ Receiver buttons per group	8	
■ Groups per QCC receiver	30	
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		
■ Number of directories	1	
■ Listings per directory	130	
■ Extension Directory		
■ Number of directories	1	

Continued on next page

Table 1-3. - *Continued*

	Limit	Constraining Factor
■ Listings per directory	144	
■ Personal Directory (MLX-20L only)		
■ Number of directories	48	
■ Listings per directory	50	
Disallowed Lists		
■ Number of lists	8	
■ Entries per list	10	
■ Digits per entry	11	
100D module (maximum 2 per carrier)	3	
Endpoints (devices)	255	
Fax machines with message-waiting	16*	
Lines/Trunks	80	Software real-time limit
Night Service		
■ Groups	8	
■ Members per group (including one group calling number)	144	
■ Groups per member	8	
■ Emergency Allowed List entries	10	
System operator consoles		
■ **DCLs**		
■ MLX-20L or MLX-28D	8	2 per MLX module
■ BIS-22D, BIS-34, BIS-34D, or MERLIN II System Display Console	8	2 per analog module
■ QCCs	4	2 per MLX module
■ DSSs	16	2 per MLX module (built into MERLIN II System Display Console)
■ Combination of DLCs and QCCs	8	
■ Number of consoles per module	2	
Park codes		
■ Number of codes	8	
Personal lines	64	
Pickup		
■ Number of groups	30	
■ Members per group	15	
■ Groups per member	1	
Pools (trunk groups)		
■ Maximum number of pools	11	
■ Maximum number of trunks in a pool	80	
Pool buttons	64	

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

Continued on next page

Table 1-3. - *Continued*

	Limit	Constraining Factor
Ports (not achievable simultaneously)		
■ Total	224	Software real-time limit
■ Voice and data (physical ports)	144	Software real-time limit
■ Voice Announce to Busy stations	127	RAM limit
■ Voice-mail interface	20*	
■ Digital data via 7500B Data Module	127	RAM limit
■ Paging	3	Software real-time limits, loop-start only
■ Delay announcements	32	Software real-time limits
Remote Access		
■ Number of barrier codes	16	
■ Digits per code	4	
Shared System Access buttons		
■ Number of buttons per principal station	16	
Speed Dial		
■ Personal Speed Dial		Single-line and 5- or 10-button telephones only
■ Entries per telephone	24	
■ Entries per system	1200	
■ Digits per entry	28	
■ System Speed Dial		
■ Entries per system	130	
■ Digits per entry	40	
Stations		
■ Total physical jacks	144	
■ Total endpoints	255	
System programming equipment		
■ MLX-20L	1	Remote access overrides on-site
■ RS-232 jack (for connection of PC with SPM)	1	programming except during backup
■ Modem (built-in processor module)	1	or restore
Telephones (not achievable simultaneously)		
■ Single-line	144	RAM limit
■ Analog multiline		
Without Voice Announce to Busy	136	17 slots x 8 ports/board
With Voice Announce to Busy	68	
■ MLX-20L	48†	RAM limit
■ All other MLX telephones (with or without 7500B Data Module or MFM)	127	RAM limit
■ Power failure transfer	20	1 per 4 LS/GS trunk jacks
Traffic (hundred call seconds/hr/system)	3888 ccs/hr.	Assuming 20% internal traffic
Two-party conversations	108	216 time slots
Voice-mail systems	1	

‘ 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 stations in a calling group is 20.

† Total includes the MLX-20L telephone used for system programming.

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 such machinery that produces electrical interference.

The control unit is mounted on a customer-provided plywood backboard. The backboard should be wide enough to accommodate additional carriers if system growth is anticipated. Allow enough space on either side of the control unit for any necessary wiring fields.

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	45lb (20.4kg) 14" W x 23" H x 12"D (35.6 cm x 58.4 cm x 30.5 cm)
Fully loaded 2-carrier system (basic carrier plus one expansion carrier)	90 lb (40.8 kg) 25" W x 23" H x 12"D (63.5 cm x 58.4 cm x 30.5 cm)
Fully loaded 3-carrier system (basic carrier plus two expansion carriers)	135lb (61.2 kg) 37" W x 23" H x 12"D (94 cm x 58.4 cm x 30.5 cm)
Mean Time Between Failures (mean or average time the system is expected to operate before any type of failure occurs)	2.1 years
Mean Time Between Outages (mean or average time the system is expected to operate before a failure affecting more than 25% of extensions or lines for more than 15 seconds occurs)	4.0 years
Backboard (minimum needed)	
■ Without SYSTIMAX®	6' W x 3' H x 3/4" D(182.9 cm x 91.4 cm x 1.9 cm)
■ With SYSTIMAX	7' W x 4' H x 3/4" D(213.4 cm x 121.9 cm x 1.9 cm)

Continued on next page

Table 1-4. - *Continued*

Backboard Mounting Hardware Requirements		
Wood surface	Wood screws	
Concrete surface, brick, cinder block	Masonry anchors	
Plaster, plasterboard	Toggle bolts	
Sheet-metal surface	Sheet-metal screws	
■ Hardware should have a combined pullout force of 650 lb (294.8 kg).		
■ When mounting to sheet-metal walls, attach to structural members.		
Location		
■ Within 5 feet (1.5 meters) of dedicated AC power outlet (1 plug per carrier)		
■ Within 1000 cable feet (304.8 m) of telephones		
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	117VAC	60 Hz±5% 3A
2-carrier	117VAC	60 Hz±5% 6A
3-carrier	117VAC	60 Hz±5% 9A
Temperature/Humidity Range		
40°-104°F (4°-40°C)		
20%-80% relative humidity		
Ventilation		
1 inch (2.5 cm) on right and left sides		
Radio Frequency Interference (RFI) Tolerance 1 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 meters for a frequency of 1000 Hz).

CAUTION:

- *Do not use switch control on AC outlet for control unit.*
- *Use approved ground (AC receptacle for 3-prong plug).*
- *Do not install control unit outdoors.*
- *Do not place control unit near extreme heat (furnaces, heaters, attics, or direct sunlight).*
- *Do not expose 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 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 maybe 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 1/2-inch (30.4 cm) in diameter, and in direct contact with the earth for at least 10 feet (3 meters).

  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, the water pipe ground can be supplemented by one of the following types of grounds:

  — Other local metal underground systems or structures— local underground structures such as tanks and piping systems

  — Rod and pipe electrodes—a 5/8-inch (1.6-cm) solid rod or 3/4-inch (1.9-cm) conduit or pipe electrode driven to a minimum depth of 8 feet (244 cm)

  — Plate electrode—a minimum of 2 square feet (61 square cm) of metallic surface exposed to the exterior soil

- Concrete-encased ground--must be an electrode, consisting of one of the following:
  - — At least 20 feet (6.1 meters) of one or more steel reinforcing rods, each being at least 1/2-inch (1.27 cm) in diameter
  - — 20 feet (6.1 meters) of bare copper conductor not smaller than #4 AWG, encased in 2 inches (5 cm) of concrete.
  - — This electrode must be located within and near the bottom of a concrete foundation or footing that is in direct contact with the earth.
  - — Ground ring---consists of at least 20 feet (6. 1 meters) 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 feet (77 cm) below the earth's surface.

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

For most surge occurrences, the following standard grounding requirements provide adequate lightning and power 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 sixteen 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

  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.

- 800 DID modules

Before installation, unit load and auxiliary power requirements for a new system are computed by qualified service personnel or an authorized dealer, 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 power supply module provides 45 unit loads to each carrier. If the unit load requirement per carrier exceeds 45, an auxiliary power unit is needed to allow the carrier to support an additional 27 unit loads.

CAUTION:
Running the system with more than 45 unit loads per carrier may not appear to do harm. However, this can cause the system to malfunction, creating "NO Trouble Found" situations.

An auxiliary power uni 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.

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 Appendix B of *Installation.*

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 the 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

Release 1,1 includes all Release 1.0 functionality plus the following enhancements:

Language selection allows the system to be programmed for prompts, menus, and messages on MLX display telephones to appear in English, French, or Spanish. Each of the following can also be programmed for any of these languages, independently of the system language:

- Individual extensions with MLX telephones
- SPM
- System programming reports
- SMDR report headers

MLX-10D, MLX-20L, and MLX-28D display telephones and MLX-10 non-display telephones are available in three separate versions, with factory-imprinted buttons in English, Spanish, or French.

In addition, user and operator guides, quick reference cards, and telephone tray cards are available in all three languages.

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 Program Mode and Forced Idle tone

System operational enhancements include the following:

- Automatic selection of an SA button when Conference is invoked (in 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 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.

Additional equipment includes the 8102 and 8110 analog voice 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 backward compatible with Release 1.0.

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.)

The Release 1.1 enhancements are described in detail in *MERLIN LEGEND Communications System Release 1.1 Notes* (555-610-119).

Release 2.0 Enhancements

Release 2.0 includes all Release 1.1 functionality plus the following enhancements:

Programming enhancements include the following:

- Extension Copy is a new 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 programming.

- Integrated Administration provides a single interface through Integrated Solution Ill (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 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 force-idles only one extension. Only a few system-wide programming activities, such as setting the system mode and system renumbering, force-idle the entire system.

System operational enhancements include the following:

- Coverage VMS is a new feature that prevents incoming external 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 programming.
- A Night Service group can be programmed to include a calling group as a member. This allows a call that receives Night Service treatment to be queued when all Night Service group members are busy or unavailable.
- When AUDIX Voice Power sends a Leave Word Calling message 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 24 DID channels on a single DS1 trunk interface, instead of requiring 24 separate physical trunks.

- A telephone user can send a timed flash (switchhook flash) on a loop-start trunk call on a System Access (SA) button.

FAX Attendant System is a new 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 Answer 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 (Release 2.0 only) is a new module that combines four ports for ground-start or loop-start trunks and eight ports for MLX telephones on a single module in the control unit.

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
- Authorization Code handling for FTS2000
- Station ID (SID) as Calling Party Number for Automatic Number ID (ANI)

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

Hardware Components

2

This chapter describes the basic hardware required for the communications system. It includes the control unit, digital or MLX telephones, analog multiline telephones, single-line telephones, system operator consoles, adapters and adjuncts for system telephones, and power-related accessories.

Control Unit

The control unit connects telephone company lines with telephones and adjuncts such as answering machines and fax machines. The control unit includes the following components:

- Carriers
- Processor module (one per system)
- Power supply module (one per carrier)
- Line/trunk and station modules
- Covers

Carriers

The basic and expansion carriers each have seven slots to hold modules (see Figure 2-1). The basic carrier contains a power supply, the processor (slot 00), and line/trunk and station modules (slots 01-05).

Up to two expansion carriers can be added to the right side of the basic carrier to increase the capacity of the system. Like the basic carrier, the leftmost and widest slot of the expansion carrier holds the power supply; the remaining six slots hold the line/trunk and station modules.

Besides the slots, both basic and expansion carriers have a backplane with an input/output (I/O) bus that interfaces with the modules.

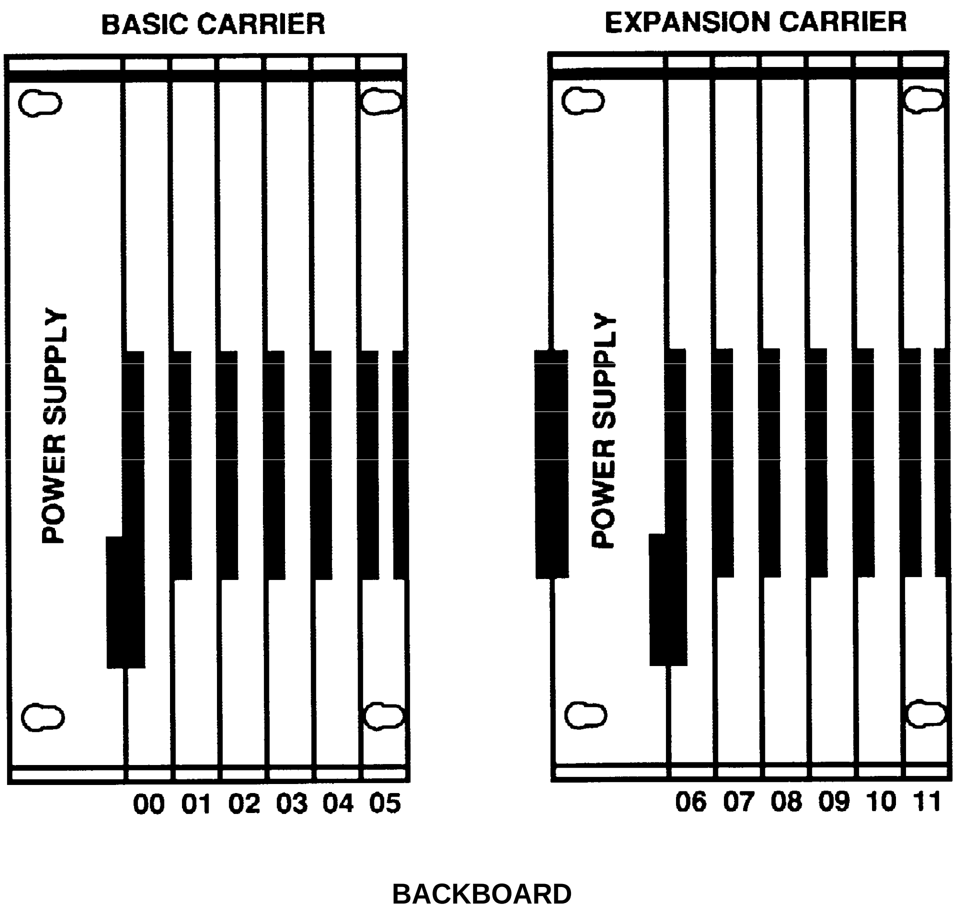


Figure 2-1. Carriers

Processor Module

Placed in slot 00 of the basic carrier, the processor module controls system features and programming, The main component of the processor module is the feature module, This component provides all the system's release-specific capabilities and features. All system programming is stored in the feature module in non-volatile memory.

The feature module plugs into the main board of the processor module, which contains the 66000 microprocessor, a built-in 1200 bits-per-second (bps) data modem, built-in diagnostics, RAM, a real-time clock, and interrupt circuitry, and interfaces to the other modules through the I/O bus on the carrier backplane.

The processor has two modular RS-232 jacks: one for Station Message Detail Recording (SMDR) and the other for system programming and maintenance via a personal computer (PC) with SPM software (see Figure 2-2).

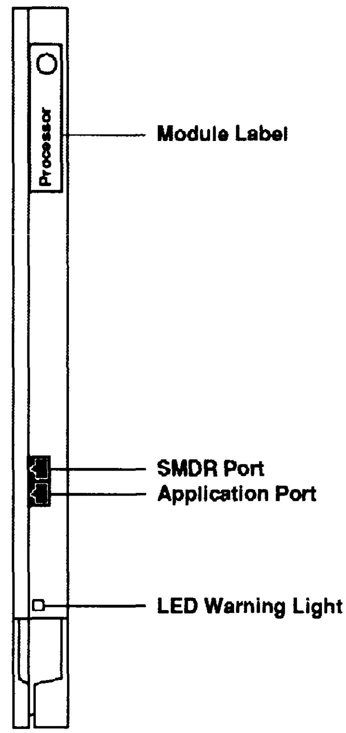


Figure 2-2, Processor Module

A NiCad battery in the processor provides backup power for the real-time clock and nonvolatile RAM in case of power failure or system shutdown. The battery provides RAM data retention for 12 to 30 days. The trickle-charge circuit can recharge the battery to 50 percent of capacity from a discharged state in 48 hours. The minimum battery life is five years.

Power Supply Module

The power supply provides power to the carrier, to each telephone, and to adjuncts—except for adjuncts such as answering machines and fax machines, which come with their own power supplies. Each carrier requires its own power supply module, which goes into the leftmost slot on each carrier.

The power supply converts 117-VAC line voltage to these outputs: +5 VDC, -5 VDC, and -48 VDC. All modules use +5 VDC and -5 VDC for logic circuits. Most line/trunk and station modules use -48 VDC for power to the stations. The Direct Inward Dialing (DID) and off-premises telephone (OPT) line/trunk and station modules also provide -48 VDC on the tip/ring (T/R) interface to the telephone company's central office (CC)) or OPT station. The 012 module basic telephone module provides 21 VDC to single-line telephones and equipment.

When the system contains a 012 or 008 OPT module, a 129B Frequency Generator (ring generator) must be installed in the power supply module of each carrier that houses one or more of these modules.

A green light-emitting diode (LED) on the power supply remains on as long as the module is receiving power. The power supply also has an on/off switch and a modular telephone jack for connecting an auxiliary power unit as needed (see Figure 2-3).

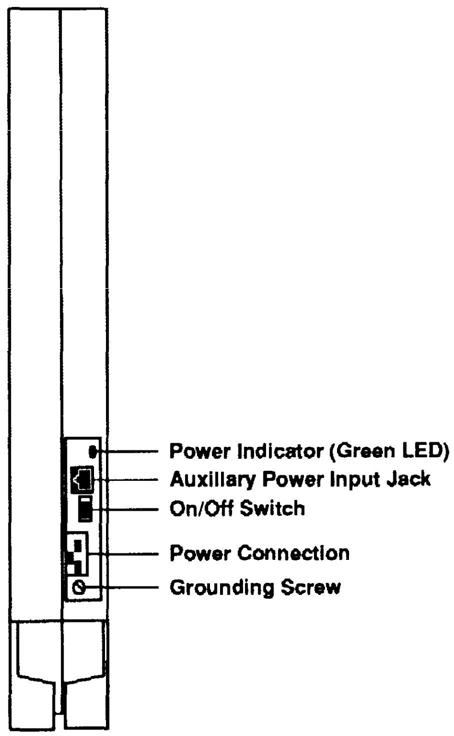


Figure 2-3. Power Supply

Line/Trunk and Station Modules

The line/trunk and station modules have jacks for connecting the telephone company lines/trunks and the station wires to the control unit. The station wiring connects to individual telephones and to adjuncts such as answering machines and fax machines.

A system with a basic carrier has five slots for modules. Up to two expansion carriers can be added, each one adding six slots for line/trunk or station modules.

Different line/trunk modules support different types of telephone company trunks. The types of trunks include the following:

- **Loop-start trunks—** (incoming and outgoing calls) the simplest and most common facilities in the nation-wide telephone network. They provide incoming and outgoing calls and are intended primarily for single-line telephones and older private branch exchanges (PBXS). A potential problem of loop-start trunks is glare, that is, picking up the telephone to make a call and another caller on an incoming call is already on the line. Glare occurs because of delays in the telephone company's ringing cycles and disconnects. This is normally not a problem for a residential, single-line telephone, but can be more serious and complex with an automated PBX.
- Ground-start trunks—(incoming and outgoing calls) specifically introduced to solve the problems that PBXs encounter on Imp-start trunks, They provide an immediate signal when the trunk is seized and when the call is completed and disconnected. Ground-start trunks can be used only if registered with the FCC. Depending on the mode of operation and the hardware strap in the processor module, the registration number (provided by the account representative or authorized dealer) includes the letters KF, MF, or PF. See FCC Registration in Chapter 1 for details.
- **Tie trunks-** a private line that 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. In more complex tie trunk configurations, a user can access a facility on the other system that does not exist on their own system.
- **Direct Inward Dialing (DID) trunks---** (incoming calls only) provide fast access to specific individuals, that is, incoming calls can be routed directly to the called extension, a calling group, or an outgoing trunk without system operator assistance, These trunks are reliable and efficient, but are more complex than loop-start or ground-start trunks. Therefore, their installation and maintenance must be coordinated with the telephone company,
- **DS1 connectivity** programmed for either T1 or Primary Rate Interface (PRI) operation-(incoming and outgoing calls) provides two-way connection and high speed transmission of analog and digital signals simultaneously. One trunk provides 24 channels; services provided on the channels can be assigned and subsequently changed by you instead of the telephone company, T1 operation enables the system to transmit and receive voice and analog information; PRI enables the system to transmit and receive voice, analog, and digital data, PRI provides a wide range of benefits not available on any other single type of 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.

The system supports 14 types of line/trunk and station modules. Figure 2-4 shows the line/trunk and station modules. Table 2-1 lists the type and number of jacks for each type of module. The names of modules are numbers that identify their connectivity and port capacities. The first digit is always the number of trunk jacks, while the third (last) digit is the number of station ports supported. For example, the 408 GS/LS module provides four trunk jacks and eight station jacks and supports ground-start or loop-start trunks.

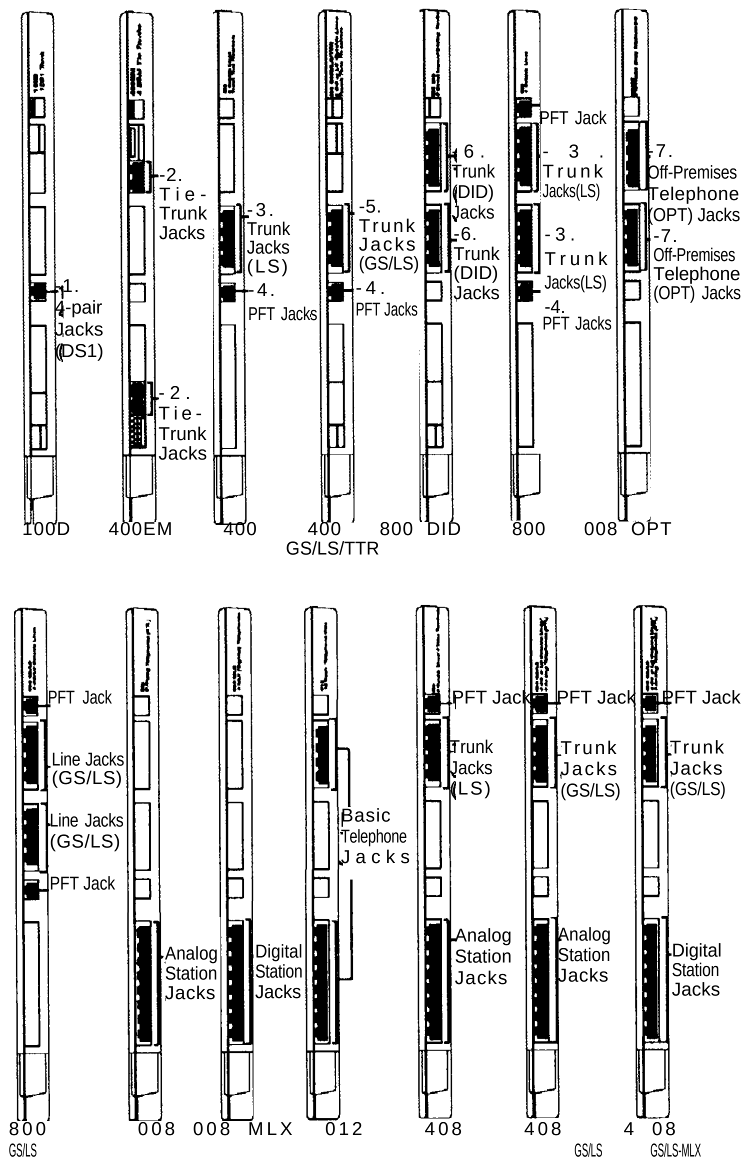


Figure 2-4. Line/Trunk and Station Modules

Table 2-1. Line/Trunk and Station Modules

Module	Line/Trunk Type	Station Type	Specifications
008	**N/A**	Analog multiline telephone; CMS	**Capacity: 8** analog stations **Signaling:** analog multiline telephone protocol **(40** kbps) **Loop range: 1000** feet **(305** meters). For In-Range Out-of-Building service, use analog I ROB protectors, over 1000 feet (305 meters), use Off-Premises Range Extender (OPRE).
ma MLX	N/A	MLX telephone; digital data device (such as 7500B Data Module)	**Capacity:** 8 digital stations, each with 1 or **2** endpoints (each endpoint is assigned an individual extension number), including the following station types: ■ digital voice only ■ digital voice with Voice Announce to Busy feature ■ digital voice and digital data (via the 7500B Data Module) ■ digital voice and Multi-Function Module (MFM) ■ digital data only (7500B Data Module) **Signaling:** BRI S/T 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 feet (305 meters) For In-Range Out-of-Building service, use MLX IROB protectors, over 1000 feet **(305** meters), use OPRE.
008 OPT[1]	N/A	On-premises or off-premises single-line telephone	**Capacity: 8** T/R stations on two-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 earner that has a 008 OPT module. **REN:** ≤1.0 per port **Disconnect signal, 900** ms (T/R short for answering machines, Group Ill Fax, etc.) **Switchhook flash detection: 3001200 ms** **Loop resistance: serves** 2-wire loops to 1300 ohms, including stations
012	**N/A**	Single-line telephone; MERLIN Attendant; MERLIN MAIL Voice Messaging System; T/R adjunct (such as answering or fax maachine): Analog data device (such as modem)	**Capacity: 12 T/R** stations on 2-way voice transmission path with support for telephones with message-waiting LEDs, 2 TTRs **Power. 21** VDC, 800-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. **REN:** ≤1.0 per port **Disconnect signal: 900** ms (T/R short for answering machines, Group Ill Fax, etc.) **Switchhook flash detection: 300-1200 ms** **Loop range:** 1000 feet (305 meters), in-building only
100D	T1 or PRI		**Capacity:** 24 lines/trunks for voice and analog data or **23** lines/trunks for voice and data with 1 channel used for signaling 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 **Signaling:** DS1 over 4-wire; T1 uses Robbed Bit Signaling (RBS) or Common Channel Signaling (CCS): PRI has 23 B + D
400[2]	LS and TTR	Power Failure Transfer **(PFT) telephone**	**Capacity: 4 lines/trunks, 4** TTRs, 1 PFT **telephone** **Signaling: loop-start**

Continued on next page

Table 2-1. - *Continued*

Module	Line/Trunk Type	Station Type	Specifications
400EM	Tie trunk		**Capacity:** 4 tie trunks **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 **Signaling:** loop-start or GS, optioned per port
408[2]	LS	Analog multiline telephone; CMS; PFT telephone	**Capacity:** 4 lines/trunks, 8 stations, 1 PFT telephone **Station signaling:** analog multiline telephone (40 kbps) **Line/trunk signaling:** loop-start line/trunk; analog voice **Loop range:** 1000 feet (305 meters), For In-Range Out-of-Building service, use analog IR0B protectors: over 1000 feet (305 meters), use OPRE.
408 GS/LS	LS or GS	Analog multiline telephone; CMS; PFT telephone (GS button needed for PFT telephone)	**Capacity:** 4 lines/trunks, 8 stations, 1 PFT telephone **Station signaling:** analog multiline telephone (40 kbps) **Line/trunk signaling:** loop-start or ground-start line/trunk (optional per port); voice **Loop range:** 1000 feet (305 meters). For In-Range Out-of-Building service, use analog IROB protectors; over 1000 feet (305 meters), use OPRE.
408 GS/LS-MLX[3]	LS or GS	MLX telephone; digital data device (such as 7500B Data Module)	**Capacity:** 4 lines/trunks, 1 PFT telephone, 8 digital stations, each with 1 or 2 endpoints (each endpoint is assigned an individual extension number), including the following station types: ■ digital voice only ■ digital voice with Voice Announce to Busy feature ■ digital voice and digital data (via the 7500B Data Module) ■ digital voice and MFM ■ digital data only (7500B Data Module) **Signaling:** BRI S/T 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 feet (305 meters) For In-Range Out-of-Building service, use MLX IROB protectors; over 1000 feet (305 meters), use OPRE.
800[2]	LS	PFT telephone	**Capacity:** 8 lines/trunks, 2 PFT telephones **Signaling:** loop-start
800 DID	DID		**Capacity.** 8 lines/trunks, 2 TTRs **Protocol:** incoming calls only, 2-way (one-pair) fixed impedance to DID trunks; no outgoing calls 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 Signaling: loop-start or ground-start

Notes:

1. The system software recognizes the OPT moduleas a 012 module, Even though the OPT module only has 8 jacks, it uses 12 ports of capacity, thereby decreasing overall station capacity by four stations for every OPT module.
2. Although these MERLIN II modules are supported, the following are the recommended modules for the system. 400 GS/LS, 408 GS/LS, 800 GS/LS, 408 GSLS-MLX.
3 This module is not compatible with Releases 1.0 and 1.1: it applies to Release 2.0 and later.

408 GS/LS-MLX Module

WARNING:
The 408 GS/LS-MLX module applies to Release 2.0 and later. It is not compatible with Releases 1.0 and 1.1.

The 408 GS/LS-MLX module (Release 2.0 only) is similar in concept to the 408 GS/LS module by providing four line and eight station ports; however, it provides MLX ports instead of analog ports. The MLX port operation is the same as that of the 008 MLX module, and the GS/LS port operation is the same as that of the 400 GS/LS module and the trunk portion of the 408 GS/LS module. A PFT port is provided for the first CO line on the module. Touch-tone receivers are not provided on the module.

A customer can replace pairs of 400 GS/LS and 008 MLX modules with the 408 GS/LS-MLX module to obtain cost and slot savings. The 408 GS/LS-MLX module can be used in any of the 17 port board slots. All 17 slots can be simultaneously equipped with 408 GS/LS-MLX modules, but the system translates and uses only the number of ports allowed by your specific system configuration, such as number of endpoints and calls per hour.

Cover

The control unit is covered by a plastic cabinet for protection. The size of the cover increases as expansion carriers are added to the control unit. Figure 2-5 shows how the control unit cover fits around the control unit carrier.

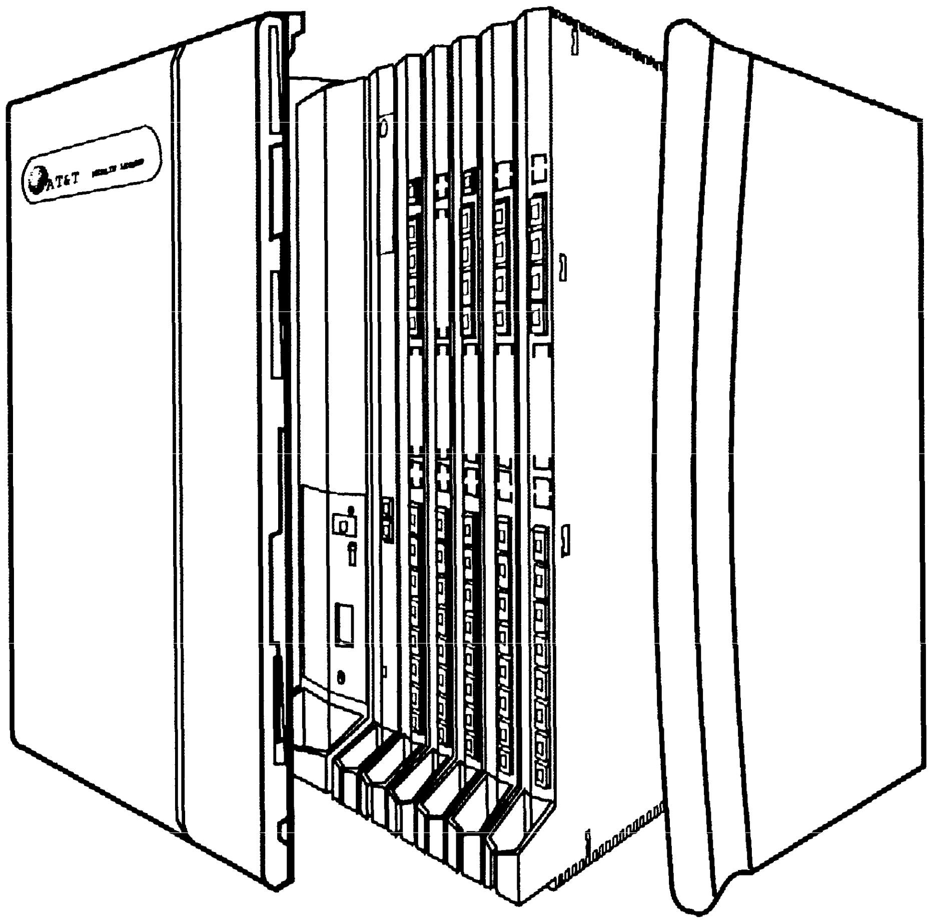


Figure 2-5. Control Unit Cover

MERLIN II Release 3 Reusable Modules

The following modules used in a MERLIN II Release 3 system can be used in this system:

- 391A1 power supply

  NOTE:
  While the 391A power supply can be reused in the system, it does not supply as much power as the 391 Al power supply module and should be replaced if anything is added to the station side of the system. The 391 A also has less protection against power surges than the 391 A1.

- 800 line/trunk module
- 400 line/trunk module
- 400 E&M line/trunk module
- 012 basic telephone module
- 008 analog station module
- 408 analog line/trunk and station module

Table 2-2 shows the reusable MERLIN II modules and their apparatus codes. Table 2-3 shows reusable MERLIN II hardware and associated apparatus codes or 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

Continued on next page

Table 2-2. - *Continued*

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

Table 2-3. Reusable MERLIN II Hardware

Type	Apparatus Code or PEC	Comments
Power supply module	391A	No surge protection; 147A protector recommended
	391 AA	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
Frequency generator (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 and Consoles

Several different analog and single-line telephones can be used with the system; the only digital telephones that can be used with the system are the MLX telephones,

NOTE:
An analog or digital multiline telephone located in a different building but within 1000 feet (305 meters) of the control unit requires an I ROB protector at each building entrance. If a single-line telephone is located in a different building from the control unit and is in excess of 1000 feet (305 meters), an OPRE or a 008 OPT module must be used.

MLX Telephones

The following are the four telephones in the MLX telephone line, all of which support the PRI services that can be used with the system:

- MLX-20L telephone
- MLX-28D telephone
- MLX-10D telephone
- MLX-10 telephone

Each of these telephones is available with factory-imprinted buttons in English, French, or Spanish, and in black or white.

The following features are common to all MLX telephones:

- Programmable line and feature buttons with two associated lights (red and green)

  NOTE:
  An MLX-20L telephone used as a Queued Call Console (QCC) has no programmable buttons.

- Fixed-feature buttons (four of them have a red or a green LED: Feature, HFAI, Mute, and Speaker)
- Red message-waiting LED
- Built-in speakerphone
- Separate volume controls for speakerphone, handset, and ringer
- A card tray under the telephone with frequently used features
- Optional internal MFM to connect to T/R equipment and alerting devices

  NOTE:
  An MLX-20L telephone used as a QCC cannot have an MFM.

- Two-position adjustable desk stand
- Four-pair modular line cord

MLX telephones with display have the following two additional features:

- LCD display
- Display-associated buttons

A list of features specific to each telephone model in the MLX family follows.

Model MLX-28D

The MLX-28D telephone provides the following features:

- Can be used as a system operator Direct-Line Console (DLC)
- 28 line buttons
- Display (2 lines x 24 characters)
- 8 display-associated buttons
- 8 dedicated feature buttons
- Accommodates one or two DSSs

This telephone is not wall-mountable.

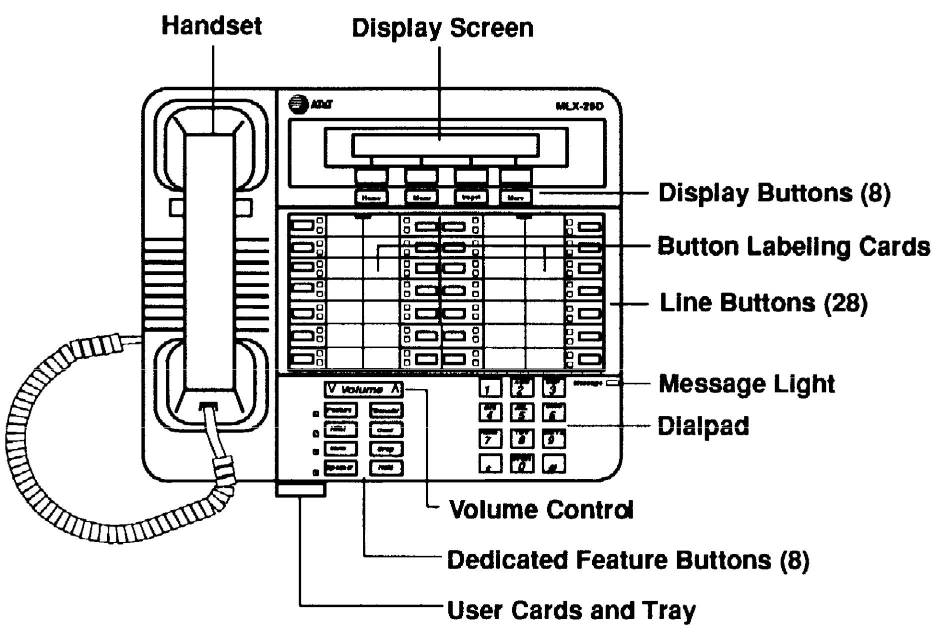


Figure 2-6. MLX-28D Telephone

Model MLX-20L

The MLX-20L telephone provides the following features:

- Can be used for system programming and as a DLC or a QCC system operator console
- 20 line buttons
- Display (7 lines x 24 characters)
- 14 display-associated buttons
- 8 dedicated feature buttons
- Accommodates one or two DSSs

This telephone is not wall-mountable.

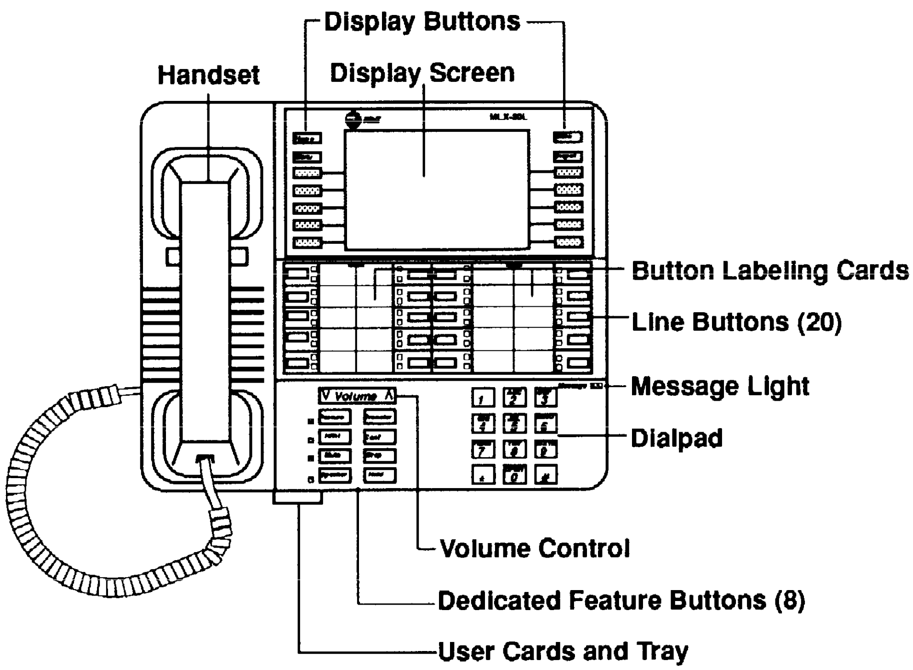


Figure 2-7. MLX-20L Telephone

Model MLX-10D

The MLX-10D telephone provides the following features:

- 10 line buttons
- Desktop or wall-mount
- Display (2 lines x 24 characters)
- 8 display-associated buttons
- 8 dedicated feature buttons

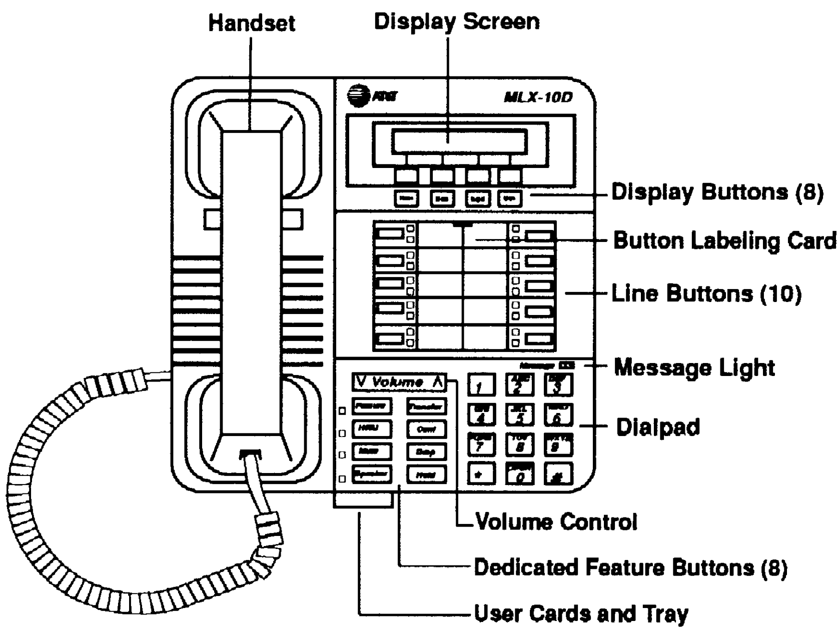


Figure 2-8, MLX-10D Telephone

Model MLX-10

The MLX-10 telephone provides the following features:

- 10 line buttons
- 8 dedicated feature buttons
- Desktop or wall-mount

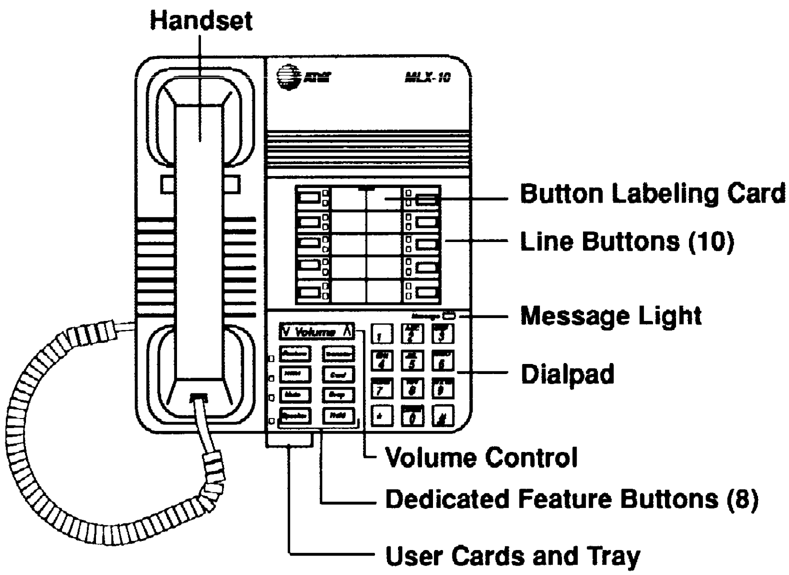


Figure 2-9. MLX-10 Telephone

Direct Station Selector

The DSS is an optional adjunct that can be connected to an MLX-20L or an MLX-28D telephone, It enhances the capabilities of both DLCs and QCCs and, when connected to the MLX-20L telephone, facilitates programming. The DSS is shown in Figure 2-10.

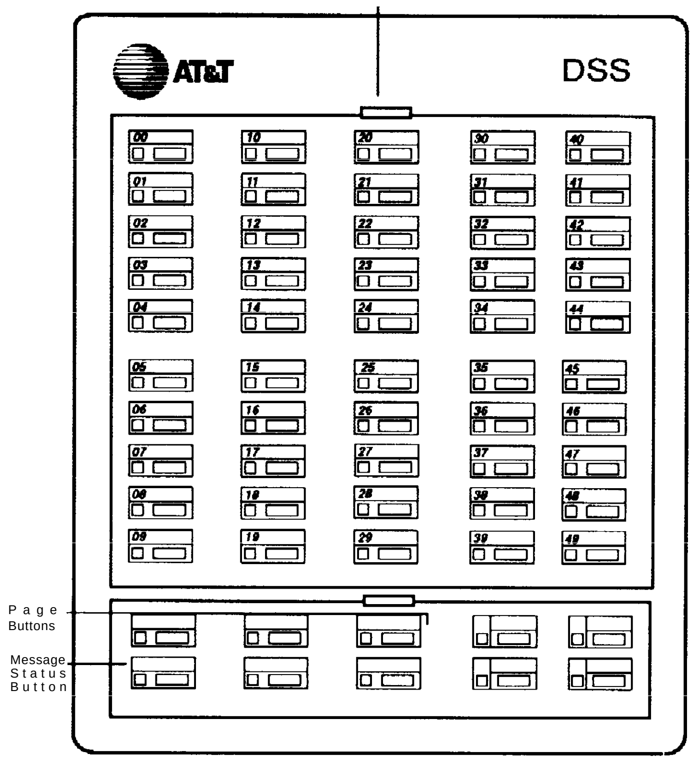


Figure 2-10. Direct Station Selector

A DSS has 50 buttons with lights that can be programmed with the following:

- 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)

These buttons are used by a system operator for one-touch dialing and call transfer. Ten additional buttons are located at the bottom of the DSS.

The 50 numbers programmed onto the DSS are considered a "page," Each DSS can have three pages of numbers, for a total of 150 numbers per DSS. Two DSSs connected together increase this capability to 300 numbers. (If two DSSs are connected, page one has the first set of 100 numbers, page two has the second set of 100, and page three has the third set.)

Three of the 10 lower buttons on the DSS are reserved as Page buttons to provide access to the different pages. The range of the numbers on the pages can be programmed to begin with any number up to 9950, but the default for the beginning Page numbers is 0, 50, and 100, respectively, for the three pages. A fourth button is reserved as a Message Status button, which, when pressed, changes the indication of the DSS button lights from telephone use to Message Status mode. The other six buttons on each DSS are for future use.

See Telephone Power Units and Table 2-9 in this chapter for more information on connecting DSSs and consoles.

NOTE:
DSSs are shipped without local auxiliary power supplies; if required, these must be ordered separately.

Analog Multiline Telephones

In addition to the MLX telephones, the analog multiline telephones in Table 2-4. can be connected to the system.

Table 2-4. Analog Multiline Telephones

Model	Description
5-button*	5-button telephone with membrane; no adjuncts supported
10-button*	10-button telephone with membrane
34-button*	34-button basic telephone with membrane
34-button Deluxe*	Deluxe 34-button telephone with membrane
10-button HFAI*	10-button hands-free-answer telephone; no adjuncts supported
34-button BIS*	34-button telephone with built-in speakerphone
34-button BIS/DIS*	34-button telephone with 16-character display and built-in speakerphone
BIS-10	10-button telephone with built-in speakerphone
BIS-22	22-button telephone with built-in speakerphone
BIS-22D	22-button telephone with 16-character display and built-in speakerphone
BIS-34	34-button telephone with built-in speakerphone
BIS-34D	34-button telephone with 16-character display and built-in speakerphone
MLC-5 Cordless	Cordless 5-button telephone (Limitations: Button assignments cannot be changed; cannot use Conference feature)
MERLIN PFC Telephone†	Analog multiline phone, fax machine, and copier

* Vintage telephone; no longer available for sale or lease
† Requires two analog multiline ports.

Single-Line Telephones

The system supports the single-line analog telephones listed in Table 2-5.

NOTE:
PFT telephones must be selected according to trunk type. If rotary trunks are used, PFT telephones must be rotary telephones (500MM is recommended). If telephones are to be connected to ground-start trunks, a ground-start button (KS23566,L1) must be added to each PFT station.

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, suet as Transfer and Hold
2500SM	Basic desk telephone used with 4A speakerphone
2514 BMW	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) stations.
CS6402U01A*	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.
	Basic desk telephone with a built-in speakerphone with volume control and **Mute** button with LED indicator.
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 model8110 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 with the system.

Telephones and Adjuncts Not Supported

CAUTION:
The following telephones and adjuncts cannot be used with the system. Connecting them can damage the telephones, adjuncts, and system.

Table 2-6. 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 that 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

System Operator Consoles

System operator consoles are telephones that are programmed for call handling and other system operator duties. They can be used in two configurations—QCC and DLC. QCCs are available only in Hybrid/PBX mode.

A system operating in Hybrid/PBX mode can include both QCCs and DLCs. The maximum numbers of both types of system operator positions is shown in Table 2-7.

Table 2-7. Maximum Number of System Operator Positions

Position Type	Type of Telephone	Maximum Positions
QCC	MLX-20L	4
DLC	MLX-20L MLX-28D	8
DLC	Analog multiline telephones	8

NOTE:
No more than eight system operator positions of any combination (QCCs and DLCs) are allowed; when used in combination, no more than four can be QCCs.

Queued Call Consoles

QCCs are available only in Hybrid/PBX mode. In a QCC configuration, incoming calls are held in a queue and calls are directed to a QCC as a position becomes available. Only one call rings at a time.

QCCs must be connected to a digital station jack on a 008 MLX or 408 GS/LS-MLX module (Release 2.0 only). The first QCC must be connected to the first station jack in the system.

The MLX-20L telephone is the only telephone that can be assigned as a QCC through system programming. A QCC system operator cannot use feature codes to activate features. The QCC system operator can only use the features that can be selected from the display or that are assigned as fixed features to buttons on the console.

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.

The buttons on the QCC are factory-set with fixed features and cannot be programmed by the system operator or through centralized programming. The QCC fixed-feature buttons areas follows:

- **Call:** Five buttons used to answer incoming calls and make inside and outside calls.
- **Start:** Initiates the call-extending 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 while the call is in a split condition.
- **Release:** Releases the system operator from a call and/or completes the call-extending process, making the system operator available for another call.
- **Destination:** Reconnects the system operator to the destination while a call is in a split condition.
- **Cancel:** Cancels call extending and reconnects the system operator with the caller (source).
- **Join:** Connects the system operator with both 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 message waiting, 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.

Each QCC can have one or two DSSs attached. The system operator can use the buttons during call handling, for example, to extend a call, make an inside call, park a call, or see the availability of an extension.

No more than four QCCs are allowed on a system. A maximum of two QCCs can be assigned on each 008 MLX or 408 GS/LS-MLX module (Release 2.0 only). No more than eight system operator positions of any combination (QCC and DLC) are allowed; when used in combination, no more than four can be QCCs.

When a system has QCCs, the first MLX module used for QCCs must be installed in the control unit to the left of any other type of module with station jacks.

A maximum of two QCCs is allowed per 008 or 408 MLX module. A QCC can be connected only on the first and fifth station jack on each module.

The following options must be assigned 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

Direct-Line Consoles

In a DLC configuration, lines/trunks are assigned to individual buttons and 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), outside lines are assigned as personal lines to individual buttons on the console. The lines assigned on an individual DLC can also be assigned 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 via the Transfer button.

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

On a system with fewer than 29 lines, Alarm, Night Service, and Send/Remove Message are factory-assigned to analog DLCs with 34 buttons or more. On a system with more than 29 lines, Alarm is replaced with line 30, Night Service is replaced with line 31, and Send/Remove Message is replaced with line 32. Alarm, Night Service, and Send/Remove Message and the first 1 through 18 lines are factory-assigned on an MLX-28D telephone used as a DLC, regardless of the number of lines/trunks connected to the system.

The following telephones can be used as DLCs:

- Digital DLC
  - MLX-20L telephone
  - MLX-28D telephone
- Analog DLC
  - MERLIN II System Display Console with built-in DSS
  - BIS-34D telephone
  - BIS-34 telephone
  - BIS-22D telephone

One or two DSS adjuncts can be added to the MLX-20L or MLX-28D telephone to provide 50x3 or 10OX3 additional extension buttons. The DSS cannot be attached to an analog DLC; however, the MERLIN II System Display Console provides a built-in DSS.

Analog DLCs are connected either to an analog station jack on a 008 or a 408 analog multiline telephone module, or to a digital station jack on a 008 MLX or 408 GS/LS-MLX module (Release 2.0 only).

When both DLCs and QCCs are assigned in the Hybrid/PBX mode, the maximum combined number of system operator positions is eight; no more than four can be QCCs. A maximum of two DLCs can be assigned per MLX or analog module.

An MLX-20 telephone used as a DLC can be used for system programming by connecting it to the first or fifth station jack on the first MLX module and designating the station jack for system programming.

Only multiline telephones connected to the first and fifth station jacks on digital or analog modules can be assigned as DLCs. This includes DLCs assigned as calling group supervisors and Call Management System (CMS) supervision.

Adapters and Adjuncts

This section describes auxiliary hardware for the communications system. It includes adapters, adjuncts, and other accessories. The following table provides information about the hardware components that can be used with the system.

Table 2-8. Adjunct Summary

			Interface				
			LS or GS/LS	T/R	MFM	GPA	SAA
Equipment Type	Specifications	AT&T products	Trunk Jack	012 or 008 OPT Station Jack	MLX Station Jack	Analog Station Jack	Analog Station Jack
Alerts (AC)'	Any audible or visual alert that operates on 20-30 Hz ringing signals. Associated with a specific station (supplemental alert) or works on a programmed trunk port (external alert).	External Ringer—Loud External Ringer		✓	✓	✓	✓
Alerts (DC)	Any audible or visual alert that operates on 48-WC ringing signals. Associated with a specific station (supplemental alert) or works on a programmed trunk port (external alert)	Alert bell Alert horn Alert strobe Alert chime Alert deluxe horn Alert switch	✓2		✓		✓
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).	Model 1300 answering machine Model 1531 Remote Answering System telephone		✓	✓	✓	
Cordless Telephone'	Must have touch-tone dialing capability when connected via MFM; rotary or touch-tone dialing can be used on T/R port. Single line	5320 Cordless Telephone 5200 Cordless Telephone 5500 Cordless Telephone		✓	✓7	✓	
Credit Card Verification Terminal'	Must have touch-tone dialing capability when connected via MFM: rotary or touch-tone dialing can be used on T/R port.			✓	✓8		
Dial Dictation' 1[2]	A device that requires contact closure can be used on LS/GS line jack only with UPAM.		✓	✓	✓	✓	

Continued on next page

Table 2-8. - *Continued*

Equipment Type	Specifications	AT&T Products	Interface: LS or GS/LS Trunk Jack	T/R 012 or 008 OPT Station Jack	MFM MLX station Jack	GPA Analog Station Jack	SAA Analog Station Jack
Direct Station Selector (DSS)	A maximum of 2 DSSs can be connected to an operator console. A 329A power unit must be added to an operator console having 2 DSSs. Connects to DSS jack on operator console,						
Fax[1]	Must have touch-tone dialing capability when connected via MFM; rotary or touch-tone dialing can be used on T/R port. Industry-standard analog interface	AT&T 3410D AT&T 3500D AT&T 35100 AT&T 3520D AT&T 3530D AT&T Fax 4515D AT&T Fax 4525D AT&T Fax 9015 AT&T Fax 9020 AT&T Fax 9022 AT&T Fax 9025FX AT&T Fax 9035FX		✓	✓ 8		
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.	Model 1330 Answering Machine DA-5 Digital Voice Announcer		✓	✓ 8	✓	
Hands-Free Unit Headset	For use with analog multiline telephones Connects directly to telephone. For use with MLX or analog multiline telephones,	S202A StarSet® Mirage® Supra® Supra NC®					
Headset Adapter	Need to program Auto Answer Ail button for use with 502B, 502C, Connects directly to telephone OTHER jack	502A 502B 502C					
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 20 PagePac 20 with Zonemate 9[9] PagePac 20 with Zonemate 39[9] PagePac 6[10] PagePac 6 Plus	✓ 9				

Continued on next page

Table 2-8. – *Continued*

Equipment Type	Specifications	AT&T {roducts	LS or GS/LS Trunk Jack	T/R 012 or 008 OPT Station Jack	Interface MFM M L X Staion Jack	GPA Analog Staion Jack	SAA Analog Station Jack
Message Waiting Indicator	For single-line telephones Connects directly to telephone	Z34A					
Modem	If the modem supports touch-tone dialing via the associated data terminal, the keyboard can be used for dailing. If the modem does not support touch-tone dialing, an associated basic (single-line) telephone can be used for dialing.	4000 Modem 2224G Modem 4024 Modem 2296A Modem 2296 Modem	✓[3]	✓	✓	✓	
Music-on-Hold[4]	Any FCC-registered 8-ohm music source or recorded announcement device.	Magic On Hold	✓[5]				
Speakerphone	Connect directly to telephone. For signle-line telephones only.	4 A[6] 203A					
SMDR Printer	Connects to upper RS-232-C jack on proccessor module Must be located within 50 feet (15 meters) of contro unit or use ADU to extend distance.	475 Printer 476 Printer 572 Printer					

Notes.

1. Cannot be connected to a QCC
2. Requires UPAM to provide 48 VDC.
3. For 2224G Modem only.
4. If you use equipment that rebroadcasts music or other copyrighted materials, you maybe 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 or an authorized dealer
5. Music Coupler required
6. Requires 2500SM telephone.
7. Device originates and receives calls independently of associated telephone when used with an MFM. When a GPA is used, calls are dialed and received via the analog telephone
8. Device originates and receives calls independently of associated telephone when used with an MFM.
9. Bi-directional paging is supported: only one line jack is needed for multizone paging.
10. Loop-start adapter (53518) is required when connected to Ioop-start line jack.

Adapters

Adapters connect adjuncts to the system and stations. They provide access for both voice and data signals.

System Adapters

Three adapters connect directly to the control unit: the channel service unit (CSU), the Loop-Start Trunk Adapter, and the Universal Paging Access Module (UPAM).

Channel Service Unit

The CSU is the interface between the 100D module and the DS1 facility provided by the telephone company. This facility contains 24 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.

There is an Extended Super Frame (ESF) T1 CSU and a 551 T1 L1 CSU. The ESF T1 CSU is recommended for this system because it allows the unit to be maintained without interrupting service and provides diagnostic and testing capabilities, and it is the only CSU that provides bipolar 8 zero code substitution (B8ZS) line coding. The lower-cost 551 T1 L1 CSU performs most of the functions of the ESF T1 CSU but does not provide the B8ZS line coding required for 64-kbps data and for maintenance features. The 551 T1 L1 CSU does not provide diagnostic and testing capabilities; also, it is not recommended for Video Conferencing applications.

Figures 2-11 and 2-12 detail the CSU connections.

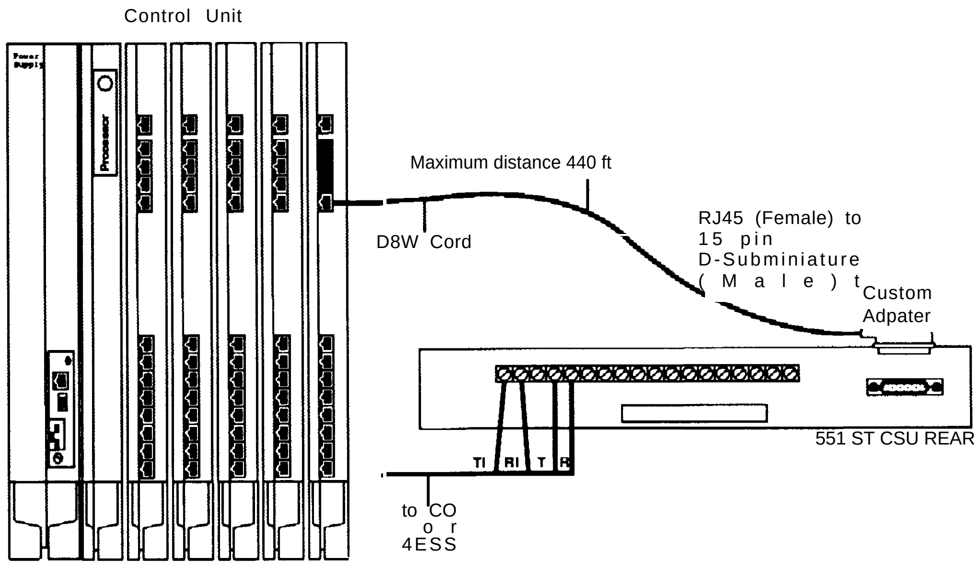


Figure 2-11. 551 T1 L1 Channel Service Unit Connections

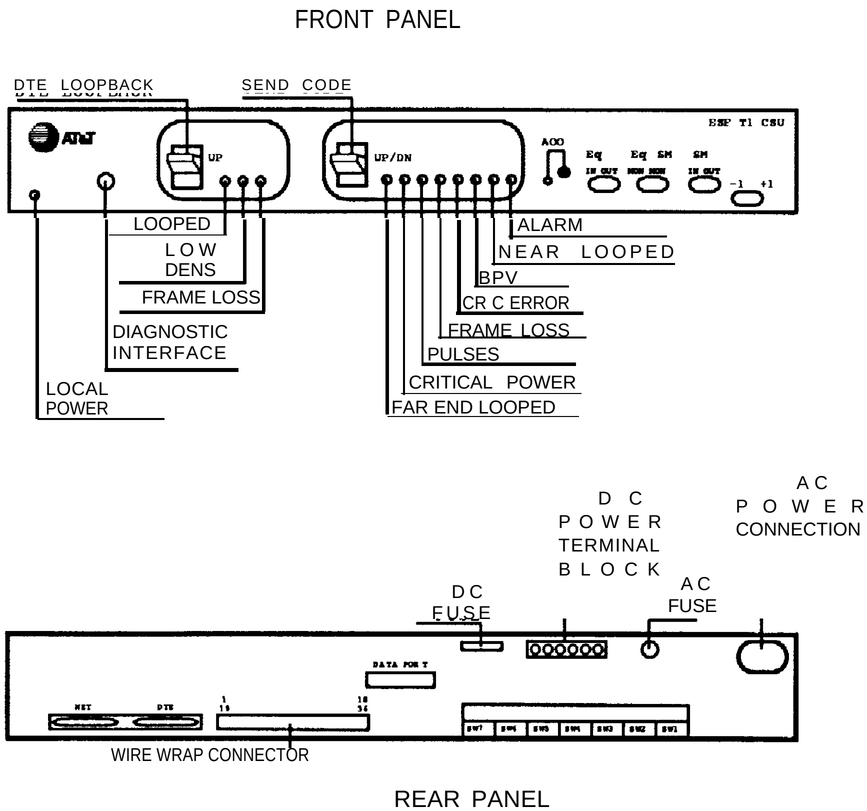


Figure 2-12. ESF T1 Channel Service Unit Connections

Universal Paging Access Module or Loop-Start Trunk Adapter

The UPAM and the Loop-Start Trunk Adapter connect paging equipment to a loop-start or ground-start jack on the control unit. A Loop-Start Trunk Adapter should be used *only* with a PagePac paging system; a UPAM should be used with a customer-provided paging system. A Loop-Start Trunk Adapter *must* be used with a PagePac paging system and no Zonemate equipment, An acknowledgement tone is not provided when an UPAM is used.

The UPAM is ordered as a D-kit, The kit contains a power source, a microphone matching transformer, and an access module.

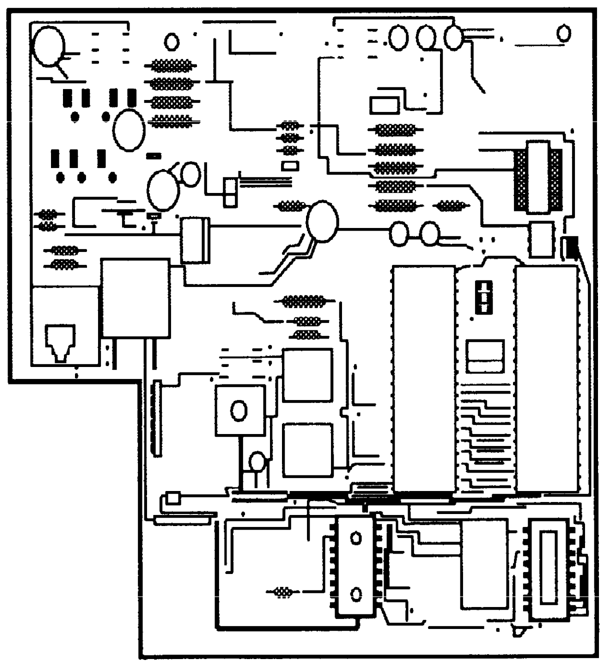
Telephone Adapters

The following adapters are used to connect various adjuncts to the telephones.

Multi-Function Module

The Multi-Function Module (MFM) is an optional adapter for connecting tip/ring (T/R) or supplemental alert adjuncts to any 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.

NOTE:
An MFM cannot be used in an MLX-20L telephone serving as a QCC.



Multi-Function Module

Figure 2-13. Multi-Function Module

WARNING:
The MFM can be installed or repaired only by a qualified technician or an authorized dealer representative. To eliminate the risk of electrical shock, the MLX telephone should not be disassembled.

T/R adjuncts operate independently of the MLX telephone. If the telephone is in use, voice or data calls can still be sent and received by the adjunct. The MFM allows the use of the following T/R adjuncts:

- Answering machines
- Fax machines
- Modems
- Credit card verification terminals
- Cordless telephones
- 2500-type (basic touch-tone) telephones

Supplemental alerts such as bells, chimes, horns, and strobes notify people in noisy areas of incoming calls.

The MFM is shipped with a KS29911 , L2 or L2 power unit that supports one MFM and one DSS. When two DSSs are connected to a telephone, a 329A wall power unit is used instead of the KS22911. With either type of power unit, the total cord length can be no more than 50 feet (15 meters) from the telephone.

The MFM supports only touch-tone dialing and does not detect pulse-dialing. Also, the MFM does not support features activated by a switchhook flash, so no transfer or conference calls can be made from a station attached to an MFM.

NOTE:
The MFM uses one of the two B-channels when it is active. This means that when an accessory, such as a fax machine, and the MLX telephone are in use at the same time, Voice Announce to Busy and speakerphone paging cannot be used, (When Voice Announce to Busy is being used, a person calling an MFM extension gets a busy signal; a person attempting to call out from a station attached to an MFM will not get a dial tone.)

Two jumper blocks on the MFM configure it for either tip/ring (T/R) or supplemental alert operation. These are preset at the factory for T/R operation.

NOTE:
Only a qualified service technician or an authorized dealer representative should change the jumper settings.

In the T/R mode, the MFM can connect to 20-Hz AC external alerting devices such as a loud external ringer and El CM-type ringer. If several devices are connected to the MFM, only one device can be off-hook at a time and the total ringer equivalent number (REN) < = 2.0.

General Purpose Adapter

A General Purpose Adapter (GPA) permits the attachment of a T/R 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 calls must be originated on the analog multiline telephone since the GPA has no pulse or touch-tone detectors.

One end of a 4-pair cord plugs into the V.T. jack on the back of the GPA, and the other end plugs into the OTHER jack on the underside of the telephone. The 1-or 2-pair cord from the T/R device plugs into the TEL. EQUIP. jack on the GPA (see Figure 2-14).

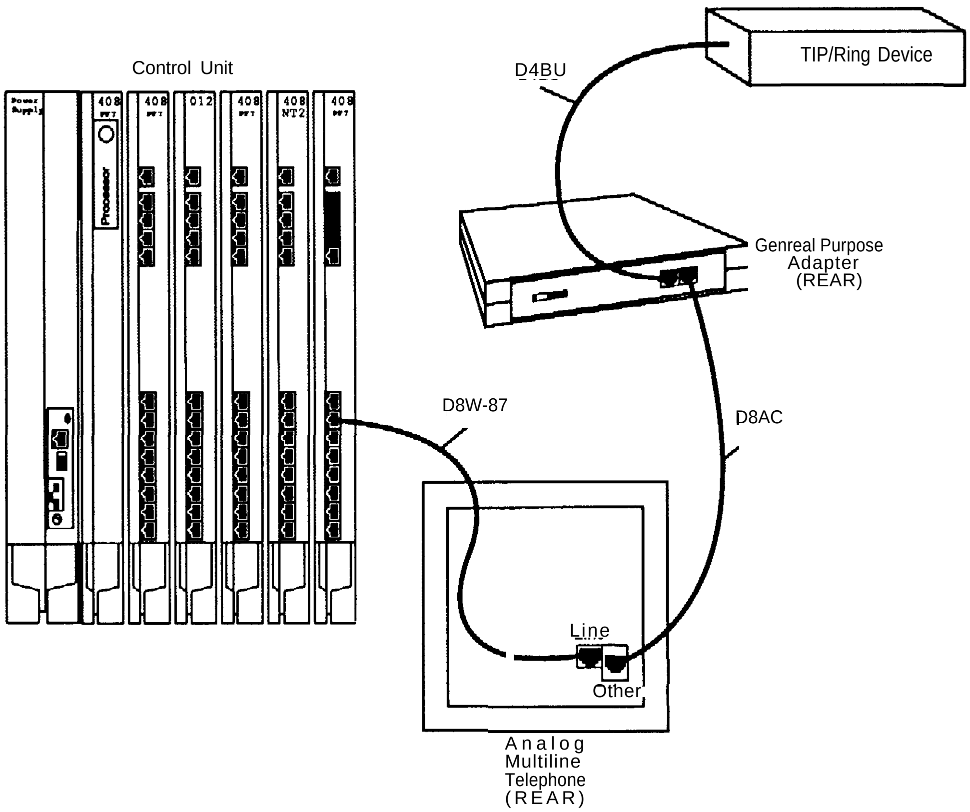


Figure 2-14. GPA Connections

Sliding the switch on the back of the GPA to the proper setting provides the GPA service required.

- Basic. This setting is used 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.** This setting is used 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.** This setting is used in two ways:
  — With a device, such as an answering machine or a modem, to answer calls. An **Auto Answer All** button is needed so that calls can be answered automatically.

  — To make and receive calls on the telephone while a computer or modem attached to the GPA is being used. This is called the *Simultaneous Voice and Data* feature.

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 only) so that a user can make and receive calls at a digital 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 digital station or over the PRI telephone network, On a data terminal, the keyboard is used to dial the number.

NOTE:
A data module cannot be used with a QCC.

The 7500B Data Module provides a RS-232 interface for asynchronous data terminal equipment operating at speeds of up to 19.2 kbps. The data module also provides a CCITT V.35 interface for synchronous data terminal equipment operating at speeds of up to 64 kbps. (Optional enhancement boards must be ordered separately.)

The data module can be set up to handle a variety of data communications equipment (DCE) and is the only digital adapter approved for use with the system.

The data module's front panel has the following features (see Figure 2-15):

- Power/Test LED—lights when power is supplied; flashes when tests are performed.
- Data LED—flashes to indicate an incoming data call and lights when a call is in progress; flashes when tests are performed.
- Display—displays status information and option settings.
- Next, Back, and Enter buttons—used to operate the data module and to adjust the screen's contrast.

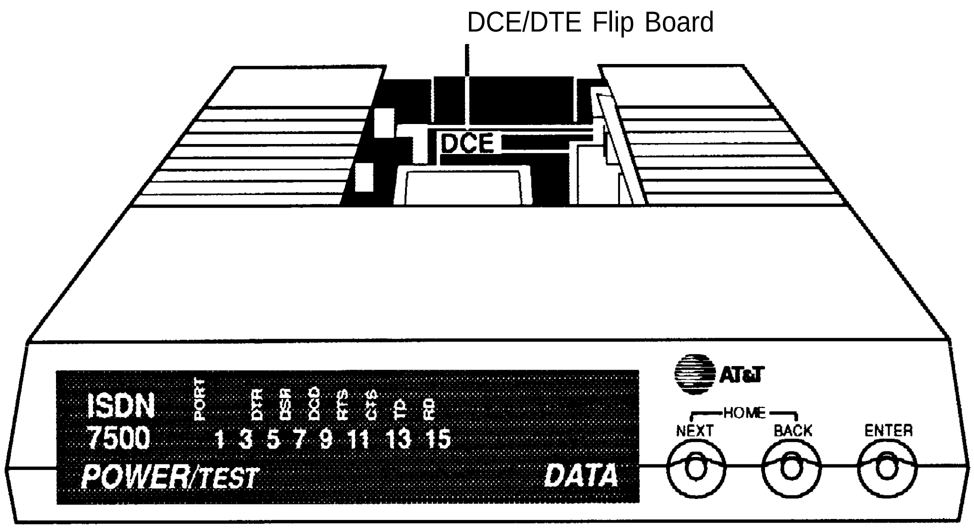


Figure 2-15. 7500B Data Module Front Panel

The data module's back panel has the following features (see Figure 2-1 6):

- Phone jack-connects an MLX telephone to the data module.
- Line jack—connects the data module to an MLX system module.
- Power connector—connects the data module to the DC power supply, which connects to an AC outlet.
- Port l—connects the 7500B to a data terminal, computer, or modem.
- Port 2—when an enhancement board is installed for synchronous operation, Port 2 connects a second data terminal, an automatic calling device (with a RS-366 interface), or a data terminal with a V.35 interface.

NOTE:
A modem can provide an analog data interface from a MLX telephone that has an MFM installed. When an MLX telephone has an MFM, the data module cannot be installed on the same line.

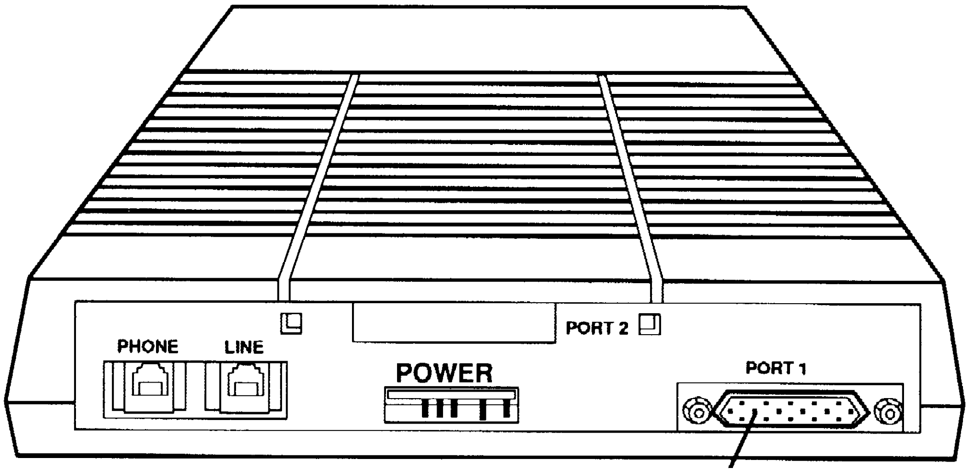


Figure 2-16. 7500B Data Module Back Panel

When the 7500B Data Module is used with an MLX telephone, one end of the D8W cord is plugged into the PHONE jack on the data module, and the other end of the cord is plugged into the LINE jack on the MLX telephone. The maximum cord length from the data module to the telephone cannot exceed 80 feet (24 meters). The MLX telephone cannot be used to dial data calls, and the data terminal equipment connected to the data module cannot be used to dial voice calls. Each device operates independently, and features are assigned to each device independently.

NOTE:
Do not connect two 7500B Data Modules on one line.

The data module can be configured as a stand-alone by ordering a WP90110,L1 power unit The data module can also be configured in a multiple-mount arrangement by ordering a Z77A data mounting, which provides a common power supply for up to eight 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 data module; both must be ordered separately.

The 7500B Data Module does not have the internal 100-ohm line termination that is provided with MLX telephones. Therefore, when it is used without an MLX telephone, a 100-ohm 440A4 terminating resistor adapter must be installed on the line near the data module.

To provide synchronous operation at speeds up to 64 kbps, the following optional circuit boards must be ordered:

- **Multipurpose Enhancement 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. A V.35 adapter cable must be ordered separately to operate 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.
- **High-Speed Synchronous Interface Enhancement Board.** Provides a V.35 interface at synchronous data rates of 48, 56, or 64 kbps. A V.35 adapter cable that converts the 25-pin male connector on the board to the industry-standard 34-pin V.35 interface is included.

Data Module Features

The 7500B Data Module offers the following features:

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 via the data terminal attached to the RS-232 interface
- Ability to change options without dropping a data call
- 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
- Auto-adjust—the ability to adjust to the speed and parity of the data terminal being used
- Call setup (dialing) from the keyboard of an ASCII data terminal by using the local command (CMD) mode or AT mode
- Automatic or manual answering of incoming data calls

Synchronous Features with Multipurpose Enhancement Board

- 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 via the V.35 interface adapter cable
- Full-duplex operation at 64 kbps via 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 data module display on the front panel
- RS-366 interface to an ACU

Synchronous Features with High-Speed Synchronous Enhancement Board

- V.35 interface (the adapter cable is provided when the board is ordered by 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 data module display on the front panel

Modems

A modem is used at an analog data station to make and answer data calls. It converts the digital signals of the data terminal into analog signals for transmission over standard telephone lines. It also converts incoming analog signals to digital signals for acceptance by the data terminal.

Most types of modems can be connected to the system unless the modem is being used in a modem pool. The recommended models are as follows:

- Modem Model 4000
- Modem Model 2224G
- Modem Model 4024
- Modem Model 2296A
- Modem Model 2296

If a modem is used with an MLX telephone, an MFM must be installed in the telephone to provide a tip/ring interface for the modem. The modem is connected directly to the MFM. If the modem is used with an analog multiline telephone, a GPA is required to provide a tip/ring interface for the modem.

When a modem is connected to an MLX telephone using an MFM, data calls are dialed using the data terminal keyboard and voice calls are dialed using the telephone dialpad. The MLX telephone cannot be used to dial data calls, and the data terminal keyboard cannot be used to dial voice calls. Each device operates independently, and features are assigned to each device independently, When a modem is connected to an analog multiline telephone using a GPA, data calls and voice calls are dialed by using the telephone dialpad. The modem and telephone do not operate independently; features assigned to the telephone also apply to the analog data station (modem and associated data terminal).

Modem Features

The modem used in an analog data station (and not in a modem pool) provides the following features:

- Dialing or ending asynchronous data calls from the keyboard when connected using a basic telephone station 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 for the call on the keyboard and, if necessary, change the options without dropping the call

Supplemental Alert Adapter

A supplemental alert adapter (SAA) allows the connection of an alerting device such as a bell or chime to an analog multiline telephone (see Figure 2-17). These alerts notify people working in noisy areas of incoming calls.

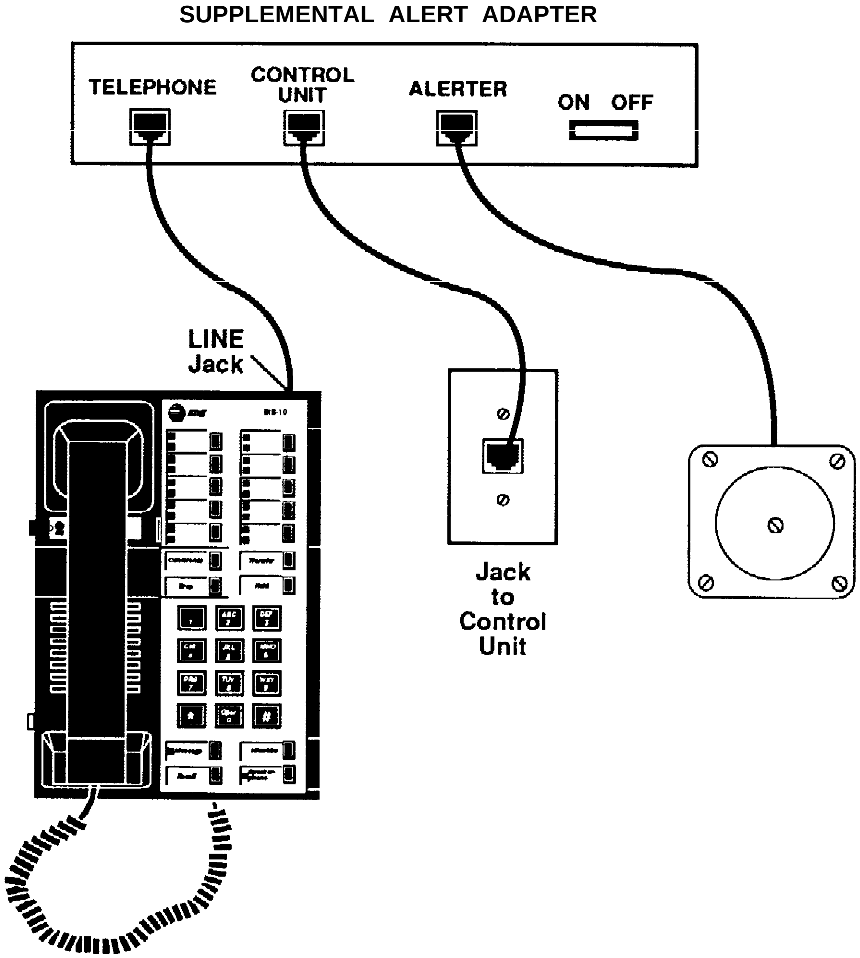


Figure 2-17, SAA Connections

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

Adjuncts

An adjunct is an auxiliary piece of equipment, connected to the system or to telephones with an adapter, for example, a fax machine or an answering machine.

System Adjuncts

System adjuncts are auxiliary pieces of equipment that connect directly to the control unit.

Station Message Detail Recording Printer

A Station Message Detail Recording (SMDR) printer can be connected to the control unit at the SMDR jack on the processor. The SMDR printer must be located within 50 feet (15 meters) of the control unit; otherwise, an asynchronous data unit (ADU) must be used to extend the distance.

The SMDR feature is used to capture detailed usage information on incoming and outgoing voice and data calls. Two SMDR report formats are available: the factory-set basic format or the PRI format. The PRI format is used when the user subscribes to the AT&T INFO2 automatic number identification (ANI). When the PRI format is selected during system programming, the number identification information is printed in the Called Number field of the call report. The remainder of the fields are identical to the basic format.

This information is sent to the SMDR printer. An SMDR record consists of the following fields:

- **Call Type** (Basic or PRI)
- **Date**
- **Time**
- **Called Number**
- **Dur** (Duration)
- **Line** (Facility Number)
- **STN** (Station Extension)
- **Account** (Account Code)

System Programming and Maintenance PC

A PC with DOS version 3.3 or higher and SPM software can be used for the programming and maintenance of the system. The PC is connected to the lower jack on the processor (the system programming/maintenance jack). See the Applications chapter for additional information.

Loudspeaker Paging Systems

Loudspeaker paging systems use a GS/LS line port. The port should be programmed for loop-start operation and programmed as a paging port. Up to three ports can be programmed as paging ports. When connecting a customer-owned paging system, a UPAM should be used. See the discussion of UPAM earlier in this chapter.

NOTE:
A ground-start/loop-start (GS/LS) line port should be programmed for loop-start operation for paging equipment. If the loop-start port is programmed for paging, it cannot be used for outside calls unless a PagePac Port Saver is used.

The PagePac 20 with Zonemate 9 or Zonemate 39 is an external paging system using dual-tone multifrequency (DTMF) signaling that can be connected to a line/trunk port programmed for paging operation. Bidirectional paging or "talkback" is available. For paging systems with multiple zones, only one loop-start or ground-start line/trunk is required.

This system is compatible with the PagePac 20 set to loop-start or ground-start operation, and a PagePac 6 Port Saver. The Port Saver allows the paging unit to be connected to the same loop-start line/trunk used for calls. Calls can be made and received when the paging unit is not active. The Port Saver is not compatible with ground-start lines/trunks.

Music-on-Hold and the Music Coupler

Background music can be provided for outside callers on hold or for selected areas of a building over the loudspeaker system.

NOTE:
If equipment is used that rebroadcasts music or other copyrighted materials, the user may be required to obtain a copyright license from, and pay license fees to, a third party such as the American Society of Composers, Artists, and Producers (ASCAP) or Broadcast Music Incorporated (BMI). Or a Magic On Hold system, which does not require the user to obtain such a license, can be purchased from an AT&T or authorized dealer.

Music can be heard by outside callers on hold by connecting a music source or Magic On Hold unit to a music coupler. The music coupler must connect to a loop-start port that has been programmed for Music-on-Hold operation and must be used with an 8-ohm music source. The loop-start port cannot be assigned to a trunk pool, cannot appear as a programmed line on system telephones, cannot be connected to a loudspeaker paging system, and cannot be on a trunk port assigned to the QCC queue or to calling groups.

When a call is transferred, it is automatically connected to Music-on-Hold if the system provides this feature.

The PagePac 20 also provides a music source for paging and Music-on-Hold without a music coupler.

Dial Dictation

Dial dictation through a customer-provided dictation unit can be used as either a system or station 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 an UPAM that is connected to a loop-start port programmed for dial dictation (similar to loudspeaker paging). The UPAM does not support dial dictation equipment requiring contact closure.

Fax Machine

A fax machine can be connected to any T/R jack on the control unit or to a MFM. Using a fax machine with a GPA is not recommended because the fax machine cannot auto dial through the GPA. Instructions are packed with the unit.

A fax machine originates and receives fax calls independently of any associated telephone. Calls are dialed with the fax machine's dialpad or from an associated single-line telephone.

If the system doesn't have direct inward dialing (DID) trunks, fax stations should be programmed to personal lines. When the system has DID service, incoming calls can be directed automatically to individual fax stations or to machines in calling groups.

NOTE:
A fax machine can also be considered a telephone adjunct, when used with an MFM.

Delay Announcements

A delay announcement recording is used when there is a delay before an incoming call is answered (such as with calling groups). Announcements can be made with industry-standard announcement devices connected to a station port on a 012 module or a 008 OPT module, or by an interface such as an MFM.

Telephone Adjuncts

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

Headsets and Headset Amplifiers

For hands-free operation of the telephone, four headsets are available:

- **StarSet Headset—** a monaural headset 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.
- **Mirage Headset—** 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.

- **Supra Headset—** a monaural headset with an adjustable headband. It offers a soft ear, comfortable cushion that reduces surrounding noise, making it easier to understand the caller.
- **Supra NC Headset—** a binaural headset with adjustable headband and soft ear cushion for working in noisy environments. Its noise-canceling microphone and voice expansion technology reduces up to 75% of the surrounding noise.

Each headset is light, comfortable and uses a transparent voice tube to eliminate the cumbersome large microphone. Each model comes with a 10-foot (305-cm) coiled cord and a Quick Disconnect latch.

These headsets work with any telephone connected to the communications system when combined with one of the amplifiers described in the following section.

See Figures 2-18 and 2-19.

The following amplifiers work with the supported headsets:

- **Modular Amplifier—** connects the StarSet, Mirage, Supra, and Supra NC headsets to virtually any telephone equipped with a modular handset. It is installed without special tools. You can adjust the incoming volume, switch between the headset and handset as needed, and temporarily mute the line.
- **Plug Prong Amplifier** (for non-MLX telephones)—connects the StarSet, Mirage, Supra, and Supra NC headsets to operator consoles, telephones equipped with a headset adapter, and many automatic call distributors. It provides switch-hook control for answering calls by pressing a button, You can also adjust the incoming volume.

Both amplifiers provide safety and comfort features that protect you from unexpected loud signals and provide clear sound. The amplifier stays with the telephone and works interchangeably with individual headsets.

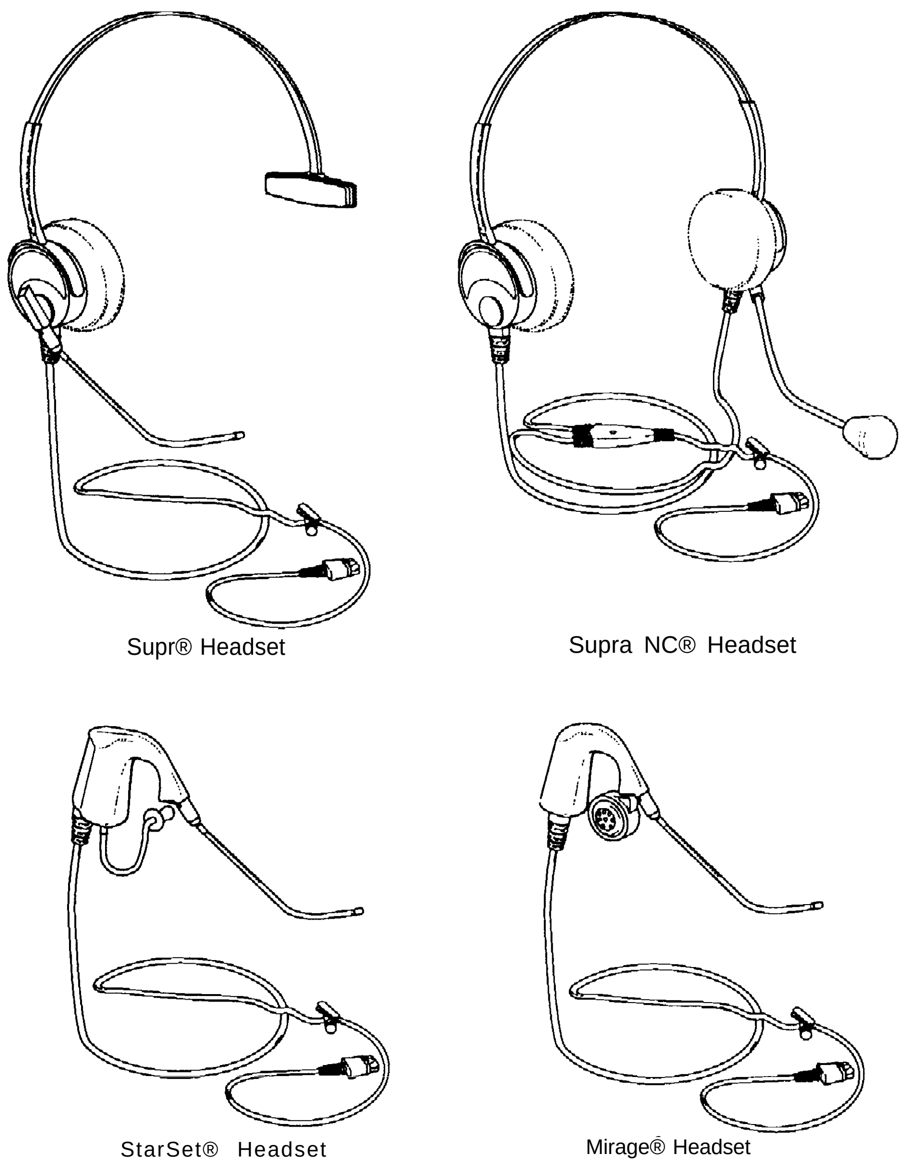


Figure 2-18. Headsets

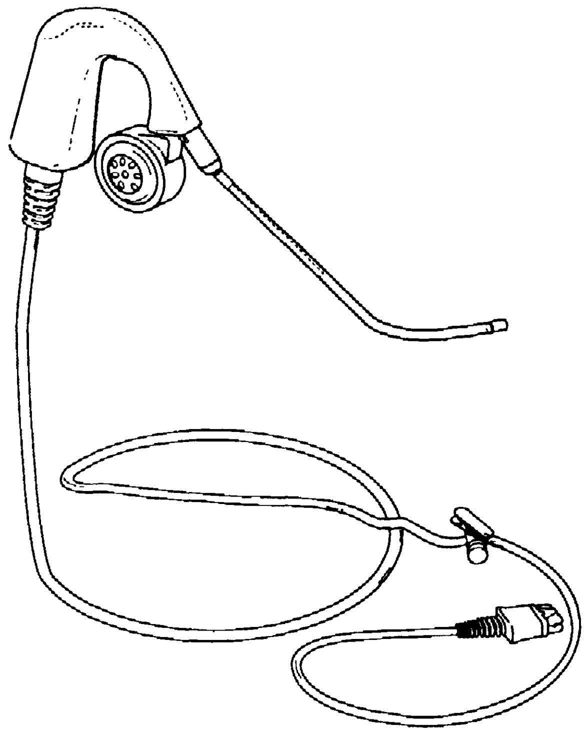


Figure 2-19. Analog Multiline Telephone Headset

Speakerphones

Separate speakerphones can be connected to single-line sets

Model S202A Hands-Free Unit (HFU) for analog multiline telephones allows calls to be made and received without using the handset. If an **Auto Answer Intercom button** is programmed and activated on the telephone, the HFU goes on automatically when a voice-announced call is received.

The S203A speakerphone allows voice transfer without using the handset. It is compatible with any 2500 or 500 phone and requires its own local power. Newer units may be shipped with a transformer, but older units require a KS21239L4 transformer and 248B adapter to be ordered separately.

The 4A speakerphone consists of a 108AA loudspeaker set, a 680-type transmitter, an 85B1 power supply, and a 223C connecting block. The 4A speakerphone must be used with a 2500SM telephone.

NOTE:
The quality of speakerphone transmission can be affected by equipment in the area of the microphone.

Specialty Handsets

Handsets for users who are hard of hearing (model K6S) are available for use with MLX telephones.

Message Waiting Indicator

The Z34A message-waiting indicator can be connected to single-line sets that do not have a Message LED.

Additional Telephone Adjuncts

The following adjuncts also can be connected to telephones:

- Answering machines
- Credit card verification terminals
- PCs (connected through modems, 75006 Data Modules, or directly with a built-in modem)

Adapters and Adjuncts Not Supported

CAUTION:
The following analog telephone adjuncts and adapters cannot be used with the system and in some cases will damage the device or the system if connected:

- 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)

Power-Related Hardware

Power-related hardware can be added to the system to provide more power and added protection from power surges. Other accessories apply to specific conditions.

NOTE:
In most cases additional power surge protection is not needed.

Power Accessories

In a power failure, battery backup units can keep the system running for several hours.

When adjuncts and adapters are connected to telephones, the power requirements of the telephones and the communications system increase. Adding a power accessory to an individual telephone or to the system accommodates these additional needs.

Battery Backup Power

Battery backup for power to the system can be provided by an optional 500 VA uninterruptible power supply (UPS) and reserve UPS units, The basic UPS provides power for 15 minutes. Reserve UPS units can be added to the basic UPS. Each reserve unit added extends backup power for an additional hour.

The holdover back-up durations for normal system operation of one full carrier at a 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

Connected between the telephone and the wall jack, the KS22911, L2 and 329A power units provide additional power to individual telephones that have adjuncts, adapters, and/or two DSSs attached, or to telephones far from the control unit. Adding local power to a few telephones can reduce the system load.

Table 2-9 shows local auxiliary power requirements. The KS22911, L2 power supply must be connected to an unstitched 117-VAC outlet.

Table 2-9. Local Auxiliary Power Requirements

Number of MLX Telephones	Number of DSSs	KS22911,L2 Number of MFMs	Number of Number of 329A Power Supplies	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 MI-X telephone after the first **2**	—

CAUTION:
When additional control unit carrier power is required and the system is backed up by an uninterruptible power supply (UPS), the carrier's auxiliary power unit (Supplemental Power Unit 9040-2) should also *be connected to the UPS.*

The total length of wire between the KS22911, L2 or 329A power supply and the MLX telephone can be no more than 50 feet (15 meters).

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

A KS22911 kit for MLX telephones comes complete with a D6AP cord, the KS22911, L2 power unit, and a 400B or 400B2 adapter. For analog multiline telephones, the KS22911 kit includes the KS22911, L2 power unit, a D6AP cord, and a Z400F adapter.

The 329A power unit does not come in a kit, so the D6AP cord and the 400B or 400B2 adapter must be ordered separately.

The MFM comes complete with a D6AP cord, the KS22911 power unit, and a 400B or 400B2 adapter. DSSs are shipped without power units. Therefore, when DSSs require local power, the KS22911, L2 or 329A power unit, D6AP cord, and 400B or 400B2 adapter must be ordered separately.

NOTE:
Telephone operation without adjuncts is guaranteed for a wiring run up to 1000 cable feet (305 meters) from the control unit.

Auxiliary Power Units

The power supply provides 54 unit loads to each carrier. If the unit load requirement for a carrier exceeds 54 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 unit loads per carrier may not appear to do harm. However, this can cause the system to malfunction, thereby creating "no trouble found" situations, such as malfunctioning LEDs on multiline telephones, or power unit failure.

Any station connected to the modules in the last two slots receives power from the auxiliary power unit instead of from the power supply.

If an auxiliary power unit is required, complete instructions are provided in *Installation.*

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

NOTE:
Only one auxiliary power unit can be connected to the 391A power unit. If additional 48-VDC power is needed, connect some telephones to KS-22911 or 329A telephone power units.

Protection Accessories

Certain accessories are used for grounding and protecting special telephone connections from power surges, electromagnetic interference, and electrostatic discharge.

IROB Protection

Equipment connected to the analog multiline telephone station jacks (on the 008, 408, 408 GS/LS-MLX (Release 2.0 only), and 408 GS/LS/TTR modules) or to the MLX telephone jacks (on the 008 MLX or 408 GS/LS-MLX module) that is located in a different building but within 1000 feet (305 meters) of the control unit requires in-range out-of-building (IROB) protection units. These units protect the equipment and the control unit from lightning strikes and power surges. Two units are required for each piece of equipment—one for the control unit end of the wire run, the other for the equipment end.

NOTE:
012 basic telephone ports may not be used for out-of-building or off-premises telephones.

Use the following IROB protectors:

- TII Model 343 for analog multiline telephones and equipment
- Model 505A for MLX telephones and equipment

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

See the documentation packaged with the IROB protector for complete installation instructions.

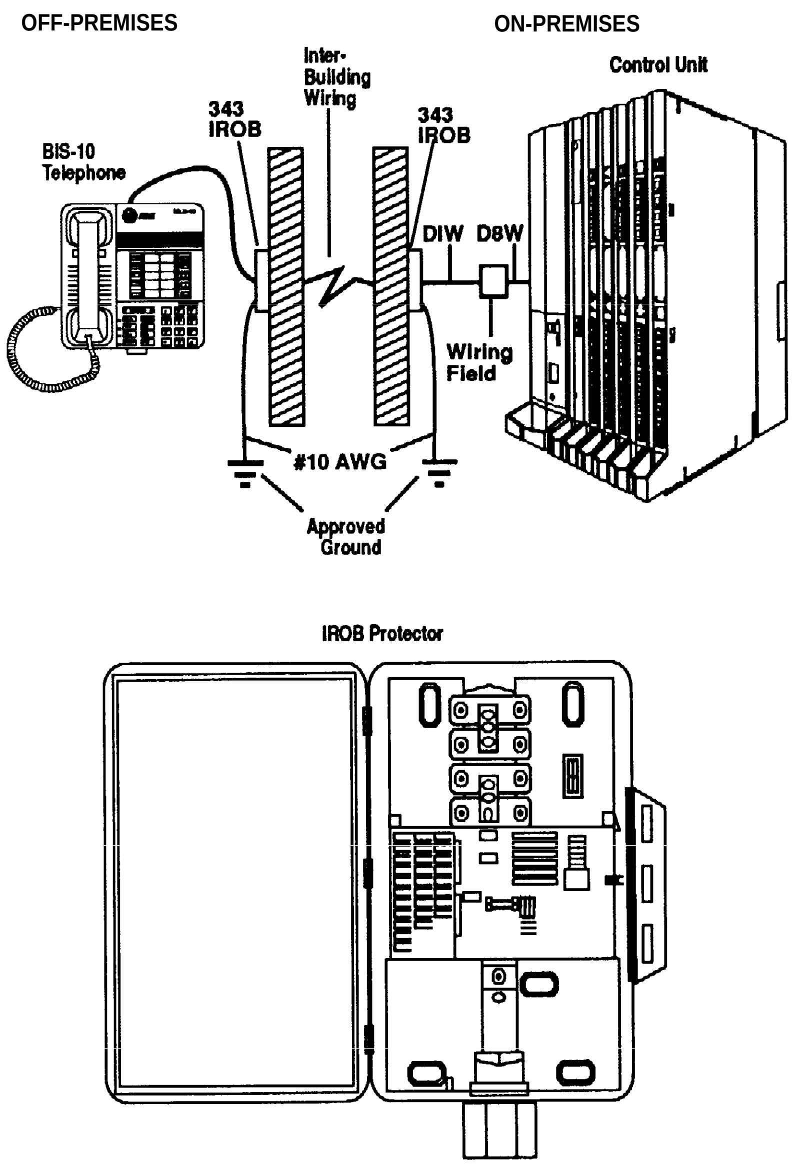


Figure 2-20. Analog IROB Connection

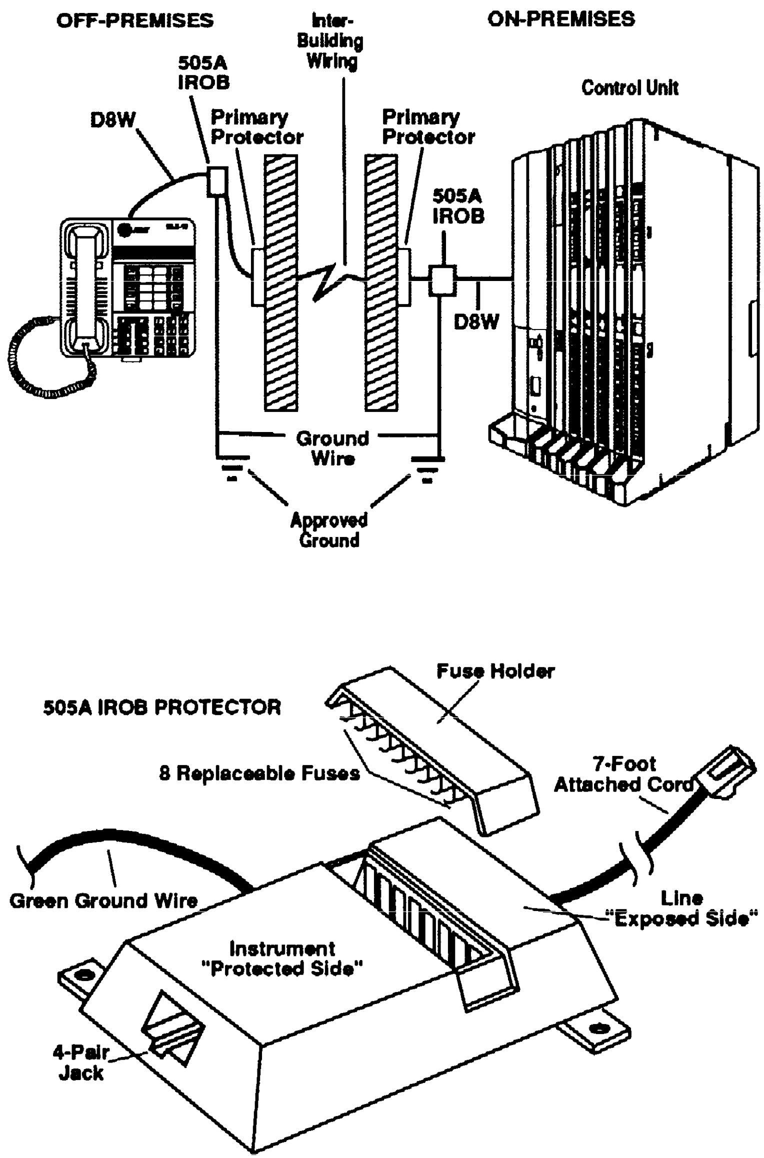


Figure 2-21. MLX IROB Connection

Off-Premises Range Extender (OPRE)

If a single-line or tip/ring telephone is located 1000 feet (305 meters) or further from the control unit, connect the telephone to the control unit using an off-premises range extender (OPRE).

Also, if the network interface is greater than 25 feet (7.6 meters) from the control unit, connect the control unit to the network interface using an OPRE.

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

146A and 147A Surge Protectors

Protection from lightning and power surges is needed to safeguard system functioning.

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 will be prevented.

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

Occasionally, additional AC line protection maybe needed if the customer is located in a heavy lightning area. A 147A protector an 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 sixteen outside lines. For more than sixteen lines, additional 147A protectors are required (see Figure 2-22).

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

Installation instructions are provided with the protectors. See Figure 2-22.

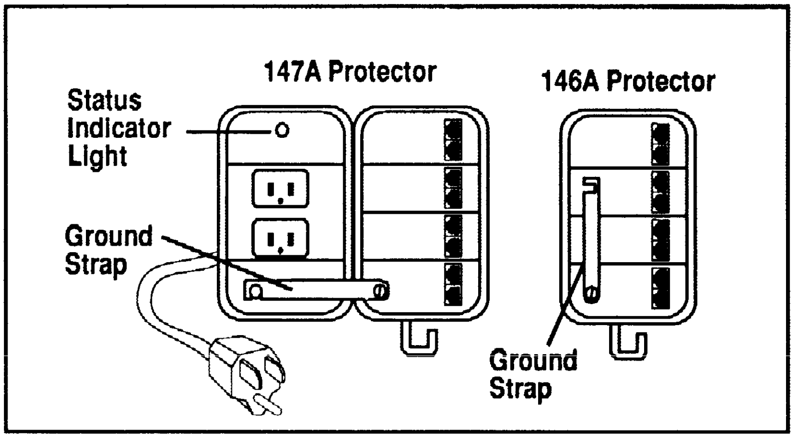


Figure 2-22. Surge Protectors

EMI Filter

The Z200A electromagnetic interference (EMI) filter can be connected to the system between the control unit and a telephone. Instead of a D8W cord, the filter cord is plugged into the telephone LINE jack. The Z200A filter *must* be installed with the SMDR printer.

ESD Suppression Kits

Electrostatic discharge (ESD) kits can be installed in older analog multiline telephones with membranes to eliminate damage to the telephone that can be caused by a voltage discharge resulting from electrostatic build-up.

Ring Generator

The 129B Frequency Generator (ring generator) must be added to the 391A1 or 391A power supply module when a 012 basic telephone module or a 008 OPT module is installed in the carrier. It provides a 105-VAC, 3(3-Hz ringing current used by the ringers on the single-line telephones connected to these modules.

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.

An UPAM is needed because 48-VDC alerting devices require four contact closures and the ground-start or loop-start ports only have two. The UPAM provides the additional two.

Trouble Alarm

A ground-start or loop-start power failure transfer (PFT) port can be used to activate an alarm by connecting the port to an UPAM (see Figure 2-23). When system trouble (software or hardware malfunction) is detected by the system operator console, a signal is sent to that port. The port's switching contacts close and send the signal on to the UPAM. The UPAM activates an external alert.

Power Failure Alarm

A ground-start or loop-start PFT port can be used to activate an alerting device during a power failure by connecting the port to an UPAM (see Figure 2-24). When a power failure occurs, the switching contacts on the PFT port close and send a signal to the UPAM, which activates an external alert.

NOTE:
A PFT telephone cannot be used on this port when the port is connected for a power failure alarm.

Power Failure DID Busy-Out

The PFT port on a ground-start or loop-start module can be programmed 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 telephone company's CO that the DID trunks are out of service. Figure 2-25 shows this connection.

NOTE:
Before the ground-start or loop-start module containing the PFT port for the DID busy-out is removed, the busy-out pair must be shorted and then the modular cord must be disconnected from the PFT jack. Otherwise, a false busy-out will occur. The short is removed after the system is powered up.

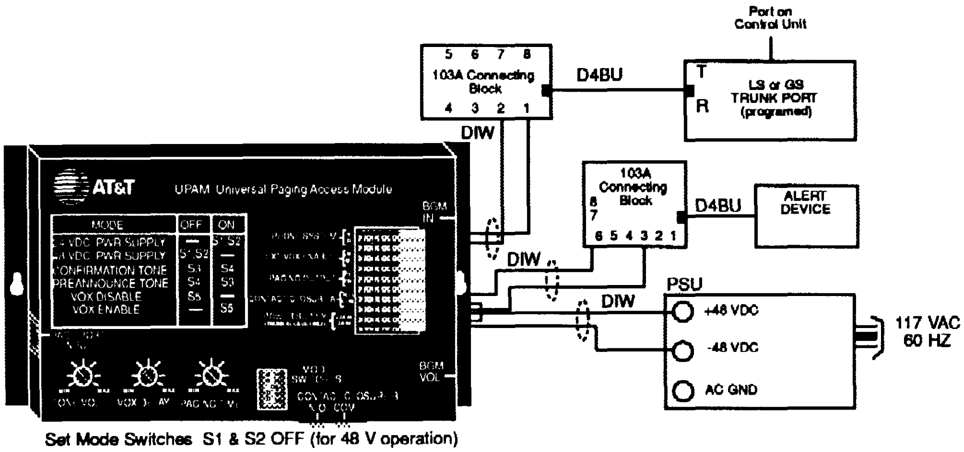


Figure 2-23. Trouble Alarm Connections

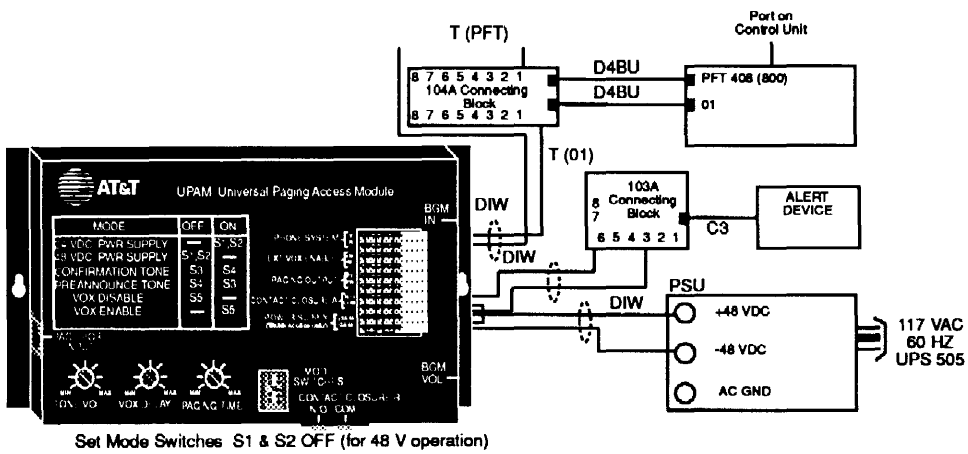


Figure 2-24. Power Failure Alarm Connections

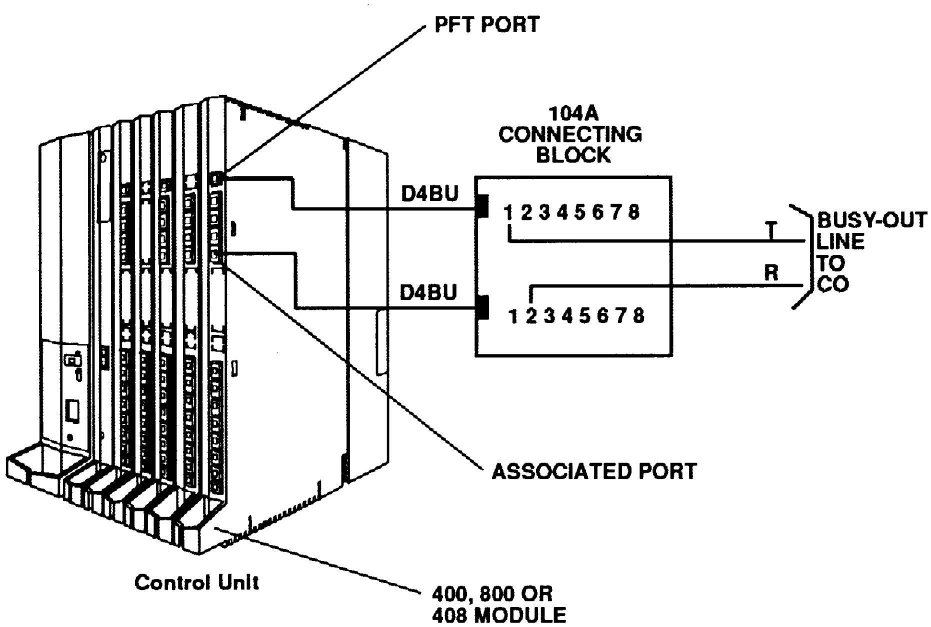


Figure 2-25. Power Failure DID Busy-Out Connections

Power Failure Transfer Telephone

A PFT telephone is a 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 only), 800, or 800 GS/LS 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.

NOTES:

- The PFT jack does not operate unless a power outage occurs or the power supply units are turned off.
- A single-line telephone connected to an MFM cannot be used as a PFT telephone.
- If PFT telephones are to be connected to ground-start lines/trunks, a ground-start button must be added to each PFT telephone. If power fails, this button is used when the number is dialed. If the button is used with modular 2500 sets, the button should be wired from the wall jack.
- If rotary lines/trunks are used, rotary telephones must be used as PFT telephones.

3 Lines and Trunks

Telephone lines and telephone trunks are facilities that carry voice or data communications. They are similar in form and function. The fundamental difference between a line and a trunk is as follows: a *line* connects a telephone to a switching system, and a *trunk* connects one switching system to another switching system. Most of the facilities that connect the system to the central office (CO) are properly referred to as trunks. However, a system that is configured for Hybrid/PBX mode supports personal lines. These facilities usually appear on a voice terminal button and pass transparently through the system to the CO. Selecting a personal line button on a voice terminal and lifting the handset brings dial tone directly from the CO.

NOTE:
The system cannot support digital data transmission via dedicated facilities (DS1/Tie/PRI) connecting two MERLIN LEGEND Communications Systems.

Loop-Start Lines/Trunks

Loop-start lines/trunks are the standard for home and small business Key systems. They are less expensive in some areas but have certain limitations:

- They do not protect against glare. (Glare occurs when a person tries to make an outside call on a line/trunk at the same time an incoming call is being received on that line/trunk.)
- They can have higher cable losses and, therefore, transmissions of less quality than ground-start lines/trunks.
- They cannot assure secure toll restriction.

Loop-Start Trunk Connection

The system's control unit can connect to another system's control unit from an off-premises telephone (OPT) line to a loop-start line/trunk (or vice versa) via analog facilities. A 008 OPT station module on system A's control unit can be connected to a loop-start port on system B's control unit. This enables the user on system B to access all the stations and facilities on system A. If System B has remote access, the user on system A can directly access stations and facilities on system B without operator intervention. Conversely, a loop-start port on system A can be connected to an OPT port on system B.

NOTE:
If the systems are on the same premises, the connection can be made to any T/R port.

With the OPT/loop-start connection, glare is more frequent as the volume of calls increases. In addition, if system B (the loop-start interface) does not have remote access, only the stations assigned to the loop-start facility on system B can be accessed by system A.

Ground-Start Lines/Trunks

Ground-start lines/trunks are outside lines/trunks used by some businesses (such as hotels or motels) where improved signaling is important. The improved signaling of ground- start allows more secure toll restriction. In addition, ground-start lines/trunks prevent glare. Ground-start lines/trunks also provide cable losses $\leq$ 4.5 dB.

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

- Basic lines/trunks (used for both local and long- distance calls)
- WATS (wide area telecommunications service)
- 800 service (in-WATS)
- Foreign exchange (FX)

Tie Trunks

Tie trunks provide private communication between two systems. Tie trunks “tie” the two systems together, making it seem that all the telephones are on the same system. 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 is connected to the control unit of another system via a 400EM module. if both systems are on the same premises, this module can be connected directly to the other system if the other system has similar tie-trunk facilities

  For off-premises connection, the 400EM module can be connected via the telephone company's facilities to another system.

  An analog tie-trunk connection can be administered 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 system's control unit is connected to the other system's control unit via a 100D line/trunk module programmed for T1-type transmission. A back-to-back connection from one DS1 facility to the other can be used when the total cable distance is less than 1300 feet (396 meters). To reach a remote system, the DS1 facility connects via a channel service unit (CSU) to the telephone company's facilities.

Tie trunks provide efficient communication between systems at different locations. These locations can be different floors of the same building, different buildings in the same campus, or different cities or states.

400EM Module Options

Tie trunks can be added to the system via the 400EM module. The 400EM module has four ports that must be programmed individually by selecting trunk options via system programming and setting the DIP switches (located on the front of the module) for E&M-protected, E&M-unprotected, or simplex signaling mode,

The following tie-trunk options need to be programmed via system programming, See Tables 3-1 and 3-2 for information on DIP switch settings.

- E&M Signaling Type

  — Type 1 Standard (the factory setting)—used when tie trunks are connected to the other system through the local telephone company.

  — Type 1 Compatible—used when tie trunks are connected directly to a system that uses type 1 S signaling and is located near this system.

  — Type 5—used when tie trunks are connected directly to a system that uses Type 5 signaling and is located near this system.

- **Direction**
  - — Two-way (factory setting)—Calls can be made in either direction.
  - Outgoing only—Calls can be dialed but not received (no ringing).
  - — Incoming only—Calls can be received but not dialed (no dialing).
- **Dial Mode**
  - — Rotary (factory setting)
  - — Touch-Tone

  NOTE:
  If the 400EM module is programmed for touch-tone dialing and there are no modules in the system that provide touch-tone receivers (TTRs) (012, 008 OPT, 400 LS/TTR, 400 GS/LS/TTR, or 800 DID), a 400 GS/LS/TTR module must be installed.

- **Dial Tone** determines whether the dial tone originates from the remote or local end of the line:
  - — 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** sets a time limit in milliseconds (ins) for the remote station to signal the calling station:
  - — 300 ms (factory setting)
  - — 20 to 4800 ms (increments of 20 ms)
- **Disconnect Time** sets a time limit in milliseconds for the release of the E or M lead:
  - 300 ms (factory setting)
  - — 140 to 2400 ms (increments of 10 ms)

- **Signaling Types** (also called *seizure type)*
  - — Wink (factory setting)—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—No start signal is necessary, and dialing can begin immediately after the tie trunk is seized.
  - — Delay—The originating end of the tie trunk transmits an off-hook signal and waits for the remote end to send an off-hook signal followed by an on-hook signal.
  - — Automatic—incoming calls are routed directly to another station without a start signal. When the user picks up the handset, the signal 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.*

If you are installing a 400EM module in the control unit, you need Form 3c, Incoming Trunks—Tie to determine the switch settings prior to installing the module in the control unit. For each 400EM module 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.

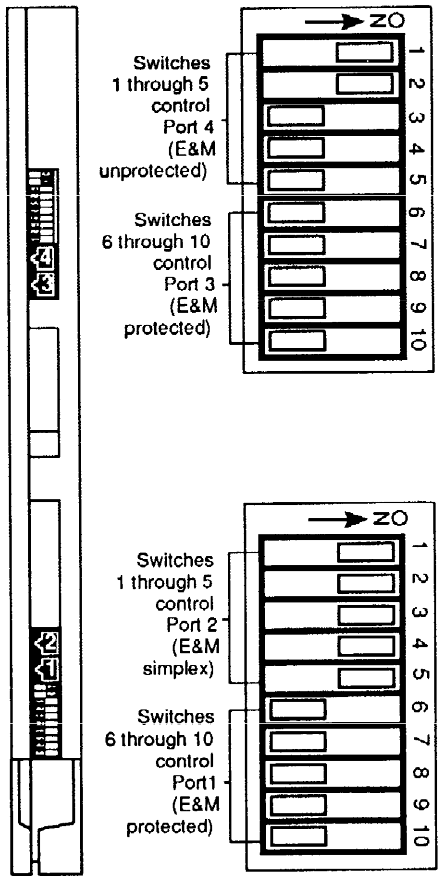


Figure 3-1. Setting the 400EM Module DIP Switches for E&M Signaling Types lC and 5

Table 3-1, Setting the 400EM Module DIP Switches

Ports		E&M Mode	E&M Signaling Type DIP Switch Position E&M Mode	1S (Default) Unprotected Mode	1C Protected
2	4	1	ON	OFF	NA
		2	ON	OFF	NA
		3	OFF	OFF	ON
		4	OFF	OFF	ON
		5	OFF	OFF	ON
1	3	6	ON	OFF	NA
		7	ON	OFF	NA
		8	OFF	OFF	ON
		9	OFF	OFF	ON
		10	OFF	OFF	ON

Table 3-2. Sample DIP Switches for the 400EM Module

Ports	E&M Signal	Switches
1 and 2	1C	Set all switches to OFF
3 and 4	1S	Default: no action required

Tie-Trunk Signaling

Tie trunks transmit via three different signaling formats, each made up of a specific mode and a specific type.

The dual in-line packaging (DIP) switches on the 400EM module select the signaling modes needed for tie-trunk transmission; the signaling type is selected during system programming.

Signaling Modes

There are two signaling mode:

- **Simplex mode.** Two signaling leads superimposed onto the analog transmission leads provide a 2-pair wire interface for connecting two local systems at minimal cost.

- **E&M mode.** This is a standard interface. The E&M signaling leads are isolated from the transmission leads, requiring a 3-pair wire interface.

In the simplex mode, protective resistance is always included in the circuit. The E&M mode can be either protected or unprotected from high-voltage transients or fluctuations. In the protected mode, a resistance is added to the leads to reduce current peaks, The protected mode is used when there is no network interface to protect the circuit from outside interference.

The unprotected mode must be used for an E&M Type 1 Standard interface (see below) to meet the specified voltage-drop criteria. This mode is used 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.

- **Type 1 Standard.** This is the factory-set type, which is used to connect two systems to the network through two intermediate telephone company COs. The switches must be set for E&M mode.
- **Type 1 Compatible.** This directly connects two systems without intermediate telephone company COs. One system is set to Type 1 Standard, the other to Type 1 Compatible. The switches must be set for E&M mode.
- **Type 5 simplex or E&M.** This type is used to connect similar systems or systems with compatible signaling that are located in the same building or on the same business campus.

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 3-3 shows how to determine tie-trunk compatibility between this system and other systems.

Table 3-3. Tie-Trunk Compatibility

Installation Situation		Preferred Signaling Format			
From system		System		Far End	
To	Location	Signaling Mode and Type	Protected or Unprotected	Signaling Mode and Type	Protected or Unprotected
System 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
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
Network Interface		E&M Type 1 Standard	Unprotected	N/A	N/A

E&M Tie-Line Ports

The E&M tie-line circuit module provides four tie-line ports. These ports may be individually configured for Type 1 Standard, Type 1 Compatible, or Type 5 (simplex or E&M).

Type 1 Standard is used to connect to the network. Type 1 compatible is used to co-locate systems that have a Type 1 Standard interface. Type 5 is used to connect co-located systems that have the Type 5 interface.

Table 3-4 shows the E&M option switch settings for the Type 1 Standard and Type 1 Compatible tie-line modules.

Table 3-4. Type 1 Standard and Type 1 Compatible E&M Switch Settings

Option	Switch Position	Switch Setting
Unprotected	1	ON
	2	ON
	3	OFF
	4	OFF
	5	OFF
Protected	1	OFF
	2	OFF
	3	OFF
	4	OFF
	5	OFF

Tie Trunk Networking

The system supports only nontandem tie-trunk networking. A nontandem tie-trunk network is used primarily to connect telephone lines at both ends; it does not connect to another tie trunk or to other facilities. See Figure 3-2.

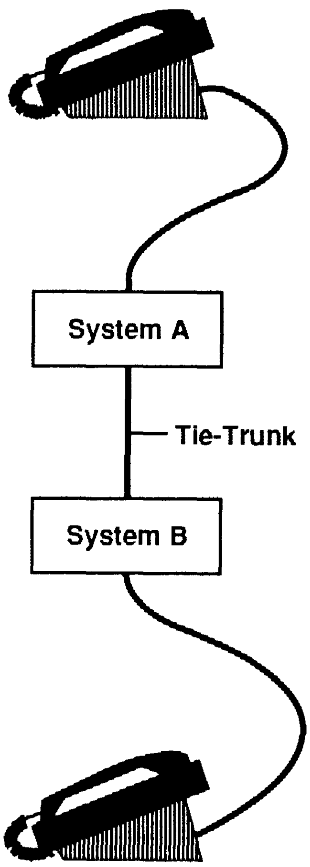


Figure 3-2. Nontandem Tie-Trunk Network

Direct Inward Dialing (DID) Lines/Trunks

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

For Release 2.0, DID functionality has been extended to DID-administered B-channels in the DS1 circuit board in T1 mode. B-channels administered for DID operation behave like tie trunks to the network and like DID trunks to the system switch.

With DID service, the customer reserves blocks of DID numbers from the local telephone company. The DID numbers should correspond to the extension number for an individual, a calling group, or a Remote Access or pool dial-out code.

CAUTION:
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 Customer Support Information in the front of this book for more information on security.)

The system can receive 1- to 4-digit extension numbers over the 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.

Because DID trunks allow calls to come directly to a telephone 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 used in the system are fewer than four digits but the CO sends four, the system can be programmed 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 number to connect.

The system also can be programmed to match more digits than are received from the CO. For example, if the system is set up 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.

No routing of calls is made until the designated number of digits is received.

Incoming DID numbers that don't match a valid extension are directed to a predesignated extension, such as the system console, or the system can be programmed to send back a reorder tone.

Options are assigned to blocks of DID trunks. A maximum of two blocks of DID trunks is allowed. Each block can be configured to match the system numbering plan. For example, the system could have both 3- and 4-digit extension numbers. Trunk block 1 could contain the options needed to reach the 3-digit numbers and trunk block 2 could contain the options needed to reach the 4-digit numbers, The following items must be programmed for each trunk group:

- Type of **DID Trunk**
  - — Wink-start (factory setting)—The preferred setting if the local telephone company can support it. It allows a greater probability of call completion during heavy calling periods,
  - — Immediate-start—The setting used when the local telephone company can support only immediate-start.
- **Signaling** sent from the local telephone company.
  - — Rotary (factory setting)
  - — Touch-tone
- **Expected Digits** sent by the local telephone company.
  - — 3 (factory setting)
  - — 1 to 4
- **Delete Digits** The number of leading digits that must be deleted from the digits sent by the local telephone company when the number of digits sent is more than in the chosen system numbering plan.
  - — 0 (factory setting)—Used when the number of digits sent by the telephone company matches the number of digits in the chosen system numbering plan.
  - — 0 to 4
- **Add Digits.** The specific leading digits that must be added to the digits sent by the local telephone company when the number of digits sent is fewer than the number of digits in the chosen system numbering plan.
  - — 0 (factory setting)—Used 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)

System Programming

The following system programming is needed in Release 2.0 to administer DID B-channels on the DS1 circuit module:

- A menu item to program a single B-channel on the circuit module as DID
- A menu item to program all B-channels on the circuit module as DID

- Expand the Inspect feature to show DID channels as well as LS/GS/TIE/Unassigned
- A new Inspect screen for DID channels

Programming a DID B-channel automatically adds it to the first DID trunk 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 circuit module and B-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 the administrator wants to move a DID trunk from one block to another, the trunk simply can be assigned to the new trunk block. The trunk is automatically removed from the previous trunk block and assigned to the new trunk block. 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.

The following options need to be programmed via system programming:

- **DID Trunk Dialing Protocol Type** describes the dialing protocol used for determining when address digits are sent from the CO to the Release 1.0 system.
  - Wink Start (factory setting)—the Release 1.0 system signals the CO when it is ready to receive incoming address digits.
  - Immediate Start—the CO sends digits about 65 ms after line seizure. (This is a criterion required of the serving CO.) Programming of the Immediate Start type for a trunk block that uses dual-tone multifrequency (DTMF) for passing address signs is not allowed.

- **DID Trunk Address Signaling Type** describes the method by which address signals are transmitted from the serving CO to the Release 1.0 system.
  - Dial Pulse (factory setting)
  - DTMF (not allowed for Immediate Start dialing protocol trunks)

- **Expected Number of Digits** (a number from 1 to 4, factory setting = 3) indicates the number of address digits expected from the CO on DID calls in this trunk block. Assignment of a value greater than 4 or less than 1 will be blocked.

- **Number of Digits to Delete** (a number from 0 to 4 digits, factory setting = 0) describes the number of digits to delete from the incoming address digits. A values greater than 4 will be blocked.
- **Digits to Add** (a number from 0 to 9999, factory setting = 0) describes the digits to prepend to the collected digits in order to determine a routing number,

The following options can be programmed on a DID trunk-by-trunk basis:

- **Trunk Disconnect Timing** (a number from 10 ms to 2550 ms, factory setting = 500 ms) is used by the DID circuit module to determine the time needed before a disconnect from the CO is considered valid.
- **Trunk Number** is the "line number, " assigned to every trunk or line in the system on startup, that serves as the trunk number. This trunk number can be changed by System Numbering.
- **Alphanumeric Label** is an ASCII string with up to seven characters (factory setting = OUTSIDE) that can be assigned to an individual DID trunk.

DID trunks can be programmed in a pool but on system resets these trunks are not placed in a default pool. The board renumbering mechanism automatically removes DID trunks from pools.

DID trunks can be assigned via system line programming as DFTs on any station in the system, but they will never be automatically assigned to any station in the system, including the operator position. The intent of having a DFT for a DID trunk is *not* for receiving or originating calls, but to allow monitoring the facility by observing the lamp.

DID trunks cannot be assigned as Music On Hold ports, page ports, or given direct access to calling groups.

You can program the DID feature in modes other than Hybrid/PBX; note, however, that the DID feature works *on/y* in Hybrid/PBX mode.

In Release 2.0, if the B-channel already has been programmed as a non-DID type trunk, the following items will block programming of the B-channel for DID operation:

- Trunk is programmed to ring into a DGC group
- Trunk is programmed for Remote Access
- Trunk is owned by a station for Coverage/Call Forwarding

DS1 Facilities

A Digital Signal 1 (DSl) facility is a transmission system that transports digital signals in the DS1 format. The interface that allows the connection of DS1 facilities to the system is the 1000 module. Through this module, voice and data calls can be made or received using a DS1 facility.

Twenty-four digital signal 0 (DS0) channels, each operating at 64 kbps, plus framing bits, are multiplexed, forming a DS1 signal of 1.544 Mbps. Each DS0 channel within the DS1 signal corresponds to a logical endpoint. Even though there is only one physical jack, the 10013 module supports up to 24 logical endpoints or ports (one for each channel).

In DS1 format, calls to other digital private branch exchanges (PBXs) or telephone company CO remain digital, and signals do not need to be converted to analog for acceptance by the connecting trunk. In addition, the 100D module can be configured to work with T1 or Primary Rate Interface (PRI) service.

To connect the 100D module to an outside DS1 facility, a CSU is used. The CSU regulates the transmission into and out of the 100D module so that the module matches the transmission of the outside facility.

Both ends of the DS1 facility must be able to communicate. To ensure this, the following options are set during system programming 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. The 100D module can be programmed to operate in either type of service. T1 service transmits and receives voice and analog data; PRI transmits and receives voice, and analog and digital data.

Any combination of the following AT&T Services Network (ASN) Services can be provided through a T1 or PRI line/trunk:

- Megacom WATS service for domestic long-distance outward voice calls; PRI on the system does not support access to international Megacom WATS service.
- Megacom 800 for domestic toll-free incoming voice calls. PRI on the system does not support access to international Megacom 800 service. T1 and PRI services support Megacom 800 with or without Dialed Number Identification Service (DNIS), also called Routing *by Dial Plan.* Dialed Number Identification Service (DNIS) is a service provided by the AT&T Switch Network that 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 should be routed to a specific individual or group responsible for accounts in the area.
- Software Defined Network (SDN)—ASN service; for 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 “uniform dialing plan,” which is necessary for complete integration with SDN. PRI on the system not support access to global SDN service.
- MultiQuest for domestic toll incoming voice calls (900 number). T1 and PRI support MultiQuest with or without DNIS.

In addition to these ASN services, T1 and PRI also support Shared Access for Switched Services (SASS) and Call-by-Call Service Selection. SASS allows both Megacom and Megacom 800 services to be offered over the same line/trunk facilities, eliminating the need to have separate incoming and outgoing line/trunk groups.

Like PRI, T1 also supports Megacom WATS and Megacom 800 on a shared line/trunk, but on a call-by-call basis. Call-by-Call Service Selection provides more than one outgoing PRI service, such as Megacom WATS, Accunet® Switched Digital 56/64, SDN, OUTWATS, Virtual Private Network Access, and Long Distance. PRI also provides Accunet switched digital service for 56-kbps, 64-kbps restricted, and 64-kbps clear circuit-switched data calls.

T1

T1 is the factory setting for DS1 facilities, allowing each of the 24 channels to be programmed to emulate tie, loop-start, ground-start, and DID lines/trunks in any combination. This means that a single 100D module can take the place of 24 regular outside lines.

If common-channel signaling (CCS) is selected, 23 channels are available for emulation and the 24th channel carries formatting signals.

The system’s control unit can be connected to another system’s control unit via a digital emulated tie trunk on a DS1 facility connected to a 100D module programmed for T1 -type transmission. A back-to-back connection from one DS1 facility to the other can be used when the total cable distance is less than 1300 feet (396 meters).

Primary Rate Interface (PRI)

The 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 through a 4ESS™ Generic 16, a 5ESS Generic 6, and a 5ESS serving the FTS2000 network.

A PRI line consists of 24 channels, sometimes referred to as DS0 channels, each with a capacity of 64 kbps. Each channel can be designated as either a B-channel (bearer channel) or a *D-channel* (data channel). DS1, then, refers to the twenty-four 64-kbps channels plus framing and signaling bits multiplexed together to form a 1,544-Mbps signal.

A B-channel is used to carry 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 of the system supports access to only one network service per B-channel. For Release 2.0, Call-by-Call Service Selection allows multiple network services over the same B-channels. The D-channel conveys signaling required to set up, control, and clear calls made over all of the B-channels.

The most common configuration of a PRI consists of 23 B-channels and 1 D-channel, although other combinations are possible. Each PRI must include a D-channel, but may include fewer than 23 B-channels. The remaining channels cannot be used for any other purpose.

Up to three PRIs can be connected to the system through separate 100D circuit modules, each of which may occupy a slot in the system carrier. 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. Their signaling is provided over three separate D-channels.

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, a framing bit appears in every 193rd bit position of the 1.544-Mbps DS1 signal.

Frames repeat at a rate of 8000 per second, with each frame repeating DS0 channels 1 through 24 sequentially.

Two methods of framing can be used by a 100D module (T1 service): D4 or Extended Super Frame (ESF). The framing method chosen must match the framing at the far end, and must be programmed to the format selected when service was ordered.

To identify the DS0 channels, the DS1 signal is segmented

- **D4 Framing Format.** The system is factory set for the most common framing format, D4 framing. A D4 frame consists of 24 eight-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 to identify which frames contain signaling information. The framing information repeats once every 12 frames; these 12 frames form the 134 superframe. This framing format is used by most DS1 equipment.

- **ESF Framing Format.** The ESF format extends the 12-frame D4 superframe to a 24-frame superframe, hence its name. 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, ESF is not used universally by DS1 equipment.

Line Coding

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

To meet the ones-density requirement, either AMI-ZCS or bipolar 8 zero substitution (B8ZS) line coding can be chosen.

AMI-ZCS

AMI-ZCS line coding monitors each DS0 channel and prevents strings of eight or more zeros. Upon detecting an all-zero channel octet, AMI-ZCS line coding forcibly changes the seventh zero (second least significant bit) to **a** one. The factory-set line coding is AMI-ZCS.

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

B8ZS

B8ZS line coding inserts eight consecutive zero bits into a unique binary sequence with a "bipolar violation" in bit positions 4 and 7. Normally for bipolar transmission, ones are encoded alternately as a positive then negative, or negative then positive, pulse. If two positive or two negative pulses are received in succession, a bipolar violation occurs,

Ordinarily, bipolar violations are caused by noise hits to the signal, However, B8ZS line coding allows the 8-bit strings to be detected at the receiving end and converted back into the original sequence.

B8ZS line coding is preferred over ZCS because it does not cause errors in data transmission.

B8ZS violations are passed by the ESF T1 CSU but not by other CSUs. The CSU is a hardware component needed when two endpoints are located in different buildings or when the distance between the two endpoints makes office or line repeaters necessary. 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:

- It terminates an outside DS1 facility on the 100D module.
- It ensures that the signals entering the public network comply with the requirements of the DS1 facility as specified by the FCC.
- It includes maintenance, diagnostic, and testing capabilities.

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. The other possible settings are shown in Table 3-5.

Table 3-5. Line Compensation Settings

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

NOTE:
Cable length in Table 3-5 is the distance between the 100D module and the CSU. If no CSU is used, the distance between 100D modules is twice these numbers.

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, the module that provides the primary synchronization for the other 100D modules and for the time-division multiplexing (TDM) bus must be identified during system programming.

Backup synchronization in the event of a maintenance failure can be provided by programming the second and third installed modules as secondary and tertiary synchronization.

In addition, the source of synchronization can be factory set to “loop clock reference source” (the clock is synchronized to the external endpoint—the factory setting) or set to “local clock reference source”

Signaling Mode

Signaling is the process of communicating channel-state information (such as dialing) from endpoint to endpoint. Two types of signaling can be used in T1 transmission: robbed-bit signaling (RBS) and common-channel signaling (CCS).

Choosing a signaling mode pertains only to T1 service; PRI always uses CCS (23 B-channels and 1 D-channel).

Robbed-Bit Signaling

RBS replaces (“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 data (up to 19.2 bps), but facilities using RBS cannot transmit digital data at 64 kbps because this bit-robbing corrupts data. Digital data at 56 kbps may be possible in certain applications by using 7-bit words.

Common-Channel Signaling

CCS is an format that places the signaling bits for channels 1 through 23 into the 8-bit word of the 24th channel. This restricts DS1 from using the 24th 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, CCS can support digital data up to 64 kbps per channel.

Recommended Framing Formats and Signaling Modes

ESF framing should be used to take advantage of its improved maintenance, diagnostic, and testing capabilities (the ESF T1 CSU is required to interface with the network). If the transmission between two systems is voice-only, RBS should be used for all 24 communication paths. For voice transmission, both ZCS and B8ZS line coding can be used to satisfy the ones-density requirement; the preferred line-coding format is B8ZS, which is needed for 64 kbps digital data.

The framing and signaling formats depend on the network and interconnection devices (CSUs) used. For example, many CSUs only support ZCS line coding.

NOTE:

- Digital data up to 64 kbps is possible only in PRI mode.
- An ESF-T1 CSU must be used for interbuilding DS1 connections.

4 Applications

> **CAUTION:**
> *This chapter is intended solely as an overview of the applications that can be connected to the system. For information about the use of any application listed here, see the documentation for that product.*

The following applications can be connected to the system for enhanced call-handling and system management capabilities:

- Standalone voice messaging applications
  - MERLIN MAIL Voice Messaging System
  - MERLIN Attendant
- Standalone call accounting and management applications
  - Call Accounting System (CAS)
  - Call Accounting Terminal (CAT)
  - Call Management System (CMS)
  - InnManager Guest Management System
- Standalone system management application
  - System Programming and Maintenance (SPM)
- Integrated applications

  Integrated Solution II (IS II) incorporates the following applications:
  - AUDIX Voice Power
  - CAS
  - SPM

Integrated Solution III (IS Ill) incorporates the following applications:

— AUDIX Voice Power
— CAS
— SPM
— Fax Attendant
— CONVERSANT Intro

- Primary Rate Interface (PRI) applications

— Group IV (G4) Fax
— Video Conferencing

- Optional telephone service

— Centrex operation

- Fax services

— MERLIN PFC Telephone
— Automated Document Delivery System (ADDS)

- **Voice response system**

— CONVERSANT Intro

This chapter provides a brief description of each of these applications, services, and systems. The descriptions are organized under the following subheadings; any subheading not applicable to a given application is omitted.

- Mode Differences—differences or limitations of the application in the Key, Hybrid/PBX, or Behind Switch modes of operation.
- Considerations and Constraints—restrictions, capacities, and other information to be considered before installing or using the application.
- Feature Interactions—system and telephone features that affect how the application works, and any features that do not work with the application.
- System Programming—an outline of the communications system programming required to set up the application. See *System P/arming* for planning instructions and *System Programming* for system programming instructions. Also see the documentation provided with the application for instructions to program the application.
- Platform Requirements—additional hardware and software required to connect the application to the system. (See *Installation* and the documentation provided with the application for connection diagrams and installation instructions.)

Table 4-1 summarizes the capacity of the system to support each application and the modes of operation in which it can be used.

Table 4-1. Application Capacities and Modes of Operation

Application	Capacity	Key	Hybrid/ PBX	Behind Switch
MERLIN MAIL Voice Messaging System	1 * (2 or 4 ports)	√	√	
Number of mailboxes	40			
MERLIN Attendant	4*	√	√	
CAS	1	√	√	√
CAT	1	√	√	√
CMS	2	√	√	
Number of lines/trunks (each)	28			
Number of agents (each)	28			
Number of external alerts (each)	4			
InnManager Guest Management System	1	√	√	√
SPM (standalone)	1	√	√	√
IS II	1	√	√	√
AUDIX Voice Power	1	√	√	
Number of mailboxes	300			
Integrated Voice Power	1	√	√	
Automated Attendant	1	√	√	
CAS—IS II	1	√	√	√
SPM—IS II	1	√	√	√
IS III	1	√	√	√
AUDIX Voice Power	1	√	√	
Number of mailboxes	300			
CAS—IS III	1	√	√	√
SPM—IS III	1	√	√	√
Fax Attendant	1	√	√	
PRI				
Group IV (G4) Fax		√	√	
Video Conferencing			√	
Centrex		√	√	√
ADDS	1 voice port and 1 fax port	√	√	√
CONVERSANT Intro	1	√	√	√

* These attendant applications are mutually exclusive.

Voice Messaging Systems

CAUTION:
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 voice messaging system (VMS) is an application that provides call answering services and may provide voice mail services as well. Each of the following VMS applications is connected to an enhanced tip/ring port, called a *voice messaging interface* (VMI) port.

- MERLIN MAIL Voice Messaging System
- MERLIN Attendant
- AUDIX Voice Power (IS III)
- Integrated Voice Power (IVP) Automated Attendant (IS III)

CAUTION:

Your Voice Mail system permits callers to leave verbal messages for system users or gain access to the back-up 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 pass words 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.*
  - *Assign toll restriction levels to all AUDIX Voice Power ports.*
  - *If you do not need to use the Outcalling feature, completely restrict the outward calling capability of the AUDIX Voice Power ports.*
- *Monitor SMDR reports or Call Accounting System reports for outgoing calls that might be originated by AUDIX Voice Power ports.*

Tip/ring ports on an 012 module can be programmed either as *generic VMI ports* or *integrated VMI ports.* The MERLIN MAIL Voice Messaging System and AUDIX Voice Power use streams of touch-tone codes, called *mode codes,* to communicate with the system's control unit. Because they use mode codes, these applications must be connected to integrated VMI ports, MERLIN Attendant and IVP Automated Attendant, which do not use mode codes, connect to generic VMI ports.

Mode codes are categorized in two classes: Call Information and Other. Table 4-2 lists the types of mode codes in each class.

Table 4-2. Mode Codes

Call Information	Other
Direct inside access	Leave word calling
Direct dial access—line/trunk	Refresh message-waiting LEDs
Call coverage—internal station	Port taken out of service
Call coverage—external (line/trunk)	Port restored to service
Call coverage-other	Day service
	Night service

A VMS requires touch-tone receivers (TTRs); the number required depends on the number of VMI ports, as shown in Table 4-3. Note that these TTR requirements are *only* for a VMS and do *not* include the TTR needs of tip/ring sets.

TTRs are supplied by the following modules: 012, 400, and 400 GS/LS/TTR. (The 008 OPT module also supplies TTRs, but does not support VMS applications.)

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.

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 or 1	0 1
8	4	2 or 1	0 1
12	6	3 or 2	0 1

VMI Port Capabilities

VMI ports use switchhook flashes in the same way single-line telephones do for Hold, Transfer, Conference, and Drop. VMI ports also have the ability to perform transfer redirection, respond to far-end disconnect, and, for integrated VMI ports only, mark a port in service or out of service. These capabilities are described in the following sections.

Transfer Redirect

If unanswered by the end of the transfer redirect time interval (programmable for 0 to 9 rings), a call transferred from a VMI port will alert at the VMS transfer redirect extension, rather than return to the VMI port that originated the transfer.

For example, suppose station port 15 is programmed as a VMI port connected to a MERLIN Attendant, and the programmed transfer redirect time interval is 4 rings. A call comes in on port 15, and after listening to the recorded prompt, the caller dials a request for extension 24 (station port 24). The call rings at station 24 for four rings without being answered. The call is then redirected to station port 10, the system operator. It is not redirected back to port 15.

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

Far-End Disconnect

When a far-end disconnect signal is detected on an outside line/trunk on which a call is made to or received from a VMI port, the system sends the disconnect signal to the VMI port, whether or not that port 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 the LS/GS/DID/Tie VMI port is programmed for reliable disconnect.)

Ports In/Out of Service

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

For an integrated VMI port, the control unit sends mode codes to inform the VMS that the port is out of service. Both the VMS and the calling group software mark the unavailable port out-of-service. If all VMI ports go out of service, a programming logic inconsistency (PLI) is generated.

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

MERLIN MAIL Voice Messaging System

CAUTION:
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 is a standalone application that provides the following integrated call-handling services:

- Automated Attendant Service
- Call Answer Service
- Voice Mail Service

Automated Attendant Service

The 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 the Automated Attendant routes the call to an internal 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 without a touch-tone 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.

Calls can be answered immediately (Immediate Call Handling) or after a delay (Delayed Call Handling), for example, if the call goes unanswered by the system operator after a specified number of rings.

The Automated Attendant Service can recognize fax tone on an incoming call and direct the call either to a single fax extension or to a calling group serving multiple fax machines. If the fax machine is busy or does not answer within four rings, the call is automatically disconnected.

The Automated Attendant Service can be programmed to transfer calls in either of three ways:

- **Unsupervised transfer—** in combination with the system's Coverage feature, the Automated Attendant dials the extension or department requested by the caller and disconnects. If the call is not answered or the extension is busy, the call is routed to the system operator or, if the user is a registered subscriber, returns to the Automated Attendant,

- **Supervised transfer—** the Automated Attendant transfers the call and can retrieve it if the transfer is unsuccessful. If the called party is not a subscriber, the caller can opt to be transferred to another location.
- **No transfer—** the Automated Attendant transfers the call, and the caller is prompted to leave a message.

Call Answer Service

When a caller reaches a busy or unanswered extension, the Call Answer Service connects him or her to the personal mailbox of the subscriber associated with that extension, where the caller can leave a message. If the subscriber has recorded a personal greeting, the caller hears it; otherwise, a general greeting including the subscriber's name is played.

If the subscriber's personal mailbox is full, the Call Answer Service connects the caller to a general mailbox and plays a message including the subscriber's name. The caller can leave a message in the general mailbox; the system operator is responsible for forwarding the message to the appropriate subscriber,

If the general mailbox is full, the Call Answer Service informs the caller and allows him or her to transfer to another extension.

When a message is left in a subscriber's personal mailbox, the system lights the message-waiting LED on his or her telephone. When a message is left in the general mailbox, the general mailbox owner's (typically the system operator) message LED goes on.

With Outcalling, when a user, or *subscriber,* receives a new message, the system can automatically call a number that he or she has programmed, for example, a beeper or a home telephone. The subscriber can then log in to the Voice Mail Service to retrieve messages.

Voice Mail Service

The Voice Mail Service allows subscribers to send messages to other extensions in the system, forward messages received with comments, and return a call to an extension that has left a message. A subscriber can also record a personal greeting and program a password to prevent others from retrieving messages from his or her personal mailbox. The system manager can broadcast a message to every system subscriber. A broadcast message does not light message LEDs and does not cause outcalling.

In addition, the system manager can create group lists of subscribers. Any subscriber can send a message to a group list.

Additional features include the following:

- **Outcalling** automatically calls the user at a number the user has programmed when a new message is received. The user can then log in to the VMS to retrieve messages.

- **Pager Notification** calls the user at a designated number when a new message is received. However, the user cannot log in to the VMS.
- **Broadcast Lists** allows the system administrator to send a message to every user on the communications system. However, this message does not light message LEDs and does not cause outcalling.
- **Fax Transfer** directs an incoming fax call to a designated fax station. This fax station can be a single machine or a calling group with several machines.
- **Announcement Service** allows a caller to enter a code to hear information about specific subjects, such as new product information or marketing programs.

Mode Differences

The system must operate in Hybrid/PBX or Key mode. The MERLIN MAIL Voice Messaging System cannot be connected to a system operating in Behind Switch mode.

Considerations and Constraints

- Only one MERLIN MAIL Voice Messaging System can be connected to the system.
- The MERLIN MAIL Voice Messaging System 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 length of the messages it can hold—can be set up to match individual needs, up to a maximum size of 60 minutes. Available options are 5, 10, or 15 minutes; 60-minute storage is available for special mailboxes.
- Callers who dial from rotary telephones cannot use the features of the MERLIN MAIL Voice Messaging System and should be directed to the system operator during business hours.
- The Automated Attendant Service can answer calls immediately *(Immediate Call Handling)* or after a delay *(Delayed Call Handling),* for example, when a call remains unanswered by the system operator after a certain number of rings.
- Programming is done via a touch-tone telephone. The MERLIN MAIL Voice Messaging System is equipped with an RS-232 serial port and an external modem to support remote diagnostics.
- Call restrictions should be assigned to the VMI ports that connect the MERLIN MAIL Voice Messaging System to the system so that toll calls cannot be dialed through this application and so that MERLIN MAIL is not prohibited from outcalling.

- The MERLIN MAIL Voice Messaging System cannot be used with MERLIN Attendant.

Feature Interactions

Coverage

- All extensions that need coverage are assigned to a coverage group through system programming The MERLIN MAIL Voice Messaging System ports are assigned to a calling group designated as the coverage receiver for the coverage group.
- An internal call on a VMI port that transfers to an internal extension will not go to coverage, but will continue to ring at that extension.
- If a sender's telephone is programmed so that only outside calls are sent to coverage, calls received on **ICOM** or **System Access** buttons are not sent to voice mail.
- For Release 2.0, outside calls that would normally proceed to the MERLIN MAIL Voice Messaging System as coverage do not do so if the telephone that sends the call to group coverage has activated Coverage VMS. No special action is needed on MERLIN MAIL administration to activate this feature.

Group Calling

- All VMI ports to which the MERLIN MAIL Voice Messaging System is connected are assigned to the same calling group through system programming.
- Calls that overflow from one calling group to another calling group with integrated VMI ports are identified as coverage calls via mode codes. As a result, the overflow calling group's number appears in the called-party field of the mode code.
- For Release 2.0, when the MERLIN MAIL Voice Messaging System sends a Leave Word Calling message 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.

Leave Word Calling

- Leave Word Calling sends mode codes to the MERLIN MAIL Voice Messaging System to deposit a canned message if the target telephone does not have display capabilities.

Night Service

- The MERLIN MAIL Voice Messaging System Automated Attendant Service works with the Night Service feature to provide specialized after-hours service. The Automated 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

- Privacy is automatically programmed for each VMI port connected to the MERLIN MAIL Voice Messaging System.

Ringing Options

- If lines set for answering by the Automated Attendant Service also appear on telephones other than the system operator console or a backup extension, they should be programmed for no ring.

Transfer

- Integrated VMI ports can transfer an incoming call to an outgoing line/trunk.
- 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 internal greeting.
- If a caller incorrectly specifies the answering VMI port as the desired transfer destination station, the VMI port may park the call.
- Any calling group, calling group member, or station can be programmed to be a VMS transfer redirect extension. If the station is a Queued Call Console (QCC), 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 operator has no indication of the origin of the call.

System Programming

- Assign all the MERLIN MAIL Voice Messaging System ports to a calling group, set the group type to VMI Integrated, and set the hunt type to linear.
- Program VMI loop-start ports for the MERLIN MAIL Voice Messaging System for reliable disconnect.
- Specify the touch-tone duration and interval between digits in codes sent between the MERLIN MAIL Voice Messaging System and the communications system.

- Specify the VMS Transfer Return Interval, that is, the number of rings before a call transferred by the MERLIN MAIL Voice Messaging System is sent to the backup position (system operator).
- Set Inside (intercom) Dial Tone to outside.
- Assign call restrictions to each VMI port used to connect the MERLIN MAIL Voice Messaging System.
- When the Automated Attendant is used only for Night Service:
  - If the lines/trunks set for answering by the 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

The following equipment is required to connect the MERLIN MAIL Voice Messaging System to the system:

- MERLIN MAIL Voice Messaging System unit and power cords
- Remote maintenance device (a modem with a wall-mounted transformer)
- Modem cable with a 9-pin connector at one end and a 25-pin connector at the other, for connecting the modem to the serial pod on the MERLIN MAIL Voice Messaging System unit
- D4BU modular cords (two for a two-port system or four for a four-port system, plus one for the modem)
- 012 module (a ring generator is required)

NOTE:
Additional TTRs may be needed to allow the 012 module to handle a large number of voice connections.

If the MERLIN MAIL Voice Messaging System is to be used only for backup call handling or night service, only one VMI port may be required. For other uses with heavier call traffic, the number of VMI ports required 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 4-4 shows these requirements.

Table 4-4. MERLIN MAIL Voice Messaging System Ports Required

No. of VMI Ports Required	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 Attendant

CAUTION:
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 Attendant is a standalone application that answers incoming calls and plays a menu of recorded prompts. A caller can respond to the prompts by dialing touch-tones, and the MERLIN Attendant routes the call to an internal extension accordingly. A caller without a touch-tone telephone is transferred to the system operator for further call handling and routing.

The MERLIN Attendant can be programmed to transfer calls in either of two ways:

- **Unsupervised transfer—** the MERLIN Attendant dials the extension or department requested by the caller and disconnects. If the call is not answered or the extension is busy, the call is routed to the system operator or goes to the redirect extension.
- **Supervised transfer—** the MERLIN Attendant transfers the call and can retrieve it if the transfer is unsuccessful. The MERLIN Attendant then directs the call to another telephone, allows the caller a second route choice, or plays a failed-transfer announcement, depending on how the application is programmed.

Calls can be answered immediately (Primary Call Handling) or after a delay (Secondary Call Handling), for example, if the call goes unanswered by the system operator after a specified number of rings.

Mode Differences

The system must operate in Hybrid/PBX or Key mode. The MERLIN Attendant cannot be connected to a system operating in Behind Switch mode.

Considerations and Constraints

- The MERLIN Attendant cannot be connected to a communications system that has an AUDIX Voice Power or MERLIN MAIL Application installed.
- A maximum of four MERLIN Attendants can be connected to the system.
- The MERLIN Attendant can be programmed to answer every incoming call or in or only calls on certain lines/trunks.
- Unanswered transferred calls do not return to the MERLIN Attendant, but are redirected to a designated extension, such as the system operator,

- If the extension called is busy or unanswered, or after business hours, calls can be directed to an answering machine to allow callers to leave messages.
- The MERLIN Attendant can transfer calls to fax machines, if the fax extension number is specified and the caller dials it.
- The MERLIN Attendant provides 64 seconds for recording up to five 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.
- When the MERLIN Attendant is set up for after-hours operation, the time on its clock must match the system clock.

Feature Interactions

Coverage

- An internal call on a VMI port that transfers to an internal extension will not go to coverage, but will continue to ring at the internal extension.
- For Release 2,0, outside calls that would normally proceed to the MERLIN Attendant as coverage do not do so if the telephone that sends the call to group coverage has activated Coverage VMS. No special action is needed on MERLIN Attendant administration to activate this feature.

Group Calling

- All MERLIN Attendants connected to the system must be assigned to the same calling group through system programming.

Night Service

- The MERLIN Attendant works with the communications system's Night Service feature to provide specialized after-hours service. The MERLIN 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

- Privacy must be programmed for each VMI port connected to the MERLIN Attendant.

Transfer

- If a caller incorrectly specifies the answering VMI port as the desired transfer destination station, the VMI port may park the call.
- Calls on generic VMI ports cannot be transferred to telephones that have Remote Call Forward activated.

System Programming

- Assign all the MERLIN Attendant ports to a calling group and set the group type to VMI Generic.
- Set Inside (intercom) 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. (Auto-logout is the default.)
- Assign Privacy to each VMI port used to connect the MERLIN Attendant.

Platform Requirements

The following equipment is required to connect the MERLIN Attendant to the system:

- MERLIN Attendant unit
- 6-wire modular telephone cord
- 012 module (a ring generator is required)

NOTE:
Additional TTRs may be needed to allow the 012 module to handle a large number of voice connections.

The number of MERLIN Attendants required 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 4-5 shows these requirements when the MERLIN Attendant is programmed for Primary Call Handling.

Table 4-5. MERLIN 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 (CAS)

CAUTION:
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.

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 Ill**
- **CAS Plus V3** for general business use, a standalone application that runs on an approved AT&T DOS personal computer (PC)
- **CAS/Hospitality (CAS/H)** for hotels and health care facilities, a standalone application that runs on an approved AT&T DOS PC

All three versions allow businesses to calculate the cost of calls by using the rates charged by long-distance carriers in one of 11 major metropolitan areas. In addition, CAS Plus V3 and CAS—IS Ill can be customized by programming additional rate tables.

CAS Plus V3, CAS/H, and CAS—IS Ill provide the following services and features:

- **Call Record Processing—** records of calls are collected and stored, and costs are calculated using the rate table selected. The system can be programmed 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, projects, or (with CAS/H) rooms.

  In addition, CAS-IS III collects and processes automatic number identification (ANI) information (caller identification), and can provide detailed information on incoming calls by point of origin. However, the availability of this information may be limited, depending on the legal jurisdiction and the equipment at the telephone company central office (CO) serving the caller.

- **Report Generation—** stored call record information can be organized and printed in the following kinds of reports:
  - 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 or (with CAS/H) by room.

— 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 perform a variety of customization and maintenance activities, such as editing tables, setting up reports, and keeping call rate information up to date.
- **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.

Considerations and Constraints

- Only one CAS can be connected 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 5,000 extensions and 15,000 account codes.
- When an industry-standard T/R device, such as a fax machine with a built-in telephone, is attached to an MLX telephone via a Multi-Function Module (MFM), the connected T/R device cannot enter an account code,

  When the T/R device is connected to an analog multiline telephone via a GPA, the same is true. However, an Account Code Entry button can be programmed in the telephone. A user can then place a call from the telephone, enter an account code, and then go off-hook on the associated T/R device. Or a user might place a call from the built-in telephone on the fax machine, press the **Account Code Entry** button on the analog multiline telephone, enter the account code, and press the **Account Code Entry** button again. The account code is then captured along with the other call information.

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

Platform Requirements

The following equipment is required to connect Standalone CAS software (CAS Plus V3 or CAS/H) to the system:

- An AT&T 286/386 PC, configured as follows:
  - — MS-DOS 3.1 or higher
  - — 64K RAM
  - — 20 MB hard disk
  - — 1 parallel port
  - — 2 serial ports
  - — CGA/VGA/EGA, Super VGA, or Hercules monochrome monitor
  - — Real-time clock card
  - — 3.5-or 5.25-inch floppy disk drive
- 132-or 80-column IBM-compatible graphics parallel printer
- D8W modular cord and 355AF adapter connecting the SMDR port on the system to the COM1 serial port on the PC (CAS Plus V3 only; CAS-IS Ill connects to the COM2 port).

Call Accounting Terminal (CAT)

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

CAT, a standalone application, is a dedicated terminal and printer designed to track, sort, and print reports on telephone charges.

Three versions of CAT are available:

- **CAT Basic** is an entry-level system for small businesses.
- **CAT Plus/Business,** for larger businesses, includes a two-line display.
- **CAT Plus/Hospitality,** for hotels and health care facilities, also includes a two-line display.

CAT can be set up to calculate the cost of calls by using toll rates or charging by the minute. Service charges and discounts can be applied to calls made to local and long-distance numbers and to directory assistance, Calls to specified area codes (such as 900) can be singled out for special treatment,

CAT is customized with current local and long-distance rates for a company's location. As rates change or a new area code or exchange is added, the rate information can be updated simply by exchanging a chip inside the terminal.

When a new telephone line or account code is added to the system, the CAT adds this information to its memory automatically the first time the new line or code is used.

The CAT provides a variety of reports that can be printed on a regular schedule or automatically when stored call information reaches 90% of the terminal's capacity. The available reports include the following, depending on the version of CAT:

- A variety of summary and detail reports. For example, reports can be printed 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 from the SMDR. The system gets such information from the AT&T Megacom 800 service and puts it into the SMDR.

CAT features an LCD display instead of a printed menu.

NOTE:
The availability of the caller identification may be limited by local-serving (caller's) jurisdiction, availability, or telephone company equipment.

Considerations and Constraints

- Only one CAT can be connected 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% of this capacity (5040 calls) is reached, When 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 PRI call report format.
- 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 users 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

The following equipment is required to connect the CAT to the system:

- One of the following CAT units:

  - CAT Basic
  - CAT Plus/Business
  - CAT Plus/Hospitality
- CAT Printer
- D8W modular cord and 355AF adapter connecting the SMDR port on the control unit to the CAT

Call Management System (CMS)

CAUTION:
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.

CMS is a standalone 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 person's extension.

CMS is appropriate for businesses with large groups of personnel who perform a common function, such as airline ticketing, filling catalog orders, or providing customer service. Agents within these groups can be divided into splits, or subgroups, to handle different kinds of calls or customers. For example, the agents in a travel agency might be divided into three splits: one that handles personal vacations, one that handles business trips, and one that handles group charters. Another split can be designated 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. The system can be setup so that agents are automatically in the ACW state whenever they complete a CMS call or so that they 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, which determines how long a call rings before it is designated as unanswered and connects to the recorded announcement.
- The Forced Delay option, which connects all calls to the recorded delay announcement regardless of whether all agents are busy.
- Designation of 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.

- Ability to connect Music-on-Hold to callers waiting for available agents.
- Ability to connect 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 the user to modify the call flow on-line.

Mode Differences

The system must operate in Hybrid/PBX or Key mode. CMS cannot be connected to a system operating in Behind Switch mode.

Considerations and Constraints

- A maximum of two CMSs can be connected to the system.
- CMS must be installed on an approved AT&T DOS PC. The PC must be dedicated to CMS. The two CMS interface card ports on the PC must be connected to two analog multiline extension jacks on the same module in the control unit (an 008 or 408). These jacks must be system operator positions. If two system operator position jacks are not available on the same module, another of these modules must be installed 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.
- Up to six agent splits can be designated for each CMS, with 28 agents per split.
- The CMS supervisor's console is any Direct-Line Console (DLC). CMS agents can have any MLX telephone or any analog multiline telephone that can be used with the system. CMS agent telephones must be connected to the first 58 extension jacks on the control unit.
- Lines/trunks ringing in to CMS can be loop-start, ground-start, T1 emulated ground-start, or PRI.
- Up to four external alerts can be used to alert agents and supervisors when the number of calls waiting to be answered reaches the programmed threshold.
- A MERLIN Attendant can be used to direct callers to the appropriate CMS group by use of loop-arounds.
- To play music for waiting callers, a Music-on-Hold product must be used that is compatible with a Music-on-Hold coupler.

NOTE:
If such equipment is used to rebroadcast music or other copyrighted materials, it may be necessary to obtain a copyright license from and pay license fees to a third party, such as ASCAP or BMI. A Magic On Hold system does not require such a license.

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 Numbering

- CMS agent telephones can use any extension. However, CMS refers to telephones using the 2-digit default numbering plan.

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 two digits.
  - 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

The following equipment is required to connect CMS to the system:

- An approved AT&T DOS PC, configured as follows:
  - — 640 kbytes RAM
  - — 3.5-inch floppy disk drive
  - — 20-Mbyte hard disk drive
  - — Monochrome or color monitor
- CMS interface card with two 14-foot (430-cm), 4-pair modular extension cords
- CMS software for the system
- DA-5 Digital Voice Announcement Unit with one 14-foot (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 supported by the system
- One analog multiline module (008 or 408) to connect the two PC ports to the extension jacks assigned as DLC ports.

InnManager Guest Management System

CAUTION:
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 InnManager Guest Management System is a standalone application that provides a complete management package for hotels with up to 100 extensions. It is a turnkey system, consisting of bundled hardware and the following menu-driven software modules. The system manager can customize each of these modules to suit the needs of the business.

- **Call Accounting and Rating System (CARS)** provides detailed call record accounting and control over call billing.
- **Guest Management System** provides front desk functions for management and tracking of room reservations, occupancy, and billing; hotel management functions such as housekeeping and bookkeeping; and general office utilities such as form letters, spreadsheet export, calculation and tracking of travel agency commissions, and credit limit reports.

Considerations and Constraints

- If the PC is more than 50 feet (15 meters) from the control unit or the PC does not share the same AC power supply as the control unit, ADUs must be used.
- If the single-user version of the InnManager Guest Management System is run on a network with multiple terminals, the data files will be corrupted and unusable.

Platform Requirements

- An approved AT&T DOS PC with at least a 40-Megabyte fixed disk, one 3 1/2-inch floppy disk drive, and at least 640K of available memory.
- MS-DOS 3.3 or higher
- 80-column printer capable of printing in elongated, bold-face, underline, and condensed modes.
- Internal or external modem
- A 355AF adapter and cords to connect the PC to the control unit.

System Programming and Maintenance (SPM)

CAUTION:
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.

SPM is a software application used for programming and maintenance of the system. It performs the same functions as an MLX-20L telephone used 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 standalone,** running on an approved AT&T DOS PC
- **SPM Integrated with IS III**

A PC with DOS-based SPM can be connected directly to the control unit or can access the system remotely in one of the following ways:

- The system programmer dials the system directly. A password can be set up 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 *IO).

SPM—IS III can be used only through a direct local connection,

SPM can be programmed to operate in English, French, or Spanish for communication with the control unit. Independent of this language setting, an on-screen option allows the programmer to select from the same three languages for the console-simulation window only for the duration of the current session.

Considerations and Constraints

- SPM *must* be upgraded to version 2. *xx* to function with Release 2.0 of the communications 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 him or her.
- A PC running DOS-based SPM connects to the lower RS-232 jack on the processor module of the control unit. This connection runs at 1200 or 2400 bps with autobaud.

- A UNIX* system-based version of SPM is available. See IS II and IS III.
- SPM reports can be printed out or can be written to the PC's hard or floppy disk drive. At the same time, the report is displayed on the screen together with prompts for browsing.
- SPM reports should not be printed while the system is handling more than 100 calls per hour.
- A printer connected to the computer running SPM can be used to print system programming reports. Reports can also be sent 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

Standalone SPM requires an approved AT&T DOS PC, configured as follows:

- MS-DOS 3.3 or higher.
- At least 128 kbytes of RAM.
- A double-sided floppy diskette drive, either 5¼-inch or 3½-inch. (A 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 feet (15 meters) from the control unit. Distances of greater than 50 feet (15 meters) require back-to-back ADUs.

Integrated Solution II (IS II)

CAUTION:
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. Callers are provided with a menu of recorded prompts that they respond to by dialing numbers on a touch-tone telephone.

  Callers without touch-tone telephones can be transferred to the system operator, who then handles their calls. Separate menus for day and night service as well as multilevel menus and corresponding announcements can be set up 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) offers all the features of the IVP AA combined with the following services:**

  — **Call Answer Service,** which 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,** which 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,** which provides a customer-oriented, call-in information service that plays a recorded message and then disconnects the caller.

— **Message Drop,** which 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)** 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. CAS—IS II provides all of the functionality of CAS along with ANI. For more information on the features of CAS, see Call Accounting System in this chapter.
- **System Programming and Maintenance-IS II (SPM)** is a programming package built into IS II that allows the system manager or a systems technician to upgrade and maintain the system and its features and to add, change, or rearrange telephones. Programming can be done 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 predefine 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** are 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 user's system. See Table 4-6.

Table 4-6. Voice Channels Required

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

Mode Differences

Only the CAS—IS II and the SPM—IS II applications can be connected to a system that operates in the 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.
- Either IVP AA or AVP can be installed 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 turn on, 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 calls go to coverage prematurely.
- SPM—IS II reports can be printed out or written to a disk (floppy or fixed drive). At the same time, the report is displayed on the screen together with prompts for browsing.
- SPM—IS II reports should not be printed while the system is handling more than 100 calls per hour.

Feature Interactions

Account Code Entry

- The account code entered by users before or during calls are used by CAS-IS II to associate calls with accounts and individuals; they appear on CAS-IS II reports.

Coverage

- An internal call on a VMI port that transfers to an internal extension will not go to coverage. It will continue to ring at the internal extension.
- If a sender programs his or her telephone to that only outside calls are sent to coverage, calls received in **ICOM** or **System Access** buttons will not be sent to voice mail.
- For Release 2.0, 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. No special action is needed on AUDIX Voice Power administration to activate this feature.

Group Calling

- Calls answered by an overflow calling group will get coverage mode codes; the overflow calling group's number appears in the Called Party field of the mode code.
- For Release 2.0, when AUDIX Voice Power sends a Leave Word Calling message 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.

Leave Word Calling

- If a Leave Word Calling message is left in a mailbox in a system with heavy VMI traffic, the user may have to dial out manually for messages.

Night Service

- If the AVP Automated Attendant handles only after-hours calls, a phantom station (an unused telephone jack) must be programmed as a member of a Night Service group associated with the system operator. In turn, this phantom station 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 station 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 PRI.

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 telephone, the VMI port can inadvertently park the call.
- Any calling group, calling group member, or telephone can be programmed to be a voice messaging system (VMS) transfer redirect extension. If a QCC is programmed 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 via voice mail to the system operator, the system operator has no indication of the origin of the call.

System Programming

The following must be programmed when IS II includes IVP AA:

- Designate Inside (Intercom) Dial Tone to be the same as the outside line/trunk dial tone.
- Assign all Automated Attendants connected to the system 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 were originally transferred to unanswered or busy extensions or 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.

The following must be programmed 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 the AVP is sent to the backup position (system operator).
- When the AVP Automated Attendant is used for Night Service only, do the following:
  - If the lines/trunks set for answering the Automated Attendant appear at other stations, set the No Ring option for the other telephones.
  - — Assign the phantom station to a Night Service group for each system operator position.
  - — Assign the phantom station 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

- An 80-Mbyte or 200-Mbyte fixed disk drive is required 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, which includes the following:
  - Master Controller II processor (with a 40-Mbyte, 80-Mbyte, or 200-Mbyte fixed disk and a 3.5-inch floppy disk drive)
  - — Video monitor (monochrome or color)
  - — Keyboard
  - — Optional tape drive (required for systems with a 200-Mbyte fixed disk for saving UNIX files, application program files, administration files, and voice system files during backup)
- A 355 AF adapter for connecting the Master Controller to the serial port on the control unit if they are within 50 feet (15 meters) of each other and are on the same AC branch circuit
- ADUs for connecting the Master Controller to the serial port on the control unit if they are not within 50 feet (15 meters) of each other and are not on the same AC branch unit
- Any additional hardware required by the individual applications 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 (IS III)

CAUTION:
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.

IS Ill is 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:
  - **Call Answer Service,** which 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, which allows subscribers to send messages to other extensions in the system, forward messages received with comments, and reply to messages received. The system manager can broadcast messages to all subscribers.
  - **Information Service,** which provides a call-in information service that plays a recorded message and then disconnects the caller.
  - **Message Drop,** which 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 their messages to specific extensions.
  - **Automated Attendant Service,** which answers incoming calls and plays a menu of recorded prompts. A caller can respond to the prompts by dialing touch-tones, and the Automated Attendant routes the call to an internal 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 without a touch-tone 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.

— With Outcalling, when a user or *subscriber* receives a new message, the system can automatically call a number that he or she has programmed, for example, a beeper or a home telephone number. The subscriber can then log in to the VMS to retrieve messages.

- **Fax Attendant** provides an integrated voice/fax mailbox, fax broadcasting, fax bulletin board, and coverage for busy or off-line fax machines, It must be installed with AUDIX Voice Power. Fax Attendant includes the following services:

  — **Fax Call Answer,** which allows Fax Attendant to receive fax messages for subscribers whose fax machines are busy or out of paper. This feature also enables subscribers who have personal fax numbers but do not have fax machines to receive fax messages. In such a case, Fax Call Answer gives the appearance of a personal fax machine by automatically answering and receiving fax messages for the specified phone number.

  — **Fax Mail,** which allows subscribers to send fax messages, get fax messages, record personal greetings, administer outcalling (standalone configuration only, create fax distribution lists, and change their account passwords, delivery report settings, and autoprint setting.

  — **Fax Response,** which allows the user to dedicate a phone number from which callers can retrieve information. This feature directs callers through a series of prompts to retrieve information on 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.

- **Call Accounting System** 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,

  CAS-IS Ill provides the same functionality as the standalone CAS Plus V3 application, and in addition collects and processes ANI information (if available from the originating CO).

- **System Programming and Maintenance** provides a maintenance and programming interface to the system. SPM—IS Ill provides the same functionality as the standalone SPM application, except for remote connection to the control unit.

In addition to these applications, IS Ill provides Extension Directory services. Extension Directory allows integrated programming of extension and subscriber information for both AUDIX Voice Power and the system from a single interface. This integration eliminates duplication of effort. When a change is made to either database, the two databases are automatically reconciled, ensuring that they remain in agreement.

Integrated Administration

Integrated Administration is the integration of AUDIX Voice Power and Fax Attendant administration with the switch parameters that are used by those two applications. Integrated Administration consists of three cooperating parts accessed by menu selection:

- Extension Directory
- Extension Directory Setup
- Integrated AUDIX Voice Power and Fax Attendant Administration (System Programming/Switch Administration)

Integrated Administration is intended to be used primarily by system technicians who are responsible for administering the applications and the switch through IS Ill. 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 via the IS Ill menu.

Initial Installation

The system technician performing initial installation of Integrated Administration logs in to IS Ill and selects `System programming and Maintenance (SPM)` from the menu, The SPM screen appears. The technician performs basic system administration, such as dial plan, mode, attendant, phantom stations, and lines in pools, for the switch and exits from SPM.

Through the `Technician Maintenance` menu item on the Integrated Solution Ill Maintenance menu, the technician selects `AUDIX Voice Power Switch Defaults` and changes the defaults for `Calling Group` if necessary. Through the same menu item, the technician performs an Extension Directory Setup, which downloads the switch dial plan and directory labels into the Extension Directory Database.

Through the Integrated Solution Ill Maintenance menu, the technician performs an Extension Directory update, that is, steps through each extension and attaches a name label and other information.

The technician then selects `System Programming/Switch Admin` from the AVP or AVP/FA main menu and performs Integrated AVP or AVP/FA Administration. (The technician is guided through a series of choices and forms that direct the administration.) The technician then administers the remaining AVP and FA administration, and performs the rest of switch administration.

Installation on Existing Switch

The system technician installing Integrated Administration on an existing switch, that is, a switch that is already functioning, selects `AUDIX Voice Power Switch Defaults from alll the Technician Maintenance item on the Integrated` Solution Ill Maintenance menu, and changes the defaults for calling `Group` if necessary. Through the same menu item, the technician performs an Extension Directory Setup. This downloads the switch dial plan and directory labels into the Extension Directory Database.

Through the Integrated Solution Ill 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 Administration. (The technician is guided through a series of choices and forms that direct the administration.) During the administration, 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 administration, including administration of greetings and other voice prompts, and through the System Programming/Switch Admin menu, goes back to each installed service and adds the appropriate lines.

Remote Operation

Remote initial installation can be provided by equipping the remote location with a surrogate switch and IS Ill. Using the remote location's switch and Master Controller II+ or Ill, the technician programs the customer's configuration, as specified earlier in Initial Installation.

Through SPM, the technician backs up the switch 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, via remote SPM, restores the customer's switch from the translations made at the remote location.

After requesting Pass Through to the customer's Master Controller, the technician, through the Technician Maintenance menu, restores the customer's database files from the database files backed up at the remote location.

NOTE:
The Extension Directory and Integrated Administration screens can be accessed remotely, but the information is stored in a file and run after the remote caller hangs up, Also, a change made to System Renumbering is not reflected immediately in the Extension Directory; reconciliation is run automatically at 3:00 a.m. or can be invoked manually via the technician menus. This reconciliation synchronizes the information in the database and the switch.

Mode Differences

- The system must operate in Hybrid/PBX or Key mode for all IS Ill applications except CAS and SPM. Those are the only two applications that can be connected to a system operating in Behind Switch mode.

Considerations and Constraints

- IS Ill can store up to 36 hours of voice-mail messages for AUDIX Voice Power with the 500-Mbyte fixed disk.
- IS III can store over 200,000 call records for CAS with the200-Mbyte fixed disk.
- Fax Attendant cannot be installed without AUDIX Voice Power.
- Automated Attendant cannot be installed as a standalone application, but only in conjunction with AUDIX Voice Power.
- When an AUDIX Voice Power subscriber receives a Voice Mail message, the message-waiting LED is lit on his or her telephone.
- If an AUDIX Voice Power mailbox is needed for a person with no telephone, a phantom station must be assigned in the system switch.
- AUDIX Voice Power time should be synchronized with the time on the system switch.
- Updating of Message Waiting lights on users' telephones is most efficiently obtained by properly linking users' AVP mailboxes with telephones in the system switch. If an AVP mailbox is desired for a person who does not have a telephone, a phantom extension must be assigned in the system switch.
- Integrated Administration is not supported for Standalone Automated Attendant.
- For Integrated Administration, the Master Controller II+ or III provides a separate Backup and Restore capability that saves the Directory information on the fixed disk. This is provided through the Backup and Restore menu options under Maintenance.
- Assigning the AVP application to a user via the Integrated Administration Extension Directory automatically sets that user to be covered by AUDIX Voice Power, if specified, and produces the AVP Subscriber Screen.
- The AVP Subscriber menu uses the name, extension, and coverage information contained in the Integrated Administration Extension Directory. This information cannot be changed on the AVP Subscriber Screen.
- No AUDIX Voice Power subscriber who does not have AVP in the Applications field of the Integrated Administration Extension Directory can be added to or deleted from AUDIX Voice Power.
- All stations and lines being programmed via Integrated Administration must be idle.

Feature Interactions

Account Code Entry

- CAS uses the account codes users enter before or during calls to associate calls with accounts and individuals; these codes appear on CAS reports.

Coverage

- An internal call on a VMI port that transfers to an internal extension will not go to coverage, but will continue to ring at the internal extension,
- If a sender programs his or her telephone so that only outside calls are sent to coverage, calls received on **ICOM** or **System Access** buttons will not be sent to voice mail,
- For Release 2.0, 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. No special action is needed on AUDIX Voice Power administration to activate this feature.

Group Calling

- Calls that overflow from one calling group to another calling group with integrated VMI ports are identified as coverage calls via mode codes. As a result, the overflow calling group's number appears in the called-party field of the mode code.
- For Release 2.0, when AUDIX Voice Power sends a Leave Word Calling message 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.

Labeling

- Names entered via the Integrated Administration Extension Directory are sent to the switch and are available through Switch Labeling screens.

Leave Word Calling

- If a Leave Word Calling message is left in a mailbox in a system with heavy VMI traffic, the subscriber may have to dial out manually to retrieve the message.

Night Service

- If the Automated Attendant handles only after-hours calls, a phantom station (an unused station jack) must be programmed as a member of a Night Service group associated with a system operator. In turn, this phantom station 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. AUDIX Voice Power must be programmed to recognize the call to be from the phantom station, and provides Automated Attendant service rather than the usual Call Answer service.

SMDR

- CAS collects call information from the SMDR output of the system.

System Renumbering

- System Renumbering can be done only via SPM or MLX-20L system programming. Integrated Administration uses System Renumbering to read extension numbers and adjuncts.

Transfer

- Integrated 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 station, the VMI port may park the call.
- Any calling group, calling group member, or extension can be programmed to be a VMS transfer redirect extension. If a QCC is so programmed, 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 via Voice Mail to the system operator, the system operator has no indication of the origin of the call.

System Programming

The following system programming is required for AUDIX Voice Power:

- Assign AUDIX Voice Power 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 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 (system operator).

The following system programming is required for AUDIX Voice Power with Automated Attendant:

- Set Inside (Intercom) 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 were originally transferred to unanswered or busy extensions or when a caller fails to respond to the announcement.
- Specify the number of rings before a call transferred by the VMS is sent to the backup position.

When the AUDIX Voice Power Automated Attendant is used only for Night Service:

- Set the no ring option for the other extensions if the lines/trunks set for answering by the Automated Attendant Service appear at other extensions.
- Specify the VMI ports that provide Automated Attendant Service to be Automated Attendant ports.

The following system programming is required for Integrated Administration with Fax Attendant:

IVP 4/6 Board Ports. For Fax Response service, the following items must be programmed:

Put the tip/ring ports 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.
- Give the lines appropriate labels.

Fax Board Ports. Each fax port on the TR112 or TR114 board is connected to a tip/ring port on the switch. These ports are regular tip/ring ports and do not have to be identified as fax ports on the communications system. The following items must be programmed:

- Give the tip/ring connections appropriate labels.
- Put the tip/ring extensions on the Night Service Exclusion list to enable Off-Site Fax Delivery to function at night.
- Identify the ports as fax ports on the switch using SPM if the user wants to use the Fax Message Waiting Light feature.

Private Fax Extensions. A Private Fax Extension is either an extension connected to an actual fax machine used by an individual or a phantom station associated with an individual's voice extension. Programming for Private Fax Extension depends on whether or not the communications system's configuration supports DID lines. In systems with DID, unique DID extension numbers are sufficient for Private Fax Extension because outside calls placed to that DID number ring the fax machine or phantom station. Systems without DID must rely on personal line appearance.

The following item must be programmed for Private Fax Extensions in DID configurations:

- Assign the DID extension of a phantom station or actual fax machine as a Private Fax Extension.

The following items must be programmed for Private Fax Extensions in non-13113 configurations:

- Assign a personal line to a phantom station or to the extension connected to an actual fax machine.
- Assign the phantom station or fax machine as the owner of that line using SPM.

The following remaining items must be programmed for Private Fax Extensions in both configurations:

- Set 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 extension of the phantom station or fax machine as a Private Fax Extension.
- Place the Private Fax Extension in 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 also be used by a group of individuals through parameters set using Fax Attendant setup screens on the Master Controller. To facilitate these configurations, a Group Fax Administrator is selected. The following items must be programmed:

- Assign a non-valid extension number as a Special Purpose extension. This will be 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 Extension.

Platform Requirements

IS Ill is delivered already installed and configured with the applications ordered. The system consists of an AT&T Master Controller II+ or Master Controller Ill running UNIX System V Release 3.2.2. Various hardware configurations are available; see the *AT& T Integrated Solution Ill Installation and Maintenance Guide* for details.

If AUDIX Voice Power is installed, an 012 module (with a ring generator) is required in the system to provide the tip/ring interface.

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 4-7 shows these requirements,

Table 4-7, Voice Channels Required

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

Primary Rate Interface (PRI) Applications

CAUTION:
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.

Group IV (G4) Fax

Group IV (G4) Fax is an application that enables the system to use the advanced Group IV (G4) Fax equipment-one of the new services accessible with the PRI trunks. Group IV (G4) Fax equipment provides several advantages:

- High speed transmission
- High quality laser reproductions
- High speed, high capacity printing
- Virtually error-free transmission
- Fax machine can double as an office copy machine.

Documents received using Group IV (G4) 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) are ideal candidates for this application.

Depending on the interface the fax machine has, the Group IV (G4) application can be connected in three configurations:

- Direct RS-232 (the recommended method)
- V.35 interface connecting to a UDM RS-232D interface
- V.35 interface connecting to a 7500B Data Module

Each of these configurations requires additional equipment.

See Chapter 5, Data, for additional information about Group IV (G4) Fax.

Video Conferencing

Video Conferencing, available with PRI service, enables groups of people in geographically dispersed locations to meet face to face. They may exchange information, documents, ideas, and data while employing a variety of visual aids to support the exchange of this information. Visual aids can include interactive writing and drawing, prepared text and graphic materials, and precorded audio and motion video material. Improved technology and superior camera optics and digital audio signals result in video pictures that are equal to commercial broadcast quality

Video conferences can be started from an easy-to-use control console and can be as easy to use as a telephone. No special technical expertise is required to operate the system.

The basic components include

- The conference control subsystem—to establish and terminate connections and allows camera control including pan, tilt, and zoom; and audio control for volume and privacy.
- The video subsystem—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.
- The audio subsystem—allows video participants to hear and speak at the same time. Includes microphones, microphone mixer, an echo canceller. Microphones may include table top microphones, wired lapel microphones, and/or wireless hand-held or lapel microphones.
- The video codec subsystem—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.

These basic components can be integrated into a mobile roll-about console that can be easily wheeled into a conference room or executive office prior to a scheduled video conference call. Alternatively, the components can be built-in to a video conference room.

Additional equipment may be required, for example, an interface converter for the video codec.

Optional applications can be added to the basic components to enhance the information being sent back and forth during an interactive broadcast.

See Chapter 5, Data, for additional information about Video Conferencing.

Centrex Operation

CAUTION
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.

Centrex is an optional telephone service for business customers. It provides an array of telephone features from the CO that formerly were available only from a PBX located on the customer's premises.

Basic Centrex features include the following:

- Transfer
- Three-Way Conference
- Drop
- Hold
- Recall
- Call Forwarding
- Call Waiting
- Call Pickup
- Group Pickup
- Automatic Callback

Additional features such as Speed Dialing and Night Service can also be added.

The system can be used with either full or limited Centrex service. Full Centrex service requires that telephones have a Centrex line and that the 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 system features for most of their communications needs.

Timed Flash

In Releases 1.0 and 1.1, a timed flash can be generated on a Direct Facility Termination (DFT) button or a Direct Pool Termination (DPT) button only, as long as the call is not a conference call and the facility is a loop-start line. A timed flash cannot be generated for an external call that terminates on a System Access button.

In Release 2.0, a timed flash can be generated on a call terminating on a System **Access** button, as well as on a **DFT** button or a **DPT** button. This includes, among others, transferred calls, group calling calls, and forwarded calls.

The following apply to timed flash (Release 2.0):

- **Dial Access to Pools:** If the **Recall** button is pressed during dialing while connected to a trunk or when end-of-dial is reached, a timed 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 flash cannot be generated. When dialing is complete, pressing the **Recall** button generates a timed flash, the accessed trunk is kept, and restrictions are applied,
- **Rotary trunks:** A timed flash cannot be generated during dialing. When dialing is completed, pressing the **Recall** button generates a timed flash, the accessed trunk is kept, and restrictions are applied.

Full Centrex Service

With full Centrex service, each telephone has a direct line to the CO Centrex. The direct line allows users to dial outside numbers directly after dialing an access code (usually a 9). The direct line is also used to call other four-digit Centrex extension numbers. System intercom lines are used to dial other telephones on the system.

Users with full Centrex service can send a switchhook flash via the **Recall** button without the system intercepting or responding to the signals,

A full Centrex service requires that the planning form for each MLX telephone using a direct line be marked for central administration of a **Recall** button (code *775).

Limited Centrex Service

Limited Centrex service is for customers who will 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 POTS, Tie, or DID lines. Others will use **System Access** buttons to access pooled facilities. Generally, users will 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

The system must be configured for Behind Switch mode for full Centrex operation.

For limited Centrex operation, if Centrex features are dominant, the system should be configured for Behind Switch mode; if system features are dominant, the system should be configured for Key or Hybrid/PBX mode.

The system can be used as a Hybrid/PBX behind a host switch by combining the features of a Behind Switch system with the ground-start capabilities of a Hybrid/PBX system. If a ground-start line is connected directly to the control unit, the FCC considers the system a Hybrid. Any Behind Switch system with full or limited Centrex service using a ground-start line must be registered as a PBX system even if it operates in the Key mode.

Considerations and Constraints

- With full Centrex service, during periods of high traffic, users may experience delays in obtaining dial tone from the host. Should a user begin dialing too rapidly, the first and second digits could be lost and the call would be misdialed. This situation is more probable when the host is another PBX, not a CO Centrex. With full Centrex service, the delay in dial tone could cause misdialing when using System Speed Dialing or Personal Speed Dialing.
- With full Centrex service, 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.
- With full Centrex service, 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.
- With full Centrex service, loop-start lines have higher cable losses than ground-start lines and cannot assure secure toll restrictions.
- With full Centrex service, single-line telephones have limited functionality when connected directly to the CO Centrex host. They cannot access system features or make inside calls. The can, however, use all of the Centrex features via code numbers.

  If the single-line telephone has the idle line preference programmed for in intercom ring line, it cannot be used for internal conference, transfer, or drop because it would result in a Centrex dial tone.
- With limited Centrex service, touching a hard feature button to call a local feature will cause misdialed calls.
- If the limited Centrex configuration is programmed for Hybrid/PBX mode, the use of calling groups, Shared System Access buttons, pools, and other features is possible, and applications such as MERLIN MAIL, CAS, and CMS can be used. If the mode is Behind Switch, the applications cannot be used.

- If a Key system is being converted to a Hybrid/PBX system for limited Centrex service, the conversion from loop-start lines to ground-start lines can cause a long delay before the system can be installed.
- Extension numbers should reflect the last four digits of the Centrex telephone line 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

CAUTION:
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 PFC (Phone-Fax-Copier) Telephone is a BIS-34D (34-button) display telephone with a built-in fax machine and personal copier that provides the convenience of a fax machine and personal copier in one compact unit.

Using the MERLIN PFC Telephone allows the user to do the following:

- Make and receive inside and outside calls using the built-in speakerphone as well as use the BIS-34D telephone features provided by the system
- Send and receive fax transmissions while using the telephone
- Make quick copies while using the telephone

The system must have two analog ports available on the control unit. In Behind Switch mode, a dedicated fax line for incoming fax calls is also required; in Hybrid/PBX or Key mode, the system can have either a dedicated fax line or direct inward dialing (DID).

NOTE:
The fax machine component of the MERLIN PFC Telephone does *not* transmit date, time, and fax number.

Mode Differences

Hybrid/PBX and Key Modes

- The dedicated fax line for incoming fax calls from the CO must be connected to a line port on the control unit, and the line cannot be assigned to any pool.
- If DID is used, a DID number must be assigned to the fax extension.
- If a dedicated private line is used, assign a fax line to the voice station.
- No lines or line pools can be administered to the fax extension.
- The dedicated fax line should be administered to Immediate Ring and any other lines to No Ring at the fax extension.

Behind Switch Mode

- The dedicated fax line can be administered only to the MERLIN PFC Telephone fax extension,
- The dedicated fax line cannot be assigned to a pool.
- The dedicated fax line should be assigned as the secondary line on the MERLIN PFC Telephone.

Considerations and Constraints

- The MERLIN PFC Telephone requires two analog ports: one for the voice line and one for the fax line.
- The telephone wiring between the system controller or control unit and the MERLIN PFC Telephone must be installed in the same building.
- The MERLIN PFC Telephone cannot be installed outside of building.
- All button assignments except the one for the fax line must be removed from the fax extension.
- The Voice Announcement feature should be removed from the fax extension.

Feature Interactions

Ringing Options

- If the dedicated fax line is shared *for outgoing calls only,* the Ringing Option must be administered to No Ring at any station except the MERLIN PFC Telephone fax extension.

Automated Document Delivery System (ADDS)

CAUTION:
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 AT&T 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.* ADDS has one voice port for handling incoming calls and one fax port for fax delivery of 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 administration 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 any message or no message after requesting a document.

A record is maintained by ADDS, including the file name 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 may need more than one system to handle the call volume efficiently.
- ADDS should be used in a two-line configuration to maximize performance and minimize busy signals.
- ADDS can be used behind an Automated Attendant.
- ADDS does not function using a BTMI, GPA, or a tip/ring adapter.

Platform Requirements

To set up ADDS, a business must have the following:

- The Automated Document Delivery System 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 (G3) Fax machine.

For backup of stored data, one of the following is required:

- AT&T 705 MT Multi-tasking Terminal
- AT&T 6386/SX WGS (or compatible) with ProComm Plus software

CONVERSANT Intro

CAUTION:
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 Intro is an entry-level voice response system that enables the user to run integrated voice response (IVR) applications. CONVERSANT Intro can automatically answer and route calls and execute telephone transactions.

The CONVERSANT Intro software can be configured in either of the following ways:

- As an application development environment in which all the tools to create an application are available
- As a platform to run applications already developed

CONVERSANT Intro 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 Intro, the application running on CONVERSANT Intro prompts the caller with 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.

Applications can be developed that allow the call 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 Intro 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
- Repeatable prompts

- Interaction of applications with voice mailboxes, with the ability to leave and retrieve messages, execute voice mail script, or get subscriber information
- Creation of tables and retrieve and update 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 Intro supports a maximum of 24 channels of analog ports, or up to 6 IVP4 boards. In a co-resident environment, such as CONVERSANT Intro 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 Intro is the Master Controller Ill, a high-performance 32-bit computer built around an Intel® 486® SX 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-inch 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 also are 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 Ill uses AT&T UNIX System V version 3.2.2, It includes a system unit, a monitor, and a keyboard.

Applications Printers

The following table shows the printers that can be used with the communications system for applications connected to the system.

Table 4-8. Applications Printers

Printer	Document No.	Description
Applications Printer	582-421-105	9-pin dot matrix printer that provides choice of print quality and speed. Uses parallel connection to the computer.
Applications Printer (Wide Carriage)	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 inches (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.

Data Communications Support

5

Using its circuit-switched connections, the communications system can establish a dedicated communications path between two data endpoints for the transfer of data. This connectivity enables the communication system to be used to share resources as well as to establish and manage connections between computers and other data input and output devices. In addition, the communications system can support advanced network services that integrate voice and data, such as Video Conferencing. The communications system's features used for voice service, such as Automatic Route Selection (ARS), Call Restriction, Idle Line Preference and others, can be used to enhance the usage of data facilities.

This chapter describes the system's data communications capabilities, the configurations and features that support those capabilities, and typical data communications applications. For instructions on making and answering data calls, see the *Data User's Guide.*

Data Communications Configuration Overview

Figure 5-1 shows how data and voice equipment connects to the communications system to provide the support for data communications. The communications system's control unit (hardware and software), in conjunction with other external hardware devices, provides data connectivity for the following:

- Internal analog data stations
- Connection to external analog data stations via analog facilities (GS, LS, Tie, DID) or via dedicated analog facilities, or via a DS1 Digital Service Link (DSL) providing emulated GS, LS, Tie, DID, or PRI
- Internal digital data stations
- Connection to external digital data stations via a Primary Rate Interface (PRI) facility
- Circuit-switched connections between two similar type data stations
- Circuit-switched connection between a digital data station (on a B-channel) and an analog data station via two-stage dialing through a conversion resource (modem pool)
- Conversion resources, such as 7500B 7500B Data Module/modem pools for making connections between Analog-to-Digital and Digital-to-Analog data stations
- Data Hunt Groups (DHGs)
- Simultaneous Voice and Data on analog and MLX ports
- On-premises host computer access
- Local area network (LAN) access via a modem or 75009 Data Module connected to an RS-232 port on a workstation on the LAN

This section describes the various equipment configurations and connectivity arrangements shown in Figure 5-1.

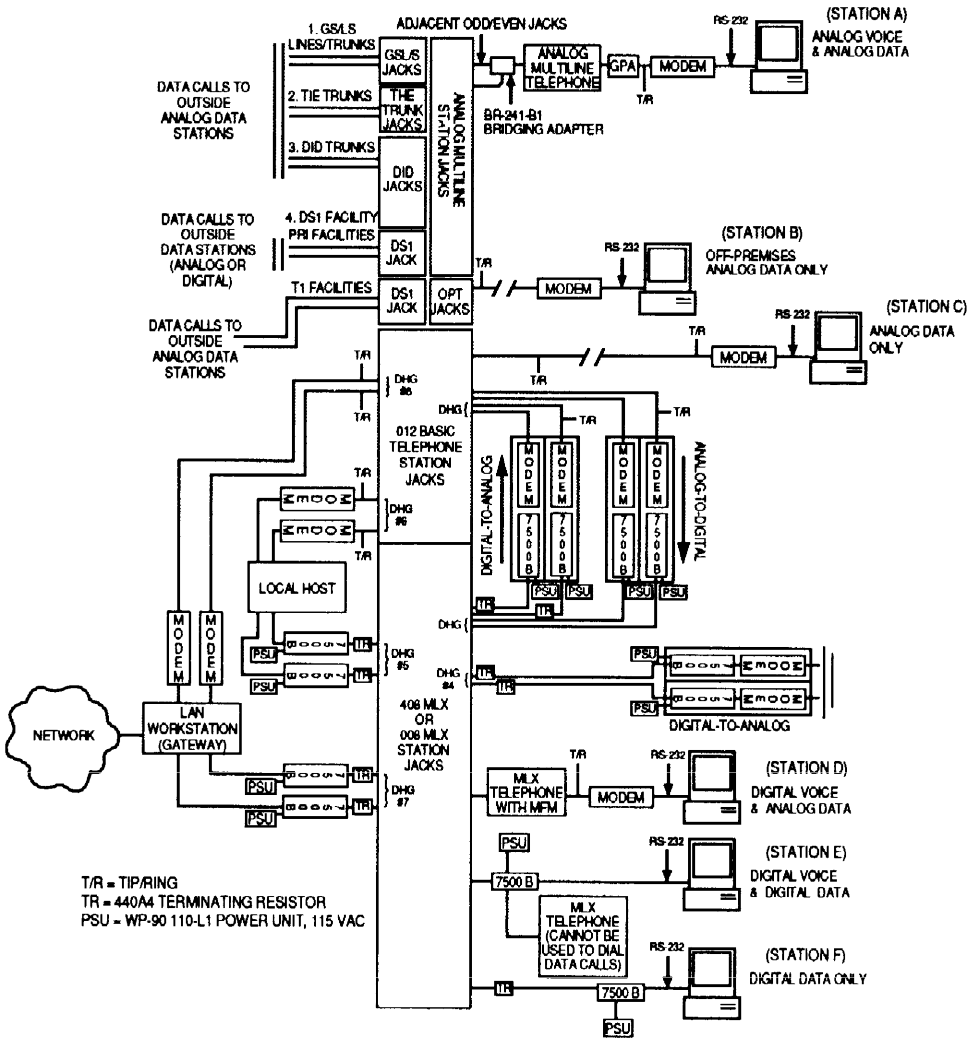


Figure 5-1. Individual Use Data Station Configurations

Data Stations

A data station consists of data terminal equipment (DTE), such as a personal computer (PC), data terminal, printer, optical scanner or video system, and data communications equipment (DCE), such as an internal or external modem or an external 7500B Data Module.

The DTE connects to the communications system via the DCE, which has capabilities similar to a telephone. The DCE places the data call, maintains its connection, and terminates the data call.

The DCE and DTE may have hardware and/or software options that can be set for transferring and receiving data, such as parity and bit rate. Other options may differ, depending on whether the transfer of data is synchronous or asynchronous. See the DTE and DCE documentation for configuration compatibility requirements, the options for transferring and receiving data, and guidelines for changing options.

Data stations (PC-based or workstation-based) require a communications software or terminal emulation package to transfer and receive data. The communications setup for each data station depends on specific configuration requirements and equipment limitations. Reference the communications **or** terminal emulation software documentation for instructions.

Data stations can be either analog or digital. Some analog data stations and all digital data stations can include a telephone for users who need simultaneous voice and data transmission.

Analog Data Station

An analog data station uses a modem as its DCE. The modem converts digital signals from the DTE into analog signals. It transmits these signals as continuously varying electrical voltages in the voice frequency band. It converts incoming analog signals into digital signals, passing them to the DTE. Most types of modems can be connected to the system. (See Chapter 2 for modem hardware descriptions.)

From the standpoint of the communications switching system, the analog data endpoint for an analog data station is the Tip/Ring (T/R) interface provided by one of the following:

- A General Purpose Adapter (GPA), connected to an analog multiline telephone
- A port on a T/R module (012T/R or 008 OPT)
- A Multi-Function Module (MFM) adjunct on an MLX telephone

From a physical viewpoint, the connected modem provides a T/R analog signal interface to the telephone network or switching system and an EIA RS-232 digital interface (or other type) to the data equipment.

Analog Data Station Configurations

As Figure 5-1 shows, the T/R interfaces support the following analog data stations:

- **Analog Voice and Analog Data** (Station A): Includes a data terminal (with a keyboard and display) or a computer connected to a modem via an ElA-type RS-232 interface. The GPA connects the modem to the analog multiline telephone to provide the T/R interface for the modem. To provide the simultaneous voice and data, the port configuration requires two adjacent odd/even station jacks on a 408, 408 GS/LS, or 008 module in the control unit. 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. The telephone provides the dialing capability for the data station.
- **Off-Premises Analog Data-Only** (Station B): Includes a data terminal (with a keyboard and display) or a computer connected to a modem via an ElA-type RS-232 interface. The modem connects to a port on a **008** OPT module in the control unit. A telephone mayor may not be connected depending on modem capabilities; simultaneous voice and data is not supported, The terminal keyboard provides the dialing capability for the data station.

  NOTE:
  This distance between the station jack and the modem, as shown in Figure 5-1, is acceptable for short distances up to 5000 feet (1524 meters). For longer distances, this configuration may need '{data grade" facilities with 4.5 dB maximum loss.

- **Analog Data-Only** (Station C): Includes a data terminal (with a keyboard and display) or a computer connected to a modem via an EIA RS-232 interface. The modem connects to a port on an 012 module in the control unit. A telephone mayor may not be connected, depending on modem capabilities. Simultaneous voice and data transmissions not supported. The terminal keyboard provides the dialing capability for the data station.
- **Digital Voice and Analog Data** (Station D): Includes a data terminal (with a keyboard and display) or a computer connected to a modem via an RS-232 interface. The modem connects to an MLX telephone configured with an MFM. The MFM provides the T/R interface for the modem. (See Chapter 2 for the hardware description of the MFM.) The MLX telephone voice capabilities operate independently from the data station data capabilities. The terminal keyboard provides the dialing capability for the data station.

NOTE
If the MLX telephone is voice-signaled while active on a call, both B-channels are required to accommodate the signaling. This may conflict with the data station.

Other Supported Analog Data Endpoints

Other data equipment that may be connected to an analog data endpoint (T/R interface) via a modem includes the following:

- A local host computer (described later in this chapter)
- Group Ill (G3) Fax terminal
- An output-only device (optical scanner)
- An input-only device (printer or display)

A personal computer containing an internal modem card can also be connected to a T/R interface.

NOTE:
A few restrictions or special conditions should be considered when configuring ports to serve a certain class of equipment or a special class of service. Port modules, station sets, and station adjuncts specifically designed for the communications system provide *equipment type* information to the system software automatically via the port *signature* or terminal/adjunct *classmark.* However, there are types of devices that can connect to a T/R interface that provide no classmark or other means of automatic identification to the system. This equipment includes Group III (G3) FAX terminals, automatic answering machines, modems, automatic dial-announce alarm-sending equipment, external alerting device, music sources, and paging equipment,

Digital Data Station

The 7500B Data Module adapts the DTE, such as a 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 port on the control unit. The 7500B Data Module does not convert the digital signal from the DTE to an analog signal, but sends it as a sequence of separate electrical impulses. (See Chapter 2 for the 75006 Data Module hardware specifications and requirements.)

Along with the feature descriptions in Chapter 2, the communications system also supports the 7500B Data Module's capability for circuit-switched data connections on the B-channel.

NOTE:
To communicate with the switch, the digital data endpoint uses D-channel messages during call setup and termination. When the call is set up, the switch establishes a connection between the calling and called endpoints on a B-channel. The switch sends the appropriate messages to drive the endpoints into the data mode. In the data mode, the 7500B Data Module transmits and receives data over a B-channel using transport modes as defined in the Digital Multiplexed Interface.

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 Digital Multiplexed Interface (DMI), and the communications 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.

NOTE:
If the MLX telephone is voice-signaled while active on a call, both B-channels are required to accommodate the signaling. This may conflict with the data station.

The 7500B Data Module may provide dialing and answering capabilities to the data station. The terminal keyboard can also provide the dialing capability for the data station.

NOTE:
Although the 7500B Data Module supports packet-switched data on the D-channel, this data mode is not supported by the communications system.

MLX Port Connection Requirements

When configuring digital data equipment connection to an MLX port, the following requirements and/or restrictions must be applied:

- Only one 7500B Data Module should be connected 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 are not a problem, incoming calls may not always be answered by the intended party.
- If a 7500B Data Module is the only digital data endpoint on the MLX port (no MLX telephone is connected), a 440A4 Terminating Resistor (TR) Adapter must be configured to provide 100-ohm termination for each transmission pair. The 7500B Data Module does not provide termination.
- An MLX telephone is independent 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, an MLX telephone should not be connected to a 7500B Data Module used in a data station configuration.

- The maximum cord length from an MLX telephone to a the 7500B Data Module is 80 feet (24 meters). This should be considered if there are plans to use the voice capability of a port by connecting an MLX telephone located some distance away from the 75006 Data Module.

Digital Data Station Configurations

As Figure 5-1 shows, the 7500B Data Module interface supports the following digital data stations:

- **Digital Voice and Digital Data** (Station E): Includes a data terminal (with a keyboard and display) or a computer connected to a 7500B Data Module via 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 only) module on the control unit. The MLX port is shared by the telephone and the 75006 Data Module but, they operate independently of each other. The terminal keyboard provides the dialing capability for the data station.

  NOTE:
  For a digital voice and digital data station, the MLX telephone *cannot* contain an MFM. The MFM interferes with communication to the switch in this data station configuration.

  If the MLX telephone is voice-signaled while active on a call, both B-channels are required to accommodate the signaling. This may conflict with the data station.

- **Digital Data-Only** (Station F): Includes a data terminal (with a keyboard and display) or a computer connected to a 7500B Data Module via an RS-232 or V.35 interface. The configuration does not include an MLX telephone, so a 440A4 terminating resistor is required. The 7500B Data Module connects to a 008 MLX, 408 MLX, or 408 GS/LS-MLX (Release 2.0 only) module in the control unit. The keyboard provides the data call dialing capability.

NOTE:
There is no analog voice and digital data station configuration.

Data Station Configurations Summary

Table 5-1 summaries the hardware requirements and port assignments needed to support the various data station configurations.

Table 5-1, Configurations of Data Stations

Type of Data Station	Equipment Configuration: Computer or Terminal	Modem	7500B Data Module	Analog Telephone	MLX Telephone	Type of Module
Analog voice and analog data	✓	✓		✓*		008, 408, 408 GS/LS
Analog data only	✓	✓		✓		012, 008 OPT
Digital voice and analog data	✓	✓		‡	✓†	008 MLX, 408 MLX, 408 GS/LS-MLX
Digital voice and digital data	✓		✓		✓§	008 MLX, 408 MLX, 408 GS/LS-MLX
Digital data only	✓		✓			008 MLX, 408 MLX, 408 GS/LS-MLX

* Requires a GPA
† Requires an MFM
‡ Telephone may be connected depending on modem capabilities.
§ Telephone cannot contain an MFM.

Other Supported Digital Data Endpoints

Other data equipment that may be connected as a digital data endpoint via a 7500B Data Module includes the following:

- Video Conferencing system
- Group IV (G4) Fax machine

See PRI Applications later in this chapter.

Data Hunt Group

A data hunt group (DHG) consists of a specified group of station ports *of the same type* (i. e., all analog or all digital) assigned a Calling Group number, When a call is placed to a DHG, the switch performs a circular search (starting with the station listed after the one that received the last call) to find the first idle station. The idle station is alerted and when it answers, it is connected to the originator. If all the stations in the DHG are busy, the originator hears ringback.

DHGs support the following:

- modem pools
- dedicated lines for data service (for example, DHG #4 in Figure 5-1)
- a host computer with multiple ports
- a workstation (gateway) on a LAN

NOTE:
One important aspect of a modem pool is the specific direction of call origination. All the units (for example, modems) on one side of the modem pool can be grouped into a DHG, so users only need to dial one number for any pair.

Figure 5-1 shows the following examples of DHG assignments for the various shared resources:

- DHG #5 is assigned to the data modules used to communicate with the local host computer
- DHG #6 is assigned to the modems used to communicate with the local host computer
- DHG #7 is assigned to the 7500B Data Modules used to communicate with the workstation (gateway) on a LAN
- DHG#8 is assigned to the modems used to communicate with the workstation (gateway) on a LAN

The communications system accommodates up to 32 calling groups (or DCG). Each calling group can have a maximum of **20** members. A data station can be a member of only one calling group.

Modem Pool

A modem pool acts as a conversion resource to accommodate communication between analog and digital data endpoints. The modem pool can be one or more pairs of DCEs; a pair consists of one 7500B Data Module connected via an RS-232 interface to a modem.

NOTE:
It is possible for a communications system user to unintentionally or inadvertently corrupt the modem or 7500B Data Module settings by changing options at the modem/7500B Data Module interface. If a problem occurs with a modem pool, the settings on each device should be checked.

Data Call Direction

An important aspect of modem pools is that call origination is in *one direction only:*

- Analog-to-digital, which enables an analog data endpoint to originate a call to a digital data endpoint
- Digital-to-analog, which enables a digital data endpoint to originate a call to an analog data endpoint

Using a DHG extension to access a modem pool ensures that the intended incoming service unit (modem or 7500B Data Module) receives the data call first. Once communication is established between the data endpoints, the communication is full duplex.

NOTE:
Modem pools should not be used to originate calls in both directions because it is possible for a modem/7500B Data Module pair to be called at both ends at the same time. This could result in connecting the wrong data endpoints or not completing either or both calls.

Modem Pool Dialing

Calls through a modem pool are placed using two-stage dialing. In the first stage, users enter the modem pool extension or DHG group extension, if available, for the analog-to-digital or digital-to-analog modem pool being called. In the second stage, users enter the extension of the analog or digital data endpoint for inside calls. For outside calls, users enter the dial-out code and telephone number of the outside analog or digital data station.

This two-stage dialing should be considered before providing modem pools dialing access to outside lines.

For example, it is possible for a data call (originated from an external data endpoint) to come into a modem/7500B Data Module pair and, through second stage dialing, access another external data endpoint.

CAUTION:
Because the communications system originates the second data call in the two stage dialing process, a call to an external data endpoint in the second stage would be charged to the communications system. If this is a concern, separate analog-to-digital and digital-to-analog modem pools should be created that service incoming data calls but have no access to outside lines for second stage dialing.

Modem Pool Configurations

Figure 5-1 shows three modem pool configurations:

1. **Digital-to-analog modem pool,** which enables a digital data endpoint to originate a call to an analog data endpoint. In this configuration, the 7500B Data Modules are assigned to a DHG.
2. **Analog-to-digital modem pool,** which enables an analog data endpoint to originate a call to a digital data endpoint. In this configuration, the modems are assigned to a DHG.
3. **Digital-to-analog modem pool on dedicated outside lines,** which converts digital signals to analog signals for data calls from a digital data station to an outside analog data station. It does this over analog dedicated outside lines used solely for data communications. In this configuration, the 7500B Data Modules are assigned to a DHG.

Configurations 1 and 2 (digital-to-analog and analog-to-digital) use system lines; therefore, the modems and 7500B Data Modules are connected directly to the control unit. Each modem is connected to a jack on a 012 or 008 OPT module, and each 7500B Data Module is connected to a jack on a 008 MLX, 408 MLX, or 408 GS/LS-MLX module (Release 2.0 only).

In configuration 3 (digital-to-analog on dedicated outside lines), the 75006 Data Module connects to the control unit via jacks on a 008 MLX or 408 GS/LS-MLX module (Release 2.0 only).

In all three configurations, a 440A4 terminating resistor is required because the 7500B Data Module connects directly to the control unit. Also, each modem/7500B Data Module pair requires a null modem connector to interconnect the RS-232 interfaces of the modem and 7500B Data Module.

The 7500B Data Modules or modems in modem pools must not share a port with an MLX telephone. Otherwise, the telephone could use both B-channels and block use by the 7500B Data Module or modem.

Other Resource Pools

Users can share a limited number of modems or 7500B Data Modules connected to a shared data endpoint. To do this, the communications system can be configured for modem-only or 7500B Data Module-only resource pools.

The modem-only pool can provide access to multiport data equipment, such as a local host computer, by assigning the T/R ports interfacing with the modems in the pool to a DHG.

Like the modem-only pool, the 7500B Data Module-only pool can provide access to multiport data equipment, such as a host computer, by assigning the MLX ports interfacing with the 7500B Data Modules in the pool to a DHG.

Connectivity to a Local Host Computer

Figure 5-1 shows DHGs assigned to modem-only and 7500B Data Module-only modem pools that provide access to a local host computer.

These modem pools and the host computer are connected by EIA type RS-232 interfaces. Each modem provides the T/R interface to the 012 module on the control unit, and each 7500B Data Module connects to the 008 MLX or 408 GS/LS-MLX module (Release 2.0 only) on the control unit. Terminating resistors are required for 7500B Data Module connection.

Connectivity to a LAN

A LAN is an interconnected chain of terminals or PCS that pass data to and from a mainframe computer or interconnected workstations using some topology arrangement.

The communications system connects to the LAN through a workstation that functions as a gateway, The gateway provides the ports for the modem and data module connections to the communications system and the connection to the LAN. It also accommodates the protocols needed for the transfer of data between a data endpoint on communications system and a workstation (or data terminal) on the LAN.

The modems connect to the gateway via EIA-type RS-232 (or other type) interfaces and provide the T/R interface to the 012 module on the control unit. The 7500B Data Modules connect to the gateway via EIA-type RS-232 interfaces (or other type) and on the communications system end connect to the 008 MLX, 408 MLX, or 408 GS/LS-MLX module (Release 2.0 only) on the control unit. Terminating resistors are required for the 7500B Data Module connection.

Once a connection is established between a data endpoint on the communications system side and a workstation within the LAN, all of the features and capabilities of the LAN environment are available to the originating data endpoint on the communications system side. However, limitations or hardware requirements may restrict the usage of some LAN facilities.

Outside Trunks

Figure 5-1 shows the types of outside trunks that can be used to make and receive data calls to and from data stations outside the system. (In the figure, the trunk types are displayed to the left of the control unit and are labeled 1-4.)

1. **Ground-start (GS) trunks** and **loop-start (LS) trunks** are used to communicate with outside analog data stations. A loop-start trunk is the standard for home and small businesses is the least expensive trunks in some areas LS trunks have the following disadvantages:

- 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.
- They cannot provide reliable far-end disconnect for toll restriction.

A **ground-start (GS) trunk** is preferred for communication with outside analog data stations. Ground-start trunks provide improved signaling and reliable far-end disconnect for secure toll restriction.

The following kinds of outside ground-start/loop-start trunks can be used for data communications:

- Basic trunks
- Wide Area Telecommunications Service (WATS)
- 800 service (inbound WATS)
- Foreign exchange (FX)

Ground-start/loop-start trunks connect to ground-start/loop-start jacks on the following types of modules in the control unit:

- 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 (Release 2.0 only)

2. A **tie trunk** provides communication between two telephone switching systems. A tie trunk “ties” the two systems together, providing access to all telephones or data stations on each system. Tie trunks are usually used for data communication with analog data stations connected to a system at a different location, such as a different floor of a building, a different building, or a different city or state.

   A tie trunk connects to a jack on a 400EM module in the control unit.

3. A **Direct Inward Dialing (DID) trunk** allows incoming calls to reach specific individuals or facilities in the system without the help of a system operator. DID trunks are available only in the Hybrid/PBX mode. A DID trunk is used to receive incoming calls from outside analog data stations; it is not used for outgoing calls.

   A DID trunk connects to a jack on an 800 DID module in the control unit.

4. A **DS1 trunk** carries digital signals in the Digital Signal 1 (DS1) format. The DS1 format multiplexes 24 Digital Signal 0 (DS0) channels of 64 kbps each and one 8-kbps framing signal, for a total of 1.5444 Mbps. A DS1 trunk can be used for communication with outside digital or analog data stations.

A DS1 trunk connects to the jack on a 100D module in the control unit. Even though there is only one physical jack, the 100D module supports up to 24 logical endpoints or ports for voice and data calls. Each DS0 channel in the DS1 signal corresponds to a trunk or logical ID.

A DS1 trunk provides either T1 or PRI access.

- T1 is the factory setting.

  A T1 facility is used for communication with outside analog data stations. The 24 channels on a T1 facility can be programmed individually in any combination to emulate a loop-start, ground-start, E&M tie, or DID trunk, so a single 100D module can replace 24 outside trunks. Digital data calls cannot be placed through this trunk.

- PRI is the standard format provided by connection to a 5ESS central office (CO) switch or a 4ESS toll switch. (PRI must be used for digital data calls).

  The 100D module supports any combination of the following AT&T Switched Network services:

  — Accunet switched digital service for 56-kbps and 64-kbps restricted and 64-kbps clear circuit-switched data calls

  — Megacom 800 for incoming domestic toll-free voice calls

  Megacom WATS service for outgoing domestic long-distance voice calls

  — Software Defined Network (SDN) for circuit-switched voice and data calls at up to 56 kbps

PRI service provides the following benefits:

- Speed. Data calls to outside destinations can be made on the same B-channels used for voice calls if the service allows. Modems and dedicated, conditioned trunks are not required.
- INFO-2 automatic number identification (AN I) service. Customers who subscribe to this service can identify the caller on an incoming call on a PRI trunk by either telephone number or billing number.

  NOTE
  If your system has automatic number identification (AN I), the display shows the number for the outside caller. Availability of the caller identification information may be limited by local-serving (caller's) jurisdiction, availability, or CO equipment.

- Dynamic B-channel assignment. An individual B-channel can be removed from service without blocking calls to or from any other B-channels.
- Improved toll restriction. PRI trunks severely limit the potential for bypassing of toll restrictions.
- Reliable indication of far-end disconnect. Blocking of incoming calls is prevented because a trunk is not immediately released; instead, there is a delayed indication of disconnect.
- Improved Station Message Detail Recording (SMDR). Call records provide more accurate duration information,
- Shared use of B-channels for Megacom WATS and Megacom 800 on a call-by-call basis for more efficient use of facilities.
- Supports digital data transmission at speeds up to 19.2 kbps for asynchronous and 64 kbps for synchronous.

System Features Used For Data

Some communications system features provided for voice service may also be used to enhance the usage of data facilities:

- **Account Code Entry** allows tracking of outgoing data calls for billing, forecasting, or budget reports.
- **Auto Answer All** allows a modem with automatic answering capability to answer data calls when the user is away from the station. This feature is used for Analog Voice and Analog Data configurations only.
- **ARS** (Hybrid/PBX mode only) routes calls over outside trunks according to the number dialed and the trunks available. Therefore, the system can be programmed to select the least expensive route for each data call. When using ARS with digital data calls, make sure that the calls use PRI.
- **Calling Restrictions,** such as Allowed Lists, Disallowed Lists and others, inhibits line access. These features enable companies to control and manage communications costs for outgoing data calls.
- A **Data Status** button monitors station activity (busy, not busy) of any data station. The green LED on the button lights up to indicate a “busy” status. Unlike an **Auto Dial** or **Signaling** button, which is programmed to dial a specific number when pressed, the **Data Status** button does not perform any dialing function. Pressing it, then, has no adverse effect on data calls in progress. The **Data Status** button is the *only* button which should be used to monitor station activity.
- **Dial Access to a Direct Facility Termination (DFT)** provides access from a digital data station to outside lines supporting the PRI interface.

- **Idle Line Preference** automatically selects the first available line for data calls.
- **Last Number Dialed** automatically places a call to the last number dialed from that station. Dialing sequence must include dial-out code for outside calls.
- **Personal Speed Dial** allows quick dialing (a 2-digit code) of frequently used numbers on 10-button phones. The dialing sequence requires a dial-out code for outside calls.

  NOTE:
  Use this feature on telephones that have only 10 or fewer buttons. If you have an MLX-20L telephone, you can program Personal Directory entries instead of Personal Speed Dial codes. If you have any other MLX or analog telephone, you can program Auto Dial buttons. If you program Personal Speed Dial codes on telephones with more than 10 buttons, you may delete features you have already programmed onto buttons.

- **Pool Access to external transmission facilities** (Hybrid/PBX Mode only) allows data endpoint dialing to seize access to pool numbers servicing outside lines and trunk.
- **Privacy** prevents loss of data by ensuring that data transmission is not interrupted accidentally. The Privacy feature is automatic for data calls on digital data stations and on analog data stations with analog multiline telephones. It is activated manually on all other analog data stations.
- **System Speed Dial** allows quick dialing of numbers that are used often and provide for data security. Dialing sequence requires a dial-out code for outside calls.

NOTE:
Certain system (voice) features interfere with data connections. These features should be disabled and include the following:

- Voice Announce
- Call Waiting
- Automatic Callback

Endpoint Communications Features

The communications system supports the use of data equipment or data software. These features are provided by the data station hardware/software in the personal computer, data terminal, or communicating device, such as the 7500B Data Module or modem.

- Data transport mode selection (for example, 7500B Data Modules support DMI modes 0, 1, 2, and 3)
- Data metering (speed matching) of bit rates between digital data endpoints
- Data terminal dialing
- Automatic answering of data calls

PRI Applications

The communications system provides an interface between PRI services and the small business customer. Some advanced digital applications are already supported by the communications system, which is unique because it is the first Key system to allow customers to use PRI services. These applications include high-speed fax transmissions and Video Conferencing.

This section provides the configurations supported for fax transmissions on the Group IV (G4) Fax machine and a general description of video conferencing connections.

Fax Transmissions Application

The Group IV (G4) Fax machine is a fax unit, offering 400x 100 dots per inch (DPI) in fine mode. It can operate at any speed for communications with a Group Ill (G3) Fax machine or another Group IV (G4) Fax machine. When speed is essential, it can transmit at 64 kbps and achieve speeds as fast as 3 seconds per page.

Supported Configurations

There are three ways to connect a Group IV (G4) Fax machine to the communications system for transmitting and receiving data:

- A Direct RS-232 Interface
- An RS-232 to V.35 Interface Conversion
- A Direct V.35 Interface

These connections are only a guide, not an assurance that different fax machines with other proprietary interface connections also operate properly. The configurations are based on compatibility testing that used Canon and

Ricoh fax models on the communications system. The communicating adjunct is the 7500B Data Module, (see Chapter 2 for a description of the 7500 Data Module).

Group IV (G4) Fax machine 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 before buying the fax machine what interfaces are required. This also means knowing the transmission type the machine normally operates in, asynchronous or synchronous. (When operating behind the communications system, it is operates synch ronously.)

Fax Configuration 1: Direct RS-232 Interface

This is the recommended connection. The other two methods, described later, are shown as alternatives when the EIA interface on the fax machine requires that they be used. The communications system does support the three methods, but there is virtually no advantage of using one method over another; it is only a matter of which interface the fax machine uses.

Figure 5-2 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 (G4) fax machine.

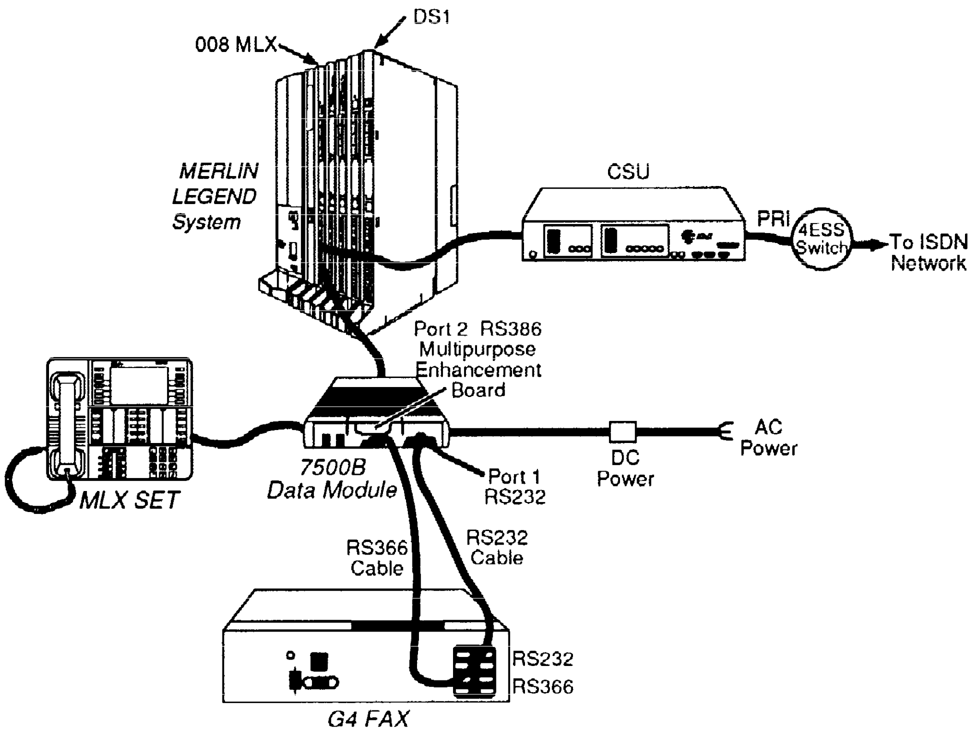


Figure 5-2. Direct RS-232 Interface

To use this method, it is important that the fax machine be ordered with an EIA-RS-232 connection, with V.28 or V.35 interface.

In this method, there is a direct connection between the Group IV (G4) Fax machine and a 7500B Data Module. The standard RS-232 interface on the back panel of the Group IV (G4) Fax machine is connected from the jack by 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 is a DCE terminal adapter that connects unattended DTE or other DCE to the Basic Rate Interface (BRI), for example, a 008 MLX or 408 GS/LS-MLX (Release 2.0 only) module in the control unit. This application describes the Group IV (G4) Fax machine as synchronous DTE sending and receiving imagery transmission through the digital network.

The dialing path in this configuration requires a Multipurpose Enhancement Board that is installed in port 2 of the 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 7500B Data Module is then connected from the Line jack on the back panel to an MLX port on an 008 MLX or 408 GS/LS-MLX module (Release 2.0 only) in the control unit, Completing the link is the PRI trunk connection that is plugged into the jack on the channel service unit (CSU) shown in the Figure 5-2. The CSU is then plugged into the 100 DSI Module in the control unit.

This connection is recommended when manual dial is used to dial the fax transmission telephone numbers from the fax dial pad.

It is recommended that the 75006 Data Module be located in a safe location, away from accidental tampering with the configuration settings, which 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 (G4) 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 carrier.

Fax Configuration 2: RS-232 to V.35 Interface Conversion

This configuration is used 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 V.35 interface.

Figure 5-3 shows the connections required for a Group IV (G4) fax V.35 interface connecting to a data module RS-232D interface.

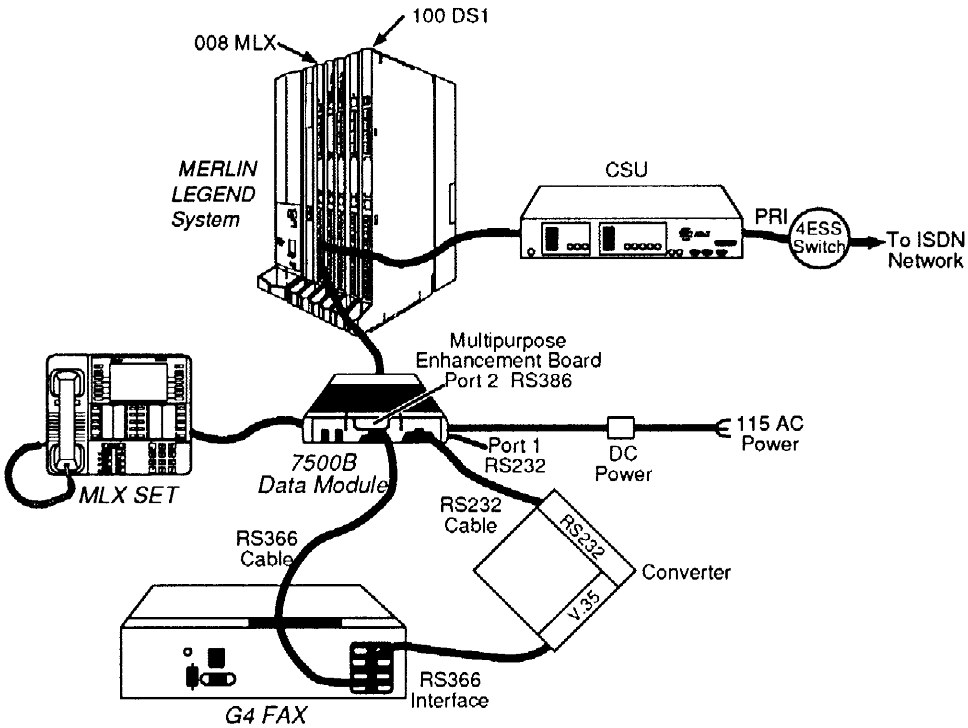


Figure 5-3. RS-232 to V.35 Interface Conversion

A V.35-to-RS-232 converter must be used between the fax machine and the 7500B Data Module. This configuration requires the use of the optional Multipurpose Enhancement Board in port 2 of the 7500B Data Module to set up an RS-366 dialing interface. The configuration allows the Group IV (G4) Fax machine to be connected to port 1 for imagery 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 75008 Data Module.

The Multipurpose Enhancement Board provides an RS-366 auto dial interface on port 2. It converts the RS-232 interace on port 1 on the main circuiit board from asynchronous to synchronous mode. The V.35 adapter cable must be ordered 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.

The 7500B Data Module is connected by a D8W cord to the MLX port on the 008 MLX or 408 GS/LS-MLX module (Release 2.0 only).

The PRI interface is established from the 100 DS1 Module to the CSU and then to the ESS switch and the PRI interface.

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.

Fax Configuration 3: 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, an optional connection board must be used with the 7500B Data Module. 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, the 7500B Data Module must be located in a work area where it cannot be accidentally reconfigured. Accidental reconfiguration could cause the 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 5-4 shows the connection between a Group IV (G4) fax machine and a 7500B Data module using an EIA-V.35 interface.

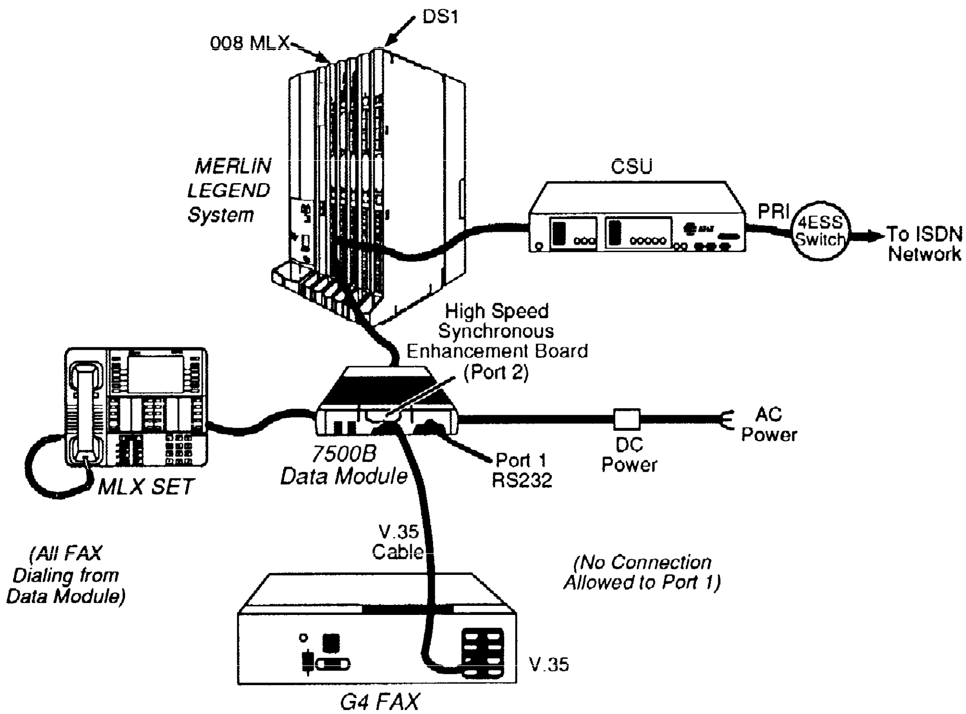


Figure 5-4. Direct V.35 Interface

This connection requires an optional High Speed Enhancement Board to be included in the 7500B Data Module.

The High Speed Synchronous Enhancement Board provides a V.35 interface at synchronous data rates of 48, 56, or 64 Kbps on port 2. The connection is via 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.

The board comes with an adhesive V.35 label that must be affixed to the back panel of the 7500B Data Module so that port 2 cannot be mistaken for a second EIA-232D interface.

When the High Speed board is used, no connection is allowed to port 1.

In this configuration, no dialing connection is made between the Group IV (G4) Fax machine and the 7500B Data Module, and all dialing must be made from the front panel of the data modem.

A D8W cord also connects from the Line port on the 7500B Data Module to the 008 MLX or GS/LS-MLX module port on the communications system. The PRI interface is made from the 100 DS1 Module.

Video Conferencing Application

This section shows an example of how to connect a 7500B Data Module with a *multipurpose enhancement board* for synchronous data communication, which allows data transmission at the speed of 56 or 64 kbps.

High-Speed Synchronous Enhancement Board

For instructions on using a high-speed synchronous enhancement board or any information on setting up the 7500B Data Module, not included in this section, see the documentation packaged with the 7500B Data Module,

This section is intended only as a guideline for connecting video conferencing equipment to the system. For any additional information, see the documentation packaged with the video codec.

Hardware Requirements

- 008 MLX or 408 GS/LS-MLX module in the control unit
- ESF T1 CSU

  NOTE:
  If any other type of CSU is used, your customer support organization cannot support installation and maintenance.
- Two Shore Microsystems SM-100 EIA-232/V.35 converters (or equivalent)
- Two 7500B Data Modules
- Two 7500B Data Module feature package 2 upgrades (user's manuals included)
- Two multipurpose enhancements boards. By installing a multipurpose enhancement board in each 7500B Data Module, you can provide synchronous communication and RS-366 ACU interface.
- Two WP901 10-L7 power supplies (one per standalone 7500B Data Module)
- Two 440A4 terminating resisting adapters
- Z77A multiple mounting (mounting for multiple 7500B Data Modules)
- Cables:
  - Two male/male EIA-232-D cables, 8 feet (24 meters), to connect the PORT 1 jacks on the data modules to the EIA-232/V.35 converters
  - Two male/male V.35 DB-37 cables, 8 feet (24 meters), to connect the V.35 communication ports on the video codec to the EIA-232/V.35 converters
  - Two male/male RS-366 DB-25 cables, 8 feet (24 meters), to connect the RS-366 dialing port of the video codec to PORT 2 on the 7500B Data Module

Video Conferencing Connections

Figure 5-5 shows an example of video conferencing connections.

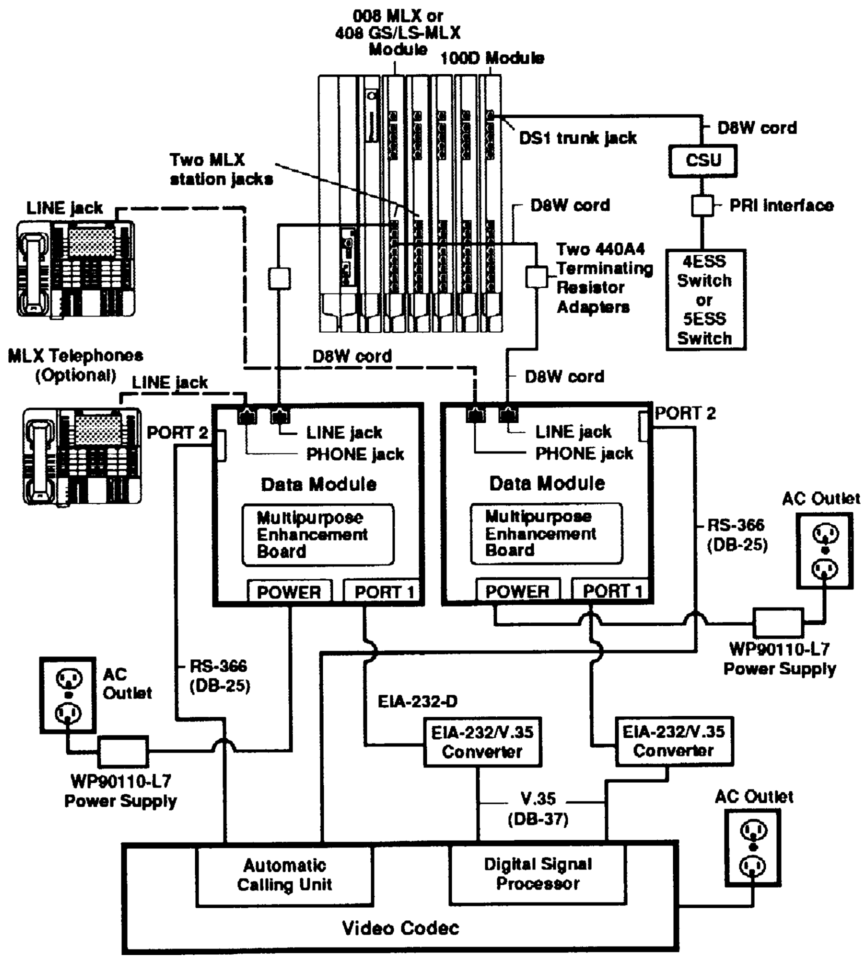


Figure 5-5. Video Conferencing Connections

NOTE:
If you are connecting MLX telephones, omit both 440A4 terminating resistors, and note that the maximum cord length from the data module to the telephone is 80 feet (24 meters).

Product Ordering Information

A

Ordering Codes

Component	PEC	Comcode	App. Code
Control Unit			
MERLIN LEGEND Control Unit	**6140-CU2**		
Basic carrier and housing		106388614	403E
Power Supply module		105743801	391A 1
Processor		106215155	517A27
Feature Module		106874738	517G25
MERLIN LEGEND Control Unit w/408 ATL	6140-CA2		
Basic carrier and housing		106388614	403E
Power supply module		105743801	391A1
Processor		106215155	517A27
Feature module		106874738	517G25
408 GS/LS		106064678	517A26
MERLIN LEGEND Control Unit w/408 MLX	6140-CD2		
Basic carrier and housing		106388614	403E
Power supply module		105743801	391A1
Processor		106215155	517A27
Feature module		106874738	517G25
408 GS/LS-MLX		106698590	517/429
MERLIN LEGEND Control Unit w/12 x 24 MLX	6140-242		
Basic carrier and housing		106388614	403E
Power supply module		105743801	391A1
Processor		106215155	517A27
Feature module		106874738	517G25
408 GS/LS-MLX (qty.3)		106698590	517A29
Expansion Unit	61490		
Expansion carrier and housing		106388630	403F
Power supply module		105743801	391A1
Expansion Unit Cover	N/A	106388259	17A

Ordering Codes (continued)

Component	PEC	Comcode	App. Code
R1/R1.1 Upgrade to R2	6141-102A	106874738	517G25
Upgrade from Merlin II (Basic)	6140-UD2A		
Cover		106388234	16A
Processor		106215155	517A27
008 MLX		105628010	517A21
Feature module		106064660	517A25
MLX-20L telephone (choose one):			
Black		106743420	7713D01 B-003
White		106743438	7713001 B-264
Expansion cover (zero, one, or two)		106388259	17A
Upgrade from Merlin II (w/MLX module)	6140-U2LA		
Cover		106388234	16A
Processor		106215155	517A27
Feature module		106874738	517G25
Expansion cover (zero, one, or two)		106388259	17A
Trunk and Station Modules			
008 MLX	61486	105628010	517A21
008 OPT	61489	106387525	517A28
012 (T/R)	61487	106553779	517E13
100D(DS1)	61491	105461560	517B15
400EM (tie trunk)	61492	105311401	517A14
400 GS/LS/TTR	61483	105628044	517B18
408 GS/LS	61481	106064678	517A26
408 GS/LS-MLX	61493	106698590	517A29
800 DID	61488	105628077	517B20
800 GS/LS	61484	105627996	517A19
Vintage Trunk and Station Modules			
400 (with TTRs)	61379	105408892	517B12
408 LS	61482	105512495	517C1
008 (ATL)	61485	105351092	517B3
800 LS	61384	105351100	517B4

Ordering Codes (continued)

Component	PEC	Comcode	App. Code
Telephones			
MLX Telephones			
MLX-10			
English (black)	3156-02B	106743024	7712 D01B-003
English (white)	3156-02W	106743032	7712D01B-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
MLX-10D			
English (black)	3156-03B	106743040	7712D02B-003
English (white)	3156-03W	106743057	7712~2B-264
French (black)	3156-F3I	106633928	7712~2A(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
MLX-20L			
English (black)	3156-05B	106743420	7713D01B-003
English (white)	3156-05W	106743438	7713D01B-264
French (black)	3156-F5I	106634421	7713D01A(29)-003
French (white)	3 156-F5I	106634439	7713D01A(29)-264
Spanish (black)	3156-S5I	106613557	7713D01A(22)-003
Spanish (white)	3156-S5I	106613573	7713D01A(22)-264
MLX-28D			
English (black)	3156-04B	106743503	7713D02B-003
English (white)	3156-04W	106743511	7713D02B-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
Analog Multiline Telephones (black)			
MLC-5	3168-MLC	105515332	7312HO1A-003
BIS-10	3165-10B	105161061	7313HO1A-003
BIS-22	3166-22B	105188809	7314HO1A-003
BIS-22D	3166-DSB	105630420	7315HO1B-003
BIS-34	3167-34D	105167027	7317HO1A-003
BIS-34D	3167-DSB	105630529	7317HO1B-003
MERLIN PFC (ATL)	3169-PF2	106681562	
PFC paper	31690	106673361	

Ordering Codes (continued)

Component	PEC	Comcode	App. Code
Telephones (continued)			
Vintage Analog Multiline Telephones (black)			
5-Button	3160-111	105217426	Z7302H01D0-003
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
Single-Line Telephones			
8110 Analog Voice	3193-001		
Black		106272321	8110A01A-003 811
White		106272339	8110A01A-264 811
8102 Analog Voice	3192-001		
Black		106272305	8102A01A-03 810
White		106272313	8102A01A-264 810
7102	3185-MWR		
Black		105335285	7102A01A-003
Misty cream		105330419	7102A01A-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	500 MM-03
Ivory		103870226	500 MM-50
Beige		103870267	500 MM-60

Ordering Codes (continued)

Component	PEC	Comcode	App. Code
Telephones (continued)			
Single-Line Telephones (continued)			
554 BMPA	3 100-ORW		
(rotary, wall)			
Black		103823498	554BMPA-3
Ivory		103823506	554BMPA-50
Beige		103823555	554BMPA-60
Cordless Telephones			
Model 5320	31CJ3-CLS	105543516	CS6300U11A-229
Consoles			
MERLIN II System Display Console	61392	105229744	7318H01A-003
DSS			
English (black)	3156-DCB	105685481	604A1-003
English (white)	3156-DCW	105685499	604A1-264
Spanish (black)	3156-SDI	106613672	604A1(22)-003
Spanish (white)	3156-SDI	106613680	604A1(22)-264
Applications			
SPM Version 2.0 — DOS	61495	106906092	
SPM Version 2.0 — UNIX System	61496	N/A	
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		
CAS Plus V3 Software	1201-DR1	406362244	
Rate Table*	12010	406158444	
CAS Plus upgrade	12009	406158394	
UNIX CAS (LEGEND)	1201-U12	406478537	
CAS/H			
CAS/H LEGEND 100S	1201-H10	405799255	
CAS/H LEGEND 200S	1201-H20	405788289	
CAS/H Rate Table*	12050	405788420	

* Consult AT&T or an authorized dealer for other area-specific information.

Ordering Codes (continued)

Component	PEC	Comcode	App. Code
Applications (continued)			
Call Accounting Terminal (CAT)			
CAT BASIC/B	3600-010		
CAT Terminal		406669762	
Printer		406637306	
CAT Basic Rate Table* (Update Chip)	36014	406669739	
CAT/B 150S	3600-023	406478800	
CAT/H 150S	3600-024	406478818	
CAT/B Rate Table* (update)	36023	406478792	
CAT/H Rate Table* (update)	36024	406478784	
Call Management System (CMS)	1207-100		
5¼* floppy disk		106496540	
3½* floppy disk		106496532	
Board	8301-100	106198815	
CONVERSANT INTRO Application	4201-100		
CONVERSANT INTRO Application Casual User	4201-101		
Conversant Intro Application Casual Dev. & Data	4201-102		
Inn Manager w/Base Software	7051-INN	406670984	
Site Specific Software			
20 station	70501	406670992	
40 station	70502	406671008	
60 station	70503	406671016	
80 station	70504	406671024	
100 station	70505	406671032	
355AF Adapter, Receptacle	2709-A25	105012645	355AF
355A Adapter, Plug	2750-A24	105012637	355A
14’ Cord	2725-07N	103687802	D8W-87 14FT

* Consult AT&T or an authorized dealer for other area-specific information.

Ordering Codes (continued)

Component	PEC	Comcode	App. Code
Applications (continued)			
MERLIN LEGEND Integrated Solution Ill			
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)		106220666	
200 MB MC-II + Processor	4200-503		
4 x 200 MB MC-II + Processor		406506337	
4MB Memory Upgrade		106219533	
COLOR MONITOR		406504571	
KEYBOARD		406504563	
9 *to* 25 PIN ADAPTER		406139394	
CARTRIDGE TAPES (qty.2)		106220866	
200 MB MC-III + Processor	4200-912		
8 x 200 MB MC-III + Processor		406700930	
COLOR MONITOR		406504571	
KEYBOARD		406504563	
9 to 25 PIN ADAPTER		406708503	
2 x 250 MB CART, TAPES		406760009	
500 MB MC-III + Processor	4200-936		
8 x 500 MB MC-III + Processor		406700914	
COLOR MONITOR		406504571	
KEYBOARD		406504563	
9 to 25 PIN ADAPTER		406708503	
2 x 250 MB CART. TAPES		406760009	
SOFTWARE APPLICATIONS			
MERLIN LEGEND IS-III		106894942	
LEGEND IS-III PLATFORM SOFTWARE			
UNIX SPM 2.0			
LEGEND IS-III System Manager's Guide			
LEGEND IS-III I+M Guide			
LEGEND IS-III Tape			

Ordering Codes (continued)

Component	PEC	Comcode	App. Code
Applications (continued)			
MERLIN LEGEND R2 AVP. 2.1.1		**106876311**	
IVPSS 2.0			
AVP2.1.1			
AVP/FA MLR2 Switch Integ. SW			
MERLIN LEGEND Integ. AVP/FA Admin.			
MERLIN LEGEND AVP User's Guide (50)			
MERLIN LEGEND AVP System Manager's Guide			
MERLIN LEGEND AVP/FA Planning Guide			
MERLIN LEGEND FAX Attendant		106876220	
FAX ATTENDANT 2.1.1 CO-RESIDENT Base PKG.			
AVP/FA MLR2 Switch Integ. SW			
FAX ATTENDANT User's Guide (50)			
MERLIN LEGEND AVP/FA Planning Guide Reference Sheet			
OTHER (Voice Boards)			
IVP4 Board LEGEND	8306-100	106248651	
IVP6 Board LEGEND	N/A	106856271	
IS-III Bundles			
100 MB MC-II + 4 *x* 4 AVP	6146-100		
100 MB MC-II + Processor	4200-503		
IVP4 Board	8306-100	106248651	
UNIX 3.2.2	N/A	106529548	
CART. TAPE UTILITIES	N/A	106632938	
MERLIN LEGEND IS-III	N/A	106894942	
LEGEND R2 AVP 2.1.1	N/A	106876311	
200 MB MC-II + 4 *X* 12AVP	6146-200		
200 MB MC-II + Processor	4200-503		
IVP4 Board	8306-100	106248651	
UNIX 3.2.2	N/A	106529548	
CART. TAPE UTILITIES	N/A	106632938	
MERLIN LEGEND IS-III	N/A	106894942	
LEGEND R2 AVP 2.1.1	N/A	106876311	
200 MB MC-111 +8 x 12AVP	6146-012		
200 MB MC-III + Processor	4200-912		
IVP4 Board (qty.2)	8306-100	106248651	
UNIX 3.2.2	N/A	106529548	
CART. TAPE UTILITIES	N/A	106632938	
MERLIN LEGEND IS-III	N/A	106894942	
LEGEND R2 AVP 2.1.1	N/A	106876311	

Ordering Codes (continued)

Component	PEC	Comcode	App. Code
Applications (continued)			
500 MB MC-III + 8 x 36 AVP	6146-036		
500 MB MC-III + Processor	4200-936		
IVP4 Board (qty.2)	8306-100	106248651	
UNIX 3.2.2	N/A	106529548	
CART. TAPE UTILITIES	N/A	106632938	
MERLIN LEGEND IS-III	N/A	106894942	
LEGEND R2 AVP 2.1.1	N/A	106876311	
100 MB MC-II + 4 *X* 4 AVP, CAS	6146-101		
100 MB MC-II + Processor	4200-503		
IVP4 Board	8306-100	106248651	
UNIX 3,2.2	N/A	106529548	
CART. TAPE UTILITIES	N/A	106632938	
MERLIN LEGEND IS-III	N/A	106894942	
LEGEND R2 AVP 2.1.1	N/A	106876311	
CAS—IS-III	1201-U12	406478537	
200 MB MC-II + 4 *X* 12 AVP, CAS	6146-201		
200 MB MC-II + Processor	4200-503		
IVP4 Board	8306-100	106248651	
UNIX 3.2.2	N/A	106529548	
CART, TAPE UTILITIES	N/A	106632938	
MERLIN LEGEND IS-III	N/A	106894942	
LEGEND R2 AVP 2.1.1	N/A	106876311	
CAS —IS-III	1201-U12	406478537	
200 MB MC-III + 8x 12 AVP, CAS	6146-112		
200 MB MC-III + Processor	4200-912		
IVP4 Board (qty.2)	8306-100	106248651	
UNIX 3.2.2	N/A	106529548	
CART. TAPE UTILITIES	N/A	106632938	
MERLIN LEGEND IS-III	N/A	106894942	
LEGEND R2 AVP 2.1.1	N/A	106876311	
CAS—IS-III	1201-U12	406478537	
500 MB MC-III + 8 x 36 AVP, CAS	6146-136		
500 MB MC-III + Processor	4200-936		
IVP4 Board (qty.2)	8306-100	106248651	
UNIX 3.2.2	N/A	106529548	
CART. TAPE UTILITIES	N/A	106632938	
MERLIN LEGEND IS-III	N/A	106894942	
LEGEND R2 AVP 2.1.1	N/A	106876311	
CAS—IS-III	1201-U12	406478537	

Ordering Codes (continued)

Component	PEC	Comcode	App. Code
Applications (continued)			
200 MB MC-III + 8 x 12 AVP, 4 x 1000 FAX	6146-212		
200 MB MC-III + Processor	4200-912		
IVP4 Board (qty.2)	8306-100	106248651	
UNIX 3.2.2	N/A	106529548	
CART. TAPE UTILITIES	N/A	106632938	
MERLIN LEGEND IS-III	N/A	106894942	
LEGEND R2 AVP 2.1.1	N/A	106876311	
LEGEND FAX ATTENDANT		106876220	
500 MB MC-III + 8 x 36 AVP, 4 x 3000 FAX	6146-236		
500 MB MC-III + Processor	4200-936		
IVP4 Board (qty.2)	8306-100	106248651	
UNIX 3.2.2	N/A	106529548	
CART. TAPE UTILITIES	N/A	106632938	
MERLIN LEGEND IS-III	N/A	106894942	
LEGEND R2 AVP 2.1.1	N/A	106876311	
LEGEND FAX ATTENDANT		106876220	
200 MB MC-III + 8 x 12 AVP, CAS, 4 x 1000 FAX	6146-312		
500 MB MC-III + Processor	4200-936		
IVP4 Board (qty.2)	8306-100	106248651	
UNIX 3.2.2	N/A	106529548	
CART. TAPE UTILITIES	N/A	106632938	
MERLIN LEGEND IS-III	N/A	106894942	
LEGEND R2 AVP 2.1.1	N/A	106876311	
LEGEND FAX ATTENDANT		106876220	
CAS—IS-III	1201-U12	406478537	
500 MB MC-III + 8 x 36 AVP, CAS, 4 x 3000 FAX	6146-336		
500 MB MC-III + Processor	4200-936		
IVP4 Board (qty.2)	8306-100	106248651	
UNIX 3.2.2	N/A	106529548	
CART. TAPE UTILITIES	N/A	106632938	
MERLIN LEGEND IS-III	N/A	106894942	
LEGEND R2 AVP 2.1.1	N/A	106876311	
LEGEND FAX ATTENDANT		106876220	
CAS—IS-III	1201-U12	406478537	
MERLIN Attendant	6125-All		
Hardware		406406090	
Documentation		106431265	

Ordering Codes (continued)

Component	PEC	Comcode	App. Code
Applications (continued)			
MERLIN MAIL™ Voice Messaging System for the MERLIN LEGEND Communications System			
Two-port	6107-005		
MERLIN MAIL unit		406824532	
Remote maintenance device		406466193	
Four-port	6107-006		
MERLIN MAIL unit		406824540	
Remote maintenance device		406466193	
Two-port line card (R2) (upgrade from two to four)	6107-007	406824524	
System Adjuncts and Adapters			
Auxiliary Power Unit 9024	61416	406467142	9024
Channel service units (CSUs)			
ESF T1	2152-ESF	405616293	305010171-001
Cable (8 mod, wire wrp)	21555	406012609	513861312-5050 FT
Cable (8 mod, 15-pin sub)	2155M	406012591	513823015-5050 FT
Stand-alone wall mount	21545	405!370104	380-100213-001
Stand-alone wall bracket		405616277	380-100542-001
551 T1 L1	2152-15T	403768179	305-10097-001
Power unit	21530		
Unit		403242639	KS22911LI
Cord		103895660	DW4A-SE 10FT- IP
Stand-alone wall mount	21545	405970104	380-100213-001
Stand-alone wall bracket		405616277	380100542-001
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)	21691	N/A	
Transformer		102600517	
Adapter (2486)		102802113	
Cord		102937620	
Adapter (4006)		103848859	

Ordering Codes (continued)

Component	PEC	Comcode	App. Code
System Adjuncts and Adapters (continued)			
Electrostatic discharge (ESD) suppression kits			
D-181574	N/A	105179329	D181574
D-181589	N/A	105201891	D181589
D-181590	N/A	105201909	D181590
D-181591	N/A	105201917	D181591
D-181593	N/A	105201933	D181593
EMI filter		103965206	Z200A
In-Range Out-of-Building-343B (IROB) unit — analog multiline*	32918	406721738	343B
IROB unit — MLX*	32919	106417447	505A ASSY OA 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	277A
7500A upgrade kit	21627	105688501	D 182208
Digital Magic On Hold® player (replaces analog unit)	3128-020	406659326	PAKG DMOH1 DIG L
Digital Announcement Unit	3119-001	406747774	ATTDAU
Modem 2224G	2224-CEO	105659965	2224C-L1 D/2
Music Coupler	61398	406143925	ASSY-K23395 L3
PagePac® 6	5323-006	405701277	22052-006 PG PC
PagePac 6 Pius	5323-008	405701608	22052-000 PG PC6
PagePac 20 PowerMate™	5323-005	403308026	ADP06
Trunk Adapter	53518	405223298	22050-900
PagePac6 Portsaver	53519	405703026	220520020

* 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)			
AT&T Door Phone Speaker	53240	406269860	
PagePac 20 Talkmate	53501	403307994	PE53501 AT
PagePac 50	5322-051	403305444	5322-051 VC PG
PagePac 200	5322-201	403305469	5322-201 VC PG
PagePac VS	5322-700	403307192	5322-7003500 VC PG
200 WATT AMPLICENTER	52120	403305493	52120 AT
50 WAIT AMPLICENTER	52150	403305501	52150 AT 50W
Ring generator unit	61388	105213201	129B RING GEN
SMDR Printers			
Parallel Printer (80-column)	4200-570	406637314	
Parallel Printer (132-column)	4200-571	406712067	
CAT Printer (serial)	4200-572	406716464	
Uninterruptible Power Supply (UPS)			
500 VA (15min)	2403-050	105610141	515005C111
Reserve (1 hr)	24035	105610174	0053150
Universal Paging Access Module (UPAM)	58500	405891698	KIT-UPAM
TAM-B	N/A	405899972	D181900
PRS-48	N/A	405742735	D181900
WMT- 1A	N/A	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

Ordering Codes (continued)

Component	PEC	Comcode	App. Cods
System Adjuncts and Adapters (continued)			
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
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	N/A	406648469	N/A
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 Suppiy		405331711	KS22911 L2
Modular Power Cord		102937620	D6AP-87
Z400F Adapter		103942857	Z400F
Single-line telephones			
Program, Pause, and Auto Dial button conceal kit for 8100 series telephones	31931	106248370	Kit-D 182363 Analog
4A Speakerphone	3120-02W		4A
Power unit		102139938	PWR UNIT-85B1
Block connector		102434925	BLK CON-82B-49
Adapter for single-line telephone		102813888	ADPTR-223C
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
Telephone Adjuncts and Adapters (continued)			
Single-line telephones (continued)			
S201 Speakerphone	31 52-007A	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	MOD3-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		105058340	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® Headset	3122-040	406445791	KS23822L5
Supra NC® Headset	3122-060	406741900	KS23822L12
Headset Adapter	3164-HFA	105752042	ADPTR-502C-003
500A Headset Adapter	3152-001	106690043	Adapter EL-500A-266
		405331711	Pwr Sup-KS22911L2
		102479904	Cord-D4BU-29 Std 7FT
		104152558	Adaptr-40082
Modular Amplifier	3122-020	406445619	KS23822L2
Plug Prong Amplifier	3122-010	406445601	KS23822L1
MLX Telephones Miscellaneous Add-Ons and Replacement Parts			
Handsets and Cords			
Handset (black)	N/A	106050065	K2S1-003
Handset (white)	N/A	106053408	K2S1-264
Handset, amplified hearing	31052		
Black		105581896	K6S2-003
White		106248248	K6S2-264
Misty cream		105581904	K6S2-215

Ordering Codes (continued)

Component	PEC	Comcode	App. Code
MLX Telephones Miscellaneous (continued)			
Handsets and Cords (continued)			
Handset cord, 9’ (2.74 m), black	N/A	105635429	H4DU-003 9FT
Handset cord, 9’ (2.74 m), white	N/A	105701809	H4DU-2649’BULK
Handset cord, 12’ (3.66 m), black	N/A	102401445	H4DU-3 12FT IP
Handset cord, 12’ (3.66 m), white	N/A	102402609	H4DU-264 12’IP
Handset cord, 25’ (7.62 m), black	N/A	105523666	H4DU-3 25’
DSS line cord, 2’ (61 cm)	N/A	106187545	CORD D8AC-87
Desk Stands and User Trays			
Stand (large, black)	N/A	846320851	STAND-LARGE BL
Stand (large, white)	N/A	846320844	STAND-LARGE WH
Stand (small, black)	N/A	846320810	STAND-SMALL BL
Stand (small, white)	N/A	846320802	STAND-SMALL WH
User tray (black)	N/A	846320240	USER TRAY DWR B
User tray (white)	N/A	846320232	USER TRAY DWR W
Designation (Button Assignment) Cards and Covers			
Card*—MLX-1O, MLX-1OD	N/A	846865939	
Card*–MLX-20L	N/A	846865947	
Card*–MLX-28D	N/A	846865954	
Card set†—DSS	N/A	106448756	KIT-D182464
Card covers†—DSS (black)	N/A	106448731	KIT-D 182462 PRT
Card covers†—DSS (white)	N/A	106448749	KIT-D 182463 PRT
Card set‡—QCC	N/A	106561673	KIT-D182562 PRT
Card covers§—MLX-10, MLX-10D, MLX-20L	N/A	106448681	KIT-D 182457 PRT
Card covers§—MLX-28D	N/A	106448699	KIT-D 182456 PRT
Analog Multiline Telephones Miscellaneous Add-Ons and Replacement Parts			
Desk Stands and Wall Mounts			
Adjustable desk stand, 10-button	32002	103746855	11A
Adjustable desk stand, 34-button	32003	103746863	11C

* 10½ 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 and Replacement Parts (continued)			
Desk Stands and Wall Mounts (continued)			
Fixed desk stand, 5- & 10-button	32004	103746848	10A
Desk stand/wall mount 14A, BIS-10	N/A	103804290	14A-003
Desk stand/wall mount 14B, BIS-22	N/A	103964458	Z14B-003
Desk stand/wall mount 14C, BIS-34	N/A	103979837	14C-003
Fixed desk stand and wall mount, 5-button	32000	103804290	14A
Kit of parts		103995882	D-181230
Wail mount, 10-button	32001	103747846	201A
Kit of parts		103995882	D-181230
Wail mount, 34-button	32006	103747853	203A
Kit of parts		103995882	D-181230
Faceplates			
BIS-10	N/A	105203186	KIT PRTS-D-181582
BIS-22	N/A	105336986	KIT PRTS-D-181786
BIS-22D	N/A	105690762	KIT PRTS-D-182210
BIS-34 and BIS-34D	N/A	105203194	KIT PRTS-D-181583
Button Label Sheets			
BIS-10	N/A	105336978	KIT PRTS-D-181785
BIS-22	N/A	105336960	KIT PRTS-D-181784
BIS-22D	N/A	105690770	KIT PRTS-D-182211
BIS-34 and BIS-34D	N/A	105336956	KIT PRTS-D-181783
Display console (FM1) (includes one faceplate)	N/A	105299754	KIT PRTS-D-181727
Display console (FM2 & R3) (includes one faceplate)	N/A	105486252	KIT PRTS-D-182041
Single-Line Telephones Miscellaneous Add-Ons			
Ground-Start Button	31021	405792839	Key-KS23566L1

Ordering Codes (continued)

Component	PEC	Comcode	App. Code
Miscellaneous Parts			
Interconnect Wiring Kit			
110AB1-100JP12	N/A	104409396	
110A1 trough	N/A	104407960	
D-Rings	N/A	842139248	
D8W cords	N/A	103786802	
Parts list	N/A	N/A	
SYSTIMAX MERLIN Wiring Kit	3103-MER	106393671	
110A1 trough (5)	N/A	104407960	
110AB1-100JP12 modular block (2)	N/A	104409960	
110AB1-100FT punch down block (1)	N/A	103823845	
D-Rings (6)	N/A	842139248	
patch cords 12 cords, 4-pair, 5' (1.5 m)	N/A	846619989	
D8W cords 24 cords, 14' (4.3 m)	N/A	103786802	
Template	N/A	846613933	
Instruction sheet	N/A	846613941	
Parts List	N/A	846623924	

Glossary

GL

7500B Data Module — A data communications device that allows connection between RS-232 data terminal equipment (DTE) and the communications system control unit via MLX station jacks on the 008 MLX or 408 GS/LS-MLX module. The 7500B Data Module is used together with a modem in a modern pool to change digital signals to analog signals, and vice versa, which allows transmission between digital and analog data stations.

account code — A 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 or the originating terminal of data, such as the dialed extension number assigned to the data terminal, Multiple terminals on one communications line, for example, must each have a unique address.

adjunct — Optional equipment used with the communications system such as an alerting device that connects to a multiline telephone or to a telephone jack.

alternate mark inversion — See AMI.

AMI — (Alternate mark inversion) A line-coding format in which a binary 1 is represented by a positive or negative pulse and a binary 0 is represented by no line signal, Subsequent binary 1's must alternate in polarity or a bipolar violation will occur. AMI is used in the DS1 interface.

analog transmission A mode of transmission in which information is represented in continuously variable physical quantities such as amplitude, frequency, phase, or resistance. See *a/so* digital transmission.

ANI (Automatic Number Identification) The 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. INFO-2 (Information Forwarding-2) is AT&T's ANI service.

application Software and/or hardware that adds functional capabilities to the communications system. For example, the 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 telephones.

ARS (Automatic Route Selection) Routes calls over outside trunks according to the number dialed and the trunks available.

ASCAP American Society of Composers, Artists, and Producers

ASN (AT&T Switched Network) AT&T telecommunications services provided through a PRI line or trunk: Accunet switched digital service, Megacom WATS, Megacom 800, Software Defined Network (SDN), MultiQUEST, and Shared Access for Switch Services (SASS).

asynchronous data transmission A method of transmitting a short bit stream of 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, permitting data to be transmitted at irregular intervals. *See also* synchronous data transmission.

AT&T Switched Network *See* ASN.

AUDIX Voice Power A voice-processing application, part of Integrated Solution II or Ill (IS II/III), that provides automated attendant, call answering, voice mail, message drop, and information services for use with the communications system.

Automated Attendant An IS II, 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.

auxiliary power unit A device that provides additional power to the communications system.

B8ZS	(Bipolar 8 zero substitution) A line-coding format that encodes a string of 8 zeros in a unique binary sequence using bipolar violation. *See also* bipolar signal and bipolar violation.
B-channel	A 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 with other digital information streams. Also called *bearer channel.*
barrier code	A password used to limit access to the Remote Access feature of the communications system.
basic carrier	A piece of hardware that holds and connects the processor module, power supply module, and up to five line/trunk or station modules in the communications system. *See also* expansion carrier.
Basic Rate interface	*See* BRI.
baud rate	A unit of transmission speed equal to the number of signal events per second. *See also* bit rate *and* bits per second,
bearer channel	See B-channel.
Behind Switch mode	A mode of operation in which the communications system control unit is connected to (is “behind”) another communications system.
binary code	An electrical representation of quantities or symbols expressed in the base-2 number system.
bipolar 8 zero substitution	*See* B8ZS.
bipolar signal	A digital signal in which pulses (1's) alternate between positive and negative. *See also* AMI, B8ZS, and bipolar violation.
bipolar violation	A condition that occurs when two positive or two negative pulses are received in succession. *See also* AMI, B8ZS, and bipolar signal.
BIS	Built-in speakerphone
bit	(Binary digit) One unit of information in binary notation, having two possible values: zero or one.
bit rate	The speed at which bits are transmitted, usually expressed in bits per second. Also called data rate. *See also* baud rate and bits per second.
bits per second	(bps) The number of binary units of information that are transmitted or received per second. *See also* baud rate and bit rate.

blocking A condition in which end-to-end connections cannot be made on calls because of a full load cm all possible services and facilities.

BMI Broadcast Music Incorporated

BRI (Basic Rate Interface) A standard ISDN frame format that specifies the protocol used between the communications system and a terminal. BRI runs at 192 kbps and provides two 64-kbps voice or B-channels and one 16-kbps signaling or D-channel per port. The remaining 48 kbps are used for framing and D-channel contention.

bus A multiconductor electrical path used to transfer information over a common connection from any of several sources to any of several destinations.

button A key on the face of a telephone or console that is used to access a line, activate a feature, or enter a code on a communications system.

byte A sequence of bits (usually eight) processed together. "Octet" is used instead of “byte” in CCITT (International Telegraph and Telephone Consultative Committee) documentation.

calling group A team of agents who answer the same types of calls.

CAS (Call Accounting System) A DOS or UNIX-based application that monitors and manages telecommunications costs.

CAT (Call Accounting Terminal) A 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 DS1 module, channel 24 can be designated as the signaling channel for channels 1 to 23 by selecting “common channel” for emulated service when programming the system. CCS must be PRI service.

central office *See* CO.

Centrex A set of communications system features a user can subscribe to on telephone lines from the local telephone company.

channel	A telecommunications transmission path for voice and/or data.
Channel Service Unit	*See* CSU.
circuit-switched data call	A data call made via a connection exclusively established and maintained between data stations for the duration of the data call.
clock synchronization	The operation of digital facilities from a common clock.
CMS	(Call Management System) A DOS-based application that simulates the actions of a system operator by answering and distributing calls. CMS also produces management reports for call analysis.
CO	(Central office) The location of telephone switching equipment that provides local telephone service and access to toll facilities for long distance calling.
codec	(Coder-decoder) A 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	The software–controlled processor complex that interprets dialing pulses, tones, and/or keyboard characters and makes the proper interconnections both inside and outside the system. The communications system itself consists of a digital computer, software, a storage device, and carriers with special hardware to perform the actual connections. A communications system provides voice and/or data communications services, including access to public and private networks, for telephones and data terminals on a customer's premises.
control unit	The housing, carriers, power supply, processor, and modules of a communications system.
conversion resource	See modem pool.
CRC	(Cyclic redundancy check) An error-detection code used on DS1 facilities with the ESF format.
CSU	(Channel Service Unit) Equipment used at a customer's premises to provide DS1 facility terminations and signaling compatibility.

cyclic redundancy check	See CRC.
D4 framing format	A 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.
D-channel	The 16-kbps or 64-kbps channel carrying signaling or data on a Basic Rate Interface (BRI) or 64–kbps channel carrying signaling or data on a Primary Rate Interface (PRI).
data channel	*See* D-channel.
data communications equipment	*See* DCE.
data hunt group	*See* DHG.
data rate	*See* bit rate.
data terminal	An input/output device that can be connected to the communications system control unit via an interface.
data terminal equipment	*See* DTE.
DCE	(Date communications equipment) Equipment such as modems or data modules used to establish, maintain, and terminate a connection between the communications system and DTE, such as printers, host computers, or workstations.
DCP	An AT&T proprietary protocol to transmit both digitized voice and data over the same communications link. A DCP link is made up of two 64-kbps information (1) channels and one signaling (S) channel similar to the B- and D-channels used in an ISDN.
dedicated feature buttons	The imprinted feature buttons on a telephone: **Conf** or **Conference, Drop, HFAI** (Hands Free Answer on Intercom), **Hold, Mute** or **Microphone, Speaker** or **Speakerphone, Transfer, Message, and Recall.**
DFT	(Direct facility termination) A CO line/trunk that terminates directly on one or more telephones; in Hybrid/PBX mode, a DFT cannot be part of a trunk pool.
DHG	(Data hunt group) A 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	A code (usually a 9) dialed by single-line telephone users and multiline telephone users with System Access buttons to get an outside line.
DID	(Direct Inward Dialing) A service that transmits the called extension to the communications system from the central office and routes incoming calls directly to the called extension, calling group, or outgoing trunk pool, bypassing the system operator.
DID trunk	An incoming trunk that receives dialed digits from the local exchange, allowing the communications system to connect directly to an extension without assistance from the system operator.
digital	The representation of information in discrete elements such as *off* and *on* or 0 and 1. *See also* analog transmission.
Digital Communications Protocol	*See* DCP.
digital switch element	See DSE.
digital transmission	A mode of transmission in which information 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) A switch on a 400EM module used to select the signaling format for tie-line transmission. DIP switches are also used on other equipment for setting hardware options.
direct facility termination	*See* DFT.
Direct Inward Dialing	*See* DID.
Direct Station Selector	*See* DSS.
display buttons	The buttons on an MLX display telephone used to access the telephone's display.
DLC	(Direct-Line Console) An answering position used by system operators to answer calls, transfer calls, make calls, set up conference calls, and monitor system operations. Calls can ring on any of the line buttons, and several calls can ring simultaneously (unlike the QCC where calls are

	sent to a common QCC queue and wait until a QCC is available to receive a call).
DNIS	(Dialed Number Identification Service) A service provided by ASN that routes incoming 800 or 900 calls according to customer-selected parameters, such as area code, state, or time of call.
DOS	Disk operating system
DPI	Dots per inch.
DS0	(Digital Signal 0) A single 64-kbps voice or data channel.
DS1	(Digital Signal 1) A bit-oriented signaling interface that multiplexes 24 64-kbps channels into a single 1.544-Mbps stream.
DSE	(Digital switch element) A device in each jack on each module in the communications system control unit that interfaces with the TDM bus.
DSS	(Direct Station Selector) A 60-button adjunct that enhances the call-handling capabilities of an MLX-20L telephone or MLX-28D telephone when used as an operator console.
DTE	(Date terminal equipment) The equipment that makes up the endpoints in a connection over a data circuit, for example, a data terminal, host computer, or printer.
DTMF signaling	(Dual-tone multifrequency signaling) Touch-tone signaling from telephones using the voice transmission path. The code for DTMF signaling provides 12 distinct signals, each composed of two voice–band frequencies.
Dual-tone multifrequency signaling	*SEE* DTMF signaling
E&M signaling	Trunk supervisory signaling, used between two communications systems, in which signaling information is transferred through two-state voltage conditions (on the E&M leads) for analog applications and through two bits for digital applications. *See also* tie trunk.
EIA	(Electronic Industries Association) A trade association of the electronics industry that establishes electrical and functional standards.
EMI	Electromagnetic interference.
endpoint	The final destination in the path of an electrical or telecommunications signal.
ESD	Electrostatic discharge.

ESF	(Extended superframe format) A framing format consisting of individual frames of 24 eight-bit slots and 1 signal bit (193 bits) in a 24-frame extended superframe. *See also* D4 framing format.
expansion carrier	A carrier added to the control unit when the basic carrier cannot house all the modules needed. An expansion carrier houses a power supply module and up to six additional line/trunk and station modules.
extended superframe format	*See* ESF.
facility	The equipment constituting a telecommunications path between the communications system and the central office.
factory setting	The default state of a device or feature if the user does not choose an optional setting.
fax	(Facsimile) A processor the result of a process in which graphic material is scanned and the information converted into electrical signal waves to produce an exact likeness.
feature	A function or service provided by a hardware or software product.
feature code	A code entered on a dialpad to activate a feature. For example, a user might press the **Feature** button and dial 33 or might dial `#33`.
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 octets, which equals 193 bits.
frequency generator	A circuit pack added to the power unit module that generates a high-voltage, 20-30 Hz signal to ring a telephone. Also called a *ring generator.*
FX	(Foreign exchange) A central office (CO) other than the one providing local access to the public network.
gateway	A workstation on a local area network (LAN).
general purpose adapter	*See* GPA.
glare	The loud dual-tone multifrequency (DTMF) signal an incoming caller hears when another caller tries to call out on a line/trunk at the same time the call is coming in on that line/trunk.
GPA	(General Purpose Adapter) A device that connects an analog multiline telephone to optional equipment, such as an answering machine or a fax machine.

ground-start trunk	*See* GS trunk.
Group IV (G4) Fax machine	A fax unit offering 400x 100 dots per inch (DPI) in fine mode, which can operate at any speed for communication with a Group Ill (G3) Fax machine or another Group IV (G4) Fax machine,
GS trunk	(Ground-start trunk) A trunk on which the communications system, after verifying that the trunk is idle (no ground on tip), transmits a request for service (puts ground on ring) to a distant central office (CO).
headset	An ultralight earpiece and microphone for hands–free telephone operation.
HFAI	Hands Free Answer on Intercom
HFU	Hands-free Unit
Hybrid/PBX mode	A mode of operation in which the communications system uses trunk pools and Automatic Route Selection (ARS) in addition to personal lines—that is, direct facility terminations on line buttons. The Hybrid/PBX mode also provides a single interface to users for both internal and external calling.
immediate-start tie trunk	A tie trunk on which no start signal is necessary and dialing can begin immediately after the tie trunk is seized.
in-band signaiing	*See* robbed-bit signaling.
In-Range Out-of-Building protector	*See* lROB protector.
inside dial tone	A tone the user hears when connected to an Intercom line.
Inspect screen	A display screen on digital telephones that allows users to preview incoming calls and see a list of the features programmed on line buttons.
integrated Solution II/III	*See* IS II/IS Ill.
integrated Voice Power Automated Attendant	An IS II/III application that automatically answers incoming calls with a recorded announcement and directs callers to a department, an extension, or the system operator.
interface	Hardware, software, or both 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 (in-Range Out-of-Building protector) A surge protection device for off-premises telephones at a location within 1000 feet (305 meters) of cable distance from the communications system control unit.

IS II/III (Integrated Solution II/III) A UNIX-based platform of applications for improving voice and data communications and automating office operations.

jack A device, accessed by inserting a plug, that is used to terminate the permanent wiring of a circuit.

kbps Kilobits per second

Key mode A mode of operation in which the communications system uses direct facility terminations on line buttons with a separate path for internal calling.

LAN (Local area network) A networking arrangement designed for a limited geographical area.

LED (Light-emitting diode) A semiconductor device that produces light when voltage is applied. LEDs show the operational status of hardware components, the results of maintenance tests, the alarm status of circuit packs, and the activation of telephone features.

line and trunk assignment The assignment of lines and trunks connected to the communications system control unit to specific buttons on each telephone.

line coding The pattern data assumes as it is transmitted over a communications channel,

line compensation An adjustment for the amount of cable loss in decibels (dBs), based on the length of cable between a DS1 module and a Channel Service Unit (CSU) or other far-end connection point.

line/trunk and station module A module on which the jacks for connecting central office (CO) lines/trunks and/or the jacks for connecting the stations are located.

local area network See LAN.

local host computer access A method for connecting a station jack to an on-site computer for data-only calls through a modem or data module.

logical ID A numbering sequence used to identify station and line/trunk locations on the communications system control unit.

loop-start line/trunk	*See* LS line/trunk.
LS line/trunk	A line/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 from the central office (CO) signals an incoming call.
Magic On Hold	A customized Music-on-Hold system enhancement that promotes the customer's products and services.
Mbps	Megabits per second
Megacom	AT&T's tariffed digital WATS offering for outward calling.
Megacom 800	AT&T's tariffed digital 800 service for inward calling.
MERLIN Attendant	An application with equipment that connects to one or more tip/ring station ports and automatically answers incoming calls with a recorded announcement. In response to touch-tone digits dialed by the caller, MERLIN Attendant directs the caller to a department, an extension, or the system operator.
MERLIN MAIL Voice Messaging System	An application that provides automated attendant, call answering, and voice-mail services on the communications system.
MFM	(Multi-Function Module) An adapter that provides a tip/ring interface for the connection of optional equipment such as answering machines, external alerts, and fax machines to an MLX (digital) telephone. The optional equipment and the MLX telephone operate simultaneously and independently. The MFM is installed inside the MLX telephone.
MLX-lO/MLX-10D telephone	A 10-button telephone offered with or without a 2-line by 24-character menu-driven display.
MLX-20L telephone	A telephone with 20 programmable line or feature buttons and a 7-line by 24-character menu-driven display.
MLX-28D telephone	A telephone with 28 programmable line or feature buttons and a 2-line by 24-character menu-driven display.
mode codes	Streams of touch-tone codes used by voice messaging applications to communicate with the communications system's control unit.
modem	A device that converts digital data signals to analog signals for transmission over telephone lines. The analog signals are converted back to the original digital signals by another modem at the other end of the line.

modem pool A 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 digital data stations to communicate with users who have analog stations.

module A module in the control unit provides the capability to connect central office trunks and/or telephones to the system.

Multi-Function Module *See* MFM.

multiplexing A process in which a transmission channel is divided into two or more 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 Magic On Hold or a customer-provided music source connected to the communications system via a loop-start jack, Most Music-on-Hold equipment is designed for loop-start operation.

network A 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.

Off–premises telephone *See* OPT.

ones density The requirement for channelized DS1 service to the public network that eight consecutive zeros cannot be in a digital data stream.

OPT A telephone located in a building other than where the control unit is located.

outcalling A feature of the MERLIN MAIL Voice Messaging System application. When outcalling is activated, the user is automatically called by the system at a programmed number when a new message is received in the user's mailbox.

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.

parity The addition of a bit to a bit string so that the total number of 1s is odd or even. Parity can be used to detect and correct transmission errors.

PBX	Private branch exchange
PC	Personal computer
PCM	Pulse code modulation.
personal line	A central office line that rings only at the user's telephone.
pool	On a Hybrid/PBX system, a grouping of outside trunks that users can choose with multiple pool buttons or by dialing access codes on a System Access button on the telephone. Pools are also used by the Automatic Route Selection (ARS) feature to choose the least expensive method to route a call.
port	A point of access into a communications system, computer, network, or other electronic device.
power supply module	A device that directs electricity to modules and telephones on the communications 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) A standard interface that specifies the protocol used between two or more communications systems. PRI runs at 1.544 Mbps and, as used in North America, provides twenty–three 64-kbps B-channels (voice or data) and one 64–kbps D-channel (signaling). The D-channel is the 24th channel of the interface and contains multiplexed signaling information for the other 23 channels.
Primary Rate Interface	*See* PRI
prime line	An individual extension number assigned to a telephone in a Behind Switch System. Each telephone user has a prime line and is automatically connected to that line upon lifting the handset.
processor module	The module in the second slot of the basic carrier that contains the software that runs the communications system.
protocol	A set of conventions governing the format and timing of message exchanges between devices, such as an analog multiline telephone and the communications system control unit.
public network	A network that is commonly accessible for local or long-distance calling. Also called *public switched telephone network (PSTN).*
QCC	(Queued Call Console) An answering position available to MLX-20L telephone users only in the Hybrid/PBX mode. The QCC is used by system operators to answer and direct (transfer) calls, serve as a message center, and monitor system operation. Calls are sent to a common QCC queue

where they wait until a QCC is available to receive a call (unlike the DLC where calls can ring on any of the line buttons, and several calls can ring simultaneously).

Queued Call Console — *See* QCC.

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 the memory.

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.

RS–232 — A physical interface, specified by the EIA (Electronics Industries Association), that transmits and receives asynchronous data at speeds of up to 19.2 kbps over cable distances of 50 feet.

SAA — (Supplemental alert adapter) A 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.

SDN — (Software Defined Network) An AT&T private networking service created by specialized software within the public network.

SID — Station ID.

signaling — The sending of control and status information between devices to set up, maintain, or take down a connection.

simplex signaling — The transmission of signals in one direction only across a telecommunications channel.

single-line telephone — An industry-standard touch-tone or rotary telephone that only handles one trunk and is connected to the communications system via a jack on a basic telephone module.

SMDR — (Station Message Detail Recording) Captures detailed usage information on incoming and outgoing voice and data calls.

SMDR printer — A printer used for Station Message Detail Recording (SMDR) that is connected to the communications system via an RS-232 jack on the processor.

Software Defined Network — *See* SDN.

SPM	(System Programming and Maintenance) A DOS-or UNIX-based application for programming and maintaining the communications system.
Square Key	A way of configuring the communications system in Key mode so that all lines appear on all telephones.
SSN	(Switched service network) A network consisting of terminals, transmission lines, and at least one exchange on which a user can communicate with any other user at any time.
station	The endpoint on the internal side of the communications system, A station can be a telephone with or without an adjunct or can be a data terminal with a modem (analog) or a 7500B Data Module (digital) attached.
station jack	An analog, digital, or tip/ring interface on the control unit module for connecting telephones and other equipment.
Station Message Detail Recording	*See* SMDR.
Supplemental Alert Adapter	*See* SAA.
switched service network	*See* SSN.
switchhook flash	Operation of the telephone switchhook in which the on–hook period is in the range of 250–500 ms.
synchronous data transmission	A method of transmitting a continuous digital data stream in which the transmission of each binary bit is synchronized with a master clock.
system date	The date that appears on MLX display telephones and Station Message Detail Recording (SMDR) reports.
System Programming & Maintenance	*See* SPM.
system renumbering	A process used to change the extension numbers assigned to telephones, adjuncts, calling groups, paging groups, Call Park Zones, Remote Access, and lines/trunks.
system time	The time that appears on MLX display telephones and is printed on Station Message Detail Recording (SMDR) reports.
T1	A digital transmission carrier path that in North America transmits at the DS1 rate of 1.544 Mbps.

TDM	(Time-division multiplexing) A process by which the transmission channel is divided.
telephone power supply unit	Equipment that provides power to an individual telephone.
tie trunk	A private line directly connecting two communications systems.
Time-division multiplexing	*See* TDM.
timer	A built-in timing device in a display telephone.
tip/ring	The contacts and associated conductors of a single-fine telephone plug or jack.
touch-tone receiver	*See* TTR.
T/R	*See* tip/ring.
trunk jack	A jack that connects an outside trunk to the communications system control unit.
TTR	(Touch-tone receiver) A device used to decode touch–tones dialed from single–line telephones or Remote Access telephones.
Uninterruptible power supply	*See* UPS.
unit load	A measure of the power load drain of a module, telephone, or adjunct,
UPS	(Uninterruptible power supply) A device that connects to the communications system to provide 117 VAC to the equipment when the commercial power source fails.
VAC	Volts AC.
VDC	Volts DC.
VMI	Voice messaging interface.
voice-band channel	A transmission channel, generally the 300-3400-Hz frequency band.
voice-only	A telephone that is set up for making and receiving voice calls but not data calls.
voice signal pair	A pair of leads on an analog multiline telephone used for the Voice Announce to Busy feature.
WATS	(Wide Area Telecommunications Service) A service that allows calls to certain areas for a flat–rate charge based on expected usage.

wink-start tie trunk A 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.

ZCS (zero code suppression) A binary coding scheme that ensures a data stream contains at least a minimum number of information bits (1s) for receiver synchronization.

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