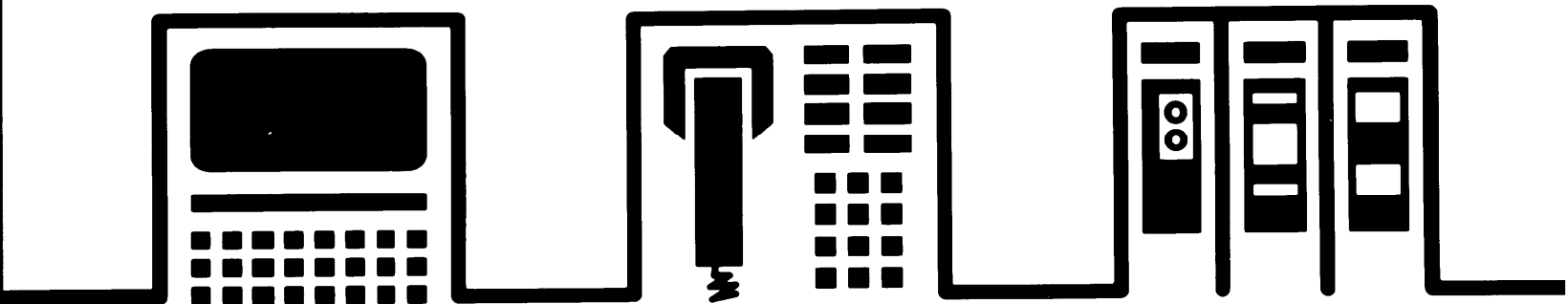




555-520-200
Issue 1, June 1987

AT&T System 25 Reference Manual



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Reference Manual
Prepared by System 25
Document Development Group and the
AT&T Documentation
Management Organization

FCC NOTIFICATION AND REPAIR INFORMATION

AT&T SYSTEM 25

This telephone equipment is registered with the Federal Communications Commission (FCC) in accordance with Part 68 of its Rules. In compliance with the Rules, be advised of the following:

MEANS OF CONNECTION

Connection of this telephone equipment to the nationwide telecommunications network shall be through a standard network interface USOC RJ21X jack. Connection to private line network channels requires USOC RJ2GX jack for tie lines or USOC RJ21X jack for off-premises station lines. These can be ordered from your telephone company.

NOTIFICATION TO THE TELEPHONE COMPANY

If the system is to be connected to off-premises stations (OPSs), you must notify the telephone company of the OPS class of service, OL13C, and the service order code, 9.OF.

Upon the request of the telephone company, inform them of the following:

- The Public Switched Network “lines” and the Private “lines” to which you will connect the telephone equipment.
- The telephone equipment’s “registration number” and “ringer equivalence number” (REN) from the label on the equipment.
- For private line connections, provide the facility interface code, TL31M for tie lines. You must also specify the service order code, 9. OF.
- The quantities and USOC numbers of the jacks required.
- For each jack, provide the sequence in which lines are to be connected; the type lines and the facility interface code and the ringer equivalence number by position, when applicable.

This telephone equipment should not be used on coin telephone lines. Connection to party line service is subject to state tariffs.

REPAIR INSTRUCTIONS

If you experience trouble with this telephone equipment, contact the AT&T Business Customer Service Center on 1-800-242-2121. The telephone company may ask that you disconnect this equipment from the network until the problem has been corrected or until you are sure that this equipment is not malfunctioning.

RIGHTS OF THE TELEPHONE COMPANY

If your telephone equipment causes harm to the telephone network, the telephone company may discontinue your service temporarily. If possible, they will notify you in advance. But if advance notice isn't practical, you will be notified as soon as possible. You will be informed of your right to file a complaint with the FCC.

Your telephone company may make changes in its facilities, equipment, operations, or procedures that could affect the proper functioning of your equipment. If they do, you will be notified in advance to give you an opportunity to maintain uninterrupted telephone service.

HEARING AID COMPATIBILITY

The voice terminals described in this manual are compatible with inductively coupled hearing aids as prescribed by the FCC.

FCC REGISTRATION INFORMATION	
Registration Number	AS593M-71565-MF-E
Ringer Equivalence	0.5A
Network Interface	RJ21X or RJ2GX

PRIVATE LINE SERVICE	
Service Order Code	9.OF
Facility Interface Code	
● Tie Lines	TL31M
● Off-Premises Stations	OL13C

FCC WARNING STATEMENT

Federal Communications Commission (FCC) Rules require that you be notified of the following:

- This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause interference to radio communications.
- It has been tested and found to comply with the limits for a Class A computing device pursuant to Subpart J of Part 15 of FCC Rules, which are designed to provide reasonable protection against such interference when operated in a commercial environment.
- Operation of this equipment in a residential area is likely to cause interference in which case the user at his or her own expense will be required to take whatever measures may be required to correct the interference.

DANGER

The AT&T System 25 cabinets are not user serviceable. Some voltages inside the cabinets are hazardous. This equipment is to be serviced only by qualified technicians.

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SECTION 1—OVERVIEW

This reference manual provides general technical information on AT&T System 25 (System 25). It includes a description of the system, its hardware and software, features and services, environmental requirements, and technical specifications. This manual is intended to serve as an overall technical reference for System 25.

This manual replaces *AT&T System 25 Reference Manual* (555-50-200, Issue 1), which covered Release 1 Version 1 (R1V1) of System 25. This new issue contains the original coverage plus complete information on R1V2, a more powerful and versatile configuration of the system. R1V2 provides new features and services that enhance system operation, particularly in the area of networking. Here are some of the improvements:

- The Virtual Facilities addition enhances the customer's outgoing network capabilities.
- The AT&T STARLAN NETWORK (STARLAN NETWORK) Access feature provides connectivity between System 25 and an associated STARLAN NETWORK.
- Switched Loop Attendant Console operation makes the handling of incoming calls from the network more efficient than in R1V1.
- With the Tandem Trunking feature, tie trunks can be used to call through System 25 to reach another switching system (CO or PBX).
- New voice terminals are provided for meeting specific system user needs.
- A new voice feature, Last Number Dialed, is added, and existing voice features are improved.
- Data services are enhanced with new options for placing and controlling calls.

Both R1V1 and R1V2 of System 25 are described in this manual. Unless specifically marked as "V1" or "V2", all of the information pertains to both versions. V1 information applies only to R1V1 systems; V2 information applies only to R1V2 systems.

Organization

The manual is divided into 10 Sections. The remaining Sections are as follows:

- SECTION 2—FEATURES AND SERVICES
- SECTION 3—FUNCTIONAL DESCRIPTION
- SECTION 4—HARDWARE DESCRIPTION
- SECTION 5—TECHNICAL SPECIFICATIONS
- SECTION 6—ENVIRONMENTAL REQUIREMENTS
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System 25 Description

System 25 (Figure 1-1) is an advanced digital switching system that integrates voice and data communications. It not only provides the features of a state-of-the-art PBX, but goes a step further by allowing data to be switched point-to-point without first being converted to analog format. This capability can be used to set up connections between data terminals, word processors, personal computers, and host computers.

System 25 uses intelligent port circuits equipped with distributed network processor elements to provide (essentially) nonblocking voice and data switching.

Voice communications features combine traditional telephone features, such as Call Transfer and Hold, with advanced features, such as Individual and Group Call Coverage, Hands-Free-Answer On Intercom, and Speed Dialing (see Section 2, "Features and Services").

Data communications features provide switched data connections supporting transmission of voice and data over Premises Distribution System wiring. Data connections can be made between two digital data modules (asynchronous data units), two analog modems, or between an analog modem and a digital data module. Release 1 Version 2 (R1V2) also provides access to STARLAN NETWORKs.

The system has data modules that provide an RS-232 interface for full duplex, asynchronous, transmission of data up to 19,200 bps, and an integrated 212A-compatible modem pool resource.

System 25 supports the following:

- Trunk and Network Facilities—Dual Tone Multifrequency (DTMF) and Dial Pulse Signaling on incoming and outgoing trunks (dial pulse only on DID trunks).
 - Loop Start (LS)
 - Ground Start (GS) (Strongly Preferred over Loop Start in most installations)
 - Tie Trunks (Type I and Type I Compatible E&M, Type V Simplex)
 - Direct Inward Dialing (DID)
- Voice Terminals – Single-Line Touch-Tone, Single-Line Rotary (V2), MET, and MERLIN®
- Data Facilities
 - Digital Data End Points – RS-232 Interfaces via Asynchronous Data Units
 - Analog Data End Points — Tip/Ring-Type Modem Interfaces
 - STARLAN NETWORK Access (V2 only).
- Networking Capability
 - Tie Trunks
 - Tandem Trunking
 - Endpoint in Electronic Tandem Network (Tributary only, not Satellite)
 - Endpoint of Enhanced Private Switched Communications Services (EPSCS)
 - Endpoint of Tandem Tie Trunk Network (TTTN)
 - Endpoint of Common Control Switching Arrangement (CCSA).

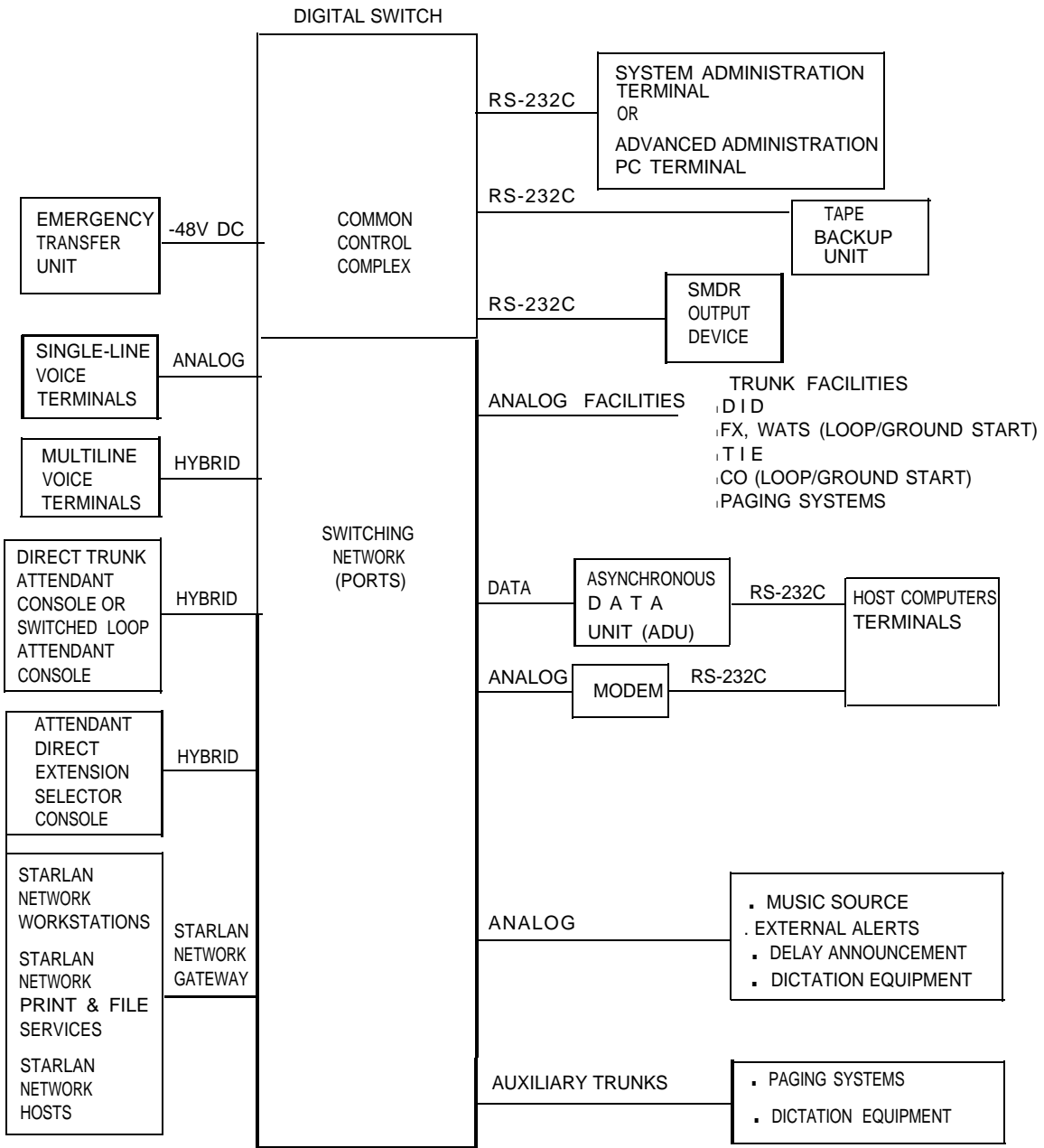


Figure 1-1. System 25 Block Diagram

Call Handling Capabilities

System 25 can be arranged as a stand-alone system or can be part of a private network. The system provides 256 ports to support the following:

- 115 simultaneous two-party conversations
- Traffic Handling Capacity of 4140 CCS/Hour (Trunking Limited)
- Busy Hour Call Capacity of 2500 calls (DTMF Register Limited)
- Up to 104 trunk ports including Central Office (CO), DID, Tie, Foreign Exchange (FX), Wide Area Telecommunications Service (WATS), and 800 Service
- An Auxiliary Trunk interface for paging (V1 and V2) and dictation systems (V2).
- Up to 240 ports that support a combination of the following:
 - Up to 200 ports for voice terminals and auxiliary feature port equipment.
 - Up to 104 data ports providing RS-232C connections to data terminals, personal or multiport computers.

Refer to Hardware and Software Parameters as provided in “Technical Specifications” (Section 5) for detailed specifications.

Safety

System 25 meets all requirements found in Underwriters Laboratories Standard for Telephone Equipment (1459).

Business Communications Needs

The remainder of this Section describes how System 25’s R1V2 features may be used to satisfy a customer’s communications needs. This material may be thought of as the reverse of the “Features and Services” section which follows.

The business communications capabilities of the majority of small businesses with more than thirty phones are provided by a Private Branch Exchange (PBX). System 25 is a PBX designed to meet the business communications needs of customers in the 30 to 150 station range.

The communications needs of most business customers may be broken down into five basic categories. Customer experience has shown that a PBX needs to provide–

- Prompt handling of incoming calls to maximize revenue opportunities and client satisfaction,
- Ease of access to and cost control of outgoing calls over public network and private facilities,
- Easy movement of calls between on-premises phones and between on-premises and off-premises phones,
- Sharing of data between PCs and/or host computers and data terminals, and
- Growth and rearrangement of facilities.

The following pages outline System 25’s outstanding ability to provide these services.

Incoming Business Communications

Successful call termination is the key to capturing all incoming communications associated with revenue issues, client inquiries, decision data, etc. Call termination involves identifying the called party and routing the call to a primary or secondary answering position. System 25 provides powerful tools for both call screening and call termination.

- Attendant Consoles allow one or two attendants to answer, screen, and steer incoming calls using either Direct Trunk or Switched Loop operation. With attendant operation, incoming calls can be screened and forwarded to the appropriate party for resolution, messages taken for absent clients, or forwarded to alternate locations. Calls may arrive over any of the network facilities described in later sections of these notes.
- Direct Inward Dialing allows incoming callers to reach specific individuals or facilities without attendant assistance. This allows specific numbers to be advertised for direct customer access to brokers, emergency services, etc. over a shared pool of DID trunks.
- Direct Group Calling (DGC) allows incoming calls to be directed to a specific group of stations. Calls to a DGC group hunt for an idle station in a circular manner, starting at the station following the last one called. If all group members are busy, calls are queued and can be sent to a delay announcement. A DGC group can terminate calls to sales, services, computer, announcement, etc. over either ordinary CO trunks or DID trunks.
- Personal Lines provide dedicated outside lines for multiline voice terminal users and are accessed via a dedicated button for both incoming and outgoing service. Up to sixteen terminals may share a Personal Line with up to four parties simultaneously off-hook. A personal line provides direct access to brokers, emergency service, etc. over a dedicated loop start or ground start trunk.

Frequently, the called party is not available to handle an incoming call. System 25 provides a number of methods for redirecting incoming calls to alternate resources.

- Call Following allows users who are away from their phone to receive calls at another phone. Users may login their Personal Dial Code (PDC) at any other System 25 voice terminal and receive their calls at that terminal. This feature supports roving personnel and shared office space for company staff.
- Call Coverage allows calls that are not answered within a specified number of rings to be redirected to an individual covering station and/or a group of covering stations. This is especially useful for Boss-Secretary arrangements, staff backup, and message service. This feature is versatile enough to permit suitable alternate answering arrangements for virtually every level of employee. Special features, such as the Send All Calls feature, which routes a user's calls directly to covering station(s), accommodate the day-to-day variations that occur in an employee's work schedule.
- Station Hunting provides automatic redirection of incoming calls to an idle member of a hunt group when the called party is busy.
- Call Pickup allows a user to answer a call ringing at another terminal. Directed Call Pickup allows a user to answer a call ringing at any terminal by dialing the call pickup code and the PDC of the ringing station. Group Call Pickup permits calls to any other terminal in the call pickup group to be answered by dialing the group call pickup code. With Call Pickup, users do not have to leave their phone to answer other's calls. This feature is especially useful for local coverage in group offices not

supported by secretarial service and equipped with economical single-line phones.

When alternate resources are not available to handle an incoming call, System 25 provides for attendant handling of the call utilizing camp-on, redirection and/or message service.

- Camp-On allows the attendant to extend an outside call to a busy station. A burst of tone is heard at the called station to notify the user of the camped-on call. The caller is placed on hold and hears music-on-hold, if available. When the user hangs up, the camped-on call begins ringing immediately. Only one call may be camped on at a time. The Return Coverage on Busy feature returns unanswered camped-on calls to the attendant for service after a specified interval.
- Return Coverage on Don't Answer returns unanswered attendant-extended calls for additional service (redirection/messaging).
- Messaging Service supports activation of an LED at the called station to indicate that the attendant, message desk, or another station has a message for the user.

Special arrangements are needed to handle incoming calls during periods when the normal staff is not available, for example at night and on weekends. System 25's Night Service feature allows on-duty personnel to answer incoming attendant-seeking calls when the attendant is not on duty. Directed Night Service redirects incoming attendant-seeking calls to designated voice terminals, such as a guard desk or coverage position. Trunk Answer From Any Station allows users to answer incoming calls from any station by dialing the Night Service access code. Night personnel can be alerted by a Night Bell.

Outgoing Business Communications

One of the key functions of a customer premises communications system is to provide easy access to the most cost effective network facilities for outgoing calls. The system needs to be capable of steering calls based on cost, and must also keep records of incoming and outgoing calls and associated costs. Building on its ground start trunk capability, System 25 features control costs and record usage as follows.

- Call Restrictions allow the manager to restrict users from making certain types of calls. Restriction is administered through outward restriction, toll call restriction, and facility access restriction.
- Automatic Route Selection provides manager defined routing of calls over the telecommunications network based on preferred routes (normally the least expensive route available at the time the call is placed) with capacity for multiple common carriers and routing through tandem switch points. The user dials a standard DDD number and the system selects the call route.
- Station Message Detail Recording (SMDR) generates detailed call information on all incoming and outgoing calls and sends this information to an output device (PC or printer).
- Call Accounting Systems provide multiple types of customer reports on communication costs and usage.
- Account Code Entry allows a user to associate calls with an account code for charge-back purposes.

Ease of access to multiple types of network facilities (provided for minimum cost) is managed by:

- Automatic Route Selection (ARS) allows the customer to dial a standard DDD number. ARS selects the preferred route and does any number conversions required for the facilities selected.
- System 25's Virtual Facility feature provides convenient and inexpensive access to OCCs. This feature provides access to OCC facilities over a user specified *physical* facility; dedicated OCC trunks are not needed. Local OCC access numbers and account codes are automatically added by System 25. System 25's Virtual Facility feature is fully integrated with its ARS, Toll Restriction, and SMDR/CAS features.
- Last Number Dialed automatically saves the last number dialed and allows the user to retry the number without redialing. (Multiline voice terminals only)
- Repertory Dialing allows multiline voice terminal users to store a telephone number or account and associate that number with a button on their voice terminal. Pressing a Repertory Dial button is equivalent to dialing the stored number (one-touch dialing).
- System Speed Dialing allows all users to dial 90 selected numbers using three-digit codes. Users can also program up to seven Personal Speed Dial Numbers which are accessible only from their terminals. System Speed Dialing can be used by the system administrator to hide business account codes from users.
- Pooled Facility-Dial Access allows both multiline and single-line voice terminal users to access a common pool of trunks for outgoing calls by dialing a facility access code, or, on multiline voice terminals, by pressing a button . This grouping provides resource pooling which results in better service with a given number of trunks.
- Personal Lines provide dedicated outside lines for multiline voice terminal users. Personal lines are accessed via a dedicated feature button. Up to sixteen terminals may share a personal line.
- Third-Party Call Setup allows PCs to set up calls between a System 25 voice/data terminal and any other facility. A PC application program could use this capability to retrieve information from a database.

Last Number Dialed, Repertory Dialing and Speed Dialing are also applicable to dialing and managing internal calls. Personal lines provide both incoming and outgoing service.

Internal Call Movement

Typically, about 40 percent of PBX calls are internal calls, call transfers to another location, conference of multiple locations, temporarily suspended calls, page to locate the called party, etc. Rapid placement of internal calls and easy call movement from the answering station to a new station are supported in System 25 with numerous features.

To provide easy internal call setup, System 25 provides the following features.

- Direct Station Selection (DSS) allows one-button access from a multiline voice terminal to another voice terminal, a pooled facility, paging zone or DGC group. The DSS status LED reflects the idle/busy status of the associated termination point. This feature is used to track and contact frequently called associates.

- Automatic Intercom allows multiline voice terminal users to call each other by use of a dedicated line appearance. A private dedicated path ensures that a path is always available. This feature is frequently used in Boss/Secretary arrangements.
- The Dial Plan for System 25 is based on the concept that, whenever possible, calls should be placed to individuals rather than to pieces of equipment. To implement this concept, individuals are assigned Personal Dial Codes (PDCs) and are allowed to login those PDCs at other terminals. The system automatically routes the call to the home terminal or logged-into terminal. This significantly increases the probability of reaching the called party. In addition, the dial plan is built on a flexible numbering scheme which allows the number of dialed digits to match assigned PDCs (2/3/4 digit dial plans) and to be administered to match telephone company assigned Direct Inward Dialing numbers.

Efficient internal call termination is supported with the following features.

- Distinctive Ringing provides two types of ringing to allow users to distinguish between outside calls and inside calls,
- Hands-Free Answer on Intercom (HFAI) allows Speakerphone and HFAI terminals to auto-answer inside or attendant extended calls. With HFAI active, the set generates a tone burst over its speaker to alert the calling and called party of the call completion. Both parties may then converse; no action by the called party is required.

Frequently, the first termination point for a call is not its final destination. To support internal call movement, System 25 provides the following features.

- Transfer allows a user to transfer any call to another voice terminal. This feature supports transfer of calls from the answering position to another location for completion of a transaction. Examples are secretary to boss, office to lab, broker to specialist, etc.
- Conference allows up to five parties (maximum two outside), including the originator, to participate in a call. This feature supports add-on of additional parties to a call for joint consultation, crisis management, schedule coordination, etc.
- Hold allows a user to suspend a call. The Hold feature allows users to temporarily disconnect from one conversation and either place or answer another call. Music or information bulletins may be provided to the held party. Called parties frequently use the hold period to access computer data bases, search categories and/or consult with others via a second phone call.
- Call Park allows a user to place a call or conference on hold and then pick up the call from any voice terminal. The user can page another party to pick up the parked call or may move to another location and then re-access the call.

Data Communications

Small Business customers have started to integrate PCs into their day-to-day business operations. Businesses have found a need to access the data bases (sales, inventory, personnel) in these PCs from more than one location (both on- and off-premises). System 25 data features are specially engineered to enhance a user's ability to access data from multiple locations. System 25 has been designed to help these businesses use their personal computers, data terminals and host computers more effectively by providing the following features.

- Circuit switched data communications up to 19,200 bps (RS232 interface). This provides circuit switched connections from asynchronous data terminals, PCs, or host computers to host computers or network facilities. Users can be located and/or moved to any on-premises office equipped with the standard AT&T four-pair wiring plan. Thus an asynchronous terminal or PC can have access to multiple host computers, remote data bases via a modem pool, and a local area network (STARLAN) via System 25's STARLAN NETWORK gateway.
- Packet switched data connections at 1 million bps over AT&T's STARLAN NETWORK local area network. This provides data transfer between client PCs and servers (PCs/host computers/printers, etc.) on the local area network (LAN). LAN users can be located and/or moved to any on-premises office equipped with standard AT&T four-pair wiring. The LAN allows PCs to share facilities (printers, disk systems, modem pools, etc.)
- System 25's STARLAN NETWORK ACCESS software and STARLAN NETWORK gateway provide access to the STARLAN NETWORK for off-premises and occasional on-premises users. These users do not need to install a Network Access Unit (NAU) in their PCs to use the STARLAN NETWORK ACCESS software. The data transfer rate is limited to 9600 bps or, for off-premises users, by the modem.

LAN users can access hosts connected to System 25, or, if their System 25 is equipped with a modem pool, remote hosts. Finally, terminals and PCs connected to System 25 data ports can access host computers on the LAN.

Frequently a user needs to access a LAN data base at or from a remote location (home, motel, client office, branch, etc.). To support out-of-building access to computer data over network facilities or OPS lines, System 25 provides the following features.

- Modem pooling allows data terminals to communicate over analog facilities utilizing the standard dialing plan and provides full access to all network facilities, cost control mechanisms, ARS, and incoming call management tools (DID/attendant/DGC, etc.).
- Transfer to data allows a data call to be set up on a voice terminal and then be transferred to a data terminal or computer. This feature can also be used to enter an account code for the data call.
- The System 25 STARLAN NETWORK gateway allows the LAN environment to be extended to occasional users or remote locations. Off-premises users can access the LAN utilizing all the network features, cost control mechanisms, and incoming call management facilities of System 25. The data transfer rate is governed by the modem.

Setting up data communications with PCs, host computers, and/or remote access can be a source of confusion for occasional users. Special data features are provided by System 25 to assist the user in utilizing its rich set of data communications capabilities.

- The integrated voice-data dialing plan recognizes the different types of data endpoints (digital/ analog and remote/local) in a connection and automatically inserts the required data communication equipment. In addition, autobauding supports the alignment of equipment with the capacity to transmit at different data rates.
- Station Hunting supports the use of a single dial code to access a group of host computer ports.

- Terminal Dialing provides the user with fast access to data communications via keyboard dialing at a terminal or PC.
- Command Mode provides a menu of data services supporting terminal dialing, and display and control of user data port options. A user friendly Change Options menu is provided for user administration of data options.
- Expert Mode is an enhancement that provides an alternative method of accessing Command Mode functions. Expert Mode eliminates the display of menus and allows multiple commands to be entered on a single line. Expert mode lends itself well to computer-driven scripts for call setup.
- Communication Access Manager (CAM) is an MS-DOS* software application that provides a phone manager for placing voice and data calls for the user and VT100† terminal emulation. CAM may be used on either STARLAN NETWORK client workstations or on PCs connected to System 25. CAM has a 200-entry directory with one-touch dialing for both voice and data calls and auto-login capability for data calls to host computers. CAM's Remote Access feature provides password protected unattended access to PC files and electronic mail. File transfer is supported with the popular XMODEM protocol.
- STARLAN NETWORK ACCESS is an MS-DOS application that allows PCs not connected to the STARLAN NETWORK to call through the System 25 STARLAN NETWORK Interface and run STARLAN NETWORK client software to access file and printer servers on the STARLAN NETWORK. ACCESS uses a PC's serial communications port to communicate with the STARLAN NETWORK Interface. ACCESS is compatible with NETBIOS, permitting execution of most applications written for the IBM‡ PC Network and IBM Token Ring Network.

Growth & Rearrangement

Historical data indicates that clients in the System 25 station range have a need for communications systems capable of significant growth and rearrangement. Clients need flexibility over the life of the system to easily add capacity, move stations, modify cost control options, etc. The architecture of System 25 was implemented with the objective of meeting this need.

- Advanced Administration (optional) is an easy-to-use, menu driven personal computer software package for configuring the rich set of system options.
- Uniform Wiring Plan (four-pair) allows a building to be prewired for the rich set of AT&T Small Business PBX service offerings. This modular wiring plan supports client reconfiguration of an office with variations in station type (Analog, MET, MERLIN, futures) and data configurations (LAN, asynchronous, synchronous). It supports simultaneous voice and data from standard four-pair modular jacks.
- System 25/75/85V2 Standard Architecture supports efficient growth with modular cabinets, universal carrier slots, non-blocking network and uniform wiring plan. (See

* Registered trademark of Microsoft Corp.

† Trademark of Digital Equipment Corp.

‡ Trademark of International Business Machines Corp.

Figures 1 and 2.) Every circuit slot in the system can be used for trunk cards or voice/data station cards. All these attributes allow the client to add future capability without breakage and re-engineering of existing equipment. Thus, the client is able to minimize initial investment while not restricting future growth.

Over time, the type of tools and facilities that a business utilizes changes. It is important that a PBX provide support for the full set of Telco network options over its installed life, even when only a subset is initially used. Trunks link two switching systems, such as System 25 and the local Central Office or System 25 and another PBX. System 25 supports five different telephone company trunk interfaces to provide desired connectivity at minimum expense. Thus the opportunity exists to select the best trunk types, depending on tariffs and customer needs.

- Loop Start (LS) trunks for public network access at minimum tariff. These trunks handle outgoing and incoming attendant calls, incoming DGC calls, outgoing pooled facility calls, and personal line calls.
- Ground Start (GS) trunks for public network access. These trunks handle the same type of calls as LS trunks. They provide protection against call reorigination without toll restriction, more reliable automatic route selection, virtual facilities, SMDR and CAS. Simultaneous incoming and outgoing call seizure of the same trunk under heavy traffic conditions is essentially eliminated with ground start trunks. GS trunks should usually be selected in preference to LS trunks unless tariff considerations are overriding. Note, however, that Centrex Service requires LS trunks.
- Direct Incoming Dial (DID) trunks for dialing a station directly from outside (attendant assistance not required). Outside dial access to stations, trunks (optional), and answering groups (Direct Group calling) is provided.
- Tie Trunks for linking PBXs with dedicated private circuits for high volume calling. Dial access to stations, other trunks, answering groups (Direct Group Calling) and an Electronic Tandem Network endpoint capability are provided.
- Off-Premises Stations (OPS) allow a single-line voice terminal to be located remotely and connected to System 25 via arrangements with the local Telco. This service is used to provide users at secondary sites (or their residences) many of the same features as an on-premises single-line station.

To enhance the usage and control of the above set of network facilities, System 25 provides the rich set of access features outlined in the Outgoing Business Communications Section. In addition, System 25 can support networking between systems by:

- Serving as an endpoint on an electronic tandem network (ETN) using its tie trunks and flexible dialing plan.
- Serving as an off-network or on-network access point with its dial access/transfer between tie-trunks and Telco trunks (LS/GS/DID). This allows usage of tie trunks to reach a distant System 25 and then connect through that System 25 to local Telco facilities to complete the call.

To support efficient utilization of trunks, they can be grouped together (up to 16 groups) if all trunks in the group perform the same function. This resource pooling provides better service with a given number of trunks, and simplifies administration and calling.

Types of trunks which can be assigned in System 25 are:

- Central Office, which provide a link with the local telco for incoming and outgoing calls (LS/GS)
- Foreign Exchange (FX), which connect to a CO other than the local CO for high volume calling from a distant location
- Wide-Area Telecommunications Service (WATS), which connect to an Outward WATS office or a dial "800" Service Office
- Direct Inward Dial (DID), which provide incoming service from a CO to directly access a station or facility (STARLAN NETWORK interface, trunk group)
- Tie trunks, which provide a link with another private switching system.

To support efficient utilization of this rich set of network options, System 25 provides the functions outlined in the Outgoing and Incoming Business Communications sections.

Conclusions

System 25 has been targeted at providing excellent small business communications capability at the right price. The thousands of systems in service in the first year of production have confirmed that these capabilities are an excellent match with small business customers' communications needs.

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SECTION 2—FEATURES AND SERVICES

This section describes the System Features, Network Features, Data Features, Station Features, and Attendant Features of AT&T System 25. It also covers certain services that support and implement the features; included in this category are the digital tape unit, the dial plan, system administration, and system maintenance. A general discussion of data topics is also provided.

The feature descriptions are arranged in alphabetical order, regardless of the feature group to which they belong. Information for each feature is presented under five headings: Description, Considerations, Interactions, Administration, and Hardware Requirements.

•Description

Defines the feature, describes what it does for the user, and how it is used.

•Considerations

Discusses the applications and benefits of the feature, followed by feature parameters and factors to be considered when the feature is used.

•Interactions

Lists and briefly describes other features that can affect the feature being described. Interacting features are those that:

- Depend on each other—One of the features must be provided if the other one is.
- Cannot coexist—One of the features cannot be provided if the other one is.
- Affect each other—The operation of one feature modifies, or is modified by, the operation of the other.
- Enhance each other—The features, in combination, provide improved service to the user.

•Administration Requirements

States whether or not administration is required and lists items requiring administration.

•Hardware Requirements

List any additional hardware needed to use the feature.

Tabular listings of features by group (System, Network, Data, Station, or Attendant) immediately follow this introduction. Each type, standard, or optional of the feature, is also noted on these lists:

- Standard features—Built into each system (always provided but can require administration to make them operational)
- Optional features—Such as Music-On-Hold, require both administration and additional hardware.

Features restricted to single-line or multiline voice terminals are noted where applicable. The MET sets operate the same way as 5-button 7300H series voice terminals, unless otherwise noted.

System Features

System features (Table 2-A) are those that affect the entire operation of the system. All system features are available with both V1 and V2.

TABLE 2-A. System Features

FEATURE NAME	FEATURE TYPE *
Call Accounting	O
Dial Plan	S
Dictation System Access	O
Digital Tape Unit	O
Direct Group Calling	S
Direct Group Calling Delay Announcement	O
End-to-End Signaling	S
Extended Stations	O
External Alerts	O
Intercept Treatment With Reorder Tone	S
Interdigit Timeouts	S
Music-On-Hold	O
Night Service (Directed and TAAS)	S/O†
Night Service Delay Announcements	O
Out-Of-Building Stations	O
Paging System Access	O
Personal Dial Codes	S
Pooled Facility-Dial Access	S
Power Failure Transfer	O
Remote Administration Interface	O
Station Message Detail Recording	O
System Administration	O
System Maintenance	S
Touch-Tone and Dial Pulse Service	S

* Feature types: S= Standard; O= Optional (requires additional equipment).

† S/O - Standard for Directed, Optional for TAAS Night Service.

Network Features

This group of features (Table 2-B) supports communications with the public network and with other locations in the private network of which System 25 can be a part.

TABLE 2-B. Network Features

FEATURE NAME	FEATURE TYPE *
Automatic Route Selection	S
Direct Inward Dialing	O
Off-Premises Stations	O
Tandem Trunking (V2)	O
Tie Trunks	O
Trunk Groups	S
Virtual Facilities (V2)	S

Data Features

Data Features (Table 2-C) support the switched data services of the system. Data services provide switched connections between analog and digital data endpoints.

TABLE 2-C. Data Features

FEATURE NAME	MULTILINE TERMINAL BUTTON LABEL †	FEATURE TYPE *
Command Mode		S
Communications Access Manager (V2)		O
Data Call Setup		S
Data Services Overview		
Data Terminal Dialing		S
Expert Mode (V2)		S
Modem Pooling		O
AT&T STARLAN NETWORK Access (V2)		O
Third-Party Call Setup (V2)		S
Transfer to Data	[DATA]	S
User Changeable Options (V2)		S

* Feature types: S= Standard; O= Optional (requires additional equipment).

† Bracketed items are associated voice terminal feature button labels; these labels are also used in feature descriptions where applicable.

Station Features

The many Station Features (Table 2-D) available allow individual user needs to be met. As these needs change, assigned features can also be changed. Station Features provide many important services that help save time and make calling more convenient.

TABLE 2-D. Station Features

FEATURE NAME	SINGLE-LINE TERMINAL	MULTILINE TERMINAL BUTTON LABEL †	FEATURE TYPE*
Account Code Entry	x	[ACCT ENTRY]	s
Automatic Intercom		[AUTO ICOM]	s
Busy-to-Idle Reminder		x	s
Call Accountability	x	x	s
Call Coverage-Group	x	[COVER-GRP]	s
Call Coverage-Individual	x	[COVER-IND]	s
Call Following (Forwarding)	x	x	s
Call Park	x	x	s
Call Pickup	x	x	s
Call Progress Tones	x	x	s
Calling Restrictions	x	x	s
Conference	x	x	s
Conference Drop	x		s
Direct Station Selection (DSS)		[DSS or FLEX DSS]	s
Distinctive Ringing	x	x	s
Exclusion		[EXCLUSION]	s
Hands-Free-Answer On Intercom		[AIJTO ANS]	o
Headset Adapter Adjunct	x	x	o
Hold	x		s
Last Number Dialed (V2)		[LAST # DIALED]	s
Line Selection		x	s
Line Status And I-Use Indications			s
Manual Signaling		[SIGNAL]	s
Messaging Services	x	x	s
Personal Lines		[PERS LINE]	s
Pooled Facility-Button Access		[FACILITY]	s
Program	x	x	s
Recall	x	x	s
Repertory Dialing		[REP DIAL]	s
Send All Calls		[SEND ALL CALLS]	s
Speaker (Spokesman Service)		x	s
Speakerphone Adjunct	x	x	o
Speed Dialing	x	x	s
Station Hunting	x		s
Station-To-Station Message Waiting		[MSG WAIT]	s
Test		x	s
Transfer	x	x	s
Trunk-To-Trunk Transfer	x	x	s

* Feature types: S= Standard; O= Optional (requires additional equipment).

† Bracketed items are associated voice terminal feature button labels; these labels are also used in feature descriptions where applicable.

Attendant Features

Attendant Features (Table 2-E) are available to the attendant using the Direct Trunk Attendant Console (DTAC) or the Switched Loop Attendant Console (SLAC) (V2 only) and (optionally) a Direct Extension Selector Console. In addition, most multiline voice terminal station features are available to the attendant.

TABLE 2-E. Attendant Features

FEATURE NAME	CONSOLE BUTTON LABEL †	FEATURE TYPE *
Attendant Call Extending	[START]	S
Attendant Camp-On		S
Attendant Cancel	[CANCEL]	S
Attendant Console, Direct Trunk		O
Attendant Console, Switched Loop (V2)		O
Attendant Display (V2; SLAC only)		S
Attendant Direct Extension Selection		O
Attendant Forced Release V2; SLAC only)	[FORCED RELEASE]	S
Attendant Join (V2; SLAC only)	[JOIN]	S
Attendant Message Waiting (DTAC)	[ATT MSG]	S
Attendant Message Waiting (SLAC)	[ATTENDANT MESSAGE WAITING]	S
Attendant Position Busy	[POS BIJSY]	S
Attendant Release	[RELEASE]	S
Attendant Return-Coverage-on-Busy	[RTN-BUSY]‡	S
Attendant Return-Coverage-on-Don't-Answer	[RTN-DA]‡	S
Attendant Source/Destination (V2; SLAC only)	[SOURCE]	S
	[DEST]	S
Attendant Splitting One-Way Automatic		S
Attendant System Alarm Indication	[ALARM]	S
Message Center-Like Operation (V2; SLAC only)		S
Night Service	[NIGHT]	S

* Feature types: S=Standard; O=Optional.

† Bracketed words are the labels for button-activated features; these labels are also used in feature descriptions where applicable.

‡ This button is assigned on the DTAC only.

ACCOUNT CODE ENTRY

Description

Allows voice terminal users to associate an account code with incoming and outgoing calls. This is accomplished by entering the account code at the voice terminal before hanging up. The account code is appended to the SMDR call record and can be used later for accounting or billing purposes.

To association account code with a call, the user, after completing a call but before hanging up, must:

- Single-Line Voice Terminal User:
 - Flash the switchhook and dial *O; then dial the account code directly or dial a System or Personal Speed Dial Number that contains the account code. If the number is entered incorrectly, redial *0 and the correct number before hanging up.
- Multilane Voice Terminal User:
 - Press Account Code Entry (ACCT ENTRY) button and then dial the account code directly or dial a System or Personal Speed Dial Number that contains the account code. A Repertory Dial (REP DIAL) button can also be used to enter an account code. If the number is dialed incorrectly, press ACCT ENTRY again (before hanging up) and dial the correct number.
- When the correct number of account code digits have been entered (or r# is entered to signal end-of-dialing), Confirmation Tone followed by Dial Tone is returned to the user and the account code is appended to the SMDR call record.

Account Code Entry is optional.

Considerations

Account Code Entry provides an easy method of allocating the costs of specific calls (and associated staff time) to the correct project, department or user. The account code is appended to the SMDR call record and sent to the SMDR output channel.

Account Codes can include up to 15 digits.

The validity of the entered account code is not checked by the system.

If the user is active on a call, invoking the feature will drop the call.

Incorrectly dialed codes (prior to last digit entry) can be corrected by dialing *0 or pressing ACCT ENTRY and reentering the code. Partial account codes entered by going on-hook before completing entry are recorded and cannot be corrected.

If, before all digits have been entered, (1) the user goes on-hook, (2) a button other than ACCT ENTRY is pressed, or (3) 30 seconds have elapsed since the feature was invoked, the SMDR call record will show the digits dialed up to that point.

If a call is on hold, this feature cannot be reinvoked.

Interactions

- **Conference:** If more than one user attempts to enter an account code on a Conference Call, the first to enter a code will prevail.
- **Repertory Dialing:** An Account Code can be stored on a REP DIAL button. Press REP DIAL after ACCT ENTRY has been pressed.
- **Speed Dialing:** An Account code can be stored in System or Personal Speed Dial Number.
- **Transfer:** A user can transfer a call to another user, then, before hanging up, enter an account code. Subsequent account code entries for the same call will be ignored, even though confirmation tone has been returned.

Administration Requirements

System:

- Maximum number of Account Code digits (0-15)--Default = 15.

Voice Terminal: (Station Port)

- Multiline terminals--Account Code Entry Button is required.
- Single-line terminals--none.

Hardware Requirements

Requires an RS-232 compatible 80-column ASCII (serial) printer or other device to output Station Message Detail Recording (SMDR)/Account Code entries.

ATTENDANT CALL EXTENDING

Description

Allows the attendant to put a call in a special hold condition, call another station, then connect the two calls together. The attendant can withdraw from the connection and separate the call from the console or remain connected to the other parties. Attendant Call Extending is a feature used at either a Direct Trunk Attendant Console (DTAC) or a Switched Loop Attendant Console (SLAC).

Note: In general, the attendant *should not* use the TRANSFER button, which invokes the standard multiline voice terminal Transfer feature, to extend calls. If Transfer is used, busy or unanswered calls cannot return to the attendant console for further handling.

The attendant, after placing or answering a call, can use Step 1 or 2 to extend this call to an inside extension or Step 1 to extend it to an outside number:

1. Press START to place the incoming call on hold via the Attendant Splitting One-Way Automatic feature. After receiving Dial Tone, the attendant then dials the requested inside or outside number.

or

2. Press the Selector Console Group Select and Direct Extension Selection (DXS) buttons associated with the requested inside station. This operation is equivalent to pressing START and dialing the extension.

If ringing tone is heard, the attendant presses RELEASE (Manual Release) to connect the caller to the ringing line and separate the call from the console. As an alternative, the attendant can press any facility button such as System Access, Automatic Intercom, or an outside line (Attendant Automatic Release) to complete the call extending procedure.

The attendant has the option of staying connected to the ringing line to announce the call before connecting the two parties. The attendant can then release or (SLAC only) join the other parties in a 3-way connection by using the Attendant Join feature.

If busy tone is heard and Attendant Camp-On (see associated feature description) is not desired, the attendant presses CANCEL and is reconnected to the calling party.

If busy tone is heard on a call to an inside station and Attendant Camp-On is desired, the attendant presses RELEASE or any facility button. The called party hears a tone burst, and the call waits at the called voice terminal. When a busy single-line station goes on-hook, or a busy multiline station System Access button becomes idle, the call automatically begins ringing at the station. Only one Camped-On call is permitted per voice terminal.

Calls extended to an *idle* voice terminal that are not answered within a specified time return to the Attendant Console on an idle LOOP button (SLAC only) or on the Return-On-Don't-Answer (RTN-DA) button (DTAC only). Calls camped-on at a *busy* voice terminal that are not answered within a specified time return to the Attendant Console on an idle LOOP button (SLAC only) or on the Return-On-Busy (RTN-BUSY) button (DTAC only). If a SLAC is not available to incoming calls (busy on another call, in Position Busy mode, etc.), a returning call remains in the console queue until the console can handle it. If the Return buttons on a DTAC are busy, an extended call remains at the called terminal until that button becomes idle.

Considerations

Attendant Call Extending allows the attendant to utilize the additional attendant related features such as Attendant Splitting One-Way (automatically places incoming canon hold), Release, Cancel, Return-On-Don't-Answer, Return-On-Busy, Forced Release (SLAC), Join (SLAC), and Source/Destination (SLAC).

Interactions

Refer to the other Attendant Feature descriptions for information on related features (Table 2-E).

Administration Requirements

System:

- Number of seconds before a Camped-On call returns to the Attendant Console (1-120 seconds), or No Attendant Camp-On (0) - Default = 30 seconds
- Number of rings before unanswered call returns to the Attendant Console (1-31) - Default = 5.

Hardware Requirements

Selector Console (optional)

ATTENDANT CAMP-ON

Description

Allows the attendant to extend a trunk call to a busy voice terminal and leave it waiting or “camped on” there. After hearing busy tone, the attendant presses RELEASE to camp-on this call at the busy terminal. When this is done, a burst of tone is heard in the handset of the called terminal and the caller is placed on hold (hearing music-on-hold if available). When a System Access button at a multiline set becomes idle or a single-line terminal hangs up, the camped-on call is connected automatically and ringing begins. Only one call can be camped-on to a voice terminal. This feature is referred to as a “Waiting Call” in the *User Guides* for the System 25 voice terminals.

Considerations

A camped-on call can be answered by a busy single-line user without losing the current call by momentarily pressing the switchhook (which places the current call on hold) and then dialing *9. Multiline terminal users cannot do this. However, if they have a System Access-Originate Only button, they can place both calls on hold, go off-hook on that button and dial *9 to pick up the camped-on call.

If the camped-on call is not answered within a specified time, the call will be returned to the Attendant Console in one of the following ways:

- Switched Loop Attendant Console:** The call returns to the common queue, where it remains until the console can receive it at a LOOP button.
- Direct Trunk Attendant Console:** The call returns to the Return-On-Busy (RTN-BUSY) button. If that button is busy, the call remains camped-on at the called terminal until the RTN-BUSY button of the console becomes idle.

Interactions

- Call Coverage/Direct Group Calling (DGC):** If the called party is a member of a hunt or Call Coverage group (or, for V1 systems only, a DGC group) and all members of the group, or all receivers of the Coverage group are busy, the call will not hunt or receive coverage. Once camped-on, calls will no longer hunt or receive coverage even if the hunted-to station or group member becomes idle.
- Direct Group Calling:** For V2, the attendant can camp-on more than one call per DGC group. For VI, the attendant can camp-on only one call per DGC group; if the attendant attempts to camp-on a second call, it is immediately returned on the RTN-BUSY button on the DTAC. Voice terminals in the group do not receive a burst of tone when a call is camped on.
- Direct Inward Dialing (DID):** DID calls can be covered by the attendant and then given Camp-On treatment. They do not automatically receive Call Waiting.

Refer to the Attendant Feature descriptions for information on other related features (Table 2-E).

Administration Requirements

System:

- Number of seconds before a camped-on call returns to the Attendant Console (1-120 seconds) or No Attendant Camp-On allowed (0) - Default = 30 seconds.

Hardware Requirements

None

ATTENDANT CANCEL

Description

Allows the attendant to terminate an attempt to extend any incoming call if the called station does not answer, or if the station answers but declines to accept the call. Before pressing RELEASE, the attendant presses CANCEL and is automatically reconnected to the calling party.

Pressing CANCEL when the Start facility is not active will be ignored.

Considerations

Attendant Cancel allows the attendant to terminate a call transfer attempt and return to the incoming held party via a one-button operation. This enhances the attendant's ability to handle calls quickly and efficiently.

Interactions

None

Administration Requirements

None Required

Hardware Requirements

None

ATTENDANT CONSOLE, DIRECT TRUNK

Description

In System 25, the Attendant Console is used to answer incoming trunk calls that are not directed to specific user stations, to answer calls from inside users, to extend calls to inside stations and outside numbers, and to assist system users in placing outgoing calls and setting up conferences. The attendant can also manage and monitor some areas of system operation. System 25 R1V1 supports only the Direct Trunk Attendant Console (DTAC). The R1V2 supports either the DTAC or the Switched Loop Attendant Console (SLAC), that is described in the next subsection of this manual.

The DTAC (Figure 2-1) is a 34-Button Deluxe Voice Terminal administered with special features, buttons, and capabilities to serve as an attendant position. In addition to the attendant features, all standard multiline terminal features are also available. (Refer to the Hardware Description section of this manual for a complete identification of the external controls, indicators, and components of the basic voice terminal.)

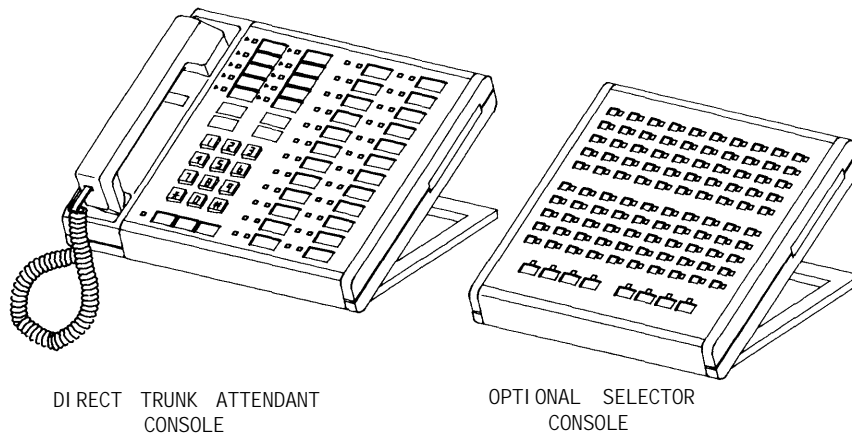


Figure 2-1. Direct Trunk Attendant Console

The DTAC is always equipped with the following feature buttons that provide unique attendant console functions. Each button has a green status LED that indicates when the feature is activated.

- **Start [START]** Initiates the call extending process by placing a caller on hold and providing internal dial tone to the attendant
- **Cancel [CANCEL]:** Terminates the “Start” operation and reconnects the attendant to the calling party.
- **Release [RELEASE]:** Releases the attendant from an active call and completes the call extending process.
- **Return-On-Busy [RTN-BUSY]:** Camped-on calls are returned to the console on this button if not answered within a specified interval.

- **Return-On-Don't-Answer [RTN-DA]:** Extended calls not answered are returned to the console on this button if not answered within a specified interval.
- **Attendant Message Waiting [ATT MSG]:** Used by the attendant to remotely control Message LEDs on voice terminals.
- **Alarm [ALARM]:** The associated status LED flashes when a system trouble has been detected; the LED can be changed from flashing to steadily lit by pressing the button.

Two other attendant-only features are assigned to console feature buttons if required, Position Busy [POS BUSY] and Night Service [NIGHT]. In a dual attendant console system, Position Busy removes an Attendant Console from service. Only one of two consoles can be in the “Position Busy” mode at a time. When Night Service is activated, attendant-seeking calls can ring a night bell, can be directed to assigned voice terminals, or can be sent to a night service announcement.

Considerations

Direct trunk operation means that trunks are terminated on individual buttons, called Personal Line buttons, where outside calls are answered and originated. The console can have several incoming calls ringing simultaneously.

Each console can also have an optional Attendant Direct Extension Selector Console to enhance internal calling. The Selector Console is described in the “Attendant Direct Extension Selection” subsection.

Dual Console Operation:

A System 25 can be equipped with up to two DTACs that operate simultaneously when both are in service. If the system has two attendant consoles, one is called the first attendant console (primary attendant console in VI); the other is called the second attendant console (secondary attendant console in Vi). The calls in the following list will be routed to the first attendant console.

- Dial “0” calls
- DID calls to unassigned numbers (when administered to route to the attendant)
- Calls to Floating Personal Data Codes (FPDCs) not logged in (when administered to route to the attendant)

If the first attendant has activated the Position Busy feature or is busy on both System Access buttons, these calls will be routed to the second console. If that console is also busy on both System Access buttons, busy tone is provided to the calling party.

For V2 only: See the “Call Coverage—Individual” feature description for information about *simulating* additional System Access buttons for handling more incoming calls.

System users and DID callers can reach a particular attendant by dialing that personal Data Codes (PDC).

Position Busy:

A POS BUSY button can be assigned to each console; this permits selection of one of two modes of operation: (1) simultaneous operation or (2) only one Attendant Console active. (Note that only one console is allowed to be inactive at any given time.) An associated POS BUSY status LED is lighted when the console is inactive. Ringing is disabled on all trunk

terminations on the rightmost two columns of buttons of the inactive console. Ringers disabled on an inactive console will be enabled on the active console for those trunks with dual appearances (appearances on both consoles). All other features on all buttons, including those on the associated Attendant Direct Extension Selector Console will continue to function normally even though the console is inactive.

Interactions

A DTAC cannot operate in the same system with a SLAC.

Refer to the Attendant Feature descriptions for information on other related features as listed in Table 2-E.

Administration Requirements

System:

- Assign Primary and Secondary Attendant Positions (Vi)
- Display attendant position number (first or second) (V2)
- Assign number of rings before unanswered calls return to the Attendant Positional-31) -Default = 5 rings
- Send DID calls to unassigned numbers to the Attendant Position (Yes, No) -Default = Yes
- Send calls to Floating Personal Dial Codes that are not logged-in to the Attendant Position (Yes, No) -Default = Yes
- Assign number of seconds before an unanswered Camped-On Call returns to the Attendant Console (1-120 seconds), or No Attendant Camp-On (0) - Default = 30 seconds.

Attendant Console: (Station Port)

- Voice terminal type = 309 (V2)
- Special Programmable Buttons:
 - Night Service
 - Position Busy
 - Attendant Message Waiting (assigned by default).

Note: The following buttons are predefined on the Attendant Console and are not administrable:

- Alarm
- Return-On-Don't-Answer
- Return-On-Busy
- Start
- Cancel

- Release
- Trunk terminations—The following is required for each trunk terminated on the console (administered as Personal Line appearances; DID trunks cannot be terminated on a DTAC):

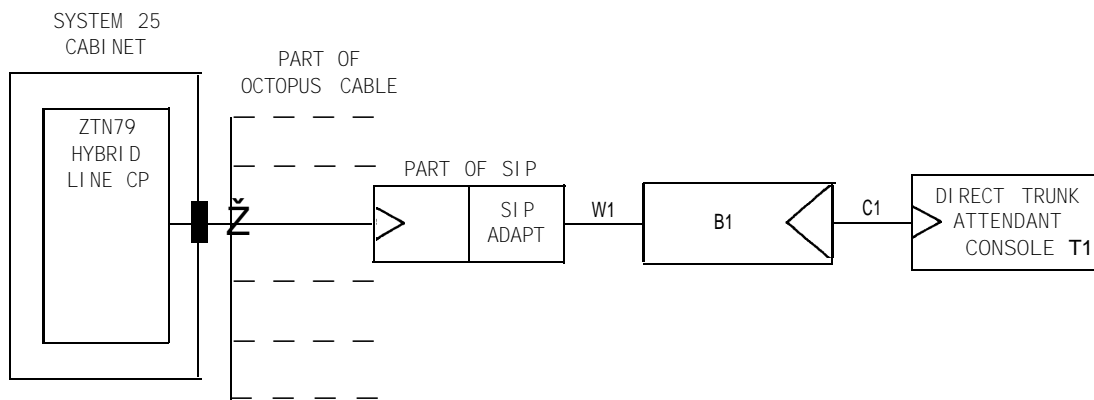
Trunk Number

- Make this the Principal Station (owner) of the trunk (Yes, No)
- Enable Ring (Yes, No).

Hardware Requirements

Each console requires a port on a ZTN79 ATL Line circuit pack.

Figure 2-2 provides a connection diagram for the DTAC.



LEGEND :

- B1** — TYPICAL-103A CONNECTING BLOCK*
- C1** — MODULAR CORD (D8W-87) - FURNISHED WITH SET
- C2** — OCTOPUS CABLE (WP90780) - PEC 2720-05P
- T1** — 7305H02B DELUXE 34-BUTTON VOICE TERMINAL - PEC 3162-417
- W1** — 4 PAIR INSIDE WIRING CABLE*
- * - FURNISHED BY INSTALLER

RANGE: WITHIN 2000 FEET OF SYSTEM CABINET (LOCAL POWER REQUIRED >1000 FEET)

Figure 2-2. Direct Trunk Attendant Console Connections

ATTENDANT CONSOLE, SWITCHED LOOP (V2)

Description

In System 25, the Attendant Console is used to answer incoming trunk calls that are specified to ring at an attendant position, to answer calls from inside users, to extend calls to inside stations and outside numbers, to assist system users in placing outgoing calls, and to set up conferences. The attendant can also manage and monitor some areas of system operation. System 25 R1V1 supports only the Direct Trunk Attendant Console (DTAC), which is described in the preceding subsection of this manual. The R1V2 supports either the DTAC or the Switched Loop Attendant Console (SLAC), but not both in the same system.

The SLAC (Figure 2-3) is a 34-Button Built-In Speakerphone (BIS) Voice Terminal with a 16-character display module. It is administered with special buttons, features, and capabilities to serve as an attendant console. In addition to the attendant features, most standard multiline terminal features are also available. (Refer to the “Hardware Description” section of this manual for a complete identification of all the external controls, indicators, and components of the basic voice terminal.)

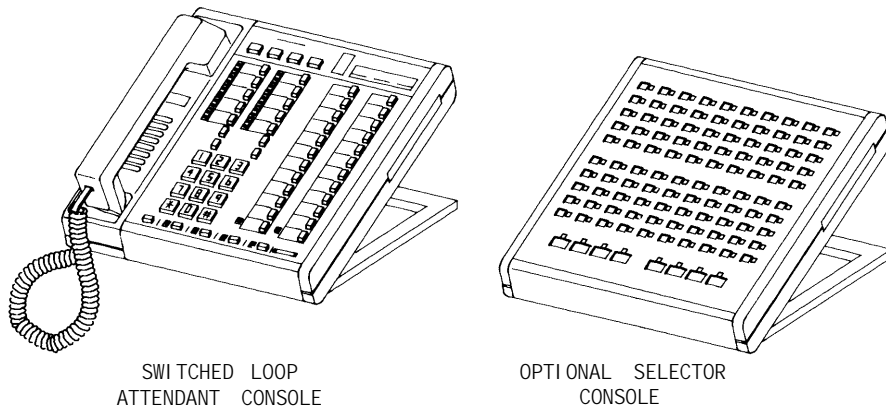


Figure 2-3. Switched Loop Attendant Console

Associated with the SLAC are message center-like capability and display support. The message center feature provides for efficient handling of calls that should be sent to message takers. These calls are directed to a message center console position through administration of call type translations. Display service allows identifiers (names) to be assigned to extension numbers and trunks. The system then displays the appropriate information to the attendant when calls are processed at the console.

Fixed Buttons (Figure 2-4)

The SLAC has five fixed line appearance, or “LOOP,” buttons where all incoming calls are answered. Each button has a red I-use LED and a green status LED. These buttons represent voice links (loops) between the console and the switch. The loops also provide the paths for outgoing calls.

In addition to the LOOP buttons and standard multiline terminal buttons (HOLD, TRANSFER, etc.), the console is equipped with the following feature buttons that provide unique attendant functions. Unless noted, the buttons have green status LEDs.

- **Start [START]:** Initiates the call extending process by placing a caller on hold (on the Source button) and providing internal dial tone to the attendant. No LED.
- **Cancel [CANCEL]:** Terminates the “Start” operation and reconnects the attendant to the calling party (on the Source button). No LED.
- **Release [RELEASE]:** Releases the attendant from an active call and completes the call extending process. No LED.
- **Source [SOURCE]:** Reconnects the attendant to the calling party after a call has been initiated to the called party but before the two parties have been connected together.
- **Destination [DEST]:** Connects the attendant to the called party again after the attendant has operated the Source button to speak to the calling party.
- **Join [JOIN]:** Joins together (in a 3-way connection) the attendant and the other parties in an extended call. No LED.
- **Forced Release [FORCED RELEASE]:** Drops all active parties from a call. No LED.
- **Last Number Dialed [LAST # DIALED]:** Redials the last number dialed. No LED.
- **Position Busy [POS BUSY]:** Temporarily removes the attendant position from service.
- **Attendant Message Waiting [ATTENDANT MESSAGE WAITING]:** Used by the attendant to remotely control Message LEDs on voice terminals.
- **Alarm [ALARM]:** The associated status LED flashes when a system trouble has been detected; the LED can be changed from flashing to steadily lit by pressing the button.
- **Inspect [INSPECT]:** Puts the display into a mode for inspecting the status or stored information of certain buttons.
- **Scroll [SCROLL]:** Causes display to present additional call information, when available. No LED.
- **Local [LOCAL]:** Allows display to be used for clock and calendar functions. No LED.

The buttons not assigned to normal voice terminal functions or to attendant functions are defaulted to the Flex DSS feature. One of these programmable buttons can be assigned to Night Service, if the feature is required, and any of the others to multiline voice terminal features.

Programmable Feature Buttons (Figure 2-4)

The features in the following list can be assigned to the programmable feature buttons. On the SLAC these buttons are not equipped with lamps for indicating feature status conditions.

- Exclusion

- Manual Signaling
- Transfer to Data
- Account Code Entry
- Auto Intercom
- Auto Answer
- Direct Facility Access
- Direct Station Selection (DSS)
- Flex DSS
- Repertory Dialing.

The last two features can be programmed with dialable numbers by the attendant. When a call is placed using a Flex DSS button or a Repertory Dial button, one of the five switched loops is automatically selected for routing the call to the switch.

Display (Figure 2-4)

The SLAC contains an alphanumeric call information display. This module is built into the top of the console. It contains a 16-character 5x7 dot matrix liquid crystal display, timer controls, and a thumbwheel Contrast adjustment. Timer functions are available only when the attendant presses the Local button. The Time/Timer Exit button allows the user to select ordinary clock/calendar display or a timer. In the Time mode, Set, Fwd, and Rev are used to set the clock. In Timer mode, Start and Stop are used to time events.

The primary purpose of the console display is to provide the attendant with descriptive information about incoming and outgoing calls. This information includes extension numbers and associated names, trunk identifiers, reasons for call return and redirection, and number of calls waiting in the queue for service. Refer to the “Attendant Display” feature description, for a detailed discussion of call information displays.

Considerations

An R1V2 system configuration can support either one or two SLACs or one or two DTACs, but not a combination of a SLAC and a DTAC.

The Switched Loop Console derives its name from the ability of the system to hold incoming attendant-bound calls in a queue and switch them on voice loops to an available console. Calls are directed to a console in a preadministered, prioritized sequence. The SLAC differs from the DTAC in the following basic respects:

- It receives calls one at a time, regardless of the number of incoming calls to the system (at the DTAC, many incoming calls can be ringing simultaneously).
- It displays pertinent information about incoming and outgoing calls.
- It can serve as an attendant console, a message center, or a combination of both.
- It has speakerphone and Hands-Free Answer on Intercom (HFAI) capabilities.

The optional Direct Extension Selector Console can be connected to a SLAC to provide busy/idle status of inside stations and quick calling of their extension numbers. In a system with two consoles, either or both can have a Selector Console as an adjunct. The Selector Console is described in the “Attendant Direct Extension Selector Console” subsection.

Switched Loop Operation

All calls that are intended for an attendant position are first routed by the system to a common queue where they wait to be sent to a console. In a configuration having two consoles, the same queue serves both consoles. When an attendant console becomes available to receive a call, the system removes a call from the queue and directs it to an idle loop on the console. Calls are selected from the queue on the basis of “first in/first out” and in accordance with administered priorities. An available attendant console is one that is not active on a call, has no calls ringing, has at least one LOOP button idle, is not in Position Busy or Inspect mode, and is not in a split condition.

In a two-console arrangement, each console can be administered to receive all types of calls or to receive only specific types. A call that can be received by either position goes to the first available attendant; when both are available, the call goes to the attendant who has been idle the longest time. If one of the two consoles is in “Position Busy” mode, all calls (except Attendant PDC, Attendant PDC via DID, and DGC calls) direct to the other one.

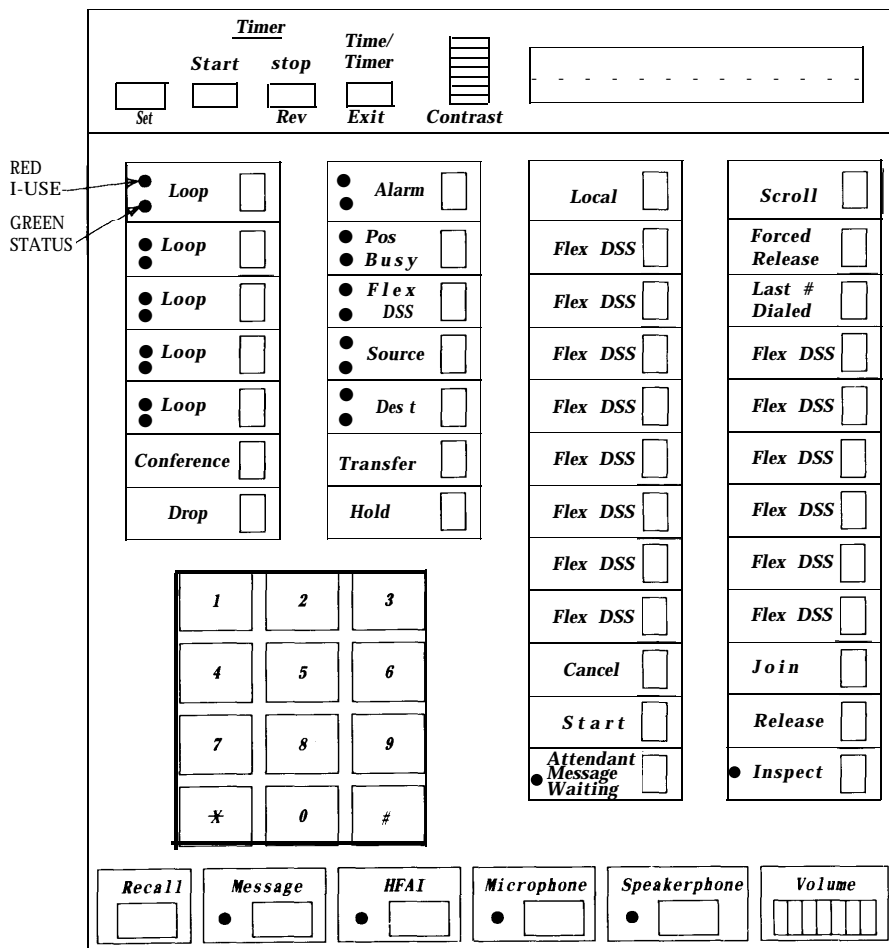


Figure 2-4. Console Buttons and Display

An incoming call from the queue to a console appears on one of the five LOOP buttons; the attendant is alerted to the call by audible ringing, a steadily lighted red lamp, and a flashing green lamp. While the call is ringing and while the attendant is handling the call, the system will direct no more calls to the console. After the attendant ends or releases the call or puts it on hold, another call can come in on an idle button.

It should be emphasized that even when all LOOP buttons on a console are idle, only one call can be directed from the queue to the console. If the attendant puts a call on hold, that LOOP button is no longer available, but a new call can come in on another button that is idle.

Answering a Call on a LOOP Button

At an available SLAC, an incoming call will ring at an idle LOOP button automatically selected by the system. The attendant has only to lift the handset to answer the call; pressing the button is not necessary.

Placing a Call on a LOOP Button

In general, originating a call at an idle SLAC involves going off-hook and then dialing the desired number. However, if Prime Line Preference has been changed to a non-LOOP button, the attendant will have to press an idle LOOP button before dialing.

If the console is not idle, the attendant can generally use one of these procedures:

- Split the active call (that is, put it on temporary hold by pressing the Start button) and place another call on the same button; this is the normal call-extending procedure.
- If the Automatic Hold feature is enabled, press another LOOP button to place a new call; the first call goes on hold.
- If the console does not have Automatic Hold (that is, it has the default Automatic Release), use the Hold button to put the active call on hold; then select a new loop to place a new call.

Dual Console Operation

A System 25 can be equipped with up to two SLACs, which operate simultaneously when both are in service. Both consoles can receive the same types of calls, or each can be administered to receive only certain types. When one console is out of service (see Position Busy below), most calls are directed to the other. Either or both consoles can function as a message center.

Inside users can reach either attendant by dialing O, or a particular one by dialing the attendant's PDC. The DID callers use the Attendant DID number.

Position Busy

Operation of the Position Busy button by the attendant makes the console unavailable to most incoming calls from the common queue and directs the calls to another answering station. The only types of calls that are not diverted by Position Busy are Attendant DID, DGC, and PDC calls. The placing of outgoing calls is not affected. When the Position Busy condition is active, the green status lamp of the button lights steadily. Position Busy is similar to the Send All Calls feature, that is not administered on the SLAC.

The Position Busy feature is automatically assigned to a button position (see Figure 2-4) when the console is administered. In a one-console configuration, however, the feature is enabled only if a multiline voice terminal in the system is administered as a receiver of calls

from the common queue while the console is unavailable. If this is not done, the button should be reassigned to another of the permissible features.

In a two-console configuration, an attendant in Position Busy mode will be covered by the other attendant. Only one console can be in Position Busy mode at a time.

Call Types

The following types of attendant-seeking calls are sent to the common queue and then directed to an idle LOOP button at a console:

- Dial Operator calls (placed from inside stations by dialing O)
- Call Following calls logged in at the console
- Calls to Floating Personal Dial Codes (FPDCs) that are not logged in at a specific station
- Direct Inward Dialing (DID) calls to numbers that are not assigned to specific stations
- Attendant DID calls, including Direct Group calls
- Calls to the attendant's Personal Dial Code (PDC), including DGC calls
- Coverage calls for which the common queue is a covering receiver
- Returning calls.

The order in which calls are serviced is established by system administration. Each type of call is assigned a priority that determines its position in the common queue with respect to other types. System administration also establishes where the calls go. Obviously, in a one-attendant system, all calls automatically go to that attendant. If a system has two attendants, however, administration can direct calls of each type (with the exceptions noted below) to either position or to both positions.

Call Following and Attendant PDC calls can be assigned priorities but cannot be directed to a specific attendant in a two-position system. Any trunk except types 901-902 and 1003-1008 (DID and Dial-in Tie Trunks) can be assigned a priority and be directed to a specific attendant position or to both.

Calls accessed by dialing a code [call pickup at other extensions, Trunk-Answer-from-Any-Station (TAAS) Night Service calls, and calls parked by other stations] are originated at a LOOP button and brought to the console on that same button. These calls do not enter the common queue.

Ringling

The SLAC receives normal ringling on incoming calls. Abbreviated alerting (one short burst of ringling), accompanied by a change in the LOOP button wink rate, indicates to the attendant that a held call has exceeded the preset hold time interval. Abbreviated alerting can also be administered as a reminder for new calls entering the queue.

Interactions

A SLAC cannot operate in the same system with a DTAC.

Connection of a headset adapter to the SLAC allows the optional use of a headset instead of the handset in handling calls.

Call Originations

Placing a new call from an active console causes interactions with the currently active call. At a console that has the default Automatic Release feature, the active call is dropped when the attendant presses anew LOOP button. The new loop becomes the active one; dial tone is provided, and the attendant can dial a number. If the optional Automatic Hold feature is administered, the interrupted call is put on hold instead of being lost.

If anew call is originated with the START button or at the Selector Console, the active call is split. The current loop becomes the active loop for the new call. The display shows the split call information. This is the normal operating procedure for extending calls.

When a Rep Dial button is pressed while the console is active on a call, the active party is not dropped or split, and the display does not change. If the active call is with an inside station, pressing Rep Dial has no effect. But if the active call is on an outside trunk, then pressing Rep Dial will cause the digits stored on the button to be sent out over the trunk (“thru-dialing” or “end-to-end signaling”).

When an active call is put on hold manually by operation of the HOLD button, the system does not automatically select a new loop for placing a call. In this case, the attendant can select a new loop by pressing an idle LOOP button, then dial a number. Pressing a DSS, Flex DSS, Auto Intercom, Last Number Dialed, Rep Dial, or Selector Console button will select a new loop and dial a number in a single operation.

Attendant Features

Refer to the Attendant Feature descriptions for information on other related features (Table 2-E).

Administration Requirements

Station Port:

- Assign voice terminal type.
- Assign Prime Line Preference (default: top LOOP button).
- Assign flexible buttons.
- Enable ring reminder when calls enter queue.

Trunk Port:

- Assign priorities to calls directed to the console queue.
- Assign the attendant(s) to handle calls from this trunk.
- Assign unique trunk identifiers.

System:

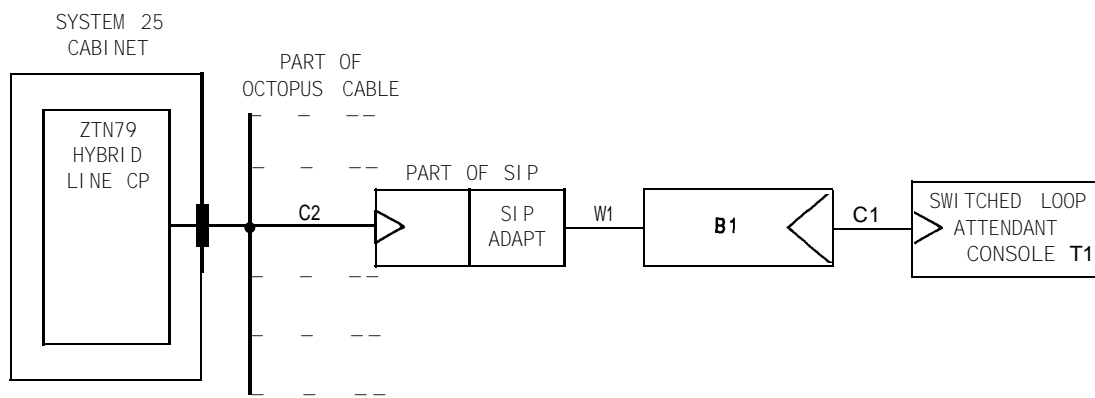
- Assign DID number for attendant “0” treatment.
- Assign Call Coverage Group number for which the console queue is to serve as a receiver.
- Assign Automatic Hold or Automatic Release.
- Assign Hold timer interval.
- Assign destination of held calls that time out.

- Assign call type priorities and attendant specification.
- Assign call types and attendant specification for Message Center-Like operation in a two-console configuration.
- Assign Position Busy “backup” station, inapplicable.

Hardware Requirements

Each console requires a port on a ZTN79 ATL Station circuit pack.

Figure 2-5 provides a connection diagram for the SLAC.



LEGEND :

- B1** - TYPICAL-103A CONNECTING BLOCK*
- C1** - MODULAR CORD (D8W-87) - FURNISHED WITH SET
- C2** - OCTOPUS CABLE (WP90780) - PEC 2720-05P
- T1** - 7305H04C BUILT-IN SPEAKERPHONE VOICE TERMINAL - PEC 3162-DIS
- W1** - 4 PAIR INSIDE WIRING CABLE*
- * - FURNISHED BY INSTALLER

RANGE : WITHIN 2000 FEET OF SYSTEM CABINET (LOCAL POWER REQUIRED >1000 FEET)

Figure 2-5. Switched Loop Attendant Console Connections

ATTENDANT DIRECT EXTENSION SELECTION

Description

Permits the attendant to extend calls to stations by pressing one or two buttons instead of pressing START and dialing the PDC or DDC. Each attendant console in a system can have an associated Direct Extension Selector Console. A Selector Console can be used with either a Direct Trunk Attendant Console (DTAC) in an R1V1 or R1V2 system or with a Switched Loop Attendant Console (SLAC) in an R1V2 system.

The Selector Console is also used by the attendant for simply calling inside stations, in addition to “extending” calls.

The Selector Console (Figure 2-6) has an array of 100 Direct Extension Selection (DXS) buttons plus seven Group Select buttons and a Test button. The DXS buttons are labeled 00 through 99. Pressing a Group Select button causes the DXS buttons to be associated with PDCs from an associated hundreds group. Default assignments for the Group Select buttons are 200-299, 300-399, etc., up to 800-899. Group Select buttons can be assigned any hundreds group in the dialing plan.

Pressing a DXS button when off-hook on an incoming call is equivalent to pressing START and dialing a station. Such action will busy out the Start facility until the call is released. The Selector Console can be used to monitor the on-hook/off-hook status of stations in the system. If the attendant, while on-hook, presses a Group Select button, the Group Select LED and the LEDs of any busy stations in that group will light steadily.

The DXS button LED (and, in R1V2 systems, the Group Select button LED) associated with a particular station will flash when one of the following events occurs:

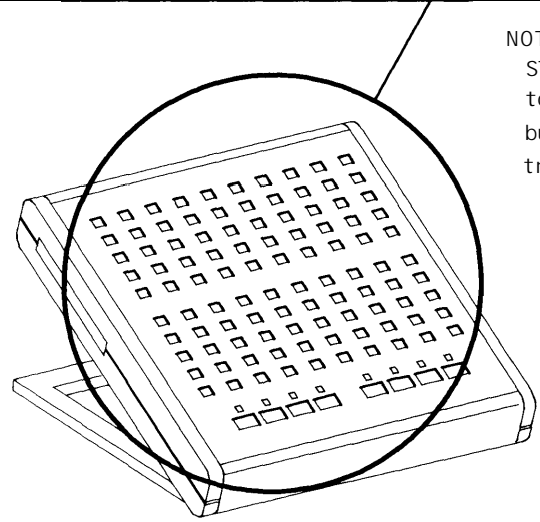
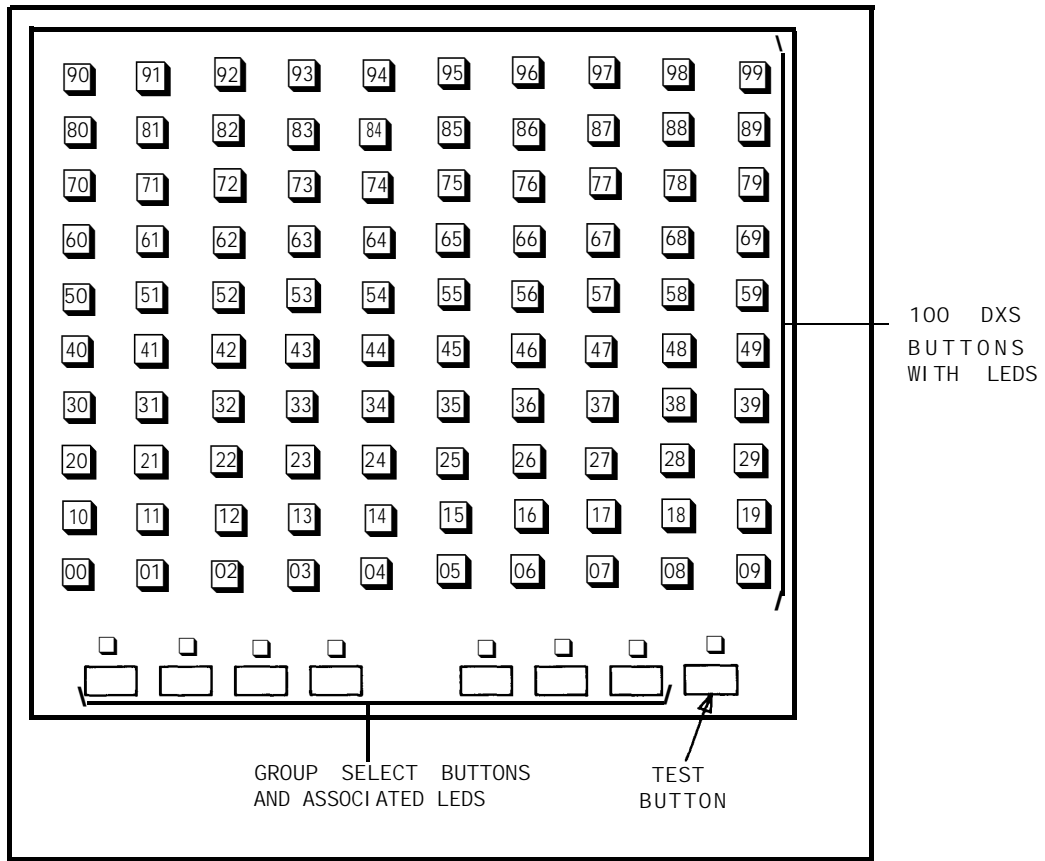
- The station calls the attendant
- A call extended by the attendant to the station returns on a RTN-BSY or RTN-DA button (DTAC only) or on a LOOP button (SLAC only)
- The station is covered and a call to it is redirected to a COVER button (DTAC only) or to a LOOP button (SLAC only).

The LEDs stop flashing when the call is answered. When the attendant answers a returning call, the LEDs will return to the state that reflects the current on-hook/off-hook status of the station. In all of the above cases, the Group Select lamp associated with the current “hundreds page” remains lighted steadily.

An outside call can be parked via the Selector Console by pressing one of the eight DXS buttons that can be programmed with Call Park extension numbers. On the DTAC, the facility status LED of the parked call winks (to indicate that the call is on hold) and the status LED on the Selector Console lights steadily. On the SLAC, the call is removed from the attendant console, with the Selector Console LED lit steadily.

A call parked via the Selector Console and not picked up within 2 minutes will return to the console. The status LED of the parked-on button will flash while the call is ringing the attendant.

A call parked via the Selector Console can be picked up at any voice terminal by dialing the Call Park retrieval code (*8) and the number on which the call is parked.



NOTE:
STATUS LEDs are located
to the left of each DXS
button (00-99) under
transparent front cover.

Figure 2-6. Model 23A1 Attendant Direct Extension Selector Console

The rightmost button on the bottom of the console is a Test button. When it is pressed, all DXS LEDs will light sequentially; a second press allows individual LEDs to be tested and a third press ends the test.

Considerations

When there are two Attendant Selector Consoles in the system the Group Select button assignments are identical. Whenever an administrative change is made to one console, the other console is automatically changed.

Buttons on the Selector Console point to either station PDCs, FPDCs (FPDCs), Call Park codes, DGC access codes, or (V2) pooled facilities. Calls extended by the Selector console are directed as described in the “Personal Dial Codes” feature description.

When a station calls the attendant, the associated LED on the Selector Console will flash while the call is ringing and will light steadily when the attendant answers the call. The LED will light steadily whenever the terminal is off-hook. Station busy indication is not provided for buttons pointing to FPDCs.

If a call to a PDC is directed to a COVER or LOOP button on the Attendant Console, the covered status LED of the voice terminal on the Selector Console will flash and then go dark when the call is answered by the attendant. If the covered call was intended for a FPDC that was logged in at a terminal with attendant coverage, the Selector Console status LED associated with the FPDC (if assigned) will flash. In this case, the Cover button status LED will also flash (DTAC only).

A call can arrive at an Attendant Console SYSTEM ACCESS or LOOP button because the PDC or FPDC is logged in at the Console or because the FPDC is not logged in. For these calls, the status LED on the Selector Console will not light.

If the attendant extends a call to a station or DGC group and that call returns to the attendant, the status LED of the called station or group on the Selector Console will flash and then go dark when the call is answered by the attendant. This is true regardless of the login status of the PDC.

Interactions

- **Attendant Position Busy:** The Selector Console functions normally when the associated Attendant Console is in the inactive mode.
- **Attendant Return-Coverage-On-Busy/Or-Don't-Answer:** If a call to a FPDC is returned to the attendant on a RTN-BUSY or RTN-DA button or on a LOOP button, the status LED of the FPDC on the Selector Console will flash during ringing and go dark when answered.
- **Call Coverage:** If the attendant receives a coverage call for a FPDC, the associated status LED on the Selector Console will flash and then go dark when the call is answered by the attendant.
- **Direct Group Calling:** When all stations in a DGC group are busy, the DXS status LED on the Selector Console lights.

Administration Requirements

Special Feature Ports:

- Requires a port assignment on a ZTN79 ATL Station Circuit Pack (CP) for each Selector Console.
- Assign Group Select button hundreds groups.

System:

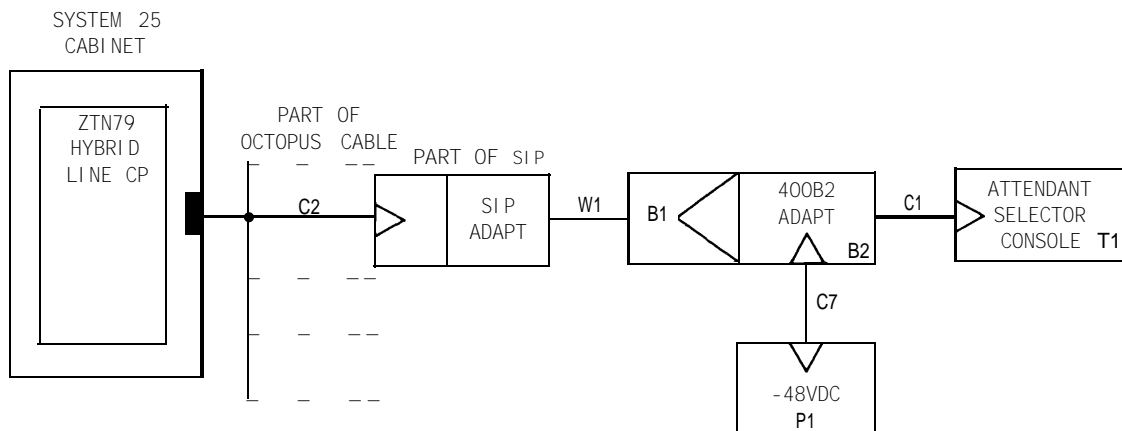
- Assign Attendant Call Park codes.

Hardware Requirements

Requires an Attendant Selector Console, and a port interface on a ZTN79 ATL Station CP.

The Selector Console requires a KS-22911, List 1 Power Supply, associated 115V ac power outlet, and a 400 B-type Adapter. The 400B2 Adapter provides power to the console at the wall jack. The Console connects to a port on the ZTN79 ATL Station CP.

Detailed connection information is provided in Figure 2-7. Descriptions of the Station Interconnect Panel(SIP), Trunk Access Equipment (TAE), and associated cables and adapters, as shown on the figures, are provided under the heading “Connectivity” in Section 4.



LEGEND :

- B1 - TYPICAL-103A CONNECTING BLOCK*
- B2 - 40062 ADAPTER - FURNISHED WITH CONSOLE
- C1 - MODULAR CORD (D8W-87) - FURNISHED WITH CONSOLE
- C2 - OCTOPUS CABLE CABLE (WP90780) - PEC 2720-05P
- C7 - CORD D6AP-87 - FURNISHED WITH CONSOLE
- P1 - KS22911 POWER SUPPLY, - FURNISHED WITH CONSOLE
- T1 - 23A1 DXS CONSOLE - PEC 62509
- W1 - 4 PAIR INSIDE WIRING CABLE*
- * - FURNISHED BY INSTALLER

RANGE : WITHIN 2000 FEET OF SYSTEM CABINET
(LOCAL POWER REQUIRED >1000 FEET)

Figure 2-7. Attendant Direct Extension Selector Console Connections

ATTENDANT DISPLAY (V2; SLAC Only)

Description

Provides displays of call-related information on a 16-character screen at the top of the Switched Loop Attendant Console (SLAC). The following types of data are automatically presented for calls handled at the console:

- The extension number and name of an inside party called from the console
- The extension number and name of an inside party calling the console
- Called extension number and trunk identification on incoming trunk calls
- Digits dialed on outgoing trunk calls
- Calling and called party identification on internal coverage calls
- Call type and reason for return or redirection
- Number of calls waiting in the common queue to be serviced.

A secondary, or *Inspection*, mode allows the attendant to make the console temporarily unavailable to incoming calls and get information about held calls, trunk availability, numbers assigned to Repertory Dialing, Automatic Intercom, DSS, or Flex DSS buttons, and the number currently stored on the Last Number Dialed button. The display module provides clock/timer functions in the Local mode.

Call Information Display

The console display can generate up to two screens of information. Screen 1 is the normal display presented for all calls. No action by the attendant is required to access this screen. Screen 2, activated by pressing the SCROLL button, contains support information about a call, such as the calling party on a coverage call. If Screen 2 does not appear after SCROLL is pressed, then no additional information is available for viewing.

- Extension number and calling/called party name
- Redirection descriptors:
 - > for a coverage call
 - } for a returning call
 - & for a call that is being extended.
- Call type indicators that track call status or inform the covering attendant why the user did not answer:
 - s for Send All Calls
 - b for call to busy station
 - n for no answer
 - d for DGC call
 - f for Call Following

p for Call Park

o for not logged-in FPDC.

- Number of calls waiting in the common queue (1-9 or ! for 10 or more)
- Trunk identification for incoming outside calls
- Conference information.

Considerations

The attendant display provides pertinent information about the currently active call. The following examples show some typical screens.

Incoming DID call:

2085 DID	3
----------	---

The first four digits are the DID number assigned to the attendant. The digit 3 is the number of calls waiting in the queue.

Incoming call from a system user:

1463 Pearson, M

The first four digits are the calling user's extension number. No calls are in the queue.

Incoming coverage call:

>1566 Davis, T 5

A call to T. Davis on extension 1566 is directed to the common queue for coverage. The attendant can determine the caller's identity by pressing SCROLL. The following display appears on screen 2.

2381 Harmon, B s

The call type indicator s means that Harmon's call was sent to coverage because Davis activated the Send All Calls feature.

Returning parked call:

}801 Park	4
-----------	---

The call parked on extension 801 at the Selector Console is returning to the attendant.

Call from console to system user:

1728 Burns, R !

The extension number and name of the called party are presented. The ! symbol indicates that 10 or more calls are waiting in the queue.

Outgoing trunk call:

912126378888

The display shows the digits (up to 14) of the dialed outside number. Digits in excess of 14 are displayed on screen 2.

Call from inside station extended to another inside extension:

2344 Carter, M 7

The attendant answers the incoming call, and the caller, Carter, asks to be connected to another inside party. When the attendant presses START, screens 1 and 2 provide the following information:

&	7
---	---

2344 Carter, M &

The call from Carter has been split while the call is extended. After the attendant dials the desired PDC, screen 1 displays called party identification, and screen 2 remains unchanged:

&1397 Phelps, T 7

2344 Carter, M &

If the attendant now presses RELEASE, both screens go blank. If the attendant presses JOIN, the two parties and the console are connected together; screen 1 displays conference information:

CONFERENCE 3 7

Conference set up by attendant:

After the attendant has called all the conferees and connected them together on a LOOP button, screen 1 displays the type of call and the number of parties, including the console:

CONFERENCE 4 7

If the attendant releases, the screen goes blank.

Inspection Mode

Pressing the INSPECT button puts the console in the inspection mode. In this condition the console is unavailable to incoming calls and the attendant can get information about held calls, trunk availability, and numbers assigned to certain buttons. Pressing the INSPECT button a second time will restore the console to normal operation, as will a change of switchhook state (on-hook to off-hook, or vice versa), selection of an idle loop, or 15 seconds of inactivity (timeout).

While the inspection mode is active, the attendant can press any of the following buttons to provide special information on display screen 1.

- Loop button where a call is ringing or on hold—The original message that appeared when the call first arrived at the console is displayed; for a multiparty call, “CONFERENCE N” N=number of conferees) is shown.
- Direct Facility Access button—The display shows the number of trunks that are busy within the selected trunk group; the format of the message is “XX of YY BUSY. ”
- Repertory Dial, Automatic Intercom, DSS, Flex DSS, or Last Number Dialed—The number stored on the button is displayed.

Pressing any button other than those listed above causes a message such as “NO INFORMATION” to be displayed.

Timer Functions

The LOCAL button allows the attendant to have access to the Timer controls on the left side of the display module. In the local mode, the screen can provide clock, calendar, timer, and alarm functions. The local mode is canceled when the LOCAL button is pressed again, a new call arrives, or a call is placed. Refer to the customer instruction booklet packed with each console for detailed information on using the Timer controls.

Interactions

The display provides a display for each call placed or answered at the console.

Administration Requirements

Directory type information must be administered in the form of user and group names corresponding to extension numbers and identifiers for incoming trunks. Although system administration allows up to eleven characters for display IDs, only the first nine characters are shown on the SLAC display.

Hardware Requirements

None

ATTENDANT FORCED RELEASE (V2; SLAC Only)

Description

Drops all active parties from a call in which the attendant and one or more other parties are connected together. The attendant uses the feature by pressing the FORCED RELEASE button while connected to other callers in a conference-type call. The other parties will be disconnected from the console and from each other. After Forced Release has taken place, the attendant can receive anew call from the console queue or place a call.

Considerations

Forced Release differs from Release in an important respect. Simple Release separates the attendant from an extended call or a conference call, but leaves the other parties connected together; Forced Release completely disconnects all parties.

When the attendant is connected to only one other party, Forced Release has the same result as Release (that is, the call is ended). If the attendant has already Released, Forced Release has no additional effect.

Interactions

Refer to the Attendant Feature descriptions for information on related features (Table 2-E).

Administration Requirements

None

Hardware Requirements

None; the FORCED RELEASE button is a standard fixed feature button on the SLAC.

ATTENDANT JOIN (V2; SLAC Only)

Description

Allows the attendant, while extending an incoming call, to connect together the calling party, the called party, and the console in a 3-way call. All parties can talk to each other. The attendant activates the feature by pressing the JOIN button while still connected to one of the other parties.

Considerations

The join feature cancels a split condition.

The attendant can use the Join feature to stay on unextended call and give assistance to the other parties. A joined call can be expanded into a conference call by adding more parties.

Interactions

Once the JOIN button has been pressed, there is no way to “unjoin” the calling and called parties (that is, separate them back into SOURCE and DEST).

Refer to the Attendant Feature descriptions for information on related features (Table 2-E).

Administration Requirements

None

Hardware Requirements

None; the JOIN button is a standard fixed feature button on the SLAC.

ATTENDANT MESSAGE WAITING

Description

Allows the attendant to remotely control the status of Message LEDs on user stations.

Considerations

This feature allows the attendant to notify stations that a message is available for them. The attendant can activate the Message LED of the station while either (1) ringing, (2) receiving Busy Tone, or (3) talking to a station. The status of the Message LED of the called party is reflected by the green status LED of the Attendant Message Waiting button in any of these cases.

Note: The Attendant Message Waiting button on the Direct Trunk Attendant Console (DTAC) is labeled ATT MSG. On the Switched Loop Attendant Console (SLAC), the name is completely spelled out.

To activate (light) a user's Message LED in any of these cases, the attendant presses the Attendant Message Waiting button. (If the signaled voice terminal is not equipped with a Message LED, the attendant's LED will remain dark.)

If the attendant presses the button a second (or third) time before hanging up, the user's Message LED will turn Off (and back On), etc.

The red I-Use LED associated with the Attendant Message Waiting button on the DTAC is inoperative.

The attendant can turn On or turn Off a user's Message LED without disturbing the user by going off-hook on a System Access or Loop button, pressing the Attendant Message Waiting button, and then dialing the station. Confirmation tone is returned, and the user's Message LED will turn on. To turn it off, press the button again.

This feature is not the same as the Station-To-Station Message Waiting or the Call Coverage Message Waiting features. Refer to the "Messaging Services" feature description for a summary of all system Messaging Services.

Interactions

- **Conference:** Pressing ATT MSG while on a conference call will be ignored.
- **Coverage Calls:** The attendant can light the Message LED of the covered station when receiving a coverage call for the station.
- **Hands-Free Answer On Intercom:** If the attendant lights the Message LED on a voice terminal with AUTO ANS button active, the auto-answer function will turn off, allowing subsequent calls to receive coverage as assigned.

Administration Requirements

Attendant Position (Station Port):

- Assign ATT MSG button (defaulted).

Hardware Requirements

Stations must have a Message indicator (not assignable).

ATTENDANT POSITION BUSY

Description

Allows an Attendant Console to be placed in an inactive mode.

Systems with Direct Trunk Attendant Console(s) (DTAC)

There must be two Attendant Consoles in the system before this feature can be activated. A Position Busy (POS BUSY) button can be assigned on each of the consoles. Pressing POS BUSY at one of two active consoles causes the POS BUSY status LED to light and the console to be placed in the inactive mode. Pressing POS BUSY a second time causes the LED to go dark and the console to be reactivated. Pressing POS BUSY when only one Attendant Console is active is ignored (i.e., only one console is allowed to be inactive at a time.)

When a console is in the inactive mode, ringing is disabled on facility appearances on the two rightmost button columns only. The (green) status LEDs will continue to operate normally. Calls to floating PDCs not logged in, DID calls, and dial "O" calls will be transferred to the active console. Internal calls to the PDC of the inactive console will still be directed to that console.

Incoming calls on lines that normally ring at only the inactive console will now ring at the active console if they have an appearance there.

All buttons on the inactive console will continue to function normally, including the Selector Console buttons. Calls can be originated by the inactive console. Call appearances in the leftmost two columns of buttons on the inactive console are not affected by the Position Busy feature.

The attendant can press a Direct Station Selection (DSS), Automatic Intercom (AUTO ICOM), or a Direct Facility Access (FACILITY) button and then receive busy-to-idle reminder when the facility becomes idle.

All dial "O" calls, calls to FPDCs not logged in, calls to unassigned DID numbers, and calls to facilities in the rightmost two columns of buttons of the console that appear at both consoles will be directed to the active console.

Note that if a personal trunk appears on only one DTAC, incoming calls on those trunks will not receive service when the console is inactive. For this reason, it is strongly recommended that each DTAC attendant be assigned a Call Coverage-Individual (COVER-IND) button for the other console so that these calls can be covered. Also, be sure to make the Attendant Console the principal station (owner) on all trunks that are to receive coverage by the other attendant.

Systems with Switched Loop Attendant Console(s) (SLAC)

In order for the Position Busy feature to be operational, the system must have either two attendant positions or one position plus a multiline voice terminal administered as a "backup." If the system has two consoles, pressing the Position Busy button on one will make it inactive and cause most calls in the common console queue to be directed to the active console. Each attendant covers for the other. Only one console can be in Position Busy condition at a time. If the system has one console with an administered *backup voice* terminal, pressing the POS BUSY button will make the SLAC inactive, and most calls from the common queue will be directed to the *backup* terminal.

A console in Position Busy mode can receive attendant PDC, DID, and DGC calls, and outgoing calls can still be placed. Local functions can be activated.

Considerations

Position Busy allows one of two attendant positions to be made inactive when not required. This is useful in situations where calling traffic requires only one console operator.

Interactions

- **Attendant Call Extending:** Unanswered calls extended by an inactive console will return to the active console on the Return-On-Don't-Answer (RTN-DA) button (DTAC only) or on a Loop button (SLAC only).
- **Attendant Camp-On:** Calls Camped-On by an inactive console will return to the active console when Camp-On timeout occurs.
- **Attendant Message Waiting:** An inactive attendant is permitted to control voice terminal Message LEDs.
- **Automatic Intercom:** The inactive attendant is permitted to place Automatic Intercom calls. DTAC only: Automatic Intercom calls to the inactive attendant will not ring at the console or be transferred to the active attendant when the AUTO ICOM button is located in one of the two rightmost button columns
- **Call Coverage:**

DTAC only: If the active attendant is a coverage receiver for the inactive attendant, coverage is invoked and calls will appear at the active attendant's Cover button. If the inactive attendant is a coverage receiver for the active attendant, coverage, when activated, is invoked at all coverage stations including the inactive attendant. However, if the Cover button is located in one of the two rightmost button columns, coverage calls will not ring at these buttons.

SLAC only: All Calls covered by the common queue will be directed to the active console.
- **Call Park:** A call parked on the inactive attendant console will return to the inactive console if the call times out; calls parked via the Selector Console will return to the active console.
- **Direct Group Calling:** If the inactive attendant is a member of a DGC Group, calls directed to the group will be routed to the inactive attendant. The attendant must dial *4 (activate DGC Group "Make Busy") to busy out from the group. Dialing *6 deactivates the "Make Busy" function.
- **Direct Inward Dialing:** All DID calls to unassigned DID numbers will be transferred to the active attendant.
- **Night Service:** An inactive attendant that is a Directed Night Service receiver will receive Night Service calls.
- **Personal Dial Codes:** If a PDC/FPDC is logged in at an inactive attendant console, then calls to this PDC/FPDC will go to the active attendant (SLAC) or to the inactive attendant (DTAC only). All calls to floating PDCS not logged in will be transferred to the active attendant.

- **Personal Lines (Trunk Appearances) on DTACs :** All calls to trunks having an appearance in either of the two leftmost button columns will ring normally at the inactive console. All calls to trunks having appearances in either of the two rightmost button columns will not ring. If these trunks also have an appearance at the active console, they will ring there even if they do not normally.
- **Program:** The Program feature remains active at the inactive console.
- **Programmable Buttons:** All DSS, Flex DSS, and REP DIAL buttons remain active on the inactive console.

Administration Requirements

Voice Terminal: (Station Port)

- Assign Position Busy button on DTAC. A Position Busy button is defaulted to the SLAC; it can be assigned to another feature if desired.
- Assign COVER-IND buttons between consoles (DTAC only).

Hardware Requirements

None

ATTENDANT RELEASE

Description

Releases the attendant from an extended call. There are two forms of Attendant Release: Manual and Automatic. This feature applies to the Direct Trunk Attendant Console (DTAC) and the Switched Loop Attendant Console (SLAC).

Manual Release:

Pressing RELEASE releases the attendant from an extended call and completes the associated call transfer. The status LED of the original calling facility will change from hold to busy for direct trunk terminations and from hold to idle for other call facilities (e.g., Loop, Return-On-Busy, Return-On-Don't-Answer, Cover, Automatic Intercom, DSS, and System Access).

Calls cannot be released to Reorder or Dial Tone.

Pressing CANCEL terminates the destination call and reconnects the attendant to the calling party. If the attendant goes on-hook without first releasing a call, the call extending operation will be terminated (the calling party remains on hold). In this case, the attendant can go off-hook and press the held call appearance button to reconnect to the calling party.

Automatic Release:

This feature simplifies the attendant procedures by eliminating the need for the attendant to press RELEASE when releasing from one call to handle another. Selection of any new line facility while active on the Start button will automatically release the first call. At release, the status LED of the first calling facility will change from hold to busy for direct trunk terminations and from hold to idle for other call facilities (e.g., Loop, Return-On-Busy, Return-On-Don't-Answer, Cover, Automatic Intercom, DSS, and System Access).

The Automatic Hold feature can be administered for the SLAC as an alternative to Automatic Release. If the attendant, active on a loop call, presses another loop button to place a call or pick up a held call, the active call is put on hold—not released.

Considerations

Attendant Manual Release improves attendant efficiency in handling calls by allowing the attendant to release an extended call without having to wait for the called station to answer.

Attendant Automatic Release enhances the attendant's ability to handle many calls by eliminating the Release operation when answering a second call.

The Release function is inhibited whenever the Start facility is connected to Reorder or Dial Tone. Pressing CANCEL will terminate the destination call and reconnect the attendant to the calling party.

Administering the Automatic Hold option instead of Automatic Release reduces the occurrence of accidentally dropped calls.

Interactions

- **Attendant Camp-On:** External calls that are released when Busy Tone is heard will be camped on.

Administration Requirements

None.

Hardware Requirements

None.

ATTENDANT RETURN COVERAGE ON BUSY

Description

Allows a camped-on call at a busy station or DGC Group to be returned to the attendant for service after a specified time period.

A camped-on call not answered within 1 to 120 seconds (administrable) after the attendant releases the call will return to the console in one of the following ways:

- On the Return-On-Busy (RTN-BUSY) button at a Direct Trunk Attendant Console (DTAC).
- On a LOOP button at a Switched Loop Attendant Console (SLAC; V2).

To answer a returned call at a DTAC, the attendant presses RTN-BUSY (if not selected by Ringing Line Preference.) A returned call can be reextended via the START button or a Selector Console button. In either case, the Return-On-Busy button is idled as soon as the attendant releases.

To answer a returned call at a SLAC, the attendant merely lifts the handset to be connected to the ringing loop.

When the RTN-BUSY button is busy at a DTAC, the calling party will remain on-hold; if a loop is not available at a SLAC, the returning call remains in the console queue. The system will continue to attempt to ring the called station until the RTN-BUSY button is idle or a loop is open. When Attendant Camp-On is not provided (Camp-On return time set to zero seconds), calls released by the attendant to busy tone are returned to the console immediately.

Considerations

Attendant Return-Coverage-On-Busy allows the attendant to service calls not answered within specified time intervals. This provides the calling party better service and results in fewer lost calls.

Interactions

- **Attendant Camp-On:** External calls that are released when Busy Tone is heard will be camped on.
- **Attendant Console, Direct Trunk:** As long as an Attendant Console remains active, the call will return to the attendant who transferred it.
- **Attendant Console, Switched Loop (V2):** A returning call is directed from the console queue to a LOOP button on any available console that is administered to receive it.
- **Attendant Direct Extension Selection:** If a call to a Floating PDC (FPDC) is returned to the attendant, the FPDCs status LED on the Selector Console will flash during ringing and go dark when the call is answered.
- **Direct Group Calling:** External calls that are camped onto a DGC group that does not have a delay announcement will return to the attendant console after the specified number of rings.

- **Send All Calls (DTAC only):** If Send All Calls is activated, returning calls will ring at the DTAC.

Administration Requirements

System:

- Assign number of seconds before unanswered camped-on calls return to the Attendant Position (1-120 seconds, or O for No Camp-On) - Default = 30 seconds.

Hardware Requirements

None

ATTENDANT RETURN COVERAGE ON DON'T ANSWER

Description

Allows unanswered calls extended by the attendant to be returned to the attendant for additional service.

Calls that are not answered after a specified number of rings will transfer ringing to the Return-On-Don't-Answer (RTN-DA) button on a Direct Trunk Attendant Console (DTAC) or to a LOOP button on a Switched LOOP Attendant Console (SLAC). If the called voice terminal has call coverage, the timing for return begins only after the coverage station begins ringing.

When the RTN-DA button is busy, calls will continue to ring at the called station until the button is idle. If a LOOP button is not available, the returning call stays in the common queue until it can be serviced.

To answer a returned call at a DTAC, the attendant presses RTN-DA (if not selected by Ringing Line Preference.) The call can be reextended via the START button or Selector Console. In either case the button is RTN-DA button is idled as soon as the attendant releases.

To answer a returned call at a SLAC, the attendant merely lifts the handset to select the line.

Considerations

Attendant Return-Coverage-On-Don't-Answer allows the attendant to service calls not answered within a specified number of rings. This provides the calling party better service and results in fewer lost calls.

Interactions

- **Attendant Console, Direct Trunk:** As long as an Attendant Console remains active, the call will return to the attendant who extended it.
- **Attendant Console, Switched Loop:** A returning call is directed from the common queue to a LOOP button on any available console that is administered to receive it.
- **Call Coverage:**

DTAC only: Whenever a DTAC attendant is a call coverage receiver for a particular call coverage group and a call is placed from the attendant position via the Start button or the Selector Console to a voice terminal in that group, the Call Coverage-Group (COVER-GRP) button on the Attendant Console will not track the call (COVER-GRP button status LED will not flash). If the call remains unanswered, it will return to the Attendant Console on the RTN-DA button rather than the COVER-GRP button.

SLAC only: Whenever the common queue is a receiver for a particular call coverage group and a call is placed from the attendant position via the START button or the Selector Console to a voice terminal in that group, an unanswered call will return to the attendant queue as a Return-On- Don't-Answer call (instead of as a coverage

call).

- **Send All Calls (DTAC only):** If Send All Calls is activated, returning calls *will* ring at the DTAC.

Administration Requirements

System:

- Assign number of rings before call return to the Attendant Position (1-31) - Default = 5 Rings.

Hardware Requirements

None

ATTENDANT SOURCE AND DESTINATION (V2; SLAC Only)

Description

Allows the attendant, while extending a call, to switch back and forth between the calling party (the source) and the called party (the destination) before connecting them together.

Pressing the SOURCE button on the SLAC after the called party has been reached has these results:

- The called party (the destination) is put on hold.
- The attendant is reconnected to the calling party (the source).
- The green status lamp of the DEST button starts winking to indicate that the destination is on hold.
- The green status lamp of the SOURCE button goes from winking to dark.

Pressing the DEST button after the source has been reconnected has these results:

- The source is put on hold (again).
- The attendant is reconnected to the destination.
- The green status lamp of the SOURCE button starts winking to indicate that the source is on hold.
- The green status lamp of the DEST button goes from winking to dark.

Considerations

The Source/Destination feature is useful when the attendant needs to talk to each party privately before connecting them.

Interactions

The Source/Destination feature can only be activated before the two parties are connected together.

When the attendant presses JOIN, the other parties and the attendant are joined in a 3-way connection.

When the attendant presses RELEASE, the other parties are connected, the call is separated from the console, and the attendant is free to handle other calls.

After the source and destination parties are connected together, the SOURCE and DEST status lamps go dark.

Administration Requirements

None

Hardware Requirements

None; the SOURCE and DEST buttons are standard fixed feature buttons on the SLAC.

ATTENDANT SPLITTING ONE-WAY AUTOMATIC

Description

Allows the attendant to converse privately with a called party while the calling party is split away on hold.

When the attendant presses **START** (or a **DXS** button) to extend an incoming call to a called party, the calling party is automatically split away from the connection and placed on hold. This allows the attendant to talk privately with the called party before extending the call. The attendant can then press **RELEASE** to complete the transfer or **CANCEL** to drop the called station and return to the incoming call.

If the console is a **SLAC**, the attendant can also use the **Join** and **Source/Destination** features while in the **Start** mode. Refer to the descriptions of these features for details.

Considerations

Attendant Splitting One-Way Automatic allows the attendant to (1) announce a call, (2) determine privately whether the called party is available to receive the call, and (3) obtain information if necessary to redirect the call or take a message.

Interactions

- **Attendant Source/Destination (V2; SLAC only):** This feature can be used after reaching the called party. It allows the attendant to speak privately to both the calling party and the called party before connecting them together.
- **Music-On-Hold:** Music-on-hold is not provided to the calling party while the call is split from the console.

Administration Requirements

None

Hardware Requirements

None

ATTENDANT SYSTEM ALARM INDICATION

Description

Provides an Alarm on the Attendant Console to alert the attendant to problems detected by the system software. The ALARM LED on the Attendant Console will flash whenever a detected fault persists longer than four minutes, or if more than five transient faults per hour are detected. *The alarm indication should be reported immediately to your AT&T Systems Technician.*

The alarm type that causes an alarm indication is referred to as a Permanent System Alarm. These alarms are faults that can cause degradation of service and require immediate attention.

If a flashing ALARM button is pressed, the LED will change from flashing to steadily lit. A new trouble situation will cause a steady ALARM LED to start flashing again. Only when the trouble has been corrected will the LED turn off.

Considerations

The ALARM LED on the Attendant Console provides a warning as soon as the fault is detected. This permits a quick response to system detected faults.

In a two-attendant system, both consoles track problems.

Interactions

None

Administration Requirements

None

Hardware Requirements

None

AUTOMATIC INTERCOM

Description

Allows a multiline voice terminal user (or attendant) to place and answer calls to and from another station by use of a dedicated button appearance.

Automatic Intercom provides a private path between two designated multiline voice terminals. To place an Automatic Intercom call, the calling party presses the Automatic Intercom (AUTO ICOM) button and goes off-hook. The calling party hears ringback tone and the called party receives standard ringing. The status LED associated with the button is steadily lit at the calling voice terminal and flashing at the called voice terminal. To answer an Automatic Intercom call, the called party presses AUTO ICOM (not necessary with Ringing Line Preference) and goes off-hook.

The AUTO ICOM status LED lights steadily whenever the other party is off-hook. This provides each party with a station-busy indication for the other. To activate the busy-to-idle reminder, the user can press AUTO ICOM (remaining on-hook). A short burst of tone is provided when the other party goes on-hook. The user can then go off-hook, and the call will be placed; the user does not press the AUTO ICOM button again.

Pressing AUTO ICOM to invoke the busy-to-idle reminder overrides Prime Line Preference. Once activated, the feature can only be canceled by preelection of another button or answering an incoming call. See the “Busy-to-Idle Reminder” feature description for more information.

At a Switched Loop Attendant Console, operation of an AUTO ICOM button seizes an idle loop button for the outgoing call. An incoming intercom call arrives on a loop button and does not flash at the AUTO ICOM button of the console.

Considerations

The intercom feature should not be confused with ordinary station-to-station calling inside the system using dialed PDCs. With Automatic Intercom, users who frequently call each other can do so by pressing one button instead of dialing a PDC. In addition, the station-busy indication and busy-to-idle reminder provide additional utility to users.

This feature is similar to Direct Station Selection (DSS), except that the buttons must always be assigned in pairs (i. e., between two sets.) Hence, an AUTO ICOM button cannot point to a single-line set. Also, Automatic Intercom calls arrive at the AUTO ICOM button, thereby providing calling party ID; DSS calls arrive on System Access buttons.

Interactions

- **Attendant Position Busy:** The inactive attendant is permitted to place Automatic Intercom calls. For DTAC only: Automatic Intercom calls to the inactive attendant where the AUTO ICOM button is located in one of the two rightmost button columns will not ring at the console, nor can they be covered by the active attendant. For SLAC only: Automatic Intercom calls to the inactive attendant *will* ring at the inactive SLAC.
- **Call Coverage:** Automatic Intercom calls are considered private and do not receive call coverage.

- **Call Pickup:** When an Automatic Intercom call is picked up via Call Pickup, the AUTO ICOM status LED on the called voice terminal lights steadily. The called party can press AUTO ICOM to enter the call at any time.
- **Direct Group Calling:** Automatic Intercom calls cannot be directed to DGC groups.
- **Exclusion:** Any attempt to engage Exclusion while active on an Automatic Intercom call will drop the other party.
- **Last Number Dialed (V2):** Numbers called using an AUTO ICOM button are not saved by the Last Number Dialed feature.
- **Line Selection (Prime Line Preference):** When the Automatic Intercom line is assigned Prime Line status, the AUTO ICOM button must be pressed to activate the busy-to-idle reminder even though the I-use LED is already lighted steadily.

Administration Requirements

Voice Terminal: (Station Port)

- Assign AUTO ICOM buttons to voice terminals in pairs. Voice terminals can have several AUTO ICOM buttons assigned for direct access to multiple stations.

Hardware Requirements

None

AUTOMATIC ROUTE SELECTION (ARS)

Description

Provides for the routing of calls over the telecommunications network based on preferred routes (normally the least expensive route available at the time the call is placed.)

An ARS pattern can be composed of two subpatterns (time of day determines which subpattern is selected), each consisting of up to three routes, associated Facility Restriction Level (FRL) codes (described below), and CO overflow flags. A route is identified by specifying a Facility Access Code for the pooled facility (trunk group); for V2 systems only, a route may also be identified by specifying a Virtual Facility code.

A trunk group (or virtual facility, for V2) can be used in more than one ARS pattern and more than once within a pattern.

Call routing can be specified by as many as eight routing patterns. Each pattern contains a sequential list of routes (for example, trunk groups) the system can use to complete a call. Number translations (deletion and addition of dialed digits) necessary to route the call is determined on a trunk group basis. Overflow to the local CO when all trunks in a pattern are busy or the route FRL is too high is optional. If all trunks in a pattern are busy (including CO trunks if overflow is allowed), the call queues on the first route in the pattern.

All calls placed using the ARS access code (default = 9) are routed via the feature. The dialed numbers that follow the ARS access code are generally seven or ten digit DDD numbers preceded by a "1" if required by the serving Central Office. Numbers preceded by a "0" are routed over the local CO pooled facility.

Typically, a dialed 7-digit number consists of a CO code and exchange number in the form NXX-YYYY where N = 2-9, X = 0-9, and Y = 0-9. A 10-digit number consists of an area code, CO code, and exchange number in the form NPA-NXX-YYYY where N = 2-9, P = 0-1, A = 1-9, X = 0-9, and Y = 0-9.

Each route in a pattern has an associated FRL (0-3). This FRL may differ each time the facility is specified as a route. A facility with a FRL of "0" is least restricted to callers; a FRL of "3" is the most restricted. Similarly, each station in the system is assigned an FRL (0-3). A terminal assigned an FRL of "0" has the least ARS privileges (i.e., routes with FRLs of 1-3 are restricted); a FRL of "3" provides the most privileges. A station's FRL must be equal to or greater than the routes FRL to use the route.

The ARS feature, when accessed, selects a pattern as follows:

- Emergency Number Calls (routed via the local CO facility)
- International Calls (routed via the administered international pattern)
- Calls made to specified COs or seven digit telephone numbers within the Home Number Plan Area (HNPA). These calls are routed as specified in the HNPA Exception Lists, or else via the NPA Routing Table or (by default if not otherwise specified) the local CO facility.
- Calls made to NPAs outside the HNPA, sometimes referred to as Foreign NPAs (FNPA). The route selected depends on the type of call, as follows:
 - FNPA special number calls (includes all "800", "900", and Telex 510, 610, 710, and 810 numbers). For V2 systems, each FNPA of the form N00 and N10 may be assigned to a routing pattern. For V1 systems, these calls are routed via the local CO facility.

- FNPA calls made to numbers specified in the FNPA Exception List.
- All other FNPA calls.

ARS Flow Chart

Figure 2-8 provides a simplified ARS flow chart. Bracketed numbers (e.g., [401], [601]) provide a link between ARS administrable action numbers and the associated item on the flow chart. Certain readers may find this reference useful when reading the following description in association with the *System 25 Administration Manual*. Administrable System, Station, Toll Allowed, and Trunk action numbers are also noted where applicable.

The ARS feature is accessed when a user dials the ARS access code. As shown on Figure 2-8, the number dialed is first checked against the Emergency Numbers List. This list consists of special service codes (911) and up to three customer-defined seven digit numbers. If the number dialed matches one of the numbers on the list, the call is immediately routed via the local CO facility. All user call restrictions are disregarded.

If the number dialed is not on this list, a check is made to determine if the terminal is allowed to originate outside calls. If the terminal is outward restricted, the caller receives Reorder Tone; otherwise, the dialed number is checked against any toll restrictions that apply.

Terminals may be assigned a Toll Restriction Class (1-4), or be unrestricted (Class 0). Terminals assigned Toll Restriction Class 1 have the most privileges, those assigned Class 4 have the least privileges. There are four associated Toll Call Allowed Lists (1-4) in the system. Up to 64 3-digit CO codes and 6-digit NPA plus CO codes may be divided among the four lists.

Numbers dialed from voice terminals assigned Toll Restriction Class 1 are checked against all four Toll Call Allowed (TCA) Lists; numbers dialed from Class 2 terminals are checked against TCA Lists 2-4; numbers dialed from Class 3 terminals are checked against TCA Lists 3-4; and numbers dialed from Class 4 terminals are checked against List 4 only. If the number dialed does not appear on a list, the user receives Reorder Tone. Calls originated at unrestricted (Class 0) terminals are not screened.

Calls are checked to determine if they are international calls or operator calls. Dialed numbers "01" or "011" signify international calls, "0" plus a number other than "1" signify operator calls. If the call is an international call, the international routing pattern is selected and the call routed accordingly. Operator calls are routed via the local CO facility.

Calls within the HNPA are checked against the HNPA Exception Lists. There may be up to four of these lists, each with an associated ARS Routing Pattern. Up to 800 (64 in R1V1) 3-digit office codes may be divided among the four lists (eight entries may be 7-digit numbers.) If a match is found, the call is routed via the associated ARS Routing Pattern. If no match is found the dialed number is routed via the HNPA pattern (specified in the NPA Routing Table), or if none is specified, via the local CO facility.

If a number is entered more than once in the exception list, the pattern used will be the pattern associated with the more specific number.

The NPA Routing Table is simply a listing of North American Plan NPAs and Special Number NPAs (V2), each having an associated ARS Routing Pattern (all North American NPAs are assigned routing pattern 1 by default). A dialed NPA that is listed in the table is routed using the associated Pattern. Calls to NPAs not listed are routed via the local CO facility.

V1 Systems Only: For calls outside the HNPA (FNPA calls), a check is made against a non-administrable Special Numbers List. The numbers in the list include all “800” and “900” numbers and Telex codes 510, 610, 710, and 810. If a match is found, the call is routed via the local CO facility.

For FNPA calls not on the Special Numbers List (Vi), the dialed numbers are checked against the FNPA Exception List (V1 and V2). Up to 32 entries may be assigned to the list. Each entry must consist of a 3-digit NPA code, 3-digit CO code, and two additional digits (for a total of 8 digits). The last two digits may be “.”, which match any digit. Each entry has an associated ARS Routing Pattern. If a match is found, the call is routed using this pattern. If no match is found, the call is then checked against the NPA Routing Table. A dialed NPA that is listed in the table is routed using the associated Pattern. Numbers that don’t match are routed via the local CO facility.

ARS Routing Pattern Table

Figure 2-9 provides a block diagram of an ARS Routing Pattern. Up to eight of these patterns may be administered in the system. Each pattern consists of two subpatterns that may be chosen based on the time of day. Each subpattern (A and B) can contain up to three allowed routes. If all routes in a subpattern are busy, a CO overflow flag (when set) allows the call to be routed via the local CO facility; otherwise, the call will queue on the first route in the subpattern.

Administrable Start and Stop times (Hour and Minute) for Routing Subpattern A specify when Subpattern A should be used to route calls. Subpattern B is used to route calls at all other times.

Each route is specified by its trunk group facility access code (or Virtual Facility code in V2) and an associated FRL.

An FRL is typically lower for the first route in a subpattern and increases with each additional route in the pattern. A terminal’s FRL must be equal to or greater than the route FRL for the route to be selected. The system first checks the Route #1 for an available trunk on which to route the call. If the route is busy, Route #2 is checked, then Route #3, if required. If all routes in the subpattern are busy and the CO overflow flag is set, the voice terminal FRL is checked against an associated Overflow FRL before routing the call. For V1 only: If the voice terminal is allowed access to the local CO trunk pool, the system, on overflow, will attempt to place the call regardless of the associated FRLs.

If all routes in a subpattern are busy and the CO Overflow flag is not set, or all CO trunks are busy, the call returns to the first route in the subpattern and is queued (if the station FRL permits access to the first route). The caller is placed in an off-hook queue indicated by five short bursts of tone (Preferred Routing Tone). The order of service in the queue is First In First Out (FIFO). An ARS user placed in an off-hook queued state always has precedence over users with a Direct Facility Access buttons. There are 16 slots available in the ARS queue. A route #1 must be specified in the subpattern for a call to queue. If it is not, the caller receives Reorder Tone instead of being queued.

Once a route has been selected, the entries in a Digit Translations Table associated with the selected route’s trunk group (or Virtual Facility in V2) is checked. Based on an associated NPA and the NPA dialed, the system can remove up to 10 digits and then add a pattern of up to 5 digits as specified to route the call.

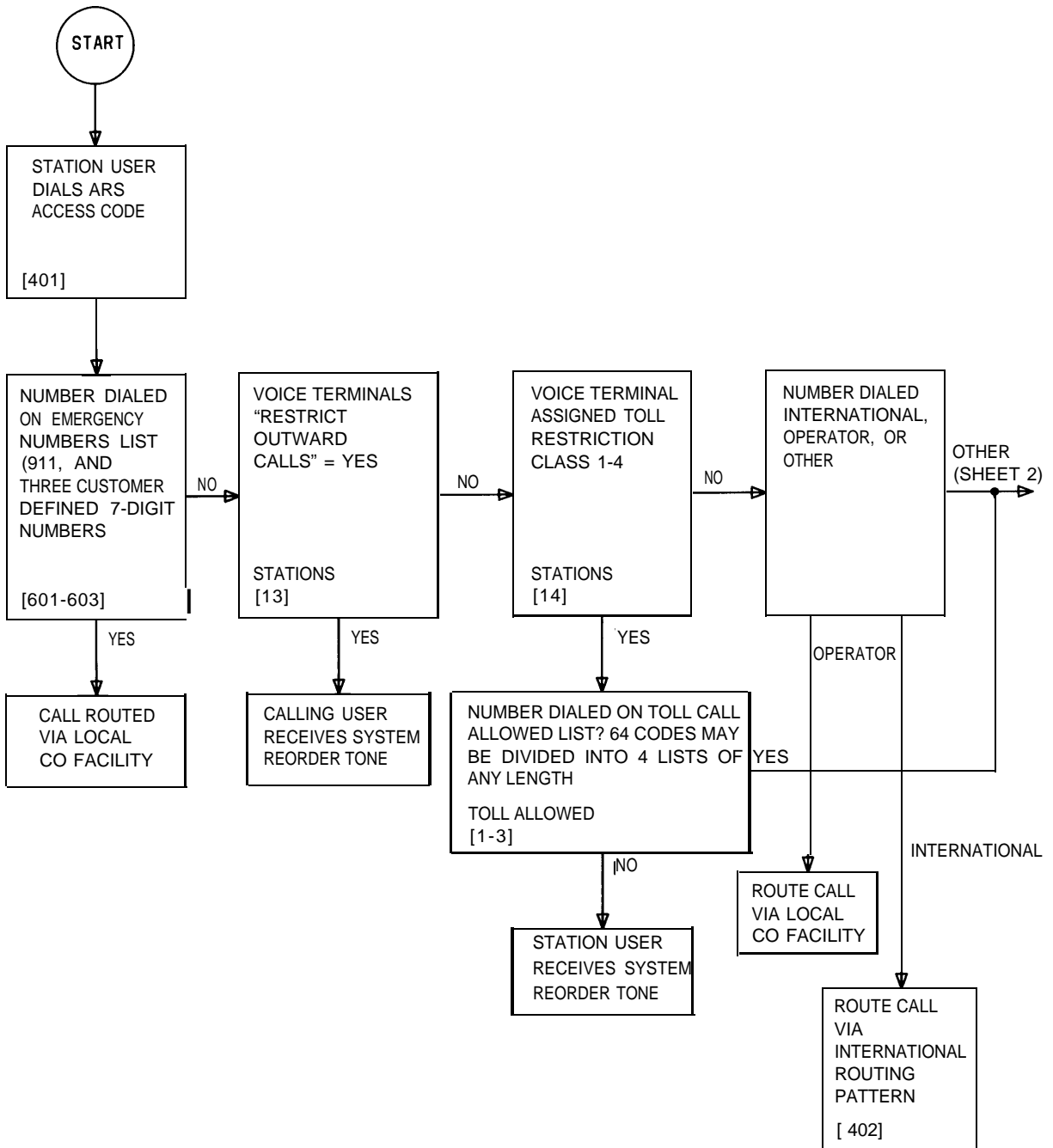


Figure 2-8. Automatic Route Selection Flow Chart (Sheet 1 of 2)

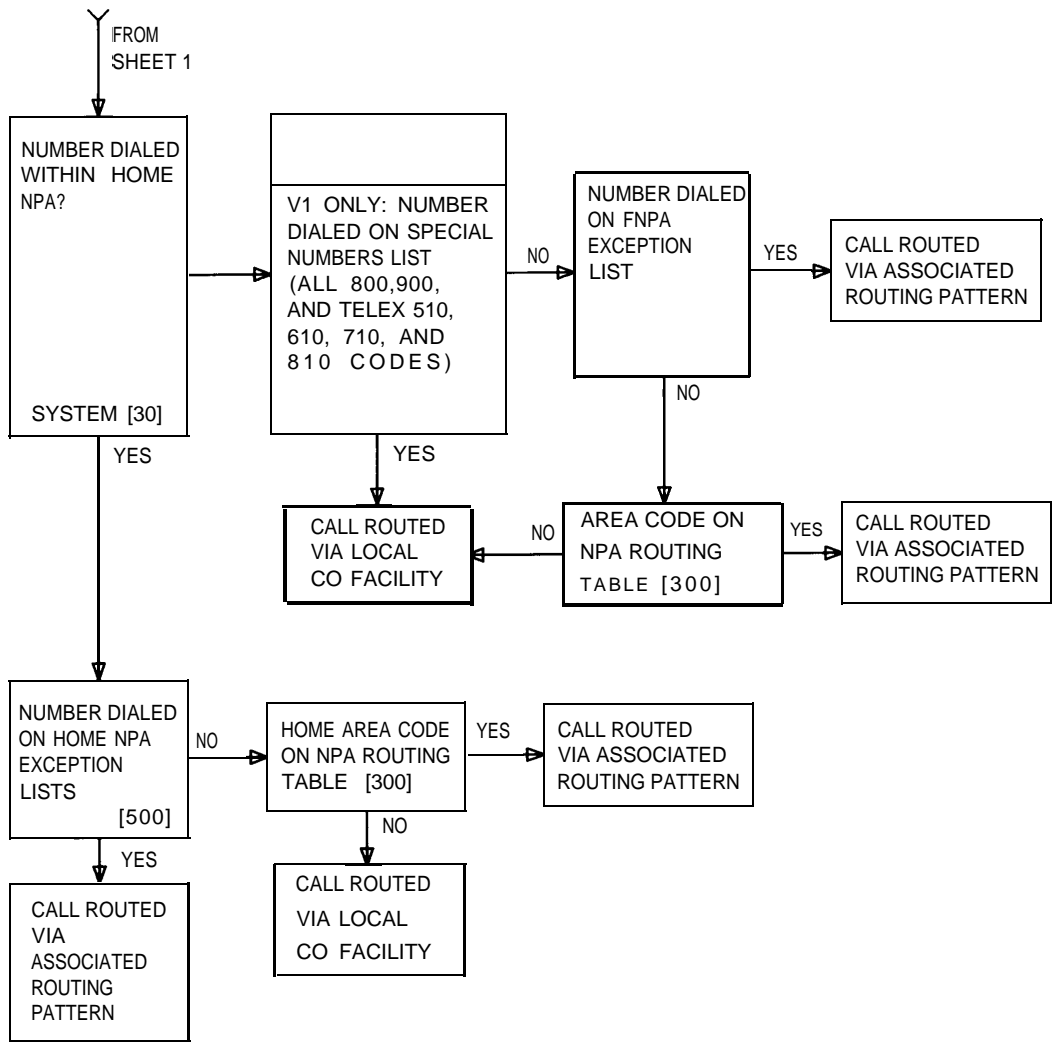


Figure 2-8. Automatic Route Selection Flow Chart (Sheet 2 of 2)

The following tones are associated with ARS:

- Confirmation—Indicates that a queued call is being serviced (trunk available to route call)
- Busy—Indicates that the called number is busy
- Reorder—Indicates that all trunks are busy, the ARS queue is full, or that ARS calling is denied.
- Preferred Route Tone—Five very short tones that indicate that your call has been queued for the preferred route.

Considerations

With ARS, users do not have to worry about accessing a particular pooled facility to make a long distance call. The user simply dials the ARS access code and the desired number. The system then routes the call via the facility best suited for that call.

The following provides a summary of the ARS call routing controls provided by the feature:

- **Emergency Numbers List:** 911, and up to three customer-defined, 7-digit numbers.
- **Toll Call Allowed Lists:** 1-4 lists, 64 entries maximum between all lists. Entries may be 3-digit CO codes or 6-digit NPA plus CO codes. (Administrable for uses besides ARS)
- **Station Toll Restriction Class:** 1-4 Classes (Administrable for users besides ARS)
- **HNPA Exception List:** 1-4 Lists, each with an associated ARS Routing Pattern. 800 (64 in V1) 3-digit CO codes entries maximum between all lists (eight of the entries may be 7-digit numbers.)
- **NPA Routing Table:** Entries may include every North American NPA (and Special Number NPAs in V2), each with an associated ARS Routing Pattern. All NPAs are assigned Routing Pattern #1 by default.
- **FNPA Special Numbers List (V1 only):** Routes all “800”, “900”, and Telex 510, 610, 710, and 810 numbers via the local CO trunk group (not administrable)
- **FNPA Exception List:** One List with up to 32 eight-digit numbers. Each entry has an associated ARS Routing Pattern.
- **Digit Translations Tables:** One per trunk group (V1 and V2) or Virtual Facility (V2 only).

A system can have up to eight ARS Routing Patterns assigned. Each pattern can contain up to six routes (three per subpattern).

Interactions

- **Calling Restrictions:** Outward Restriction and Toll Restriction, when administered, prevent calls from routing via ARS. Pooled facility access restrictions do not apply. In fact, the recommended arrangement to ensure that users make maximum use of ARS is to block dial access to most trunk groups, so that users must dial the ARS access code to place calls.

- **Direct Facility Access Button:** Multiline voice terminal users who have pressed FACILITY to activate the busy-to-idle reminder must wait until all queued ARS users have been serviced.
- **Virtual Facilities (V2 only):** Virtual Facilities may be used in place of trunk groups in ARS routing patterns. When used with ARS, a digit translation scheme may be associated with each virtual facility. See the “Virtual Facilities” feature description for more information.

Administration Requirements

System:

- Specify your area code (HNPA)

ARS

- ARS Access Code (1-9999) - Default = 9
- International ARS Routing Pattern Number (1-8) - No default
- Three Emergency Numbers Lists (7-digit numbers) - No default
- NPA Routing Table (NPA code and associated pattern number 1-8) - Default = 1
- HNPA Exception List:
 - List Number (1-4)
 - Pattern Number (1-8)
 - Exception Numbers (NXX or NXX-YYYY). Last three digits may be “.”
- FNPA Exception Telephone List
 - Pattern Number (1-8)
 - Telephone Number (8-digits in the form NPA-NXX-YY). Last two digits may be .
- Digit Translations Table
 - Trunk Group Facility Access Codes (1-9999) or, in V2, Virtual Facility Codes (#190-#199)
 - Associated NPA (NPA)
 - Number of digits to remove for calls within associated NPA (1-10 digits, none)
 - Digit pattern to add for calls within associated NPA (maximum of 5 digits) - Default = none
 - Number of digits to remove for calls not in associated NPA (1-10, none) - Default = 0
 - Digit pattern to add for calls not in associated NPA (maximum of 5 digits) - Default = none
- ARS Routing Pattern:
 - Pattern Number (1-8)

- Subpattern A Start and End Time (Hour: Minute)
- Subpattern A and B—Route 1, 2, and 3 Facility Access Codes (1-9999) or, in V2, Virtual Facility Codes (#190-#199)
Subpattern A and B—Route 1, 2, and 3 FRLs (0-3)
- Subpattern A and B Overflows to CO facility (Yes, No)
Subpattern A and B Overflow FRL (0-3).

Terminal (Station Port)

- ARS FRL Level (0-3).

Hardware Requirements

None

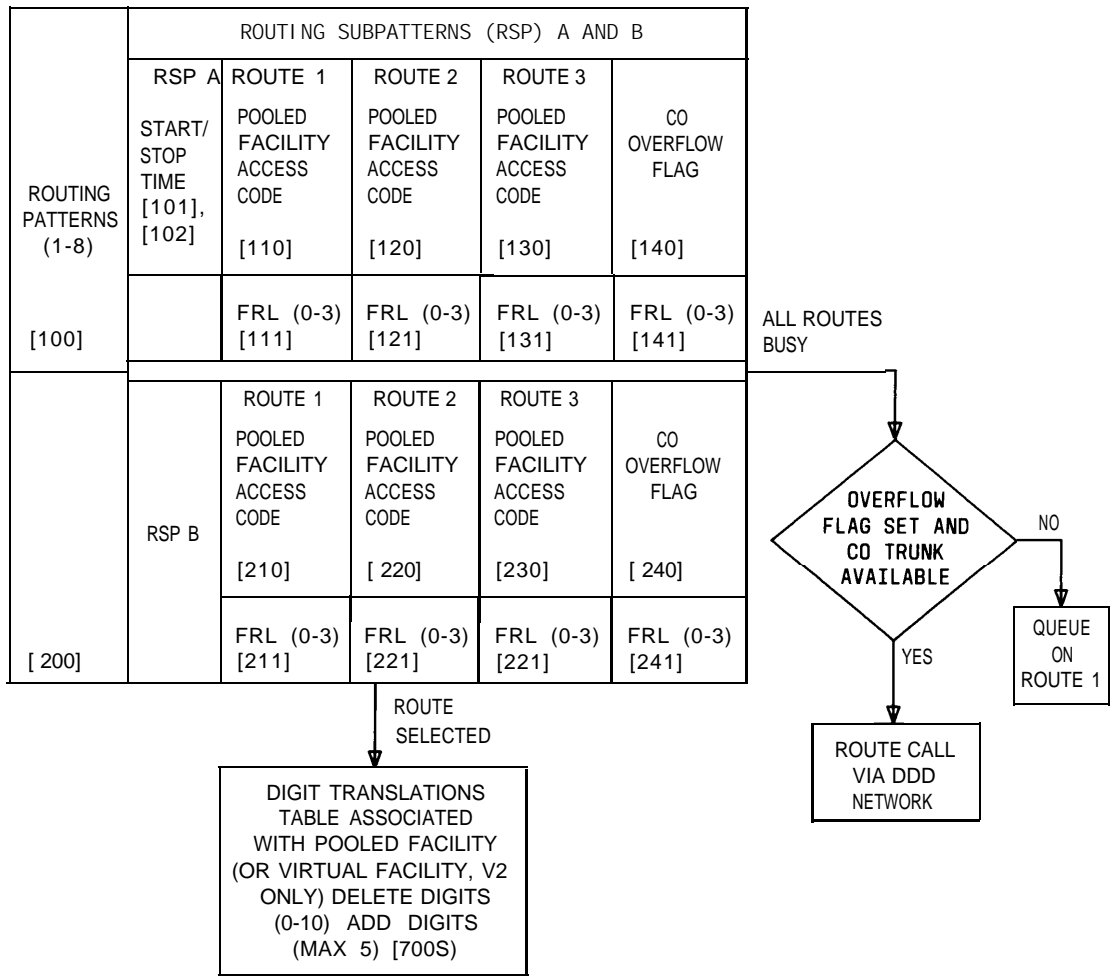


Figure 2-9. Automatic Route Selection Routing Pattern

BUSY-TO-IDLE REMINDER

Description

Enables a multiline voice terminal user to be alerted by a single ring as soon as a busy internal station, DGC group, or facility (trunk group) becomes available. The feature can be activated only for stations, DGC groups, and trunk groups represented on the terminal by DSS, FLEX DSS, AUTO ICOM, and FACILITY buttons.

Before making a call to a station, the multiline voice terminal user can check the green status LED of the station button. If it is lit, the station party is off-hook. To be alerted when the party hangs up and is available again, the user (while on-hook), presses the button of the station. The red I-use LED lights, indicating that Busy-to-Idle Reminder is in effect. When the other party hangs up, the user's terminal rings once. The user simply goes off-hook, and the station is called; the user does not have to press the button again.

If the user calls a station by pressing a FLEX DSS, DSS, or AUTO ICOM button and receives busy tone, the user must hang up before activating Busy-to-Idle Reminder.

When all the trunks in a pool represented by a FACILITY button are busy, the green status LED is lighted. The user can activate Busy-to-Idle Reminder in the same way as for a station call, by pressing the FACILITY button while on-hook. When a trunk becomes idle, the terminal rings once. The user goes off-hook and is automatically connected to the trunk. To complete the call the user dials the desired outside number.

Considerations

Busy-to-Idle Reminder gives the multiline voice terminal user a way to get quick access to a station or trunk group that has just become available after being busy. Access to the station or trunk is not reserved for the user who activates this feature; any other user has equal access to the idle facilities.

On some multiline voice terminals, FLEX DSS, DSS, AUTO ICOM, and FACILITY buttons do not have status and I-use LEDs. This makes Busy-to-Idle Reminder less convenient to use because the user must first call the facility to determine if it is available. If it is not, the user activates the feature by hanging up and pressing the button again. When the reminder ring sounds, the user must then remember which button was used to initiate the call.

Interactions

- Busy-to-Idle Reminder cannot be used with the Last Number Dialed, Personal Line, or Repertory Dialing features.

Administration Requirements

None

Hardware Requirements

None

CALL ACCOUNTABILITY

Description

Allows system users to charge outside calls made from other users' stations to their own PDCs.

To charge a call to his or her own PDC, the user dials ## followed by the PDC immediately upon receiving first dial tone to place a call. When second dial tone is returned, the user dials the desired number in the normal way.

After completion of the call, the SMDR record will reflect the "accountable" PDC (that is, the caller) in the "PDC" field, and the PDC of the voice terminal used in the "STN" field.

Considerations

Call Accountability, if used consistently, helps to ensure that calling costs are attributed accurately to the personnel who incur the costs. Users do not use this feature when calling from their own stations or when making inside calls from any station.

Interactions

All of the following conditions apply only when a user is calling from another user's station.

- **Account Code Entry:** This feature can be used on the same call with Call Accountability.
- **Call Accounting:** The caller's PDC that is entered by the Call Accountability procedure is integrated into the reports generated by Call Accounting systems.
- **Direct Facility Access:** When a call is made using a Facility button, ##PDC must be dialed before pressing the button.
- **Repertory Dialing:** When a call is made using a Rep Dial button, ##PDC must be dialed before pressing the button.
- **Speed Dialing:** When a call is made using Speed Dialing, ##PDC must be dialed before dialing the Speed Dialing code.

Administration Requirements

None

Hardware Requirements

None

CALL ACCOUNTING

Call Accounting is the collecting, processing, and use of information about all trunk calls placed from and received by System 25. It is intended to help customers control telephone use and manage associated costs.

Detailed call data is available at Interface Port 3 of the Digital Switch's Call Processor Circuit Pack. This data can be fed to one of the following peripheral equipment systems:

- Station Message Detail Recording (SMDR) equipment for printing a standard call report.
- A Call Accounting System (CAS) for preparing a variety of cost estimate reports and for providing management and directory type services.

Two station features of System 25 are also related to Call Accounting and are covered in separate subsections of this manual. Account Code Entry allows individual voice terminal users to associate specific account codes with their calls, when necessary. Call Accountability provides users with the means to properly identify calls they make from stations other than their own. The information gathered from these two features is part of the data output from the processor to the SMDR or CAS.

The remainder of this subsection is dedicated to coverage of SMDR and CAS.

Station Message Detail Recording (SMDR)

Description

SMDR records detailed call information on all incoming and outgoing (external) calls and sends this information to an (optional) output device. Data on inside calls is not collected.

The call records can be used to compute costs, allocate charges, and analyze calling patterns. The output device can be any serial RS-232 compatible DTE device capable of receiving the data (must supply DTR on pin 20) and either printing the call records or storing and analyzing them. (80 character ASCII records are sent to the output device.)

The SMDR RS-232 port interface is provided by a DUART driver (68681). It is a one-way port transmitting data to the output device. No characters are read by the port interface, and no flow control mechanisms are provided. The standard data transmit rate is 1200 bps. (Also operates at 300 bps.)

Call Records

The call records provide detailed information concerning both incoming and outgoing calls. Call detail records are generated during call processing and are sent to the SMDR output device in ASCII format. SMDR records are provided for:

- **Voice Records:** The system prints call records for incoming calls and for outgoing calls that exceed a specified duration. For special types of calls such as conference or transferred calls, one call record is reported for each trunk seized, regardless of the number of parties connected to the call. The call's duration is from the time the last digit was dialed until the last person hangs up. No indication is provided that trunks have been bridged together.

- **Data Records:** The system prints call records for incoming and outgoing (external) data calls. Calls are considered data calls if they involve a data extension.

The following list describes the SMDR data collected for each call and the number of characters in each field. All information is right justified in its field, unless otherwise indicated. The record is provided in a standard 80-column format. The headings for each record item are noted in bold type. These headings are printed across the top of each page. Page advance is determined by counting lines based on a fixed page length. Each record is followed by a carriage return and a line feed.

The system can provide for the storage of up to 100 SMDR records. If more than 100 records are received while the printer is disconnected, a message “Calls Lost Due To Call Record Overflow” is provided when a printer is re-attached.

The SMDR call detail (Figure 2-10) contains the following information for each call record:

- **TYPE** (Column 1)
All voice calls are labeled C, data calls are labeled D. (“TYPE” is not printed as a column heading)

- Blank (Column 2)

- **DATE** (Columns 3-10)

The date the call is originated.

- Blank (Column 11)

- **TIME** (Columns 12-16)

The time the trunk is seized is listed using a 24-hour clock. For example, 2:01 PM is listed as 14:01. Seconds are truncated.

- Blank (Column 17)

- **CALLED NUMBER** (Columns 18-35)

For outgoing calls, up to 15 digits may be recorded, excluding the ARS or facility access code but including the O or 1 prefix (to identify local and toll calls) and 950-10xx and 10 xxx interconnect access codes. Space is allotted for three dashes, one between the fourth and fifth digits from the right, one between the seventh and eight digits from the right, and the other between the tenth and eleventh digits from the right. Numbers longer than 15 digits will be truncated. For Repertory Dial and Speed Dial numbers, the facility will be extracted from the stored number and reported under the FAC heading; the number remaining after the facility is extracted will be reported as the called number.

An incoming call is identified by the word IN.

- **Error Character** - Question Mark or Blank (Column 36). Indicates number dialed exceeded 15 digits.

- Blank (Column 37)

- **DUR** (Columns 38-45- Duration)

For incoming calls, this provides the time between trunk seizure and disconnect, rounded to the nearest second. For outgoing calls, it provides the time between the last digit dialed until the last station on the call hangs up, less an estimated time for call setup (15 seconds), rounded to the nearest second. A call transferred between a

number of voice terminals will reflect the total call duration. The maximum time that can be reported is 95 hours, 59 minutes, and 59 seconds.

- Blank (Column 46)
- FAC (Columns 47-51- Facility)

Indicates the facility used to place the call. For outgoing calls including speed dialed numbers, the pooled facility selected by ARS or the facility access code that was dialed (or that corresponds to the facility button that was pressed) is identified. For incoming calls and Personal Line calls, the trunk number is identified.

For V2 only: If a virtual facility was used to complete the call, the applicable Virtual Facility Code (#190-#199) is identified in this field.

- Blank (Column 52)
- **STN** (Column 53-56)

Identifies the voice or data terminal responsible for the call. If an account code is entered, the voice terminal where the code is entered is reported. If no account code is entered, the terminal originating an outgoing call is identified, or the last terminal connected to an incoming call is identified.

For an incoming call to a DGC group that is connected to an announcement but is never answered, 0 will be recorded in the STN field. If the call is answered by a station after receiving announcement, the station answering the call will be recorded.

For V2 only: If an outgoing call is originated by a tandem tie trunk, the tandem trunk's Facility Access Code (FAC) is recorded in this field. If no FAC exists for this trunk, then the 4-digit trunk number (9xxx) will appear.

For V2 only: If an outgoing call is originated by a DID trunk, 0000 is recorded in this field.

- Blank (Columns 57 ,58)
- ACCOUNT (Columns 59-73)

Lists the Account Code associated with the call, if one was entered. On conference and transferred calls, the first account code entered is recorded and subsequent account code entries are ignored.

- Blank (Column 74)
- PDC (Columns 75-78)

Identifies the user responsible for outgoing calls. The user is identified by the call accountability login (##PDC) entered at the originating voice terminal. *If no call accountability is entered, the PDC field is blank.*

Figure 2-11 and Figure 2-12 summarize the Call Record and Call Record Header formats.

Considerations

SMDR provides detailed call information on incoming and outgoing calls. This information can be used to facilitate cost allocation, traffic analysis, and detection of unauthorized calls.

Interactions

- **Account Code Entry:** Allows users to have an account code or project number associated with each call record.
- **Direct Group Calling (DGC):** For an incoming call to a DGC group that is connected to an announcement and never answered, "O" will be reported in the "STN" field of the call record. If the call is answered by a station after receiving the announcement, that station will be listed in the "STN" field.
- **Modem Pooling:** SMDR records do not reflect modem pool resource usage.
- **Tandem Trunking (V2 only):** If an outgoing call is originated by a tandem tie trunk, the tandem trunk's FAC is recorded in the STN field. If no FAC exists for this trunk, then the 4-digit trunk number (9xxx) will appear. No other SMDR fields are affected.

Administration Requirements

System:

- Send SMDR records To SMDR Port (Yes, No) - Default = Yes
- Minimum length (seconds) of calls that are reported by SMDR (10-255) - Default = 40.

Hardware Requirements

An AT&T Model 475 printer or any standard RS-232 serial 80-column ASCII printer is required for printing the SMDR output. The printer must be dedicated to SMDR to ensure that all calls are recorded. An AT&T Call Accounting System may also be used as the SMDR output device (see below).

The printer can be directly connected to Port 2 of the Call Processor ZTN82 (V1) or ZTN128 (V2), or switched access (either on- or off-premises) can be provided. Connection is the same as described for the SAT.

Detailed connection information is provided in Figures 2-13 through 2-17

Maximum cabling distances are provided in Section 5, "Technical Specifications."

Call Accounting System (CAS)

Two types of CASs can be used with System 25:

- CAS Model 100—A discrete microprocessor unit with cartridge-packaged software and a built-in power supply.
- CAS Model 200, 300, 500, or 2000 Software Package associated with an AT&T Personal Computer (PC) 6300.

Description

CAS Model 100:

The CAS 100, when connected to a serial printer and the System 25 SMDR port, calculates the cost of calls made to outside numbers, stores the cost information, and generates chronological and summary reports.

After each incoming or outgoing call is completed, a call record is printed on the Chronological Report. Each call record includes the date and time of the call, the Personal Dial Code (PDC) of the station involved, the call's duration (in minutes), the call type (voice or data), the calculated cost, and the number of the facility on which the call was made. If the call is outgoing, the call record also includes the number dialed, a two-character abbreviation for the area to which the call was made, an account code, and the PDC claiming accountability for the call.

When a call record with a new date is received by the CAS 100, the previous day's Chronological Report is ended and reports containing summary information are printed. Four summary reports are produced:

- Calls by Hour of the Day
- Calls by Facility Used
- Calls by Extension Number
- Calls by Account Code.

These reports summarize the information presented in the Chronological Report and list the total number of calls, their total duration, and the total cost.

Switches inside the CAS Model 100 cabinet allow the System Administrator to control the system options that determine report parameters.

CAS Models 200, 300, 500, and 2000:

The System 25 SMDR interface provides direct output to an AT&T PC 6300 Personal Computer equipped with MS-DOS (V2.11 or later) and a CAS software package (i.e., Model 200, 300, 500, and 2000) that calculates the cost of calls and provides basic and sophisticated call reports.

After a telephone call is completed, System 25 sends a call record to the AT&T PC 6300 via the SMDR interface channel. The PC must be equipped with CAS software. Call records are collected by the PC and held in a buffer until they are processed. When a call record is processed, a cost is calculated and assigned to it. That cost, along with other call record information, is then stored on a hard disk for subsequent retrieval.

Two modes of operation are available for PC 6300 operation:

- **Dedicated Mode:** The PC is dedicated to one and only one task-processing call records.
- **Multi-Function Mode:** Allows the user to print reports, edit files, and run other PC-based programs while the CAS continues to collect and buffer call records in the background. The user must enter the Dedicated Mode to process calls and generate reports.

The CAS performs three main functions; (1) call record processing, (2) report generation, and (3) CAS system management. In addition, a limited directory lookup and message center is provided. The followings a brief description of each function:

1. **Process Calls:** Involves screening call records, calculating the cost of valid calls, anti storing the call records.
2. **Generate Reports:** Allows the user to print the stored call record information organized in one of several different ways. Users can select a report or set up their own special combination of reports from the following:
 - Summary Reports—A collection of reports that condense and summarize call record information by total number of calls, duration, and cost. The reports can be organized by department, call type, cost center, trunk, extension, cost, duration, time of day, date, and account code.
 - Organization Detail Report—A detailed report of each call record in the system, sorted by department, cost center, and extension.
 - Selection Report—This report can contain at a user's option, summary or detailed information based on any combination of the following items: time of day, date, cost, duration, extension, access code, account code, dialed number, call type, department, or cost center.
 - Account Code Detailed Report—A detailed report on call records sorted by account code. This report can be used for billing clients for calls made in their behalf.
 - Preselected Reports—Allows up to five predefine reports, which can include any of the above mentioned reports. These reports can run upon request or at a specified time and date.
3. **System Management:** Allows the user to perform several functions. These include editing the table of departments, cost centers, and extensions; setting up account codes; defining preselected reports; and keeping call rate information up to date. System configuration may be changed. This allows the user to inform the CAS of changes in System 25 (e.g., dial access codes, trunks) or changes in charge rates. System housekeeping may also be performed. This includes establishing passwords, deleting call records, determining call processing options, and performing various disk maintenance operations.
4. **Directory Lookup and Message Center:** Allows the user to look up anyone by last name, first name, or extension. Messages can be recorded for individuals and can be printed or displayed.

The following table summarizes CAS station and account code capacities.

AT&T CALL ACCOUNTING SYSTEMS		
CAS Model	Stations	Account Codes
200	100	5000
300	150	5000
500	500	5000
2000	2000	15000

Refer to the CAS documentation supplied with the software package for additional information.

Considerations

The CAS provides customers with an efficient tool to control and manage their telephone usage and costs. The information available can be used to facilitate cost allocation, traffic analysis, and abuse control.

Interactions

None

Administration Requirements

System:

- Send SMDR Records To SMDR Port (Yes, No) - Default = Yes
- Minimum length (seconds) of calls that are reported by SMDR (10-255) - Default = 40.

Hardware Requirements

CAS Model 100 requires a CAS Microprocessor and a Custom Cartridge containing rate information specific to your location.

CAS Model 200, 300, 500, or 2000 applications software must be run on an AT&T PC 6300 equipped with MS-DOS (V2.11 or later).

Refer to Figures 2-13 and 2-14 for typical CAS connection information.

	DATE	TIME	CALLED NUMBER	DUR	FAC	STN	ACCOUNT	PDC
C	10/08/85	11:14	1-232-566-1321	00:15:41	9	1794	123489764321341	4271
C	10/08/85	11:14	IN	00:09:05	2145	1324		
C	10/08/85	11:15	1-322-564-1376	00:29:50	9	1744	766544	3254
D	10/08/85	11:17	1-222-564-2171	00:10:45	9	2001		
C	10/08/85	11:19	IN	00:05:32	3214	1744		
C	10/08/85	11:20	IN	00:29:45	2342	3455		
C	10/08/85	11:34	1-242-563-1324	00:19:00	9	1677	76322	3422
D	10/08/85	11:44	555-4541	00:05:35	9	2312	323489764321341	3422
D	10/08/85	11:50	1-252-514-3176	00:19:45	9	3455		
C	10/08/85	12:14	IN	00:25:42	2145	1492		
C	10/08/85	12:24	1-222-566-2544	00:10:35	9	1244	123489764321341	4271
C	10/08/85	13:27	1-333-513-1376	00:15:05	9	3566		
C	10/08/85	13:14	1-244-564-3121	00:09:40	9	2001		
C	10/08/85	13:23	IN	00:15:45	3414	1566		
C	10/08/85	13:28	IN	00:19:35	3421	3421		
C	10/08/85	13:33	1-222-516-1176	00:19:40	9	1492	123489764321341	4271
C	10/08/85	13:45	555-2541	00:09:05	9	3655		
C	10/08/85	14:14	1-222-563-4321	00:20:42	9	4321	763444	3465
D	10/08/85	14:24	1-343-516-2574	00:10:05	9	1244		
C	10/08/85	14:34	555-3141	00:09:45	9	4633		
C	10/08/85	14:43	IN	00:19:32	3214	2351		
C	10/08/85	15:14	1-343-564-1321	00:20:45	9	1794	1234893764321341	4271
C	10/08/85	15:16	1-222-566-1321	00:19:45	9	1794		
C	10/08/85	15:19	IN	00:19:45	2342	1794		

Figure 2-10. Typical SMDR Call Detail Report

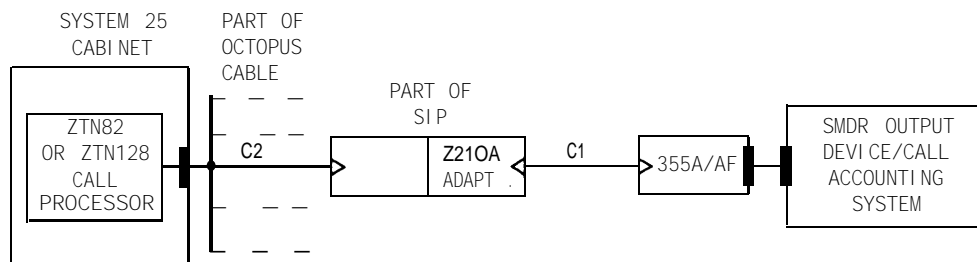
ASCII CHARACTER POSITION (Column Number)	DESCRIPTION	VALID CHARACTERS
01	Call Type	C or D
02	Space	
03-04	Date: Month	0-9
05	/	/
06-07	Date: Day	0-9
08	/	/
09-10	Date:Year	0-9
11	Space	
12-13	Time:Hour	0-9
14		
15-16	Time: Minute	0-9
17	Space	
18-35	Dialed Number	0-9, Space, -, IN
36	Error Character	?, or Space
37	Space	
38-39	Duration:Hour	0-9
40		
41-42	Duration: Minute	0-9
43		
44-45	Duration: Second	0-9
46	Space	
47-51	Facility	Space, 0-9, #
52	Space	
53-56	Station	0-9, Space
57-58	Space	
59-73	Account Code	0-9, Space, #, ?
74	Space	
75-78	Personal Dial Code	0-9, Space
79		Carriage Return
80		Line Feed

Figure 2-11. SMDR Call Record Format

ASCII CHARACTER POSITION DESCRIPTION

00	(top of form)
01-03	Space
04-07	DATE
08-12	Space
13-16	TIME
17-23	Space
24-29	CALLED
30	Space
31-36	NUMBER
37-38	Space
39-41	DUR
42-48	Space
49-51	FAC
52-54	Space
55-57	STN
58-62	Space
63-69	ACCOUNT
70-76	Space
77-79	PDC
80	(Carriage Return)
81	(Line Feed)

Figure 2-12. SMDR Call Record Header Format



LEGEND:

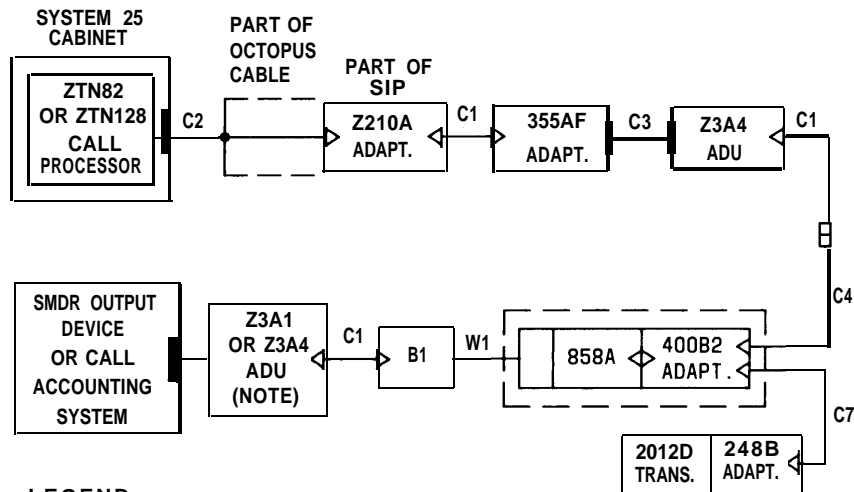
C1 -MODULAR CORD (D8W-87) - PEC 2725-07G

C2 -OCTOPUS CABLE (WP90780) - PEC 2720-05P

355A ADAPTER RS 232 PLUG TO MODULAR JACK - PEC 2750-A24

355AF ADAPTER RS 232 RECEPTACLE TO MODULAR JACK - PEC 2750-A25

Figure 2-13. SMDR Output Equipment or Call Accounting System—On-Premises Direct Connections (Sharing Same AC Outlet)



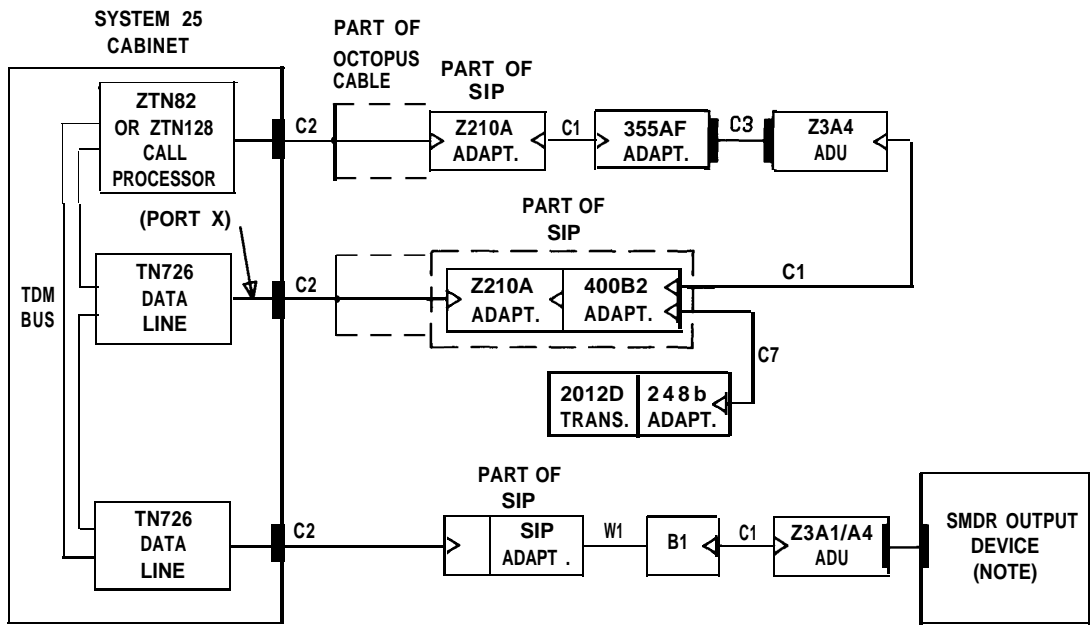
LEGEND:

- B1 - TYPICAL-103A CONNECTION BLOCK*
- C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
- C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
- C3 - EIA CROSS OVER CABLE (M7U-87) - PEC 2724-30C
- C4 - ADU CROSS-OVER CABLE (D8AM-87) - PEC 2724-38X
- W1 - 4 PAIR INSIDE WIRING CABLE*
- 355AF ADAPTER RS-232 RECEPTACLE TO MODULAR JACK - PEC 2750-A25
- Z3A1 ADU - EQUIPPED WITH A THREE FOOT PLUG-ENDED EIA CORD - PEC 2169-001
- Z3A4 ADU - EQUIPPED WITH A 3 FOOT RECEPTACLE ENDED EIA CORD
- C7 - MODULAR POWER CORD (D6AP-87)
- 248B ADAPTER - MODULARIZES 2012D TRANSFORMER } PEC 21691
- 400B2 ADAPTER - POWER ADAPTER
- 20210 TRANSFORMER - 15-18V AC TRANSFORMER

NOTE: CAS MODELS 100 AND 200 REQUIRE Z3A4 ADU.

* - FURNISHED BY INSTALLER

Figure 2-14. SMDR Output Equipment or Call Accounting System—On-Premises Direct Connections (Greater Than 50 Feet From System Cabinet)



LEGEND:

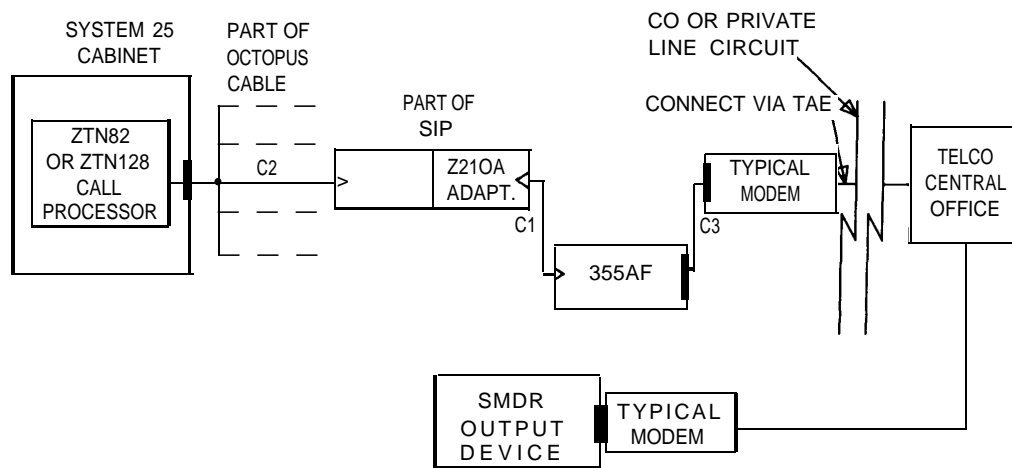
- B1 - TYPICAL-103A CONNECTING BLOCK*
- C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
- C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
- C3 - EIA CROSS OVER CABLE (M7U-87) - PEC 2724-30C
- W1 - 4 PAIR INSIDE WIRING CABLE*
- 355AF - ADAPTER (RS-232 RECEPTACLE TO MODULAR JACK) - PEC 2750-A25
- Z3A1 ADU - EQUIPPED WITH A 3 FOOT PLUG-ENDED EIA CORD - PEC 2169-001
- Z3A4 ADU - EQUIPPED WITH 3 FOOT RECEPTACLE ENDED EIA CORD - PEC 2169-004
- 248B ADAPTER- MODULARIZES 2012D TRANSFORMER
- 400B2 ADAPTER - POWER ADAPTER
- 2012D TRANSFORMER -15-18V AC TRANSFORMER
- C7 - MODULAR CORD (D6AP-87)

PEC 21691

* - FURNISHED BY INSTALLER

NOTE: OUTPUT DEVICE OR MULTILINE VOICE TERMINAL WITH DATA BUTTON DIALS PORT X DDC TO ESTABLISH DATA CONNECTION TO ZTN-82.

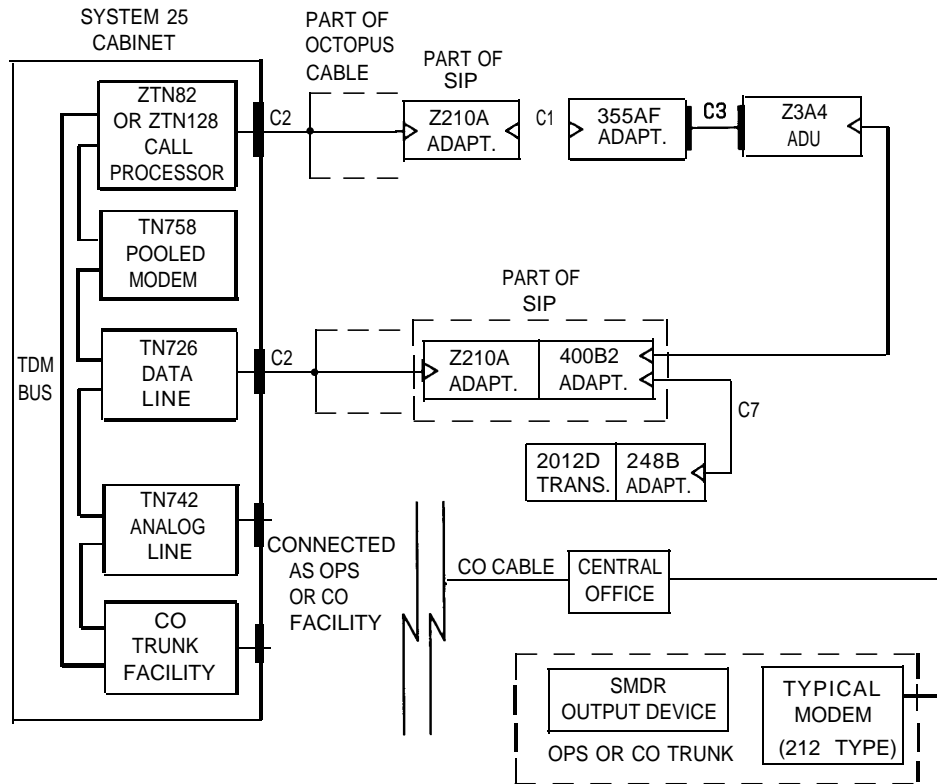
Figure 2-15. SMDR Output Equipment—On-Premises Switched Connections



LEGEND :

- C 1 - MODULAR CORD (D8W-87) - PEC 2725-07G
- C 2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
- C 3 - EIA CROSS OVER CORD (M7U-87) - PEC 2724-30C
- 355AF ADAPTER - (RS-232 RECEPTACLE TO MODULAR JACK) - PEC 2750-A25

Figure 2-16. SMDR Output Equipment—Off-Premises Direct Connections



LEGEND :

- C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
- C2 - OCTOPUS CABLE" (WP90780) - PEC 2720-05P
- C3 - EIA CROSS OVER CABLE (M7U-87) - PEC 2724-30G
- OPS - OFF PREMISES STATION
- 355AF ADAPTER - RS-23Z RECEPTACLE TO MODULAR JACK PEC 2750-A25*
- Z3A4 ADU - EQUIPPED WITH 3 FOOT RECEPTACLE - PEC 2169-004
- 248B ADAPTER - MODULARIZES 2012D TRANSFORMER
- 400B2 ADAPTER - POWER ADAPTER PEC 21691
- 20210 TRANSFORMER - 15-18V AC TRANSFORMER
- C7 - MODULAR POWER CORD (D6AP-87)

* - FURNISHED BY INSTALLER

Figure 2-17. SMDR Output Equipment—Off-Premises Switched Connections

CALL COVERAGE—GROUP

Description

Allows calls to covered stations to be redirected to a group of covering stations.

A total of 32 standard Call Coverage Groups may be assigned; for V2 systems, an additional 32 “DGC” call coverage groups may be assigned. Each standard group may be covered by up to eight call coverage receivers (buttons). There is no limit on the number of covered users (senders) that each Call Coverage Group can include, but a covered user can be assigned to only one Call Coverage Group. Each call coverage receiver must have a multiline set equipped with a Cover (COVER-GRP) button, except as noted below. A covering set may be assigned more than one COVER-GRP button for the same or different groups.

In R1V2 Systems equipped with a Switched Loop Attendant Consoles (SLAC), the console queue can serve as a standard coverage group receiver. The consoles cannot have COVER-GRP buttons, so the queue directs coverage calls to LOOP buttons.

Direct Group Calling (DGC) Groups may be designated as Call Coverage Group receivers in V2 Systems. This provides the capability for System 25 to support “non-integrated” voice mail systems as well as allow the formation of coverage pools.

Senders may be either single-line or multiline voice terminals. In V2 systems, receivers may be single-line voice terminals if part of a DGC Call Coverage Group. Multiline voice terminals may always be used as receivers

Standard Group Coverage

When a call arrives at a voice terminal that has group coverage, the COVER-GRP or LOOP button status LED at the covering voice terminals will flash. Covering voice terminals will begin to ring after a specified number of rings at idle covered voice terminals. If there is no idle system access button at the covered station (sender), the call receives coverage treatment, and the call immediately rings at the covering terminal(s). If no idle cover button is available at the covering terminal(s), the calling party receives a busy signal. In V1 Systems, the call will not receive coverage treatment if both system access buttons on the sender’s station are busy; the calling party receives busy signal.

Ringling may be turned off at standard receiver stations for each covering button, as desired (not recommended). If this option is selected, a flashing status LED will be the only indication received at the covering station. In addition, Call Coverage ringling may be turned off on internal calls (if desired) on a system-wide basis.

A member of a standard receiver group can use the Line Selection (Preelection) feature to answer covered calls even before any audible alerting has begun at the covering user’s terminal. This is useful if the user knows that the covered party is unavailable.

A covered voice terminal may elect to have calls covered while it is busy on another call. Calls directed to an idle button on a busy covered multiline voice terminal will start ringling at the covering terminal after a single burst of ringling at the busy covered voice terminal. If there is no idle Cover button on the covering voice terminal, the call will queue and will ring at the first available coverage receiver (V2 only); in a V1 system, the call will not receive coverage treatment. Calls directed to a busy single-line voice terminal will start ringling immediately at the covering terminal. If there is no idle Cover button on any covering terminal, either individual or group, the call will not receive coverage and the calling party will receive Busy Tone.

Calls sent to coverage will continue to ring at single-line sender terminals but will cease ringing at multiline sender terminals. In the latter case, the calls will remain on the incoming call appearance button, and that button's status LED continues to reflect the status of the call. In particular, covered calls to multiline sets remain available and can be entered by the called (sender) station.

If a covered station activates the SEND ALL CALLS button, calls will be directed to coverage immediately, with or without a single-ring reminder, as administered.

A station can provide (or receive) Individual Call Coverage (see Call Coverage-Individual) and also be a member of a Call Coverage Group (sender or receiver). Unanswered calls to a station, provided both Individual and Group Call Coverage, will first ring at the Individual Coverage station and then, after a second delay cycle and still unanswered, will ring at the Group Coverage station.

Calls from a covering station to a covered station will not be covered unless the covered station has additional coverage. This is an important consideration when the attendant provides coverage.

DGC Group Coverage (V2 only)

Calls proceeding to the DGC-Call Coverage Receiver Group hunt in a circular fashion for the first idle station, starting from the last station to receive a call. If all DGC members are busy, both internal and external calls continue to ring and/or flash at the covered station and any individual coverage receiver's station(s) until a DGC station becomes idle. If a DGC group is used for both DGC calls and group coverage, trunk calls into a DGC group have priority over coverage calls. Calls sent by coverage to a DGC-Call Coverage Group member station do not receive additional call coverage.

DGC groups cannot be call coverage senders to another DGC-Call Coverage Group. However, calls made directly to a DGC member can be covered by another DGC-Call Coverage Group.

Once a call has been redirected to a DGC-Call Coverage Group member, the call is transferred to the covering station. The call continues ringing until answered or dropped. The call is not accessible at the covered station nor any individual coverage receiver once it is redirected to an idle DGC station. If all DGC members are busy, the call remains accessible at the covered station until a member is available.

DGC-Call Coverage Groups count against the system specified maximum number of DGC groups, but *not* against the number of Call Coverage Groups. The limit of eight receiving stations per Call Coverage Group does not apply when administering a DGC group as a Call Coverage Group. A maximum of ten stations per DGC-Call Coverage Receiver Group is allowed.

Considerations

Call Coverage provides a way to redirect calls to alternate answering positions. The feature is versatile enough to permit suitable alternate answering arrangements for virtually every level of employee. Special functions, such as the Send All Calls feature, accommodate the day-to-day variations that occur in an employee's work schedule.

Interactions (Standard Group Coverage)

- **Attendant Console, Direct Trunk:** If the Direct Trunk attendant is a receiver for a Call Coverage Group and extends a call (using the Start button or Selector Console) that is unanswered/busy to a member of the group, the call will return On the Return-On-Don't-Answer (RTN-DA) or Return-On-Busy (RTN-BUSY) button, not on the attendant's COVER-GRP button.
- **Automatic Intercom:** Auto-Intercom calls do not receive call coverage.
- **Call Coverage—Individual:** Unanswered calls to a station, provided both Individual and Group Call Coverage, will first ring at the Individual Coverage station and then, after a second delay cycle and still unanswered, will ring at the Group Coverage station.
- **Call Coverage/Station Hunting:** A call to a busy single-line voice terminal that is both a member of a Station Hunting group and a Call Coverage sender will first hunt for an idle station to service the call. If none is available, the call will be sent to coverage.
- **Call Pickup:** Call Pickup is independent of call coverage. When a call is answered via Call Pickup, all Cover buttons associated with the called party go idle.
- **Direct Group Calling (DGC):** A call to a DGC group member will receive coverage if the member is also a Call Coverage sender. Calls to a busy DGC group do not receive call coverage. Instead, after a predefined number of rings, the call will be transferred to delay announcement (if provided), or ringing will be transferred to all button appearances of the line.
- **Exclusion:** If a call coverage receiver invokes Exclusion after answering a coverage call, all other terminals (including the attendant and the covered station) are excluded from the call. The covered user cannot enter the call until EXCLUSION is pressed a second time by the covering user.
- **Hands-Free-Answer On Intercom:** An incoming (inside) call will not receive call coverage if auto-answer is activated, since the set will answer the call (whether the user is present or not.).
- **Night Service:** Directed Night Service calls do not receive call coverage.
- **Personal Dial Codes:** Calls directed to a station because another *non-floating* PDC is logged in there *do not* receive the coverage treatment of the logged-in station. Such calls return to their home station and receive that station's coverage (immediately upon return). Calls to logged-in *floating* PDCs, on the other hand, receive *the same* coverage treatment as any other calls to the logged-into station. They, of course, have no home station to return to.
- **Personal Lines:** Personal line calls receive the coverage of the principal (owner) station for that line. Other line appearances (even if administered to ring) will not receive coverage.
- **Tie Trunks:** Tie Trunk calls directed at a user with call coverage receive normal call coverage treatment.

Interactions (DGC Group Coverage; V2 Only)

- **Attendant Console, Switched Loop:** If a SLAC is a member of a DGC-Call Coverage Receiver Group, any DGC—Group Coverage call sent to this attendant will enter the common queue and be treated as a *coverage call*, not as an Attendant—DGC call. Thus, the call will be handled by whichever attendant is administered to receive coverage calls.
- **Attendant Direct Extension Selector Console:** The Selector Console can be used to transfer and place calls to a DGC—Call Coverage Group provided the DGC group access code appears on the console. The status LED of the DXS button lights steadily whenever all stations in the DGC—Call Coverage Group are busy.
- **Automatic Intercom:** Auto Intercom calls do not receive call coverage.
- **Call Pickup:** A DGC—Call Coverage Group member can also be in a call pickup group.
- **DSS:** A DSS button can be assigned to a DGC—Call Coverage Group. The button lights whenever all DGC members are busy.
- **Flex DSS:** The Flex DSS button can be assigned to a DGC—Call Coverage Group.
- **Night Service:** Directed Night Service calls do not receive call coverage.
- **Personal Line Access:** All outside lines directed to a DGC—Call Coverage Group can be assigned to button appearances in addition to the DGC—Group Coverage assignment. If the outside lines appear at stations that also have DGC coverage by the same group, then the operation is as follows:

When an incoming call is ringing in the DGC group, the status LEDs on the appearance buttons light steadily, indicating that the line is busy. If the call goes unanswered after a system-specified number of rings, then a delay announcement is provided. The caller is subsequently put on hold and receives music if available. If the system is not equipped with a delay announcement, the call begins to ring at all line appearances after the system-specified number of rings.

If the outside lines are not directed to a DGC group, but are provided DGC Group Coverage, the feature operation is the same as for incoming calls on SA keys except that the call appearance remains accessible at the covered station after being directed to a DGC—Call Coverage Group member.
- **Station Hunting:** Calls directed to a DGC—Call Coverage Group will not hunt.
- **Trunk Groups:** Trunks can be directly assigned to DGC groups that also act as call coverage groups. Among tie trunks, only automatic incoming tie trunks can be translated as directed to a DGC Group.

Administration Requirements (Standard Group Coverage)

System:

- Provide Call Coverage ringing on internal calls (Yes, No); Default = Yes.
If No, covered calls will flash but not ring at covering stations on internal calls.

- Number of rings before call coverage ringing starts on no answer (0-31); Default = 2.
The status LEDs on Group Coverage buttons at covering stations begin flashing immediately in all cases. Ringing, in addition to flashing, is always sent on external calls, though it may not be accepted at the covering stations.

Voice Terminal: (Station Port)

- Coverage Sender group number (1-32) - Default = 1
- Provide Call Coverage ringing on no answer (Yes, No); Default = Yes.
If No, flashing LED is the only indication received at the covering station.
The calling party always hears ringing.
- Provide Call Coverage ringing on busy (Yes, No); Default = Yes.
If No, flashing LED is the only indication received at the covering station.
If the covered station is a busy single-line set or a multiline set with one System Access appearance busy, the call is sent to coverage and the calling party hears ringing. If the covered station is a R1V1 multiline set with both System Access appearances busy, the call cannot go to coverage and the calling party hears busy tone; in R1V2 the call goes to coverage.
- Call Coverage Receiver button
 - Group Number (1-32)
 - Allow Ring At Destination (Yes, No); Default = Yes.
If No, flashing LED is only indication received at the covering station.

Administration Requirements (DGC Group Coverage; V2 Only)

The parameter to assign a call coverage sender group has been expanded to include DGC groups as coverage group receivers. DGC receiver groups are first set up as regular DGC groups, numbered 1-32. DGC coverage groups can then be specified by using coverage group numbers 101-132, where coverage group 101 has DGC group 1 as its receiver group, coverage group 102 has DGC group 2 as its receiver group, etc. No button assignments are required.

Send ringing options (on busy, on no answer, system-wide for internal calls) have no effect for DGC group coverage; *all calls sent to coverage*.

The system search of group call coverage sender stations has been expanded to allow the system administrator to enter a DGC—Call Coverage Group and list all its sender stations by PDC.

Hardware Requirements

None

CALL COVERAGE—INDIVIDUAL

Description

This feature is very similar to Call Coverage—Group. The primary difference is that this is a one-on-one type coverage.

An Individual Cover (COVER IND) button can be assigned on multiline voice terminals to cover calls to a specific (single) voice terminal. The covering station can answer covered calls by pressing COVER-IND. Each button can be programmed to ring or not to ring. If ringing is selected, the covering station will begin ringing after a specified number of rings at the covered station. When the specified number of rings has occurred, multiline voice terminals will stop ringing. Single-line voice terminals continue to ring until the call is answered at a covering terminal. When the call is answered at the covering station, the call remains accessible at the call appearance button of multiline voice terminals, but is no longer accessible at single-line voice terminals.

Covered calls will appear on the COVER-IND button; all calls except Automatic Intercom and Directed Night Service calls are covered.

Each Cover button at a covering station represents one covered voice terminal. If more than one voice terminal is to be covered, multiple buttons are required, one for each station covered. A covering voice terminal may be assigned multiple COVER-IND buttons for a particular station to cover multiple simultaneous calls to that station. The first button will track the first call, the second button, the second call, etc.

Up to eight COVER-IND buttons can be assigned for each covered station.

For VI systems, up to 31 stations may receive Individual Call Coverage. There is no limitation for V2 systems.

A voice terminal can receive both Individual Call Coverage and Group Call Coverage.

Refer to Messaging Services for a description of Call Coverage—Message Waiting service, which allows the covering station to control the status of the covered user's Message LED.

Calls from a covering station to a covered station will not be covered unless the covered station has additional coverage. This is an important consideration when the attendant provides coverage.

Considerations

Call Coverage provides a way to redirect calls to alternate answering positions. The feature is versatile enough to permit suitable alternate answering arrangements for virtually every level of employee. Special functions, such as the Send All Calls feature, accommodate the day-to-day variations that occur in an employee's work schedule.

The Call Coverage—Individual feature is not administrable on the Switched Loop Attendant Console.

For V2 only: COVER-IND buttons can be used to provide the ability to receive more than two *incoming* calls at a time. Up to eight COVER-IND buttons can be administered on a multiline voice terminal, providing "individual call coverage" *for itself*. When both System Access buttons are busy, subsequent incoming calls will be directed to available COVER-IND buttons on the set, thus simulating additional System Access buttons (for incoming calls only).

Interactions

- **Attendant Console, Direct Trunk:** When a coverage call rings at a busy Attendant Console, the attendant receives a single burst of ringing. If the call is still unanswered when the attendant hangs up, the Attendant Console will resume ringing.
- **Call Coverage—Group:** Unanswered calls to a station, provided both Individual and Group Call Coverage, will first ring at the Individual Coverage station and then, after a second delay cycle and still unanswered, will ring at the Group Coverage station.
- **Direct Station Selection (DSS):** Calls placed via a DSS button to a user with Individual Call Coverage will receive call coverage. When a DSS button is used to activate the busy-to-idle reminder for the user, the reminder is returned only when the user becomes idle, not when an associated coverage user becomes idle.
- **Exclusion:** If a covering station answers a coverage call and then invokes Exclusion, all stations including the covered one, are excluded from the call.
- **Hold:** May be used to place a coverage call on hold. The COVER-IND button's status LED winks at the covering station and the call appearance's status LED winks at the covered terminal. The held call will automatically leave the coverage terminal if picked up by the covered user. The covering station will be unable to reenter the call.
- **Transfer:** When a covering station transfers a covered call to another station, the call will no longer appear at the covering station's Cover button, though it still will appear at covered multiline stations.

Administration Requirements

Voice Terminal (Station Port):

- Individual Call Coverage button (button function # 12)
 - Individual Coverage PDC (1-9999)
 - Individual Coverage Ring (Yes, No).

Hardware Requirements

None

CALL FOLLOWING (FORWARDING)

Description

Allows users who are away from their phone to receive calls at another phone. The feature is functionally equivalent to Call Forwarding.

Users can log in their PDC at another voice terminal and receive their calls at that terminal. Logging in a PDC at a terminal automatically logs the PDC out at any other terminal. PDCs will always have an associated home terminal. If a PDC is not logged in anywhere, calls to the PDC will ring at the home terminal.

Logging in a FPDC automatically logs the FPDC out at any other voice terminal. However, logging out a FPDC does not log the FPDC in at another terminal. Calls to FPDCs not logged in at a terminal may be directed to the attendant (administrable and strongly recommended).

When a PDC is logged in at another terminal (away terminal), dialing the PDC from the away terminal directs the call to the PDC's home terminal.

To log in a PDC, the user goes off-hook at the "away" terminal, enters "***" and their PDC twice. The PDC logout procedure is similar; after going off-hook and receiving dial tone, the user enters "***" followed by their PDC and then by a "0." A "***0" entered at a voice terminal will log out all PDCs and FPDCs logged in at that voice terminal. Confirmation Tone is returned to a user who correctly completes one of these procedures.

The following call types or features are voice terminal oriented (associated with stations rather than PDCs) and do not follow a user who logs in at another terminal.

- Automatic Intercom Calls
- Directed Night Service calls
- DGC Group Calls
- Manual Signaling
- Message Waiting
- Outward/Toll Restriction.
- Personal Line Calls

In R1V1 systems, Direct Station Selection (DSS)/Flex DSS calls do not receive call following treatment. This limitation does not apply to R1V2 systems.

Considerations

Call Following provides maximum flexibility to system users who are away from their voice terminals. In addition, visitors can receive calls by logging in an assigned FPDC.

For more information, see the "Personal Dial Code (PDC)" feature description.

Interactions

- **Call Coverage:** Calls to a logged-in FPDC receive the call coverage of that terminal. [Unanswered calls to a PDC at an away terminal return to the home terminal and receive the home terminal's call coverage treatment; they do not receive the away terminal's call coverage.]

Administration Requirements

None

Hardware Requirements

None

CALL PARK

Description

Allows a user to place a call on hold and then pick up the call from any voice terminal in the system. It is used in three typical applications:

- **Call Park:** A user places a call on hold and then picks it up at another voice terminal
- **Meet-Me-Conference:** A conference member places the conference on hold and pages another employee to join the conference.
- **Transfer:** A user places a call on hold and then pages another employee to pickup the call

Call Park allows a user to move a held call to a “parked” position by dialing the Call Park code (*5). The call can subsequently be retrieved from any voice terminal by dialing the Call Park retrieval code (*8) and the PDC of the parking station. In addition, any user active in a conference involving fewer than five members may park the conference, so that another user may join the conference.

A multiline voice terminal user may invoke Call Park by pressing HOLD to place a call or conference on hold, then pressing an idle System Access button and dialing *5. A single-line voice terminal user may invoke the feature by pressing the switchhook to place the call or conference on hold, then dialing *5. If the call is successfully parked, the user receives Confirmation Tone and then Recall Dial Tone. If the call cannot be parked, Reorder Tone is received. In the latter case, to return to the held call, press the held call button (multiline sets) or flash the switchhook (single-line sets).

The parking station may return to a parked call or conference without affecting the park state. The multiline voice terminal user may return by pressing the held call button. The single-line user may return by pressing the switchhook.

When the single-line user goes on-hook, the parked call is removed from the terminal and cannot be reentered.

To retrieve a parked call, a user must obtain system dial tone, dial *8 and then dial the PDC of the station that parked the call. If the call is not retrieved within 2 minutes the call will return to the user that parked the call.

A call may be parked by the attendant using the same procedure as a multiline voice terminal. In addition, if the Attendant Position is equipped with a Selector Console, up to eight additional calls may be parked by dedicating any eight of the console DXS buttons to the call park function. A call may then be parked using the Selector Console by pressing one of these dedicated Call Park buttons. The status LED of the parked line on the Attendant Console winks and the status LED of the Call Park button on the Selector Console lights steadily. The call can subsequently be retrieved from any voice terminal by dialing the Call Park retrieval code (*8) and the PDC of the DXS button on which the call is parked.

A call parked by the attendant using the same procedure as a multiline voice terminal will return to the Attendant Console’s System Access (DTAC) or LOOP (SLAC) button if it is not picked up within 2 minutes. A call parked with the Selector Console that is not picked up within 2 minutes will return to the RTN-DA button (DTAC) or a LOOP button (SLAC), in the same manner as any other unanswered call.

A call parked with the Selector Console is retrieved by dialing *8 and the access code assigned to the dedicated Selector Console Call Park button.

Each voice terminal user (except the attendant) can only park one call at a time and a maximum of 24 calls can be parked in the system at one time. A call is no longer parked when it is answered, returns to the parking terminal, or is abandoned by the caller.

Considerations

Call Park can be used whenever a user engaged on a call needs to go elsewhere, and wishes to complete the call from another terminal. Call Park also allows users to answer a call from any voice terminal when paged.

Interactions

- **Attendant Direct Extension Selection:** Station-To-Station calls cannot be parked via the Call Park buttons on the Attendant Selector Console.
- **Attendant Position Busy:** If a call is parked on an attendant console and the attendant console enters Position Busy mode, the parked call will return to the *inactive* console if not answered within two minutes.

If a call is parked on the Selector Console by a Switched Loop attendant (V2 only) and the SLAC is placed in the Position Busy mode, the parked call will return to the other *active* attendant console, if not answered within two minutes.
- **Conference:** Parked conference calls do not return to the parking voice terminal. They remain parked. Call Park may be used to place a conference on hold if it contains fewer than five parties.
- **Exclusion:** A call cannot be parked, and a parked call cannot be answered, if the Exclusion feature is invoked on that call.
- **Intercept Treatment With Reorder Tone:** An unsuccessful attempt to park a call due to misdialing or attempting to park more than one call at a voice terminal results in Reorder Tone.
- **Music-On-Hold:** Parked calls (except conferences) receive music.
- **Outward Restriction (see “Calling Restrictions”):** If the parking station is outward restricted, the Recall Dial tone following a successful parked call cannot be used to avoid restriction.
- **Personal Line:** A parked Personal Line is bridgeable by any user with a button appearance of that line. Bridging on to the connection does not answer the parked call. The parked call will not return to the parking voice terminal user in this case.
- **Toll Restriction: (see “Calling Restrictions”)** If the parking voice terminal user is toll restricted, the Recall Dial tone following a successful park cannot be used to avoid the restriction.

Administration Requirements

Attendant Selector Console:

- Assign Call Park buttons.

Hardware Requirements

None

CALL PICKUP

Description

Allows a user to answer a call ringing at another voice terminal. There are two forms of Call Pickup; (1) Directed and (2) Group.

Directed Call Pickup:

Directed Call Pickup allows calls to most other voice terminals, including Automatic Intercom calls and calls ringing at coverage buttons, to be “picked up” by dialing the Call Pickup code (*7) and the ringing terminal’s PDC. Picked-up calls remain accessible at the call appearance button of multiline terminals, but are no longer available at single-line terminals.

A ringing call can be answered at a busy- single-line voice terminal by pressing the switch hook, which will place the current call on hold, dialing *7, and the ringing voice terminal’s PDC.

This is a standard feature available at every voice terminal. No administration is required. Also, this feature cannot be turned off or restricted.

Group Call Pickup:

Group Call Pickup permits calls to another terminal in the Call Pickup group to be answered. Any call, including Automatic Intercom calls, is eligible for Call Pickup.

A member of a Call Pickup Group can answer any call to any other member of the group by dialing the Group Call Pickup code (*70).

Up to 16 groups (with up to 16 voice terminals in each group) can be set up. Each Call Pickup group can have a maximum of two simultaneous ringing calls eligible for Call Pickup treatment at a time, and the calls are picked up in order of arrival. A user can be assigned to only one Call Pickup Group.

If the picked-up call was to a multiline terminal, the called terminal can still enter the call. If the called terminal was a single-line terminal, it cannot enter the call once it is picked up.

Call Pickup cannot be invoked after the call has been answered.

If no eligible calls are ringing, attempting a call pickup results in Reorder Tone.

Considerations

With Call Pickup, users do not have to leave their own voice terminal to answer a call at a nearby voice terminal. Instead, a user simply lifts the handset and dials an access code. This allows calls that may go unanswered to be handled quickly and efficiently.

The call must be administered to *ring* at the voice terminal for which Call Pickup is attempted. Otherwise the attempt will be blocked.

Interactions

- **Call Coverage:** When a call is directed to a coverage station and the call is answered via Call Pickup, all Cover buttons associated with the call go idle.

- **Personal Line:** After a call is picked up from a Personal Line (PERS LINE) button, the called terminal can still enter the call.

Administration Requirements

Voice Terminal (Station Port):

- Assign Call Pickup Group Number (1-16, None) - Default = None.

Hardware Requirements

None

CALL PROGRESS TONES

Call Progress tones provide audible feedback on the status of calls during call set-up.

- **Busy Tone:** A slow pulsed tone indicating that all facilities for answering the call are in use.
- **Call Waiting (Camp-On) Tone:** A single short tone to a busy terminal indicating that another call is waiting (has been “camped on” by the attendant).
- **Confirmation Tone:** Three short tones indicating that the system has accepted the instruction entered or that your ARS queued call is being placed.
- **Dial Tone:** A steady tone indicating that dialing or feature activation can begin.
- **Dialing Feedback:** Indicates that a digit has been dialed.
- **Preferred Route (Queuing) Tone (ARS):** Five short tones. Indicates no facility is currently available to place your call. If you remain off-hook, your call will be placed as soon as a facility becomes available.
- **Reorder Tone:** A fast pulsed tone indicating that all trunks are busy, a dialing error has occurred, or the terminal is restricted from making this call.
- **Ringback Tone:** In general, indicates that a called terminal is ringing.

For additional information, refer to “Tones” in Section 5.

CALLING RESTRICTIONS

Description

Designated voice and data terminals can be restricted from making certain types of calls. Available restrictions are:

- Outward Restriction
- Toll Restriction
- Facility Access Restriction
- ARS Restrictions.

Note: Each of these restrictions is voice terminal oriented, not PDC oriented.

Outward Restriction:

When outward restricted, a station will be unable to place any outside calls. The station will be able to answer incoming calls and place and receive inside calls. A station that is outward restricted will be unable to use Automatic Route Selection to place external calls except to the emergency numbers.

Toll Restriction:

Allows calls by restricted terminals to be made based on as many as the first six digits of the number called (after the facility access code). Toll restricted users can make outgoing calls only to those numbers that are on the Toll Call Allowed (TCA) Lists to which it has access. TCA entries must be in the form NXX or NPA-NXX (exactly three or six characters.) The system administrator can establish up to 4 individual lists. A list can contain from 1 to 64 entries provided that the total of all four lists does not exceed 64. One character “.” can be specified as a wild card character in place of the last 1, 2, or 3-digits (e.g., “NX.”, “N.”, or “...”) of the NXX code, but not in the NPA code. When this character is used, any character in the dialed number appearing in that position is acceptable. Those stations assigned Toll Restriction Class 1 have access to all four TCA Lists; Class 2 stations just lists 2 through 4, Class 3 stations just lists 3 and 4, Class 4 stations just list 4. The entry of an area code followed by “...” on a TCA List allows access to all office codes in that area. To allow calls within a customer’s local area, individual office codes are entered; this allows the customer to restrict toll calls within the local calling area. NPA-NXX entries allow specific office codes to be called within an area. Note, NPA only entries are not permitted (use NPA-...).

Note that stations that are toll restricted are only toll restricted on CO trunks (type 701 and 801) or when they use the ARS feature. They will not be toll restricted when they dial access (or button access) any other type of trunk (e.g., FX, WATS, or Tie trunks).

Facility Access Restriction:

Any station may be denied dial access to the local CO and/or to all other pooled facilities (as a group). A station so restricted may only *dial* access those facilities via the Automatic Route Selection (ARS) feature.

In addition, each trunk (VI and V2) and Virtual Facility (V2 only) can be administered to allow or restrict dial access. If dial access is restricted, the trunk or Virtual Facility (V2) may only be *dial* accessed via ARS.

ARS Restriction:

Special restrictions on each station may be imposed when the call is routed by the ARS feature. Facility Restriction Levels (FRLs) are used to restrict access to trunk groups. An FRL is a single digit (0, 1, 2, 3). A terminal assigned an FRL of 0 has the least privileges, a terminal assigned an FRL of 3, the most. An FRL is also assigned to each route in each ARS routing pattern. The terminal's FRL must be equal to or greater than the route's FRL in order to use that facility.

Considerations

Restrictions are used whenever it is necessary to restrict certain users from accessing designated facilities. A typical application is to deny most stations dial access to all trunk groups. This forces callers to use the ARS feature, which should result in reduced toll charges.

Interactions

- **Automatic Route Selection:** The use of the ARS feature will not allow users to avoid restrictions. Outward Restriction and Toll Restriction, when administered, can prevent calls originating at associated voice terminals from routing via ARS. Facility access restrictions, however, are circumvented.
- **Direct Facility Access:** Toll restricted stations receive standard toll restriction treatment on all Direct Facility Access (FACILITY) buttons.
- **Personal Lines:** Personal Lines are subject to the toll restriction options of the stations on which they appear.
- **Repertory Dialing:** A user cannot use Repertory Dialing to access a number that he/she is restricted from dialing.
- **Speed Dialing:** A user cannot use Speed Dialing to access a number that he/she is restricted from dialing.
- **Transfer:** A non-restricted user (typically the attendant) can transfer a CO trunk to an outward restricted or toll restricted station, giving the station outward service. The toll restriction class of the *transferring* station will apply for calls placed over a transferred trunk.

Administration Requirements

Terminal (Station Port):

- Restrict access to CO trunk pool (Yes, No) - Default = No
- Restrict access to all other trunk pools (Yes, No) - Default = No
- Restrict outward calls (Yes, No) - Default = No
- ARS Facility Restriction Level (Level Number 0-3) - Default = 3
- Specify Toll Restriction Class (Class Number 1-4, None) - Default = None (not restricted).

Automatic Route Selection:

- Route Facility Restriction Levels

Trunk Port:

- Allow dial access to this trunk (Yes, No) - Default = Yes

System (V2 only):

- Allow dial access to this virtual facility (Yes, No) - Default = No

Hardware Requirements

None

COMMAND MODE

Allows data terminal users to originate data and voice calls and change (or view) their data port options.

(Refer also to the general description of the system's data features in "Data Services Overview"; users who are familiar with Command Mode options should read the "Expert Mode" subsection.)

Command Mode supports digital data endpoints connected via Asynchronous Data Units (ADUs) to ports on a Data Line CP (TN726). Command Mode also supports, with certain restrictions, users of the STARLAN Interface Circuit Pack, ZTN84 (V2 only).

Command Mode is invoked from a data terminal in the idle (on-hook) mode by:

Terminal Optioned For Autobaud

Enter Break followed by Return

Terminal Not Optioned For Autobaud

Enter Break

The terminal then displays the Command Mode menu:

<Data call> <Voice call> <Options> <Hangup> (V2 only)
<place Data call> <Options> <Hangup> (V1 only)

Menu items are chosen either by positioning the cursor under the desired item (by typing "space" characters) and entering a Return, or by typing the upper-case character in the menu field (e.g., type "D" to choose data terminal dialing or "O" to move to the options sub-menu). Once a user has entered Command Mode, the terminal is considered off-hook and busy to incoming calls until it returns to the idle mode.

<Data call> or <place Data call>:

Refer to the "Data Terminal Dialing" feature description for information on how to dial from your terminal. Once a data call has been set up, either the "Disconnect/Recall Sequence" (see the Permissible Options, Table 2-F) must be sent or <Hangup> selected from the above menu to terminate the data call and return to idle mode. If a data call is not answered, the caller must disconnect by sending a Break.

<Voice call> (V2 only):

The data terminal user can originate a call for an on-premises *source* voice or data terminal to a remote terminal by selecting <Voice call> and dialing the required digits. Refer to the "Third-Party Call Setup" feature description for a complete description of this feature.

<Options>:

If the Command Mode menu item <Options> is chosen, the terminal displays the data port's administered options. System default values for each option are also shown. The display shown below is similar to what is actually presented on the screen. (See Tables 2-F and 2-G for additional information on options.)

OPTIONS	CURRENT	DEFAULT
Speed (highest)	9600	19200 (Auto)
Parity	Even	Even
Mismatch	Yes	No
Local Echo	Yes	Yes
Answer Text	Yes	Yes
Connect Indication	Yes	Yes
Recall Sequence	Br-Br	Br-Br

<eXit> **<Change options>** **<View options>**

In V1 systems, only the System Administrator can change options. In V2 systems, the System Administrator can still perform the task or can authorize data terminal users to change their own options. Selection of <Change options> from the sub-menu shown above allows the data terminal user to change the values in the CURRENT column. Refer to the "User Changeable Options" subsection for complete information on this feature.

<Hangup>:

The <Hangup> option can be used to terminate a connection to the data port. For V2 systems only, this option is needed for AT&T STARLAN NETWORK endpoints to disconnect from a STARLAN Interface CP through Command Mode. Accessing <Hangup> provides the user with these options:

- <eXit>— Do not hang up. Return to the top level of Command Mode.
- <All>—Hang up (disconnected)

TABLE 2-F. Partial List of Permissible Data Port (TN726) Options

OPTION	DEFINITION
Speed (61-68)†	Autobaud, Low *, 300, 1200, 2400, 4800, 9600, 19200
Parity (69)	Odd, Even, 0, 1. The 0 and 1 choices are not shown on the user's display.
Enable Command Mode (70)	Yes or No. Must be On for Command Mode (i.e., Command Mode Menu display). Not shown on user's options display.
V2 only: Allow user to change data port options? (71)	Yes or No. Used to enable/disable User Changeable Options feature. Not shown on user's options display.
Permit Mismatch (72)	Yes or No. Allows two data endpoints to communicate at different rates.
Local Echo (73)	Yes or No. Determines whether characters from the data equipment will be echoed by System 25 during Command Mode.
Answer Text (75)	Yes or No. Enables call progress messages to be displayed at the <i>called</i> data endpoint.
Connected Indication (77)	Yes or No. Yes indicates that users who have Command Mode enabled will receive the "CONNECTED" message when a connection has been established. If Command Mode is disabled, the Data Line port control lead will be "raised" when a connection is established. Usually set to "No" for host computer endpoints.
Disconnect/Recall Sequence (74)	One Long Break or Two Short Breaks; the sequence used to disconnect a data call.

* A terminal whose baud rate is low cannot use the Command Mode feature. Call origination at this terminal must be via Transfer To Data.

† Numbers in () indicate the action numbers used to administer data port options.

TABLE 2-G. Typical Option Profiles for Different Types of Data Port Endpoints

Options	Data Term, or PC	Host Computer	Modem (users)	Modem (computer)
Speed (highest)	19200 (Auto)	9600 *	Modem Speeds	Modem Speeds
Parity	Even	Even	Even	Even
Enable Command Mode	Yes	Yes	Yes	Yes
Permit Mismatch	Yes	No	Yes	No
Local Echo	Yes	No	Yes	No
Answer Text	Yes	No	Yes	No
Connected Indication	Yes	No	Yes	No
Disconnect/Recall Sequence	Br-Br	Br-Br	Br-Br	Br-Br

* or 9600(Auto) if **not** not used primarily for Host-Host communications

COMMUNICATIONS ACCESS MANAGER (CAM) (V2)

The CAM program facilitates communication between workstations on the AT&T STARLAN NETWORK (STARLAN NETWORK) and workstations on System 25. Detailed procedures for using CAM can be found in the CAM User Guide. The material here provides a brief overview of CAM capabilities.

CAM is an MS-DOS application program that provides an enhanced calling interface and terminal emulation for PCs. CAM, combined with System 25's Third-Party Call Setup feature, provides the capabilities of an integrated voice /data workstation, specifically:

- A 200-entry directory for automatic dialing of voice and data calls
- VT100 terminal emulation with:
 - file transfer with error checking
 - unattended remote access operation with mail
- On-line HELP that is accessible from almost anywhere within the program.

CAM runs on the AT&T PC6300 or compatible PC with at least 384K bytes of memory, running MS-DOS Version 2.0 or later (when connected to System 25) or MS-DOS Version 3.1 or later (when connected to the STARLAN NETWORK).

The PC running CAM can be connected to System 25 in one of two ways (Figure 2-18):

1. By the PC's RS-232 COM port to the System 25 via an ADU/DLC connection
2. As a STARLAN NETWORK workstation to the System 25 via the STARLAN Interface CP.

CAM interfaces with System 25's Command Mode to provide call control. The Third-Party Call Setup feature provides voice call origination.

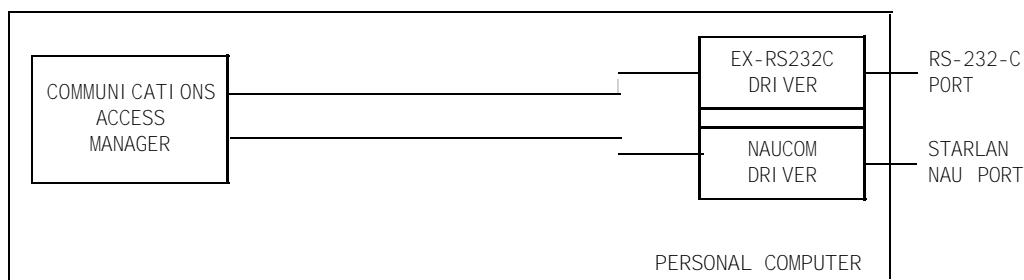


Figure 2-18. Communications Access Manager Architecture

The STARLAN NETWORK communication driver (NAUCOM) is used before CAM is run on a STARLAN NETWORK workstation. The Extended Device driver (CAM232) is used when CAM is run on a PC connected to a System 25 DLC port.

The default screen presented when the user accesses CAM is the phone directory screen. The phone screen is divided into five partitions:

- Call Appearance area—provides call appearance for voice lines and data lines for each extension shown. A call timer for each line is also displayed.
- Feature Selection area—allows the user to select the voice or data line to be used, initiate the call, and start the timer by function keys. Additional function keys may be assigned to repertory dialing numbers.
- Personal Directory area—holds a maximum of 200 entries, displayed 10 entries at a time. Each screen is arranged alphabetically.
- Message and Status area—contains prompts and messages for the user for the action being executed.
- Command Line area—contains commands available to the user for the area being worked in. Commands are executed when the user presses the <ALT> key and the first letter of the command.

The user may access the following commands:

- Data mode—provides the user with the terminal emulation screen.
- Edit—provides the user with the directory edit screen. Allows the user to add, modify, and erase directory entries, group names, and feature functions.
Directory entries contain name, number with auto login script, comment, group, and voice/data fields. Data entries also have parameter setup, a screen with fields for speed, parity, permit mismatch, and number of bits. The parameter setup allows speeds of 2400, 4800, 9600, and Autobaud.
Find—allows the user to search directory entries by name or group ID.
- Restore—displays the first 10 entries of the directory after a Find.
- Print—prints the entire contents of the directory on device LPT1.
- Setup—provides the user with the setup screen. Allows the user to view or change the following options: communications port, printer port, speed, parity, character size, return key code, autotimer, flow control, extension numbers, remote access enable, remote access password, and remote greeting.

CONFERENCE

Description

Allows up to five parties, including the conference originator, to participate in a conference call. Any voice terminal user, including operators at Direct Trunk Attendant Consoles and Switched Loop Attendant Consoles (V2), can set up conferences. Refer to the description of “(‘conference Drop” for additional information on conferencing.

Multiline Voice Terminals:

Multiline voice terminal users can add another (external or internal) party to an existing call by pressing the CONFERENCE button. This places the first party on Special Hold (indicated by a broken flutter on the line appearance button) and the system selects an idle SYSTEM ACCESS or LOOP (Switched Loop Console) button and provides system dial tone. The user may dial the desired number or select another facility to dial the party to be conferenced-in. Subsequently, pressing the held line button completes the conference. If the facility to be added is busy or has invoked Exclusion, the conference will be denied.

Users can conference up to two outside facilities (trunks), and up to five parties in all. Any attempt to add a sixth party will be denied, and the sixth party will be dropped. This limit is for the conference as a whole. Other conference inside stations are also prohibited from adding a third outside party or sixth party.

Single-Line Voice Terminals:

The single-line voice terminal user can establish a conference by momentarily pressing the switchhook, which puts the first party on hold, receiving Recall Dial Tone, and then dialing a second party. After connection to the second party, another press of the switchhook establishes the conference. A third press of the switchhook will drop the second party, restoring the original call. The user cannot put a conference that he/she has established on hold. Other internal conferees (multiline or single-line) may then add additional parties to the conference up to the five party/two outside line maximum.

Considerations

The Conference feature allows any voice terminal user to set up conference calls. Non-attendant users do not need the assistance of the attendant.

Waiting for an added party to answer and announcing the purpose of the call before adding the party to the conference is good operating practice.

Interactions

- **Account Code Entry:** If more than one user attempts to associate an account code with a Conference Call, the first to activate the feature will prevail.
- **Attendant Message Waiting:** Pressing the Attendant Message Waiting (ATT MSG) button while on a conference call will be ignored.

- **Call Park:** Call park may be used to place a conference on hold. Parked conference calls do not return to the parking station (they remain parked).

If a 5-person conference is parked, the conferee who parked the conference will be dropped when someone picks up the parked conference.

- **Exclusion:** Exclusion may be invoked before establishing a conference. If it is invoked after the conference is established, all internal conferees will be dropped.
- **Extended Stations:** An Extended Station counts as one of the two allowable outside lines on a conference call.
- **Music-On-Hold:** For V1 systems, an outside line placed on hold when CONFERENCE is pressed will hear Music-On-Hold, if provided. For V2 systems, Music-On-Hold may be enabled or disabled for “Special Hold” through a new System Administration item. However, if the outside line is already part of a conference, music is not heard.
- **Off-Premises Stations (OPS):** For conference purposes, an OPS counts as one of the two allowable outside lines.
- **Paging System Access:** A paging zone may not be conference.
- **Trunk-To-Trunk Transfer:** Trunk-to-trunk transfers may be set up using the Conference feature. The conference must include an incoming trunk call on either a ground start, DID, or tie trunk if it is to continue after all inside stations have dropped off.

Administration Requirements

None

Hardware Requirements

None

CONFERENCE DROP

Description

Allows a voice terminal user, except for the attendant at a Switched Loop Attendant Console (SLAC, V2 only), to selectively drop a previously added party from a conference call. At a SLAC, the attendant can drop conferees only before they have been added to conference.

Multiline Voice Terminals (except SLAC):

On a multiline voice terminal, pressing the DROP button and then pressing the button appearance of a conference party drops that party from the conference.

If a station called for a conference does not answer, the conferencing user should drop the call by pressing and releasing the switchhook before returning to the conference. Otherwise, the ringing line will be added to the conference.

Switched Loop Attendant Consoles:

Once a conference has been set up and all the parties can talk to each other, the SLAC attendant cannot selectively drop a conferee. Individual members of the conference wishing to drop out must hang up. However, while still setting up a conference, the attendant can drop calls before they have been conferenced in, as follows:

- A call to an inside party rings unanswered or returns busy tone—press DROP.
- A call to an *outside* party rings unanswered or returns busy tone—press another LOOP button or RELEASE or FORCED RELEASE.
- A call to an *inside* or *outside* party is completed but the person cannot participate—press another LOOP button or RELEASE or FORCED RELEASE.

It is good operating practice to wait for the called party to answer before adding the party to a conference.

All Multiline Terminals:

If all System 25 stations hang up on a conference with two outside lines, the outside parties will remain conference (until one of them hangs up) if at least one is on a ground start, tie, or DID trunk. If not, the call will be terminated when the last inside user disconnects from the conference.

Single-Line Voice Terminals:

A single-line terminal user, after having established a three-party conference, can drop the second party and retain the first party by pressing the switchhook.

Considerations

Conference Drop allows users to conference lines appearing on their terminals and then remove them from the conference when appropriate.

A user should only drop parties that *they have added* to a conference. If a user tries to drop a party who previously added them to the conference, other parties may also be dropped.

Interactions

None

Administration Requirements

None

Hardware Requirements

None

DATA CALL SETUP

Allows a user to originate data calls from a data terminal. System 25 provides three methods of data calling:

- Dialing from a data terminal, which is described in the “Data Terminal Dialing” feature description.
- Setting up data calls from a voice terminal, which is described in the “Transfer to Data” feature description.
- V2 Only: Setting up data calls (or voice calls) for another terminal from a data terminal, which is described in the “Third-Party Call Setup” feature description.

DATA SERVICES OVERVIEW

System 25's data features provide switched data transmission at up to 19,200 bps (RS-232 interface), and a 212A modern compatible conversion resource capable of handling data at 300 and 1200 bps.

The system provides switched connections between *data endpoints*. These endpoints include data terminals, personal computers, multiport computers, and modems. Data endpoints are either digital data endpoints or analog data endpoints.

Analog endpoints are connected to System 25 voice terminal or trunk port circuits through a modem in the traditional manner. Digital endpoints are connected to System 25 data port circuits on the TN726 Data Line CP. A data module (specifically, an Asynchronous Data Unit - ADU) is required in place of the modem used with analog endpoints. Section 4 of this manual shows the connections supported and required connecting equipment.

Data calls can be set up between data endpoints. Analog to analog and digital to digital connections are straightforward; calls between analog and digital endpoints are possible only if the system is equipped with a conversion resource (TN758 Pooled Modem Circuit Pack or external modem pool). System 25 data calls from analog endpoints (including those to digital endpoints) are set up in the traditional manner. The calling party should follow the procedures supplied with his/her modem. However, a Modem Request Code must be dialed when calling a digital endpoint.

Call set-up from digital endpoints is facilitated by several data features: *Command Mode*, *Expert Mode*, *Data Terminal Dialing*, *Modem Pooling*, *Third-Party Call Setup*, and *Transfer To Data*.

In the discussion that follows, it is important to understand the difference between analog voice terminology and data terminology. Refer to the "Glossary" (Section 9).

The following provides a definition of a data call in terms of its contextual components. The components are (1) data endpoints, (2) data endpoint states, (3) data call processing modes, (4) connecting configurations, and (5) controlling features.

Data Endpoints

Data endpoints are composed of data equipment, a data module or modem, and a connection to the switch via an analog or data port. A digital data endpoint is addressed by its Data Dial Code (DDC). Analog data endpoints are addressed like other voice terminals, by their PDCs. For the remainder of this description, data endpoints will refer to digital data endpoints unless stated otherwise.

Several different categories of data endpoints are supported. The categories have been divided into two general groups, those having a DTE type interface, which encompasses almost all of the data terminal devices, and a group of DCE interface devices (primarily modems). The groups have then been divided into categories based upon their functional attributes. However, it must be noted that within each category, control interfaces may vary. The following describes the categories and attributes of each:

1. DTE Devices

This group of data endpoints have one thing in common: their interface configuration (although RS-232 control signal utilization varies significantly from terminal to terminal). Some data equipment do not use any RS-232 control signals; these require only BA (Transmitted Data Ready-Tx), BB (Received Data Ready-Rx)

and AB (Signal Ground) to function, while others require more RS-232 control signals to operate. An ADU (Figure 2-19) can send Data Terminal Ready (DTR) from the data terminal to the Data Line circuit and the Data Line circuit can send a control signal to the data terminal. The signals Data Set Ready (DSR), Clear To Send (CTS), and Received Line Signal Detector (DCD) are all connected to the control signal from the DLC in the ADU and available if required by the data terminal. Refer to Section 5 (Port Specifications) for additional information.

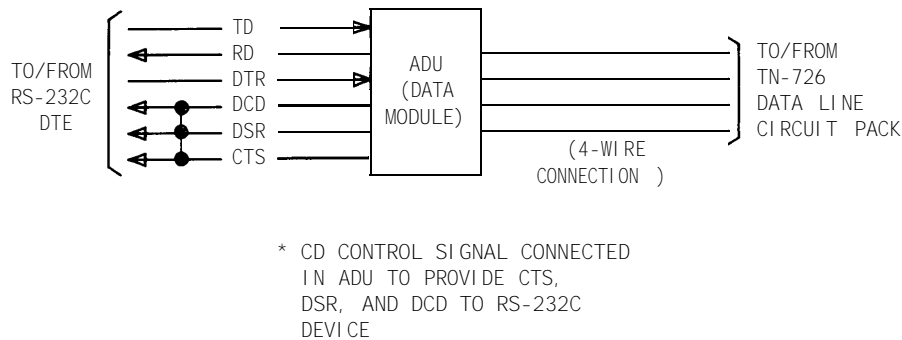


Figure 2-19. Asynchronous Data Unit Interface Signals

The following categories are part of the DTE data endpoint group:

a. Data Terminal Without ASCII Keyboard

This category includes such devices as Fax machines, EBCDIC or Baudot terminals, and receive only devices such as printers.

Once connected to an ADU and turned on, these data endpoints appear on-line, available, and ready to enter the Setup mode on auto-answered calls (modes are described below). These endpoints will display or print information received after a valid connection has been established without additional RS-232 control from that endpoint. Note that since these endpoints cannot establish calls for themselves, they must either be called by other endpoints *or* have calls established for them via the Transfer to Data or Third-Party Call Setup (V2) feature.

b. ASCII Data Terminal With Keyboard

This category can be subdivided into two classes: (1) basic terminals, and (2) intelligent, programmable data equipment such as personal computers. Basic terminals appear to the data port to be on-line and available whenever they are turned on, thus ready to enter either the

Data Terminal Dialing mode, or to enter the Setup mode on calls originating from a voice terminal or on auto-answered calls.

In the case of most personal computers, a communications program must be executed in order for it to communicate with its own RS-232 port or built-in modem. Once the communications software is running, further operation will be similar to that of the basic terminal.

c. Host Computer Endpoint

A host computer endpoint is very similar to a data endpoint with keyboard except that the host has many ports and the interface is usually capable of supporting multiple speeds and more of the RS-232 control signals. Front-end communication software running in the host is typically supplied by the computer vendor and is not designed to support the Data Terminal Dialing feature. Such software typically supports call origination through Automatic Calling Units (ACUs), which are not compatible with Data Terminal Dialing. Thus, the primary means of communicating with the host is by calling from data terminals or personal computers. Groups of host ports with matching characteristics may be members of hunt groups (referred to as host port groups).

d. Analog Data Endpoint

Data endpoints with modems are referred to as analog data endpoints. Modems connected via tip ring lines use PDCs as extension numbers rather than Data Dial Codes (DDCs). Station-to-station data calls to (or from) this endpoint from (or to) digital endpoints require a modem conversion resource to convert the endpoint's analog data to digital format. Calls from a digital endpoint to an analog data endpoint (i.e., calls to a PDC), will automatically have a conversion resource inserted in the calling path. If the called (analog) endpoint should then invoke Transfer To Data, the conversion resource will be released. Data calls originating from an analog data endpoint must first enter a Modem Request Code before addressing a digital data endpoint. This is required because the system assumes that a call originating from a voice terminal will invoke Transfer To Data. If the analog data endpoint is not going to transfer to data, they must indicate this so that a conversion resource will be included in the connection.

2. DCE Devices

This group of data endpoints consists primarily of modems. The modems are connected to a data port from their RS-232 side. The data module must be configured as a DTE interface to provide connectivity between the modem and a data port. It is possible to simulate a DTE interface from an ADU data module with a cross-over ("null modem") cable. This group of endpoints is important for users who provide their own modems, connected to dedicated trunks or private lines, for internal modem pooling.

Data Endpoint States

The data endpoint may take on three states: (1) off-line, (2) on-line (on-hook), and (3) on-line (off-hook). Off-line is when a data terminal is out of service (turned off, disconnected, etc.). The on-line (on-hook) state occurs when the terminal is turned on, is available to answer a call, but is not on a data call. Finally, the on-line (off-hook) state is when the data endpoint is actively on a data call.

Data Call Processing Modes

Data calls differ both in signaling and call setup from voice calls. For this reason, a unique set of data call processing modes have been defined to support data call operation in a manner consistent with the characteristics of data terminals.

a. Off-Line Mode

The data endpoint is considered to be in the Off-Line Mode whenever the data endpoint's DTR signal is inactive (e.g., "turned off"). The endpoint is considered unavailable and calls to this endpoint will receive the "RINGING" message or Ringback (indefinitely).

b. Idle Mode

The Idle Mode indicates that the data endpoint is in its on-line, on-hook state. While idle, call processing will allow the endpoint to:

- Enter either Data Terminal Dialing mode to originate a data call, or enter the Setup mode after a call is originated from a voice terminal (Transfer To Data) or other data terminal (Third-Party Call Setup; V2 only).
- Autoanswer a data call and go into Setup mode.

The data endpoint remains in the Idle mode while the user is establishing a data call from a voice terminal until Transfer To Data is activated.

c. Command Mode

Command Mode enables the Data Terminal Dialing feature, allows the user to view and change (V2 only) associated data port options, and provides access to the Third-party Call Setup feature (V2). Command Mode may be entered by going on-line and pressing Break or Break-Return.

- d .Expert Mode is an enhancement to the Command Mode feature that provides an alternative method of performing the full range of Command Mode functions. By eliminating the display of menus and allowing multiple commands to be entered on a single line, Expert Mode lends itself to computer-driven instructions. Individual users who are very familiar with Command Mode operations may also find it useful.

e. Data Terminal Dialing Mode

Data Terminal Dialing is a data feature accessed via Command Mode. It provides a procedure to establish data calls without the use of a voice terminal. Data Terminal Dialing supports both on-premises and off-premises data calls (with the support of the System 25 Modem Pooling feature). Dialed digits are entered from the data terminal keyboard or host computer (using a program compatible with Data Terminal Dialing protocol). Call progress text messages are sent to the terminal in place of call progress tones. Upon completion of digit entry, Data Call

Setup mode is entered.

f. Data Call Setup Mode

Data Call Setup Mode is a transitional state entered after Transfer To Data, Data Terminal Dialing, or during auto-answer; it exists during the handshake between data ports.

If the endpoints are compatible and handshaking is successful, a data connection is established. If handshake failure occurs, the user is notified and the data endpoint returns to the Idle mode. Successful handshake must occur within 15 seconds of answer at the called data endpoint. This implies that the voice terminal user must invoke Transfer To Data within 15 seconds after far-end answer. Similarly, if an originating voice user calls a voice terminal and both users transfer to data, both ends must transfer within the 15 second time limit.

If the data endpoint is optioned for Command Mode permission, the data endpoint will receive call progress text messages while in the Data Call Setup mode.

g. Data Mode

Data Mode is first entered after successful completion of Data Call Setup. Transparent communication between connected endpoints is provided in Data Mode.

Connecting Configurations

Refer to “Connectivity” in Section 4 for data equipment connections.

Controlling Features

It is possible to originate data calls from a voice terminal with a Transfer to Data button or from data endpoints that support Command Mode (i.e., ASCII data terminals with keyboards and host computers). Several controlling features are provided to allow data endpoints and voice terminals to set up data calls. The following briefly describes the Data Service features used in controlling data calls:

a. Command Mode/Expert Mode

Command Mode and Expert Mode provide an interface to the Data Terminal Dialing feature, the Third-Party Call Setup feature (V2), and permits users to display and change (V2 only) data port options.

b. Data Terminal Dialing

Data Terminal Dialing provides call setup from terminals and host computers.

c. Transfer To Data

Transfer To Data is the preferred method of data call origination from multiline voice terminals equipped with Transfer to Data (DATA) buttons and associated digital data endpoints. The DATA button is associated by DDC with a near end data endpoint. A unique DATA button must be provided for each DDC that the voice terminal is capable of controlling. Associated with each DATA button is an LED that reflects the status of data endpoints as follows:

- Dark–Data endpoint is idle
- Winking–Data endpoint is reserved (preindicated)

- Flashing–Data endpoint is being alerted of an incoming call
- On Steady–Data endpoint is either in the on-line (off-hook) state or is reserved for another user and busy.

Refer to the following feature descriptions for additional information:

- Command Mode
- Data Call Setup
- Data Terminal Dialing
- Expert Mode
- Modem Pooling
- Third-Party Call Setup.
- Transfer To Data

DATA TERMINAL DIALING

Description

When the user makes a data call from a data terminal, voice terminal dialing and call progress tones are replaced by keyboard dialing and call progress messages. The message “DIAL:” prompts the user to enter the called number from the keyboard, and “RINGING” or “DIALING . . . COMPLETED” informs the user that the dialed number is being called.

Table 2-H provides a list of the call progress messages.

Note: The following dialing procedures assume that “Command Mode” is active. Refer to the subsection on “Expert Mode” for an alternative method of dialing.

Dialed Characters:

In addition to digits and the “#” and “*” characters on the touch-tone pad, the dialed number may contain the following special characters:

- The characters “(” “)” “-” and SPACE may be used to improve legibility. These characters are ignored.
- The “%” or “,” characters may be used to cause a 1.5-second pause in dialing. Multiple pause characters can be used to produce longer pauses.
- The “\$” (mark) character indicates that the remaining digits are for end-to-end signaling.
- UNDERSCORE or BACKSPACE characters may be used to correct previously typed characters on the same line.
- The “@” character may be used to delete the entire line and start over with a new DIAL: prompt.

Each line of dialing information may contain up to 27 characters. Note that all of the dialing information, including pauses and ignored characters, must be typed on a single line following the DIAL: prompt and terminated by Return.

Dialing Correction:

The backspace character (BS key or Ctrl-H keys) or underscore (“_”) may be used to cancel the previously entered character. More than one entered character may be deleted by using multiple backspace or underscore characters. The “G” character may be used to delete the entire line of entered characters.

Pause:

To assist the completion of off-premises calls, the pause characters “%” or “,” may be used. A pause character may be used to help ensure the receipt of dial tone before continuing to dial. Each “%” or “,” causes a fixed delay of one and one-half (1.5) seconds. Pause characters may be used consecutively if a longer pause is required. Note that System 25 cannot detect tones, such as a second dial tone for end-to-end signaling.

End-to-End Signaling:

Data connections to off-premises destinations require that a conversion resource (pooled modem) be inserted into the connection. Occasionally it is necessary to send additional tones to the remote endpoint after the connection is established to signal the remote equipment. A “mark” character must be included on the DIAL: line to indicate to call processing that the remaining digits are to be sent to the far end prior to insertion of the conversion resource into the connection. The “mark” character marks the boundary between the digits dialed to reach a distant endpoint, and the digits used by that distant endpoint after it has answered. Pause characters may and usually should follow a “mark” character. An example using a “mark” character and several pause characters is shown below. Dashes are included for readability.

Examples of dialing are as follows:

- DIAL: 3478
- DIAL: 9-1-(201)-946 -8123,, \$,5678
- DIAL: 9%946-8123%%\$%5678%137%110

Call Disposition:

Call progress messages corresponding to call progress tones provided to voice terminals are listed in Table 2-H. The message supplied (indicating reorder, busy, ringback) depends on the disposition of the call.

1. When ringback is received the displayed message is “RINGING” (internal calls only). For outside calls, the corresponding call progress message is “DIALING”.
2. If the endpoint answers, the displayed message is “ANSWERED” (internal calls only). Then, if the handshake succeeds, a data connection is established. For outside calls, when the system has finished dialing, the message “COMPLETED” is displayed.
3. If the handshake fails because a connection cannot be established between endpoints (e.g., a port optioned at 9600 baud attempts to talk to a conversion resource that can only talk at 300 or 1200 baud), the user receives “INCOMPATIBLE FAR END,” “DISCONNECTED,” and the data endpoint goes on-hook.

If the far end does not answer, the caller must press Break to terminate the call attempt.

If the disposition of the call is such that TRY AGAIN or BUSY (indicating reorder or intercept and busy respectively) is received, the switch sends “DISCONNECTED” to the data terminal that goes on-hook.

Answering Endpoint:

When the dialed end point is alerted, the user receives “INCOMING CALL-”. (The called terminal will auto-answer if it is turned on.) If the handshake succeeds, a data connection is established and the “CONNECTED” message is displayed if so optioned. If the handshake fails, the user receives “INCOMPATIBLE FAR END, DISCONNECTED” and the data endpoint goes on-hook.

TABLE 2-H. Call Progress Messages for Data Terminal Dialing

Displayed Message	Application	Meaning
DIAL:	Placing a call	Equivalent to dial tone. Enter any required facility number followed by the dialed number and a RETURN.
RINGING	Placing a call	Equivalent to Ringback Tone. Called number (far-end) is being signaled. Provided on internal calls only.
BUSY	Placing a call	Equivalent to busy tone. Called number is in use, or out of service. Provided on internal calls only.
ANSWERED	Placing or receiving a call	Notifies calling and called users that call has been answered. Provided on internal calls only.
TRY AGAIN	Placing a call	Equivalent to Reorder Tone. System facilities are currently not available or invalid number.
INCOMING CALL-*	Receiving a call	Equivalent to ringing.
PLEASE ANS-	Placing a call from a voice terminal	Originating voice terminal user has transferred call to data terminal using Transfer to Data.
DISCONNECTED*	Call is terminated	Call or call attempt is disconnected.
CONNECTED, SPEED = NNNN	Call is connected	Notifies user that the call connection is established and what the baud rate is. [Provided that "Connection indication" (Data Port Action 77) is enabled.]
INCOMPATIBLE FAR END	Placing a call	Notifies user that the handshake between data end points has failed
DIALING . . COMPLETED	Placing a call	Indicates off-premises call is being dialed and that dialing is completed.
PLEASE WAIT	Placing a call	Call queued.
BAD NUMBER	Placing a call	Bad dialed number
NO MODEM	Placing a call	No modem available for a call that requires one.
(V2) SESSION 1	Placing or terminating a call	Specifies the session number (1) of the data call to the calling party

* Bell sounds when message is displayed.

Considerations

Data Terminal Dialing allows users to place data calls from their terminals using the Data Terminal Dialing feature and allows users to review the options administered for their data ports.

Interactions

- **End-To-End Signaling:** (See above text)
- **Modem Pooling:** Data calls between analog and digital endpoints require that a conversion resource (TN758) be available. If one is not, the “NO MODEM” followed by “TRY AGAIN” message will be displayed.
- **Speed Dialing:** System Speed Dial codes can be dialed from data terminals.

Administration Requirements

Data Port: See the table of Permissible Data Port Options in the “Command Mode” feature description.

Hardware Requirements

TN726 Data Line CP to support each digital endpoint.

TN758 Pooled Modem CP to support data calls between digital and analog endpoints.

DIAL PLAN

The dialing plan for System 25 is based on the concept that, whenever possible, calls should be placed to individuals rather than to voice terminals. To implement this concept, individuals are assigned Personal Dial Codes (PDCs) and are allowed to log in those PDCs at other voice terminals. There are two types of PDCs: assigned and floating. An assigned PDC is associated with each voice terminal. Floating PDCs are administered at the SAT and may (optionally) be associated with the attendant position when not logged in.

Data extensions on System 25 are assigned Data Dial Codes (DDCs).

Dial Code Assignments

System 25 dial codes are as follows:

Assignable System 25 dial codes may have 1, 2, 3, or 4 digits. These include voice terminal PDCs, data terminal DDCs, Direct Group Calling (DGC) Groups, Paging Access, Attendant Call Park, Night Service, Modem Request, Automatic Route Selection Access, Facility Access (trunk group), and Dictation System Access codes.

System 25 fixed dial codes are:

- 0—Attendant access
- *1, *2, *3—Reserved for maintenance calls
- *4—Activate Make-Busy for DGC group member
- *5—Call Park
- *6—Deactivate Make-Busy for DGC group member
- *7 + 0—Group Call Pickup Answer
- *7 + PDC—Directed Call Pickup Answer
- *8 + PDC—Call Park Retrieval
- *9—Camped on Call Retrieval
- *0—Account Code Entry from single-line voice terminals
- ** PDCPDC— Login PDC (Call Following)
- **PDC0—Logout PDC (Call Following)
- **0—Logout all PDCs (Call Following)
- ## PDC-Call Accountability
- *—PAUSE character used in programmed numbers
- #100-#189— System Speed Dial Codes
- #190-#199—Virtual Facility Codes (V2)
- #20-#26—Personal Speed Dial Codes
- #3—Start end-to-end signaling in programmed numbers (V2 only)
- #4—Activate Program mode

- #5–Insert dialed digits here (in Virtual Facility numbers; (V2 only)
- #–End of dialing

The maximum number of dial codes available for a System 25 is 600. In R1V1, each assigned code is allocated as a ten number block. In R1V2, each assigned code is stored individually in memory.

The dial codes assigned in the system must be completely unambiguous. For example, a dialing plan that contains the number “20” cannot contain the numbers “2,” “200-209,” or “2000-2099.”

PDC to Voice Terminal Association

During installation, each voice terminal is assigned one PDC that serves as its extension number. These are termed assigned PDCs, and the associated terminals are called home stations. Additional PDCs may be assigned in a system. These PDCs are termed “floating” PDCs (FPDCs). At the customer’s option, floating PDCs may have the attendant position assigned as their home station (i.e., calls to FPDCs will be directed to the attendant when they are not logged in elsewhere). A maximum of 200 assigned PDCs and 300 FPDCs may be allocated in a system.

Data Dial Codes (DDCs)

At the time of installation each digital data endpoint will be assigned a Data Dial Code (DDC) that serves as its extension number. A maximum of 104 DDCs may be allocated in a system.

Direct Inward Dial (DID) Number Assignments

Each DID number is associated with a unique PDC (floating or assigned), a DGC access code, a DDC, or a pooled facility access code. The code associated with a DID number is the last 2, 3, or 4 digits of the DID number. For example, the code associated with the DID number “NXX-2157” will be 57, 157, or 2157.

All dial codes in the system that are associated with DID numbers should have the same number of digits. However, there is no requirement that all PDCs, DDCs, DGC access codes, or facility access codes be associated with DID numbers.

Voice Terminal Directed Features

Directed Night Service, DGC calls, Personal Line Calls, Manual Signaling, Station Message Waiting, Automatic Intercom, and Outward/Toll Restriction are associated with specific terminals (stations), not with PDCs. This means that these features do not move with a PDC when it is logged in at another voice terminal. In R1V1 systems, Direct Station Selection (DSS or FLEX DSS) calls do not receive Call Following treatment.

DICTATION SYSTEM ACCESS

Description

Permits access to, and control of customer-owned dictation equipment by voice terminal users. Dictation systems may be connected either via single-line voice terminal ports (the preferred method) or via auxiliary trunk ports (if the dictation equipment requires a separate contact closure for proper operation). The dictation system is accessed by dialing a PDC oppressing a DSS button.

Considerations

Allows users to access and control shared dictation equipment.

Interactions

- **Direct Inward Dialing (DID):** A DID number may match the dictation system access code. This allows an outside caller to access the dictation equipment.

Administration Requirements

System:

- Requires a port assignment on a ZTN78 Tip Ring Line or TN742 Analog Line CP. A port on a TN763 Auxiliary Trunk CP must be used if the dictation equipment requires a separate contact closure for proper operation.

Voice Terminal: (Station Port)

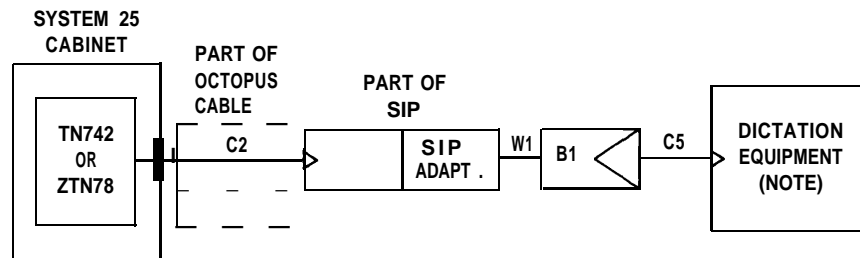
- Assign DSS access buttons as desired.

Hardware Requirements

Customer provided dictation equipment

Port on a ZTN78, TN742, or TN763 CP, as required. If the equipment requires a contact closure, the TN763 must be used.

Detailed connection information is provided in Figure 2-20. Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters are provided under the heading "Connectivity" in Section 4 of this manual.



LEGEND:

- TN742 - ANALOG LINE CP
- ZTN78 - TIP RING LINE CP
- B 1- TYPICAL-103A CONNECTING BLOCK*
- C 2- OCTOPUS CABLE (WP90780) PEC 2720-05P
- C 5- MODULAR CORD (D4BU-87)
- W1 - 4 PAIR INSIDE WIRING CABLE*
- * - FURNISHED BY INSTALLER

NOTE : IF CUSTOMER DICTATION EQUIPMENT REQUIRES A CONTACT CLOSURE, A TN763 AUXILIARY TRUNK CP MUST BE USED. REFER TO "PAGING SYSTEM USING AUXILIARY TRUNK CP" FOR TYPICAL CONNECTION INFORMATION.

Figure 2-20. Dictation System Connections (FCC Registered)

DIGITAL TAPE UNIT (DTU)

The Digital Tape Unit (Figure 2-21) is a RS-232 device used to record administration translations. The DTU does not encode the translations data as it records, nor does it require decoding circuitry when playing back (restoring) recorded data. Data is recorded and transmitted at 1200 bps.

The DTU requires 115V commercial power from a 3-wire grounded outlet. It should be located on a desk or table top. The recorder is approximately 5 inches wide, 2 inches high, and 10 inches long.

As shown in Figure 2-22, the DTU must be directly connected to port #3 on the Call Processor (ZTN82 or ZTN128) CP. Remote and switched connections are not supported. Maximum cabling distances are provided in Section 5, "Technical Specifications."

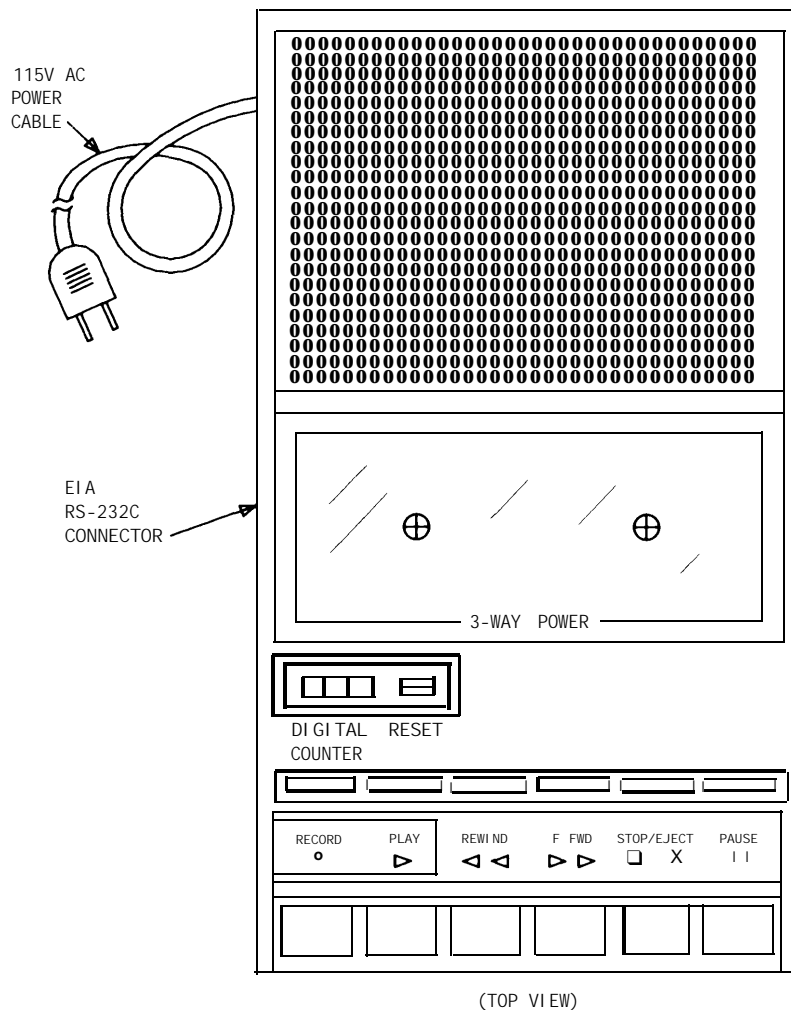
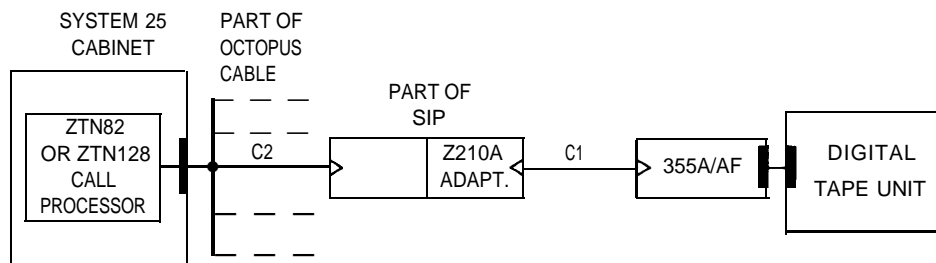


Figure 2-21. Digital Tape Unit



LEGEND:

- C1 - MODULAR CORD (D8W-87) - PEC 2725-07G**
- C 2- OCTOPUS CABLE (WP90780) - PEC 2720-05P**
- 355A ADAPTER RS 232 PLUG TO MODULAR JACK - PEC 2750-A24**
- 355AF ADAPTER RS 232 RECEPTACLE TO MODULAR JACK - PEC 2750-A25**

Figure 2-22. Digital Tape Unit—On-Premises Direct Connections (Sharing Same AC Outlet)

DIRECT GROUP CALLING (DGC)

Description

Allows incoming calls to be directed to a specific group of terminals. Up to 32 DGC groups, each including up to 10 members, may be set up. A terminal can be in only one DGC group. Incoming calls on any trunk can be directed to a DGC group. These trunks can also be used for outgoing calls. This feature is referred to as “Incoming Calls Group” in the *User Guides* for System 25 voice terminals.

DGC groups can be administered as Call Coverage groups in R1V2 systems. Refer to the “Call Coverage–Group” feature description for details.

Calls to a DGC group hunt in a circular manner, starting at the terminal following the last one to receive ringing (whether answered or not), and will ring at the next idle terminal in the group. On multiline voice terminals the calls arrive on a SYSTEM ACCESS button (LOOP button on a Switched Loop Attendant Console; V2 only).

If all group members are busy (off-hook), an outside call is queued and the caller receives ringback tone. If the system includes a delay announcement, it is played after a specified number of rings. The caller is subsequently put on hold (in queue) and will receive Music-On-Hold if available. If the system is not equipped with a delay announcement, the call will begin to ring at all line appearances after the specified interval.

An inside caller dials a DGC access code to reach a DGC group. If all members of the group are busy, the call will not go into a queue and the caller will receive Busy Tone.

Once the call begins to ring at a group member’s station, it will **not** receive announcement service or ring at a line appearance. For this reason, it is important that DGC members log out (as described below) when they will be away from their desks.

For R1V2 systems, the attendant can camp-on multiple outside (trunk) calls when all members of the group are busy. Group members do not receive camp-on indication. The camped-on calls will be queued, and are eligible for the DGC delay announcement. If no delay announcement is available, the calls will return to the attendant console after a specified number of rings.

For R1V1 systems, the attendant can camp-on only one outside (trunk) call when all members of the group are busy (assuming Attendant Camp-On is activated). Group members do not receive camp-on indication.

DGC group members may withdraw from the group by going off-hook and dialing *4. To reenter the group, the member goes off-hook and dials *6.

An off-hook multiline terminal or attendant console (even if busy on only one SYSTEM ACCESS or LOOP button) appears busy to DGC calls. However, terminals other than the SLAC (V2) may receive other (non-DGC) calls while active on a DGC call.

Direct Group Calling groups may be used for data applications to access host ports and the STARLAN Interface CP (V2). The System Administrator may disable queuing for data DGC groups, if desired (V2). Delay announcements and music-on-hold are not provided for data groups.

Considerations

DGC groups are particularly useful when the answering group receives a high volume of calls. Call completion time is minimized and attendant assistance is not required.

Any number of outside trunks may be administered to feed into a DGC group. A trunk may feed only one DGC group.

Interactions

- **Attendant Camp-On:** For V1 systems only: if the attendant attempts to camp-on a second call to a DGC group, it is immediately returned on the console.
- **Attendant Direct Extension Selection:** When all stations in a DGC group are busy, the status LED on the Selector Console lights steadily.
- **Call Coverage:** When a call rings at DGC station that has Call Coverage, the call will receive that station's coverage. Calls directed to a busy DGC group do not receive call coverage. Instead, after a predefined number of rings, the call will be transferred to a delay announcement (if provided), or ringing will be transferred to all button appearances of the line and the SLAC queue (if trunk has ringing enabled; V2).
- **Call Pickup:** A DGC group member can also be a member of a Call Pickup group.
- **Direct Group Calling Delay Announcement:** Provides a recorded announcement to an outside caller who has been placed in queue for a DGC group.
- **Direct Inward Dialing:** An incoming DID call may match a DGC group access code.
- **Direct Station Selection (DSS):** A DSS button can be assigned to a DGC group. The associated LED lights steadily when all stations in the group are busy.
- **Modem Pooling:** Modem Pooling supports calls to data endpoints that are part of a DGC group. While an incoming data call is in a DGC group queue, the caller receives ringing. The conversion resource is inserted if the call is completed to a digital endpoint.
- **Personal Lines:** An outside line directed to a DGC group can be assigned button appearances in addition to the DGC group assignment. When an incoming call is ringing at a DGC group, the status LED on the voice terminal button appearance lights steadily, indicating that the line is busy. If the call goes unanswered for a pre-determined number of rings (and no delay announcement is provided), ringing will be transferred to all button appearances of the line and the status LED will flash.
- **Station Message Detail Recording (SMDR):** For an incoming call to a DGC group that was connected to an announcement but was never answered, "O" will be reported in the "STN" field of the call record. If the call was answered by a station after receiving the announcement, that station will be listed in the "STN" field.
- **Tie Trunks:** Calls to a busy DGC group via auto-in tie trunks will be queued, but will not receive a delay announcement. Calls to a busy DGC group via dial-in tie trunks will not be queued; these callers will receive Busy Tone.

- **Transfer:** For V2 systems only: internal stations can transfer outside (trunk) calls to a busy DGC group. The transferred call will be treated as any other trunk call to a busy DGC group.

Administration Requirements

- Trunks - Assign trunks to DGC Group
- Trunks (V2) - Assign trunks to ring in SLAC queue
- System: Number of rings before DGC calls are transferred to announcement or begin ringing at button appearances or SLAC queue (V2)
- Direct Group Calling: Assign DGC access code, Add/delete DGC members, Enable/disable queuing for data DGC groups (V2).

Hardware Requirements

None

DIRECT GROUP CALLING DELAY ANNOUNCEMENT

Description

Provides a recorded announcement to an *outside* (trunk) caller who has been placed in queue for a DGC Group.

When all members in the group are busy (off-hook), the call will be queued for DGC service and the calling party will receive ringback tone. Note that no incoming call indication (ringing) is provided to the DGC group members at this point. After a specified number of rings (administrable) a recorded announcement will be played to the calling party without disturbing his or her position in queue. The caller is subsequently placed on hold and will receive music if available.

Once a call begins to ring at a DGC station, the call is no longer eligible for delay announcement service. The call will then ring until answered, covered, picked up or abandoned.

Considerations

DGC Delay Announcements provide the calling party with a message that acknowledges their call and assures them that their call will be handled in an orderly way.

Interactions

- **Tie Trunks:** Calls to busy DGC groups via auto-in tie trunks will be queued, but will not receive the delay announcement. Calls to busy DGC groups via dial-in tie trunks will not be queued (and, hence, will not receive the delay announcement).

Administration Requirements

The DGC announcement device requires a port assignment on a ZTN78 Tip Ring Line or TN742 Analog Line CP. Only one DGC Delay Announcement may be assigned in the system. Callers to all DGC groups receive the same message.

Hardware Requirements

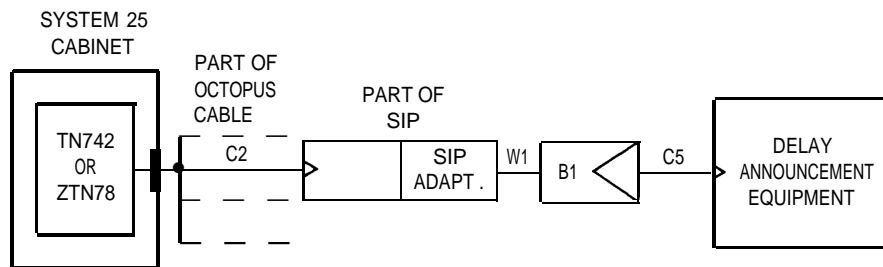
The AT&T Answer-Record 2500 or Code-A-Phone 2540 may be used as the announcement device. The announcement device *must* automatically hang up at the end of each call so that the incoming call can be returned to the DGC queue.

The equipment requires a port on a ZTN78 Tip Ring Line (or TN742 Analog Line) CP. The system supports one DGC delay announcement.

For Music-On-Hold hardware information, refer to the “Music-On-Hold” feature description.

Detailed connection information is provided in Figure 2-23.

Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters, as shown on the figures, are provided under the heading “Connectivity” in Section 4.



LEGEND:

- TN742 - ANALOG LINE CP
- ZTN78 - TIP RING CP
- B 1- TYPICAL-103A CONNECTING BLOCK*
- C 2- OCTOPUS CABLE (WP90780) PEC 2720-05P
- C 5- MODULAR CORD (D4BU-87)
- W1 - 4 PAIR INSIDE WIRING CABLE*
- * - FURNISHED BY INSTALLER

Figure 2-23. Delay Announcement Equipment Connections (FCC Registered)

DIRECT INWARD DIALING (DID)

Description

Allows incoming calls to reach specific individuals or facilities in the system without attendant assistance.

System 25 customers reserve blocks of DID numbers from the CO. The DID numbers may correspond to a PDC FPDC, DGC access code, DDC, or any facility with an access code such, as a pooled facility or a paging zone.

The system is capable of receiving either 1, 2, 3, or 4 digits over its DID trunks. The number of digits received on a specific DID trunk will be constant for that trunk; however, different DID trunks may receive different numbers of digits. The system is capable of receiving UP to four digits and then ignoring leading digits as specified to match against system dial codes. For example, the dial code matching DID number NXX-2157 can be 57, 157, or 2157. If the System 25 is administered to match on more digits than are received from the Central Office (CO), the additional leading digits are taken from the 4-digit trunk number. For example: if a call comes in on DID trunk number 1234, the CO sends two digits (77) over this trunk to identify the recipient, and System 25 is administered to match on three digits, then the call will be routed to dial code 277.

Incoming DID numbers that don't match any valid dial code may optionally be directed to the Attendant Console or to Reorder Tone.

If the DID number received is a valid dial code, the caller is provided either Ringback Tone, Busy Tone, or the tone from a pooled facility (e.g., Dial Tone) as soon as addressing is completed. Busy Tone is provided if and only if the call cannot be completed to the intended voice terminal and cannot be provided coverage.

DID calls appear at System Access buttons on multiline voice terminals (they do not have other button appearances). These calls can be transferred to a covering station, answered via Call Pickup, directed to a DGC Group, or given "call hunting" or "call following" treatment. A DID call *is not* automatically covered on the Attendant Console.

DID trunks may utilize DID Immediate Start or Wink Start protocols. Refer to Section 9, Glossary, for a brief description of each of these trunk types.

Considerations

Direct Inward Dialing frees the attendant from handling certain incoming calls.

Interactions

- **Attendant Camp-On: DID** calls are not provided Attendant Camp-On treatment. They will not appear on the Direct Trunk Attendant Console Return-On-Busy or Return-On-Don't-Answer buttons or on the Switched Loop Attendant Console Loop buttons *unless* they are first answered at the attendant position and are subsequently extended by the attendant.
- **Attendant Direct Extension Selection:** Selector Console LEDs respond to DID calls just as they do for other outside calls. When a user answers a DID call, the associated LED on the Selector Console will light steadily. When a DID call arrives at the attendant position for coverage, the LED associated with the call coverage

sender will flash and will then go dark when the call is answered. However, if the call is placed directly to the attendant position or is forwarded to the position and thereby arrives on a System Access button or a Loop button (e.g., if a DID PDC is logged-in at the attendant position), then no LED indications on the Selector Console will be provided. If a DID call is directed to the answering position and is subsequently extended to a station, then the LED on the Selector Console associated with the station will flash if the call returns to the answering position. The LED will light steadily if the call is answered by the station.

- **Call Coverage:** DID calls receive standard call coverage treatment.
- **Conference:** For conference purposes, DID calls count as one of two allowable outside parties.
- **Dictation System Access:** A DID number may be associated with the dictation system access code. This allows an outside caller to access the dictation equipment.
- **Direct Group Calling:** A DID call will be directed to a DGC group if the DID number matches the DGC group access code.
- **Night Service:** DID calls *do not* receive Night Service treatment. A DID call will ring at the appropriate station whether Night Service is activated or not.
- **Off-Premises Stations (OPS):** DID calls can be directed to OPS.
- **Paging System Access:** A DID call may access a paging zone. This allows the user to dial in and utilize the Paging feature. Dial restricting the paging code will block this interaction.
- **Personal Dial Codes:** DID calls will be redirected to PDCs logged in at other terminals in the system. DID calls to an unassigned PDC or a FPDC that is not logged-in will be either redirected to the attendant or receive Reorder Tone.
- **Pooled Facility Access:** Access to pooled facilities via DID is permitted. This includes access to WATS, FX, tie trunks, private lines, dictation equipment, and paging systems. This access is provided by selecting facility access codes so that they will match DID numbers.
- **Station Message Detail Recording (SMDR):** For V2 systems, only one SMDR record is produced if an outgoing call is originated by a DID trunk. The STN field will contain 0000, the FAC field will contain the facility access code of the trunk group used to complete the call, and the CALLED NUMBER field will contain the called number.

Administration Requirements

System:

- Send misdirected DID calls to the Attendant Console (Yes, No) - Default = Yes.
- Set number of DID digits matched against dial codes (2-4, None) - Default = 3.

Trunk Port:

- DID trunk type (Immediate Dial, Wink Start)
- Number of digits to be received from CO on this trunk - Default = 3.

Hardware Requirements:

Each DID trunk requires a port on a TN753 DID Trunks CP.

DIRECT STATION SELECTION (DSS)

Description

Allows one-button access to another voice terminal, a paging zone, or a DGC Group. This feature requires a button assignment on a multiline voice terminal.

There are two types of DSS buttons. Numbers stored on *DSS* buttons (maximum of four digits) are programmed at the SAT; numbers stored on *Flexible DSS* buttons (maximum of four digits) are programmed at the voice terminal. The procedure for programming FLEX DSS buttons is provided in the “Program” feature description.

To use DSS, the user presses DSS or FLEX DSS and goes off-hook. The caller hears Ringback Tone. DSS calls to a multiline voice terminal are received on a System Access button. The DSS status LED is lighted steadily at the calling station.

The DSS status LED is lighted whenever the pointed-to station is off-hook. The user may press DSS and remain on-hook to receive the busy-to-idle reminder. The user’s voice terminal will ring once when the other party hangs up; lifting the handset will automatically place the call.

When Prime Line Preference is assigned to a DSS button, the button must be pressed to invoke the busy-to-idle reminder, even though the I-Use LED is lighted.

Access to Paging Zones and DGC Groups:

Access is provided to an individual Paging Zone or to all paging zones or to a DGC group. The status and busy-to-idle reminder indication described above also apply to DGC groups with the understanding that they are busy if all the members are busy.

Considerations

Direct Station Selection differs from Automatic Intercom in that it provides one-button access from one voice terminal to another (one-way only), while Automatic Intercom provides similar access for each voice terminal (two-way) and must be assigned between two multiline voice terminals. A DSS button may point to a single-line station; an Automatic Intercom button may not. DSS calls receive call coverage, Automatic Intercom calls do not.

Interactions

- **Call Coverage:** DSS calls placed to an individual with Call Coverage will receive standard call coverage treatment.
- **Call Following:** For V1 systems, DSS calls do not follow users who log in at other voice terminals. For V2 systems, DSS calls do receive Call Forwarding treatment.
- **Direct Group Calling:** A DSS button can be assigned to a DGC group. The associated LED lights steadily when all stations in the group are busy.
- **Last Number Dialed (V2):** Numbers called by pressing FLEX DSS or DSS buttons are not saved by Last Number Dialed and cannot be recalled with that feature.

- **Line Selection (Prime Line Preference):** When Prime Line Preference is assigned to a DSS button, the button must be pressed to invoke the busy-to-idle reminder, even though its red I-Use LED is lighted.
- **Personal Dial Code (PDC):** An attempt to program a FPDC on a FLEX DSS button (rather than a PDC) results in Reorder Tone.
- **Pooled Facilities:** A pooled facility access code may be stored on a FLEX DSS button (but *not* on a DSS button). If so, the button will function very much like a Direct Facility Access button, with the capability of receiving a busy-to-idle reminder for the pooled facility. However, this button will not allow access to a dial-restricted facility.

Administration Requirements

Assign DSS and/or FLEX DSS buttons at voice terminal.

Hardware Requirements:

None

DISTINCTIVE RINGING

Description

Allows users to distinguish between different types of incoming calls.

Users can receive the following types of ringing:

- A repeated two-burst tone indicates an outside call or a call extended by the attendant. The two-burst tone pattern is: 0.4 seconds on, 0.2 seconds off, 0.6 seconds on, and 4.0 seconds off.
- A repeated one-burst pattern indicates a call from an internal user. The tone is one second on and three seconds off for multiline voice terminals, and 1.2 seconds on and 4 seconds off for single-line voice terminals.
- An “abbreviated alerting” signal on incoming calls to off-hook multiline voice terminals. These calls will ring just once. The status LED associated with the incoming call will continue to flash after the abbreviated ring. The user may place their current call on hold and answer the incoming call if desired.
- A single short beep at a voice terminal equipped with the Hands-Free Answer feature indicates that an incoming inside call has been answered by the terminal. Depending on the status of the terminal’s HFAI controls, the user can talk with the caller without lifting the handset.
- Priority ringing (V2) is a repeated pattern of two short rings followed by one long ring. It indicates that a data terminal has used the Third-Party Call Setup feature to originate a voice call from the voice terminal where this ringing is heard.

Considerations

Distinctive Ringing enables a user to handle each call in an appropriate manner.

Abbreviated alerting alerts the called party to an incoming call but does not provide the continued distraction of ringing.

For V2 systems only: distinctive ringing is not available at Extended Stations; All incoming calls are signaled by standard one-burst ringing, repeated.

Interactions

- **Call Coverage: Covering** stations receive distinctive ringing, depending on the origin of the call receiving coverage.

Administration Requirements

None

Hardware Requirements

None

END-TO-END SIGNALING

Description

Allows multiline voice terminals to send touch-tone (DTMF) signals over the DDD network and allows single-line and multiline users to send touch-tones over dial pulse trunks.

The 7300H series voice terminals do not generate touch-tones when a dial pad button is pressed. The End-To-End Signaling feature provides for the conversion of signals generated by these terminals to touch-tones.

Dialed numbers are toned out for a default duration of 60 ms followed by 60 ms of silence (administerable).

When using dial pulse trunks, End-To-End signaling is invoked by dialing “#” after the last digit of the *called* number or waiting for about 10 seconds after dialing the last digit (see the Interdigit Timeouts feature description). All subsequent dial pad button presses generate touch-tones on the outside line.

Considerations

End-To-End Signaling permits stations to access network services that require touch-tone signals.

Interactions

- **Command Mode And Data Terminal Dialing:** Occasionally it is necessary to send additional tones to the remote endpoint after a data connection has been established. A mark character “\$” is embedded in the dialing sequence to indicate to call processing that additional tones must be sent prior to insertion of a conversion resource (pooled modem) into the connection. The mark character “\$” is used to indicate that all the following digits are for end-to-end signaling. This character is used to mark the boundary between the digits dialed to reach the distant endpoint and the digits used by the distant endpoint after it answers.
- **Repertory Dialing:** Repertory Dialing can be programmed on the 7300H series voice terminals. End-To-End Signaling works properly with this feature.
- **Speed Dialing:** (For V2 systems only) #3 must be stored to start End-to-End Signaling.
- **Virtual Facilities (V2):** #3 must be stored to start End-to-End Signaling.

Administration Requirements

None

Hardware Requirements

None

EXCLUSION

Description

Allows multiline voice terminal users to keep other users with appearances of the same Personal Line from listening in on their calls. Exclusion allows users to exclude the attendant and other stations from an existing or held outside call, or to drop other System 25 users from a call.

The Exclusion button status and I-use LEDs are lighted steadily when the feature is invoked. When an excluded call is placed on hold, the Exclusion button's I-Use LED goes dark and the status LED winks with the LED of the held line.

Exclusion can be applied to only one line at a time. Once Exclusion is invoked on a call it will remain active until the user either presses the button a second time or goes on-hook.

Considerations

Exclusion allows the sharing of a Personal Line by several users while retaining privacy. Note that all inside calls are automatically private.

Interactions

- **Automatic Intercom:** Any attempt to activate Exclusion while active on an Automatic Intercom call will drop the other party.
- **Call Coverage:** If a call coverage receiver invokes Exclusion after answering a coverage call, all other terminals (including the attendant and the covered station) are excluded. The covered user cannot enter the call until Exclusion is pressed a second time by the covering user.
- **Conference:** When Exclusion is invoked, all other inside parties will be dropped. If a private conference including inside parties is desired, the user should activate Exclusion first and then set up the conference.
- **Hold:** A call can be placed on hold after Exclusion is invoked. The I-Use LED will go dark; the status LED of the line appearance button and the Exclusion button will wink.

Administration Requirements

Voice Terminal (Station Port):

- . Assign Exclusion button.

Hardware Requirements

None

EXPERT MODE (V2)

Description

Expert Mode is an enhancement to the Command Mode feature that provides an alternative method of performing the full range of Command Mode functions. By eliminating the display of menus and allowing multiple commands to be entered on a single line, Expert Mode lends itself to computer-driven instructions. Individual users who are *very* familiar with Command Mode operations may also find it useful.

When Expert Mode is activated, a system administrable prompt is displayed that can consist of up to nine characters (the quote character and RETURN are not allowed). **Command:** is the system default prompt. As with dialing in Command Mode, the ASCII characters backspace (BS or CTRL-H) or underscore (_) may be used to cancel a previously entered character. *When in Expert Mode, each line must be terminated with a keyboard RETURN.*

Users of Expert Mode **must** follow the exact tree structure of Command Mode (both up and down the menu tree) as shown in Figure 2-51. However, instead of moving one level at a time, Expert Mode allows the user to move up or down several menu levels at once. This can be accomplished by entering, on a single command line, the capitalized letters that define the sequence of menu selections desired. For example, to change data port parity from the tree's entry level, the user types OCPPE and presses RETURN. This requests that parity be set to "even", but does NOT enable the change. To enable this change (see Figure 2-51), the user must now type XE and press RETURN.

Activating Expert Mode

A user can move back and forth between Command Mode and Expert Mode by typing "!" (exclamation mark). For ports on a Data Line circuit pack (Data Line Card, DLC), either Command Mode or Expert Mode is presented at the start of a new session, depending upon the port's setting at the termination of the previous session. Thus, if a data session ends in Expert Mode, the next session will begin in Expert Mode. However, calls from an AT&T STARLAN NETWORK to System 25 will *always* begin a new session in Command Mode.

An alternative command, ">", can be used to *guarantee* entry into Expert Mode. Conversely, guaranteed entry into Command Mode can be accomplished with the command ">!" followed by RETURN. These commands are especially useful for computer-driven DLC endpoints that might otherwise have difficulty detecting whether a new session had been started in Expert Mode or Command Mode.

Making a Data Call

To make a data call from the entry level (see Figure 2-51), the user enters "D" following the system prompt and then the data endpoint number. For example:

Command: D9,5553822

Note: "Command:" on the above line is the default system prompt while in Expert Mode. The user enters all data following the prompt.

If the user enters “D” and then a RETURN, the system will prompt for the data endpoint number as follows:

Command: D
DIAL:

The user must then enter the digits required to complete the call.

Activating the Third-Party Call Setup Feature

The following provides an abbreviated method of using the Third-Party Call Setup feature while in the Expert Mode. A complete description of this feature is provided later in this manual.

To activate the Third-Party Call Setup feature and place a call, the user enters numbers using the following format:

Command: V{Destination} F {Source}

The V on the above command line provides access to **<Voice call>** from the Command Mode entry level menu. The balance of the dialed number is composed of *destination* and source numbers, as described in the Third-Party Call Setup feature description.

User Changeable Options

Refer to the User Changeable Options feature (discussed later) for a detailed description of the feature. The menus selected in the following discussion are shown in Figure 2-51.

To view the current Options Table (starting at the Command Mode entry level), the user simply enters “OV” following the system prompt, as follows:

Command: OV

To change the current Options Table (starting at the entry level), the user enters “OC” as follows:

Command: OC

Entering “OC” places the user at the Change Options level. At this point the user may change options by entering the appropriate letter to indicate the required option (S for Speed, P for Parity, M for Mismatch, etc.) followed by the desired setting(s). Only one Option is allowed per line. If more than one setting is selected for an Option that can only accept one setting, call processing recognizes only the last entry.

Examples:

Command: S +1200 -300 +4800 *Add 1200 and 4800 baud to the available speeds, remove 300 baud*

Command: PE

Change Parity to Even

Command: MY

Change Mismatch to Yes

If the user enters an invalid Option or setting, the system responds with **INVALID OPTION** and the entry is ignored.

Once all changes have been entered, the user enters XE to enable the options.

Considerations

Expert Mode is primarily for use by computer-driven endpoints that can store command sequences for automated use. However, a user experienced in accessing Command Mode menus may find Expert Mode to be a faster alternative when operating at slower speeds, since the time required to display each menu and to input separate commands is essentially deleted.

Interactions

• Command Mode

Refer to the Command Mode feature description for a detailed description of Command Mode and of the various menu items.

Administration Requirements

The data port associated with a data terminal can be administered to allow the user to change options when in Command or Expert Mode. Otherwise, the user may view the current options but not change them.

The default prompt for Expert Mode (**Command:**) may be changed via system administration.

Hardware Requirements

None

EXTENDED STATIONS

Description

Allows single-line voice terminals to be located at distances greater than 2000 feet from the system cabinets.

Extended stations have the same feature capabilities other voice terminals. These stations count as an outside party on conference calls.

Transmit and receive levels are increased at extended stations for conferencing.

Considerations

A single-line voice terminal must be administered as an extended station before this feature is activated.

In V2, extended stations will always receive standard (that is, single) ring for calls; System 25 will not send distinctive ringing.

Interactions

- **Call Park:** (For V1 only) unextended station cannot pick up parked calls.
- **Conference:** An Extended Station counts as one of the two outside parties allowed on conference calls.

Administration Requirements

Single-Line Voice Terminals (Station Port)

- Assign port on Analog Line (TN742) CP
- Make This An Extended Station (Yes, No) - Default = No.

Hardware Requirements

The Extended Station must be a single-line voice terminal. Requires a port on a TN742 Analog Line CP.

EXTERNAL ALERTS

Description

Provides standard station ringing at a location away from the called station. This feature can be used to activate an external alerting device such as a bell.

External Alerts supports the Trunk-Answer-from-Any-Station (TAAS) form of Night Service.

The feature can be used in conjunction with voice terminals located in noisy environments and large areas such as warehouses, etc. The alerting device is activated whenever the associated station is alerted.

A Supplemental Alert Adapter installed on a hybrid station allows the terminal user to transfer incoming ringing to an alerting device located in some remote area. When the user goes to the area, the alerting device rings for incoming calls to the user's normal station.

Considerations

External Alerting enhances user ability to recognize incoming calls. Noisy environments, large areas, and outside locations are candidates for external alerting devices.

Interactions

- **Manual Signaling:** Manual Signaling will not activate an external alerting device.
- **Night Service:** When the system is in Trunk-Answer-from-Any -Station (TAAS) Night Service mode, an incoming attendant-seeking call will activate the Night Service alerting device.
- **Power Failure Transfer:** When the system is in the power failure transfer mode, the external alerting devices are disabled.

Administration Requirements

Station Port

- An external alert operating as the endpoint device on a station line requires a port assignment on a ZTN78 Tip Ring Line or TN742 Analog Line CP. (Specify special feature port type = 253.) An xternal alerting device controlled from a Supplemental Alert Adapter operates on the same line as the associated terminal and requires no additional port assignment.
- Specify the PDC of the associated station (or 0 if alert is used with TAAS Night Service).

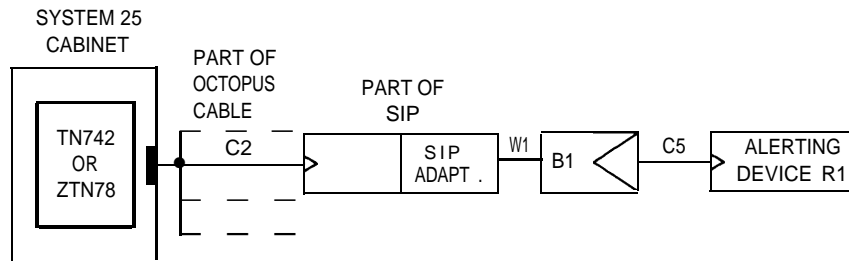
Hardware Requirements

Order line-activated alerting devices (e.g., bells) as required.

An alerting device operating on a line separate from a terminal requires a port interface on a ZTN78 or TN742. Refer to Figure 2-24 for connection information.

Order the Supplemental Alert Adapter (PEC 2301-SAA) for controlling a remote alerting device. A Supplemental Alert Adapter is installed in the line between the port CP and the user's MERLIN terminal. The line requires a ZTN79 ATL Line CP. Figure 2-25 contains connection details.

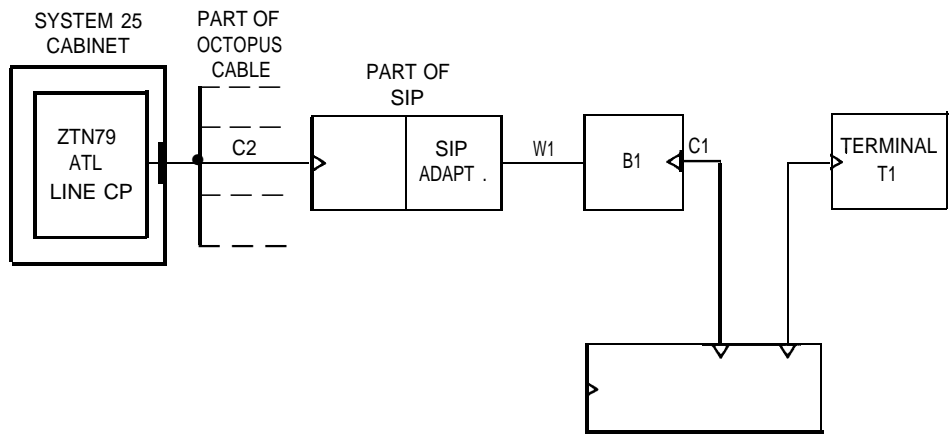
Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters are provided under the heading "Connectivity" in Section 4.



LEGEND:

- TN742 - ANALOG LINE CP
- ZTN78 - TIP RING CP
- B1 - TYPICAL-103A CONNECTING BLOCK*
- C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
- C5 - MODULAR CORD (D4BU-87)
- R1 - E1CM-50 RINGER OR EQUIVALENT PEC-31019
- W1 - 4 PAIR INSIDE WIRING CABLE*
- * - FURNISHED BY INSTALLER

Figure 2-24. External Alert Connections



- LEGEND:**
- A1 - SUPPLEMENTAL ALERT ADAPTER (2301-SAA)
 - B 1- TYPICAL - 103A CONNECTING BLOCK
 - C 1- MODULAR CORD (D8W-87)
 - C2- OCTOPUS CABLE (WP90780)
 - C8 - MODULAR CORD (D4BU-87)
 - S1 - EXTERNAL ALERT
 - T1 - HYBRID TYPE TERMINAL
 - W1 - 4 PAIR INSIDE WIRING CABLE

Figure 2-25. Supplemental Alert Adapter Connections

HANDS-FREE ANSWER ON INTERCOM (HFAI)

Description

Allows the following voice terminals to provide hands-free answer service on eligible incoming calls; each terminal must have Automatic Answer (AUTO ANS) assigned to a flexible button.

- BIS (7305H03B and 7305H04C) and HFAI (7309H01A); these sets provide full service without requiring adjuncts.
- 10-Button (7303H01B), 34-Button (7305H01B), and 34-Button Deluxe (7305H02B) equipped with a Hands-Free Unit (HFU—a 102A Speakerphone); these arrangements provide full HFAI service.
- 5-Button (7302H01C) and the 10- and 34-Button sets listed above, not equipped with an HFU; these arrangements allow callers to “voice announce” their calls, but the terminal user must use the handset to reply.

Calls Eligible for Hands-Free Service:

The following types of calls are eligible for HFAI service:

- Inside calls (that is, calls from one System 25 set to another System 25 set using a System Access, Loop, DSS, or Auto Intercom button).
- V1 Only: Incoming calls *extended* by the attendant are eligible for HFAI service at BIS sets but *not* at HFAI sets. However, HFAI sets may be translated as BIS sets and will then provide HFAI service on attendant extended calls (see “Interactions,” below, for more information).

For V2, no outside (trunk) calls extended by the attendant are eligible for HFAI service.

- Calls transferred from another System 25 set using the Transfer feature. The *transferring* station may pass both inside and outside calls in this way. Note that calls *transferred* by the attendant are indistinguishable from calls transferred by any other station.

BIS and HFAI Voice Terminals

LEDs next to the AUTO ANS button and the HFAI/Mic (HFAI set) or HFAI (BIS set) button indicate whether the HFAI feature is enabled. The LEDs are turned on and off by pressing the adjacent buttons. When both the AUTO ANS and HFAI LEDs are on, the set will auto-answer eligible calls.

The HFAI LED will wink (on HFAI sets) or light steadily (BIS sets) during HFAI calls.

The set’s response to HFAI-eligible calls depends on the status of the HFAI and AUTO ANS buttons and LEDs, as follows:

- If both HFAI and AUTO-ANS LEDs are on:
 - The set generates a tone burst over its speaker to indicate an incoming call (the calling party also hears this tone).
 - The parties may converse. The called party can speak in a normal voice toward the set. No other action by the *called* party is required.

- During the call, the called party can press the HFAI/Mic or MICROPHONE button to mute the microphone temporarily and prevent the caller from hearing. Pressing the button again turns the microphone on again.
 - The HFAI/BIS user may press the SPEAKER (HFAI set) or the SPEAKERPHONE (BIS set) button to end the call. For V2 systems only: if the calling party hangs up first, this is not necessary.
- If only the AUTO ANS LED is on:
 - The set generates a tone burst over its speaker to indicate an incoming call (the calling party also hears this tone).
 - The set's speaker turns on and the set "answers" the call.
 - Call setup is complete. However, the called party can hear, but not respond to, the calling party. To respond, the user must lift the handset or press the HFAI/Mic button on an HFAI set or press the MICROPHONE button on a BIS set.
 - The HFAI/BIS user may press the SPEAKER (HFAI set) or the SPEAKERPHONE (BIS set) button to end the call. For V2 systems only: if the calling party hangs up first, this is not necessary.
- If only the HFAI LED (or neither LED) is on:
 - The HFAI feature is disabled. The call answering procedure is the same as for a standard MERLIN set.

If, during an HFAI call, the user decides to pick up the handset, the HFAI/Mic or HFAI LED will turn off. On an HFAI set, the user is not permitted to revert to hands-free operation. (Pressing the HFAI button while using the handset will simply disable the HFAI feature for subsequent calls.) A BIS set user may transfer a call from the handset to the speakerphone by pressing the SPEAKERPHONE button and hanging up.

Voice Terminals with HFUs

These sets do not have an HFAI button. To turn on the HFAI feature the user simply presses the AUTO ANS button; the green status LED lights.

After HFAI is activated, operation is exactly the same as for the BIS set except that the SPEAKERPHONE and MICROPHONE buttons and LEDs are on the HFU.

Voice Terminals without HFUs

The HFAI feature is activated by pressing the AUTO ANS button. A beep signal announces an incoming call and the SPEAKER LED lights. A one-way talking link is established from the caller to the terminal; the user can hear the caller but cannot converse. Lifting the handset connects the user to the caller.

Considerations

The user of a HFAI equipped station should always deactivate the HFAI feature when leaving the work area. If this is not done, incoming calls will be "answered", but the callers will be talking to an unattended position.

Interactions

- **Attendant (V1 Only):** The attendant may pass calls to HFAI sets using the *Transfer* feature and button rather than *extending* these calls using the START button or the Selector Console. Such calls are then eligible for HFAI service. However, since unanswered *transferred* calls do not return to the attendant for further service (as *extended* calls do), this practice is not recommended.

A better procedure is to mistranslate the HFAI sets as BIS sets (using the BIS set code—305 at the System Administration Terminal). If this is done, attendant extended calls are eligible for HFAI service at HFAI sets.

- **Call Coverage:** When the HFAI feature is enabled at a set, calls eligible for HFAI service will not receive call coverage because the set will answer them whether the user is present or not. However, if the attendant uses the Attendant Message Waiting feature to turn on the Message indicator at the set, the HFAI feature will be disabled (the AUTO ANS LED turns off), allowing subsequent calls to receive coverage.
- **Send All Calls:** Activating Send All Calls will disable the HFAI feature (the AUTO ANS LED turns off).

Administration Requirements

Voice Terminal (Station Port):

- Assign AUTO ANS button.

Hardware Requirements:

This feature requires one of the voice terminals or combinations of terminal and HFU listed in the Description.

HEADSET ADAPTER ADJUNCT

Description

The headset adapter adjunct provides an interface for connecting a headset to an associated voice terminal. A headset plugged into the adapter is activated by switches on the adapter. The terminal operator has the choice of using either the handset or the headset for handling calls. Turning the headset on and off is equivalent to lifting and hanging up the handset.

Considerations

Use of a headset allows a voice terminal operator to carry on conversations with both hands free for writing, typing, etc. It is valuable adjunct for high traffic positions such as attendant consoles.

Use of a headset does not affect normal voice terminal operations in any way.

Interactions

- **Speakerphone Adjunct:** A voice terminal cannot have both a headset and a speakerphone. These adjuncts plug into the same jack on the voice terminal.

Administration Requirements

None

Hardware Requirements

500A/502A Headset Adapters:

The 500A adapter (Figure 2-26) is designed for use with the 12-Button (7203M) MET voice terminal. The 502A adapter, which is identical in appearance, is designed for use with MERLIN (7300H Series) terminals (with the exception of the 5-Button and HFAI sets). Most standard commercial headsets can be used with the adapters.

Each adapter has an “ON/QUIET” button, an “OFF” button, a green indicator lamp, a jack for a single headset, and two modular keyed jacks (4-wire and 8-wire). Each is equipped with an 18-inch connecting cord. Optional cords are available in lengths of 4 and 14 feet.

The 500A Headset Adapter is powered locally by a 2012D Transformer, which plugs into a 115V ac receptacle. Power from the transformer is applied to the voice terminal mounting cord via a 400B2 adapter at the wall jack and conducted to the 500A on its connecting cord. Refer to “Voice Terminal Adjunct Power Supplies” in Section 4 for additional information.

The 502A Headset Adapter does not require supplemental power, except when used with a 34-Button Deluxe, BIS, or BIS with Display voice terminal.

Detailed headset adapter connection information is provided in the following figures:

- Figure 2-27— Typical Headset Adapter Connections for 7300H Series Multiline Voice Terminals (Except 34-Button Deluxe, BIS, or BIS with Display)

- Figure 2-28 -Typical Headset Adapter Connections for 34-Button Deluxe, BIS, or BIS with Display Voice Terminals
- Figure 2-29 -Typical Headset Adapter Connections for 12-Button MET Sets

MET Headset Adapter:

Use of a headset with a 10-Button MET voice terminal requires a JS0180-3A Headset Adapter (18 inch cord) or a JS0180-4A Headset Adapter (8 foot cord).

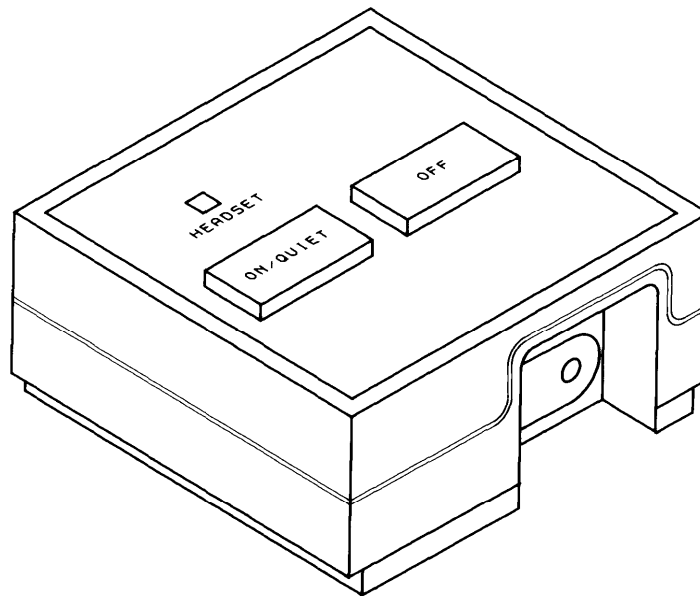
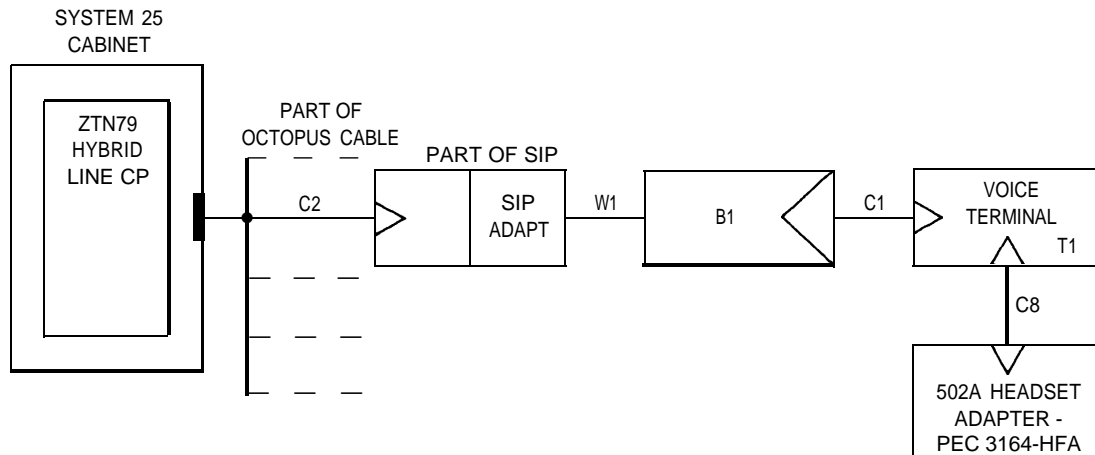


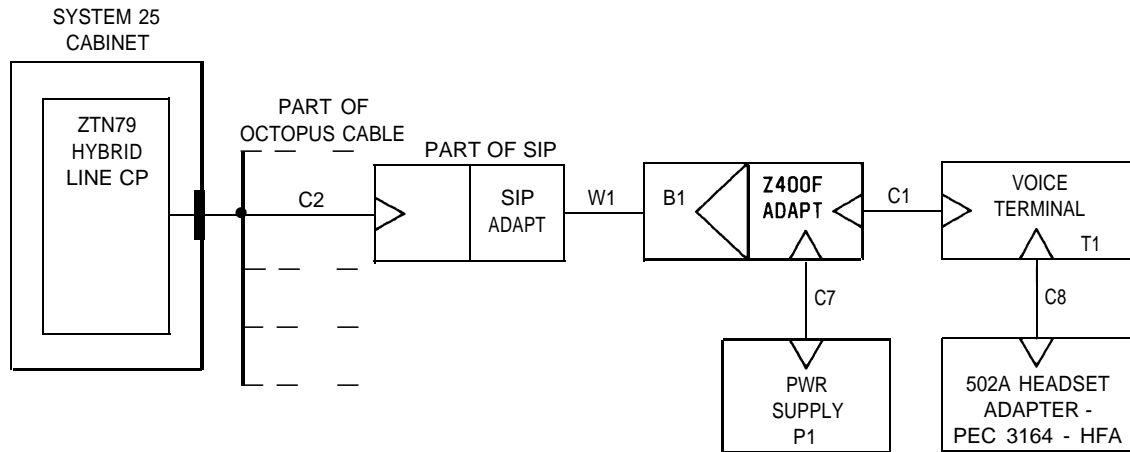
Figure 2-26. 500A/502A Headset Adapter



LEGEND:

- B 1- TYPICAL-103A CONNECTING BLOCK***
- C 1- MODULAR CORD (D8W-87) - FURNISHED WITH SET**
- C 2- OCTOPUS CABLE CABLE (WP90780) - PEC 2720-05P**
- C 8- SPECIAL CORD- FURNISHED WITH ADJUNCT**
- T 1- 7300H SERIES VOICE TERMINAL EXCEPT 34-BUTTON DELUXE, BIS, OR BIS WITH DISPLAY**
- W 1- 4 PAIR INSIDE WIRING CABLE***
- * - FURNISHED BY INSTALLER**

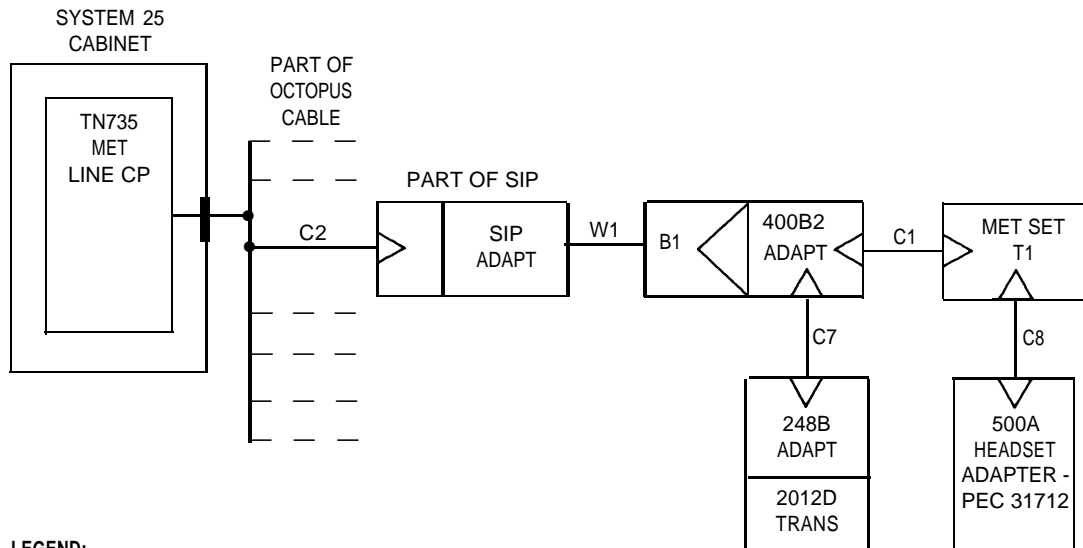
Figure 2-27. Typical Headset Adapter Connections For 7300H Series Multiline Voice Terminals (Except 34-Button Deluxe, BIS, or BIS with Display)



LEGEND :

- B1 - TYPICAL-103A CONNECTING BLOCK*
 - C 1- MODULAR CORD (D8W-87) - FURNISHED WITH SET
 - C2 - OCTOPUS CABLE CABLE (WP90780) - PEC 2720-05P
 - C8 - SPECIAL CORD- FURNISHED WITH ADJUNCT
 - T1 - 7300H SERIES VOICE TERMINAL (34-BUTTON DELUXE, BIS, OR BIS WITH DISPLAY)
 - W1 - 4 PAIR INSIDE WIRING CABLE*
 - C7 - MODULAR CORD (D6AP-87)
 - P1 -KS 22911 POWER SUPPLY } PEC 62510
 - Z400F - ADAPTER
- * - FURNISHED BY INSTALLER

Figure 2-28. Typical Headset Adapter Connections For 34-Button Deluxe, BIS, or BIS with Display Voice Terminals



LEGEND:

- B 1 - TYPICAL-103A CONNECTING BLOCK*
 - C 1 - MODULAR CORD (D8W-87) - FURNISHED WITH SET
 - C 2 - OCTOPUS CABLE (WP90780) PEC 2720-05P
 - C 8 - SPECIAL CORD - FURNISHED WITH ADJUNCT
 - T 1 - 7203M SET - 12-BUTTON MET SET
 - W 1 - 4 PAIR INSIDE WIRING CABLE*
 - 248B ADAPTER - MODULARIZES 2012D TRANSFORMER
 - 400B2 ADAPTER - POWER ADAPTER
 - 2012D TRANSFORMER - 15-18V AC TRANSFORMER
 - C 7 - MODULAR CORD (D6AP-87)
 - * - FURNISHED BY INSTALLER
- PEC 21691

Figure 2-29. Typical Headset Adapter Connections For 12-Button MET Sets

HOLD

Description

Allows users to temporarily disconnect from one call and either place or answer another call. A single-line voice terminal user can place only one call on hold and must remain off-hook to retain the held call. A multiline voice terminal user can place as many calls on hold as it has lines and can hang up without losing held calls.

Single-line users can place a call on hold by flashing the switchhook (the user receives Confirmation Tone). The user can then dial another party or return to the held call by flashing the switchhook twice. The first switchhook flash sets up a conference call, the second flash drops the third party; if System 25 Dial Tone, Busy Tone, or Reorder Tone (but not Ringback Tone) was obtained when the third party was dialed, a *single* switchhook flash will drop the tone and return the user to the held party.

Multiline voice terminal users can press HOLD and subsequently replace the handset or call another party without losing the held call. The status LED associated with the held call winks on all terminals with an appearance of the line, except in the case of a conference call. In this case, the wink indication is given only to the party who invoked hold. To return to the held call the multiline user either presses the line button associated with the held call (V1 and V2) or simply goes off-hook if the held call is on a button assigned prime line preference (V2 only).

Considerations

The Hold feature allows voice terminal users to handle several calls simultaneously. For single-line sets, placing a call on hold is the first step in transferring or conferencing the call.

Interactions

- **Attendant Console:** The Attendant does not receive hold indications for lines (trunks) appearing on the Console unless he/she placed the call on hold.
- **Exclusion:** A call can be placed on hold after Exclusion is invoked. The status LED of the line appearance button and the Exclusion button will wink.
- **Music-On-Hold:** A held party on an outside line will receive Music-On-Hold if provided.
- **Personal Lines:** A Personal Line cannot be placed on hold if any other stations are also off-hook on that line.

Administration Requirements

None

Hardware Requirements

None

INTERCEPT TREATMENT WITH REORDER TONES

Description

Reorder tone (fast busy) is provided when an unassigned or toll-restricted number is dialed, a dialing error occurs, or an attempt to park a call fails.

Any attempt to dial an unassigned code (PDC, DDC, feature or facility access code) or an unsuccessful attempt to park a call will result in Reorder Tone being provided to the caller. A DID call to an unassigned number will be routed to the attendant or will receive Reorder Tone at the System Administrator's option. Any attempt to dial a restricted call (toll or access restricted) will be intercepted and routed to Reorder Tone.

Considerations

Intercept treatment provides a calling party with positive feedback of an error in dialing or use of an incorrect code.

Interactions

- **Call Park:** An unsuccessful attempt to park a call due to misdialing or attempting to park more than one call at a voice terminal results in Reorder Tone.

Administration Requirements

None

Hardware Requirements

None

INTERDIGIT TIMEOUTS

Description

Allows an originating register to be made available for others if dialing is not completed within a set time period.

Interdigit timeouts is 24 seconds until the first five digits have been dialed, 10 seconds until the next five digits have been dialed, and five seconds thereafter.

After timeout, voice terminal dial pad button presses are interpreted as end-to-end signaling requests and touch-tones are placed on the outside line.

Considerations

Interdigit timeouts also apply to data calls.

When a user dials out over a trunk set up for dial pulse rather than Touch-Tone service, the interdigit timeout interval is involved. The caller cannot speak to the called party until the timeout expires (even though the caller may be able to hear the called party). Because the timeout interval decreases as more digits are dialed, the problem is often noticed on tie trunks since the user may dial fewer (or no) digits when using these trunks. If the user presses the # button after dialing the last digit this timeout ends and the caller may speak immediately.

Interactions

None

Administration Requirements

None

Hardware Requirements

None

LAST NUMBER DIALED (V2)

Description

Automatically saves the last number dialed from a multiline voice terminal and allows the user to place the call again without redialing the number. The feature is administered to a button on the terminal. Both inside and outside calls can be made in this way. The original call can be placed by manual dialing, by operation of a programmed button, or by speed dialing.

To use the feature, the caller first gets dial tone, and then presses the Last Number Dialed button. The I-use and status LEDs of the button selected for originating the call (for example, System Access, Loop, or Personal Line) light steadily; if the Last Number Dialed button has an LED, it lights momentarily. The call proceeds in the normal way. The number associated with the Last Number Dialed button remains saved even if the called party answers. Only the dialing of a new number changes the state of the Last Number Dialed button; the old number is then erased and the new one stored.

Considerations

Last Number Dialed is a convenience feature that is especially valuable for recalling multi-digit numbers that were first dialed manually from a terminal's pushbutton dial. However, the feature can also be used to recall numbers originally called by the following means:

- Repertory Dial buttons.
- Group Select and DXS buttons on a Selector Console.

This feature saves numbers with up to 16 digits.

The user must hear dial tone before pressing the Last Number Dialed button.

The Last Number Dialed feature cannot be activated by dialing an access code. It is not available to users of single-line voice terminals.

If a dialed number does not complete a call, Last Number Dialed still stores the digits dialed. If the user presses the Last Number Dialed button, then dials additional digits to complete the call, both the currently stored digits and the dialed digits will be stored.

Interactions

- **Account Code Entry:** Last Number Dialed does not save an Account Code dialed by operation of an Account Code Entry button.
- **Attendant Display:** When a call is placed at a Switched Loop Attendant Console (V2), using the Last Number Dialed button, the call information display has the normal format of an outgoing call. In the Inspect mode, pressing the Last Number Dialed button displays the number currently stored on the button.
- **Automatic Intercom:** Numbers called using an Automatic Intercom button are not saved by the Last Number Dialed feature. The number currently stored by Last Number Dialed is not changed by operations of the Automatic Intercom button.

- **Call Accountability:** When a station user dials ##PDC to provide accountability for a call and then dials the desired digits, the ##PDC is not saved by the Last Number Dialed feature.
- **Conference:** When a station user adds a party to a conference, the number dialed is saved as the Last Number Dialed.
- **Direct Station Selection (DSS):** Numbers called using an DSS or Flex DSS button are not saved by the Last Number Dialed feature. The number currently stored by Last Number Dialed is not changed by operations of these buttons.
- **Personal Lines:** When a user originates a call from a Personal Line, only the digits dialed after the line is accessed are saved by the Last Number Dialed feature. The same type of line must be selected to get dial tone for placing another call using Last Number Dialed. If a different type of line is used, the call may be directed to the wrong destination.
- **Pooled Facility—Direct Access:** When a user originates a call from a Direct Facility Access button, only the digits dialed after the line is accessed are saved by the Last Number Dialed feature. The same type of button should be selected to get dial tone for placing another call using Last Number Dialed. If a different type of button is used, the call may not be directed to the right destination.
- **Repertory Dialing:** When using a Repertory Dial button to place a call, the numbers dialed are saved by Last Number Dialed. When the call is redialed using Last Number Dialed, the same type of button where dial tone was originally accessed should be used again to ensure that the call is directed to the correct destination.
- **Speed Dialing:** When using a Personal or System Speed Dialing code to place a call, the code is saved by Last Number Dialed. When the call is redialed using Last Number Dialed, the same type of button where dial tone was originally accessed should be used again to ensure that the call is directed to the correct destination.
- **System Access/System Access Originate Only Buttons:** If a user originates a call from one of these buttons, the same type of button should be selected for getting dial tone to place a second call with the Last Number Dialed feature. Using another type of button, such as Personal Line or Direct Facility Access, to get dial tone may prevent the call from completing properly.
- **Transfer:** When a station user Transfers a call, the dialed number (of the party to whom the call is transferred) is saved as the Last Number Dialed.

Administration Requirements

Last Number Dialed is a default feature on all multiline voice terminals in a V2 system. One button is assigned to the feature at each set. The feature can be moved/removed by administration.

Hardware Requirements

None

LINE SELECTION

Description

Multiline voice terminals may have many line (facility) appearances. There are three methods by which a user may select a desired line: (1) Prime Line Preference, (2) Ringing Line Preference, and (3) Preelection.

Prime Line Preference:

Automatically connects a multiline voice terminal to a specified line or facility designated as preferred when the terminal goes off-hook. This feature may be assigned to System Access, Loop, Automatic Intercom, DSS, Personal Line, and Direct Facility Access buttons.

On the Switched Loop Attendant Console (V2), the topmost Loop button has Prime Line Preference by default. However, the feature can be assigned to any of the five Loop buttons.

The user may override this feature by preselecting another button (see below).

If Prime Line Preference is assigned to an Automatic Intercom (AUTO ICOM) or DSS button, the called voice terminal will ring as soon as the terminal goes off hook.

When the Prime Line Preference feature is assigned to an AUTO ICOM, DSS, or Direct Facility Access (FACILITY) button, the button must be pressed to activate the busy-to-idle reminder even though the button's I-Use LED is lighted steadily.

If Prime Line Preference is assigned to a Personal Line button, the user is connected to the button upon going off-hook, even if the line is busy. For V1 systems only: if Prime Line Preference is assigned to a button that has a call on hold, no connection will be made; to get a line for placing a call, the user must press another button. In V2 systems, the user is always connected to the button where Prime Line Preference is assigned, even if the line has a call on hold; if the user wishes to use a different line, the line may be preselected before going off-hook.

Ringing Line Preference:

Automatically connects a multiline voice terminal to an incoming call ringing at the terminal.

Line access buttons that can be selected by Ringing Line Preference include System Access, Automatic Intercom, Coverage, and Personal Lines.

If two or more lines on a multiline terminal or a Direct Trunk Attendant Console are ringing simultaneously, the user is connected to the first line to start ringing. If the user wishes to use a different line, the line *must* be preselected prior to going off-hook. If ringing ceases while the user is on-hook, line preference reverts to whichever option is applicable (Prime Line Preference or no preference).

If a line rings at a multiline terminal when the terminal is busy on another call, Ringing Line Preference will not activate, even if the user goes on-hook during the ringing cycle. However, Ringing Line Preference is not canceled at the Attendant Console while the attendant is off-hook. If a line is ringing while the attendant is off-hook, the ringing line will be selected as soon as the attendant goes on-hook.

Preselection:

Allows multiline voice terminal users to override the above line preference features.

Users may simply press a desired line access button before going off-hook. The user will be connected to the facility selected unless the facility is busy and the party using it has invoked Exclusion or is part of a conference call that is at maximum capacity.

When off-hook, a user can select a facility by pressing the associated button. (This will terminate the call the user was on.)

A user may activate the busy-to-idle reminder on a busy AUTO ICOM, DSS, FLEX DSS, or FACILITY button by pressing the button while on-hook. A burst of ringing is provided when the facility becomes idle. Refer to the subsection on the Busy-to-Idle feature for additional information.

Considerations

Prime Line Preference (on the topmost SYSTEM ACCESS or Loop button) and Ringing Line Preference are assigned by default to all multiline voice terminals. While these assignments may be changed, it is *strongly recommended* that Ringing Line Preference be retained.

It is recommended that Prime Line Preference not be assigned to a Direct Trunk Attendant Console (DTAC).

Preselection allows users to override line preference features already administered for the terminal and to activate the busy-to-idle reminder feature.

Interactions

- **Attendant Console, Direct Trunk:** If a line rings at a DTAC while the attendant is on another call, Ringing Line Preference will be invoked when the attendant hangs up.
- **Ringing Line Preference:** Ringing line preference overrides Prime Line Preference and Preselection when a call is ringing at an on-hook voice terminal.
- **Power Failure/Cold Start:** On power-up, most multiline voice terminals will have no I-use LED lit and will not draw dial tone until a button is pressed. After this, line selection should work as described above.

Administration Requirements

Voice Terminal (Station Port):

- Prime Line Preference:
Assign Prime Line Preference - Default = Top SYSTEM ACCESS button or top LOOP button (Switched Loop Attendant Console only).
- Ringing Line Preference:
Assign Ringing Line Preference (Yes, No) - Default = Yes.

Hardware Requirements

None

LINE STATUS AND I-USE INDICATIONS

Description

Provides users with a visual indication of the status of feature buttons and lines appearing at a their multiline terminals. A green status LED and a red I-Use LED are provided for each programmable button on most multiline voice terminals.

Table 2-I summarizes LED states and associated descriptions for line appearances.

TABLE 2-I. LED Indications

I-Use (Red LED)	Line Status (Green LED)	Meaning
Off	Off	Facility is idle.
On	On	If off-hook, facility is in use at this terminal. If on-hook, busy-to-idle reminder is set.
Off	On	Facility is busy or Feature has been activated.
Off	Winking	Facility placed on hold.
On	Flashing	Facility ringing; call will be answered if user goes off-hook.
Off	Flashing	Facility ringing; call will not be answered if user goes off-hook.
On	Off	Facility that will be accessed upon going off-hook.
Off	Broken Flutter	Facility is being transferred or conference.

Considerations

Line Status and I-Use indications provide the user with visible indications of the status of the lines and features.

Interactions

- **Attendant Console, Direct Trunk:** When a line that appears at both the attendant position and a multiline voice terminal is placed on hold by the terminal user, the green status LED winks at the terminal but lights steadily on the Attendant Console. When the line is placed on hold by the attendant, the green status LED winks on the console and on voice terminals on which it appears.

Trunk-to-trunk transfers will cause the affected PERS LINE buttons on the DTAC to wink.

Administration Requirements

None

Hardware Requirements

None

MANUAL SIGNALING

Description

Allows a user to signal another voice terminal. The user may do this at any time, whether on-hook or off-hook.

Multiline voice terminal users can signal another predesignated *multiline* voice terminal by pressing an associated Manual Signaling (SIGNAL) button. A single tone burst is provided at the signaled terminal. The signaling voice terminal also receives the tone and can use this feature while in any call state. No LED indication is associated with the Manual Signaling feature.

When the Manual Signaling feature is used while the called station is ringing on another call, no audible signal is received by either the signaling or the called voice terminal.

The duration of the single burst of signaling will always be the same, regardless of how long SIGNAL is pressed. The signal is repeated each time the button is pressed.

Considerations

Manual Signaling allows a user to signal another voice terminal without calling the terminal. The meaning of the signal may be prearranged between the sending and the receiving parties. Only multiline terminals may be signaled.

Interactions

- **External Alerts:** Manual Signaling will not activate external alerting devices associated with the signaled station.

Administration Requirements

Voice Terminal (Station Port):

- Assign Manual Signaling button.

Hardware Requirements

None

MESSAGE CENTER-LIKE OPERATION (V2; SLAC Only)

Description

A System 25 Switched Loop Attendant Console (SLAC) can be made to function like a message center through administration of call type translations. Certain specific types of calls in the common queue will then be directed only to a console administered as a message center. This arrangement involves no changes in equipment or in operating procedures. The message center attendant answers incoming calls of the preselected types in the normal way. No provision is made for storing messages, a capability often associated with full service message centers.

Message Center Call Types:

The Message Center receives calls of the following types:

- Returning parked calls that were originally parked from a Selector Console.
- Returning camped-on calls.
- Returning calls that were extended (transferred) from an Attendant Position to a busy station or a station that does not answer.
- Coverage calls—incoming inside and outside calls (including DID calls) covered by the common queue when the called party does not answer, is busy, or does not want to be disturbed (Send All Calls).
- Floating PDCs not logged in at a station and unassigned DID calls.

The type of each incoming Message Center call will be identified by a call type descriptor on the console's 16-character display screen. Refer to the earlier Switched Loop Attendant Console subsection for a list of descriptors.

Console Configurations:

Message Center-like operation applies only when the System 25 has two SLACs. In a one-console system, all calls are handled at the same position.

In the default condition, a SLAC is a combined Attendant Position/Message Center. It can receive any type of call. In a one-console system, there is no division between attendant and message center functions.

A dedicated Message Center is a console that is administered to receive only the specified incoming call types. Dial O calls (attendant-seeking calls from inside the system) are not directed to a Message Center, but the console has a unique PDC number that callers can use to reach the attendant.

A dedicated Attendant Position is a console that is administered to answer all of the calls not handled by the Message Center.

Message Center capability can be supported in any of the following two-console configurations:

- One dedicated Attendant Position and one dedicated Message Center. A call extended by the Attendant Position to a station that does not answer or is busy returns to the Message Center.
- One dedicated Attendant Position and one combined Attendant Position/Message Center; the Attendant at the combined position also functions as the Message Center

operator for the entire system. A call extended by either Attendant to a station that does not answer or is busy returns to the Message Center.

- Two combined Attendant Position/Message Centers; the Attendants at each combined position also function as Message Center operators. The special call types answered by Message Centers can be divided between the two consoles, or both consoles can answer all types. A call extended by either Attendant to a station that does not answer or in busy returns to the Message Center specified in translation. Three return options are provided: to the 1st attendant, to the 2nd attendant, or to either attendant.

Interactions

All System 25 console features are accessible at a Message Center. Operating procedures are exactly the same as those at a standard Attendant Position. The BIS and HFAI features can be used to answer Message Center calls.

An auxiliary Direct Extension Selector Console can be used with a Message Center. Outgoing calling, from the Basic Console or the Selector Console, is not affected by Message Center administration.

Administration Requirements

A new item in administration allows selection of an alternative set of call type defaults, making one of the attendant positions a “message center”. These defaults may be modified for individual call types, if desired, to tailor the message center-like operation for different locations. For administration purposes, the Message Center call types are divided into the following four groups:

- Nonlogged-in floating PDC calls
- Unassigned DID calls
- Coverage calls
- Returning calls.

With the standard (ie, non-message center) defaults, each of these groups is translated to be directed to all consoles. When both positions are combined Attendant/Message Center consoles, administration can direct specific types to one or both consoles.

Message Center calls are held in the same common queue as any other attendant-seeking calls before being directed to the console.

Hardware Requirements

None.

MESSAGING SERVICES

Description

Lights an LED to indicate that another station (or the attendant) has a message for the user.

The Messaging Services provide light activation/deactivation only. Users must call the sender to receive their messages.

The system supports three types of Message Waiting service:

- Attendant Message Waiting
- Call Coverage Message Waiting
- Station-To-Station Message Waiting

Attendant Message Waiting:

The Attendant can turn On (and turn Off) the Message LED at other voice terminals. When this indicator is lighted, users call the attendant for messages. The LED on multiline terminals may be turned Off by the user (by pressing MESSAGE) or by the attendant. The Message LED on most single-line terminals can only be turned On or Off by the attendant.

Refer to the Attendant Message Waiting feature description for additional information.

Call Coverage Message Waiting:

Allows a user providing Individual Call Coverage to control the Message LED on covered voice terminals. A Coverage Message button (COVER MSG) is used to display and control the status of the covered user's Message LED. The state of the COVER MSG LED reflects the state of the covered station's Message LED. The covering user can turn On or Off (toggle) the covered party's Message LED at any time during a coverage call by pressing COVER MSG. To turn On the covered user's Message LED when not on a coverage call, the covering user may go off-hook on a System Access button, press COVER MSG and then dial the covered user. The covered station's Message LED turns On if Off and stays On if On. If the covering station then presses COVER MSG a second time before hanging up, the Message LED will turn Off.

A covered party must dial the covering party to retrieve their messages. Multiline voice terminal users can press MESSAGE to turn Off their Message LED. Message indicators on most single-line voice terminals can only be controlled by the covering party or the attendant.

If a user tries to turn On the Message LED at a voice terminal for which they don't provide Individual Call Coverage, they receive Reorder Tone.

Refer to the Call Coverage-Individual feature description for additional information.

Station-To-Station Message Waiting:

Multiline voice terminals can be assigned (paired) Message Waiting (MSG WAIT) buttons with associated status LEDs. When this indicator is lighted, the user calls the other user for messages. The MSG WAIT LED can be controlled by the two associated terminals *only*; either user can toggle the state of both LEDs (e.g., both LEDs go On or Off together) at any time, whether on-hook or off-hook.

Refer to the “Station-To-Station Message Waiting” feature description for additional information.

Considerations

The Attendant and Coverage Message Waiting features light the same “basic” Message indicator on each set. The Station-To-Station feature may be assigned to programmable (MSG WAIT) buttons between two sets; it lights the LED next to the button.

Interactions

None

Administration Requirements

Attendant Position (Station Port):

- Assign ATT MSG button (defaulted)

Voice Terminals (Station Port):

- Individual Call Coverage Message Waiting - assign Coverage Message (COVER-MSG) button.
- Assign Individual Call Coverage (COVER-IND) between sets.
- Station-To-Station Message Waiting - assign Station Message Waiting (MSG WAIT) buttons. Two (multiline) terminals must share this feature.

Hardware Requirements

Z34A Message Waiting Indicator for single-line voice terminals not equipped with Message LEDs.

MODEM POOLING

Description

Allows switched data connections between digital data endpoints and analog data endpoints.

(Refer also to the description of the system's data features provided in the "Data Services Overview" subsection.)

Data transmission between digital and analog endpoints requires a conversion resource since the digital format used by the data module is not compatible with the modulated signals of an analog modem. The conversion resource translates the digital signals from the digital endpoint into analog signals and vice versa.

The modem pool is a single group of up to 12 conversion resources (3 Cabinet system) with the characteristics of a 212A full duplex asynchronous modem that can operate at speeds of 300 and 1200 bps.

The Modem Pooling feature operates transparently to the user whenever possible. The system adds a conversion resource to a connection when a digital endpoint is connected to an analog trunk or port without any explicit action by the user.

A voice terminal user who plans to use an analog modem to call a digital endpoint must first enter the Modem Request Code before dialing the digital endpoint. This is because the system assumes that a voice call to a digital endpoint will be transferred to data via the Transfer To Data feature.

A DID call terminating on a digital endpoint will be reassigned a modem resource, if available. Otherwise, the call receives Reorder tone.

For each situation that requires a conversion resource, the system:

1. Determines if a resource is required by examining the types of endpoints that are to be connected together or by user indication.
2. Once it is determined that a conversion resource is needed, it is reserved. The user receives Reorder Tone (or the "NO MODEMS - TRY AGAIN" message) if a resource is not available. The system queries the data port to determine whether its options are compatible with those supported by the modem pool. If they are not (e.g., 9600 baud and Permit Mismatch is not enabled), the originating user receives intercept treatment (i.e., INCOMPATIBLE FAR END) and call setup is abandoned.
3. At data connection time, the conversion resource is seized and placed in the connection.
4. The call is disconnected within 15 seconds if the conversion resource does not successfully handshake with both endpoints.

Conversion resources are required for:

- **Data Terminal Dialing:** To establish a data connection for calls originated via Terminal Dialing to intrapremises analog data endpoints.
- **Incoming Trunk Calls:** To establish a data connection between an incoming trunk call and a digital endpoint. Incoming trunk calls that are answered at a voice terminal can be transferred to a data endpoint using the Transfer To Data feature.

- **On-Premises Data Calls:** To establish a data connection between an on-premises analog data endpoint and an on-premises digital endpoint.
- **Outgoing Trunk Calls:** To establish a data connection between an off-premises analog endpoint (modem) and an on-premises digital endpoint.

Considerations

Modem Pooling provides a pool of conversion resources that increases data call flexibility. Conversion resources allow analog data endpoints, using modems, to communicate with digital data endpoints (using data modules). Also, modem pooling reduces costs by sharing resources.

Interactions

- **Automatic Route Selection:** Data calls may be queued. See Table 2-H, “PLEASE WAIT” message.
- **Calling Restrictions:** If a terminal is toll or access restricted, the modem resource is released when the user receives intercept treatment.
- **Direct Group Calling:** Modem pooling supports calls to data endpoints that are part of a DGC group. While an incoming data call is in the DGC group queue, the caller hears Ringback Tone. The conversion resource is inserted if the call is completed to a digital endpoint.
- **Station Hunting:** Modem Pooling supports calls to data endpoints that are part of a station hunting group.
- **SMDR:** SMDR records do not reflect modem resource usage. Interpremises data calls using a conversion resource are reported as data calls on the SMDR call record.

Administration Requirements

System (Pooled Modems):

- Modem Request Code (1-9999) - Default = 820. Allows users to indicate a need for a conversion resource on a data call originated at an analog data endpoint.
- Receiver Responds To Remote Loop (Yes, No) - Default = Yes. When active, Data Set Ready is asserted when the modem is in an analog loop test mode.
- Disconnect On Loss Of Carrier (Yes, No) - Default = Yes. When active, a loss of the received carrier will cause the modem to terminate the call.
- CF-CB Common (Yes, No) - Default = Yes. When active, Clear to Send turns off if Carrier Detect turns off. When a call is being established, Clear to Send and Carrier Detect are not allowed to turn on until carrier has been received and the Clear to Send timer has timed out.
- Disconnect On Received Space (Yes, No) - Default = Yes. When active, the modem will disconnect after receiving a “Space” signal of approximately two seconds duration.
- Send Space On Disconnect (Yes, No) - Default = Yes. When active, the modem, upon receiving a negation of Data Terminal Ready, sends approximately four seconds of “Space” signal and then disconnects. Without this option active, the modem, upon receiving a negation of the Data Terminal Ready signal, disconnects immediately.

Hardware Requirements

One TN758 Pooled Modem CP provides two conversion resources. Two TN758s are allowed per system cabinet, for a total of 12 conversion resources in a 3-cabinet system.

MUSIC-ON-HOLD

Description

Provides music or other audible indication to a held party on an outside line.

On an outside call, if the user places the call on Hold, or after a call into a DGC group receives the delay announcement, music is provided to the calling party.

Music-On-Hold is not invoked when a conference call is placed on hold or when the attendant "Start" facility is used to place a call on hold.

When a multiline voice terminal user places a call on hold, the status LED of the held line winks and music is provided to the held party. The user may return to the held party by pressing the button associated with the held call. The status LED lights steadily, music is removed from the line, and a talking connection is again established.

When a single-line voice terminal user places a call on hold by pressing the switchhook momentarily, the calling party is connected to music or a recording. The station may return to the held call by pressing the switchhook a second time. The music is removed from the line and the held party is reconnected to the user.

If a caller receives music because all members of a DGC group are busy, when a group member becomes available to answer the call, music is removed and the calling party is connected to the DGC member.

Music-On-Hold is not provided for data calls or inside calls.

Considerations

Music-On-Hold lets the waiting party know that he or she is still connected.

During the process of transferring a call or setting up a conference, the affected parties are placed in a "special hold" condition. In V1 systems, the parties on special hold will receive music. In V2, an administration option allows Music-On-Hold to be enabled or disabled for "Special Hold."

Interactions

- **Attendant Splitting One-Way Automatic:** Music-On-Hold is not provided when the attendant presses START.
- **Conference:** An outside line placed on hold when CONFERENCE is pressed will hear music if provided.
- **Call Park:** Parked calls (except parked conferences) receive music.
- **Direct Group Calling:** An incoming call to a busy DGC group that provides a recorded delay announcement will receive music after the announcement.

Administration Requirements

Assign a port on a TN742 Analog Line or ZTN78 Tip Ring Line CP as required for the music/message source (special feature port type = 254).

For V2 only: allow Music-On-Hold for special hold (Y/N)

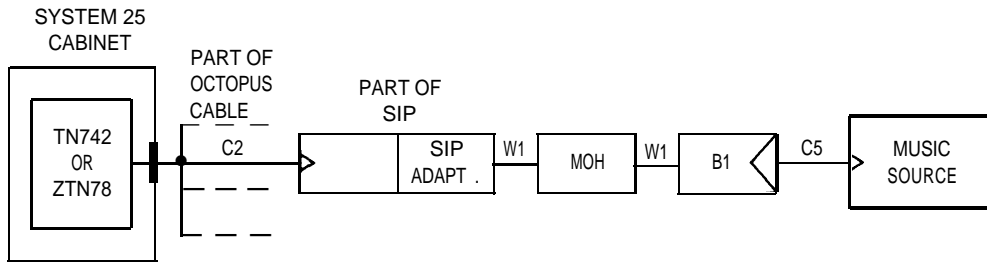
Hardware Requirements

A music source is needed to support the Music-On-Hold feature. The interface is a port on a ZTN78 Tip Ring Line CP or TN742 Analog Line CP.

Detailed connection information is provided in the following figures:

- Figure 2-30—Music-On-Hold Equipment Connections (FCC Registered)
- Figure 2-31—Music-On-Hold Equipment Connections (Non-Registered).

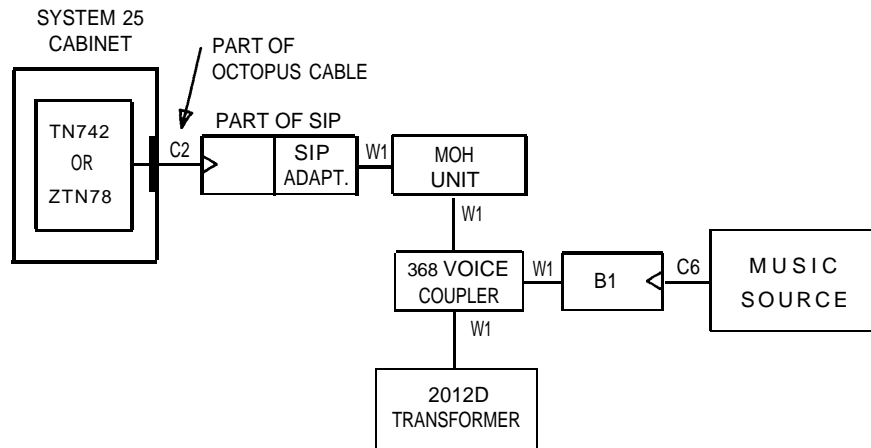
Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters, as shown on the figures, are provided under the heading “connectivity” in Section 4.



LEGEND:

- TN742 - ANALOG LINE CP
- ZTN78- TIP RING CP
- B 1- TYPICAL-103A CONNECTING BLOCK*
- C 2- OCTOPUS CABLE (WP90780) PEC 2720-05P
- C5 - MODULAR CORD (D4BU-87)
- W 1- 4 PAIR INSIDE WIRING CABLE*
- MOH - KS-23395 INTERFACE PEC 62517
- * - FURNISHED BY INSTALLER

Figure 2-30. Music-On-Hold Equipment Connections (FCC Registered)



LEGEND :

TN742 - ANALOG LINE CP

ZTN78 - TIP RING CP

B 1- TYPICAL-103A CONNECTING BLOCK*

C2 - OCTOPUS CABLE (WP90780) PEC 2720-05P

C6 - SINGLE-ENDED MODULAR CORD, D4BY

W1- 4 PAIR INSIDE WIRING*

PEC 62513 INCLUDES KS-23395 MOH UNIT, 2012D TRANSFORMER
AND 36A VOICE COUPLER

* - FURNISHED BY INSTALLER

Figure 2-31. Music-On-Hold Equipment Connections (Non-Registered)

NIGHT SERVICE

Description

Allows users to answer incoming calls on specified trunks when the attendant is not on duty. There are two types of Night Service (NS):

- **Directed NS:** Redirects incoming calls on specified trunks to designated voice terminals.
- **Trunk-Answer-from-Any -Station (TAAS) NS:** Allows users to answer incoming calls on specified trunks by dialing the Night Service access code.

Both types of NS may be provided (specified on a per-trunk basis).

To obtain Night Service, the system must be equipped with an Attendant Console, and the console administered with a NIGHT button. In a system with two Attendant Consoles, both consoles may be assigned a NIGHT button. Either attendant can press NIGHT to activate Night Service. The LEDs of both NIGHT buttons will light to indicate that the system is in the Night Service mode. Pressing NIGHT a second time (by either attendant) deactivates Night Service and turns Off both LEDs.

Directed NS:

Allows an incoming trunk call to be directed to up to four designated voice terminals. Different trunks may be directed to different voice terminals.

When the attendant presses NIGHT, incoming calls on trunks administered to receive Directed NS treatment will automatically be routed to the designated voice terminals (all designated NS stations ring simultaneously). Calls not answered within a specified number of rings will receive a Night Service delay announcement, if available. While at the announcement, they may be bridged onto by going off-hook at a station with a line appearance. The announcement is dropped at this point. If all Directed NS stations for a given trunk are busy (both System Access buttons busy on multiline sets), calls go to the announcement immediately. Directed NS calls do not hunt or receive call coverage, but they can be picked up via the Call Pickup feature.

Personal Line calls that are directed to NS will also ring at the Personal Line appearances and receive normal call coverage.

Incoming calls receiving Directed NS treatment will not activate external alerting devices associated with TAAS NS and cannot be answered by dialing the NS access code.

Directed NS is activated under the following conditions:

- An attendant has pressed NIGHT on either console.
- Directed NS has been administered for the trunk.
- Stations have been administered to receive NS calls.

Note that at least one station must be designated as a NS receiver for this feature to work properly. If only an announcement is required, administer the announcement device as a station and make this station the NS receiver.

Refer to the “Night Service Delay Announcement” feature description for additional information on the delay announcement.

Trunk-Answer-from-Any-Station:

Allows any user to answer NS calls. Incoming trunk calls activate an external alerting device such as a bell (“External Alerts” feature). A user can then dial the NS access code and answer the call. Night Service is activated under the following conditions:

- An attendant has pressed NIGHT on either console.
- TAAS NS has been administered for the trunk.
- A NS external alert has been installed and administered.

Note that TAAS NS calls *will not* activate the delay announcement associated with Directed NS.

Considerations

Directed NS provides a means of ensuring that Night Service calls are answered by designating individual voice terminals to receive the calls. In noisy environment, for example, NS via external alerting devices may not be practical. Directed NS provides a solution to the noise problem. Also, Personal Line calls to executives can receive special handling by providing Directed NS. Calls continue to ring at the attendant position or Personal Line appearances when NS is activated. They also ring the external alert (TAAS) or Directed NS station.

Trunk-Answer-from-Any -Station provides the capability for any user to answer NS calls.

Interactions

- **DID Trunks:** DID trunks are not assignable to NS. A DID call will ring at the appropriate station whether NS is activated or not.
- **Tie Trunks:** Dial-in Tie Trunks will not receive night service treatment.
- **Call Following:** Directed night service calls will not be given call following treatment if the PDC is logged in at another station.

Administration Requirements

Trunk Ports:

- Assign trunk Class of Service with Night Service - (8-15)
- Assign Directed Night Service trunk (Yes, No) - Default = Yes.

Voice Terminal: (Station Port)

- Directed NS
 - Add Night Service trunk number to station list.
- Assign External Alert for TAAS NS.

Attendant Console: (Station Port)

- Night Service is defaulted to a button on the first (VI: the primary) Direct Trunk Attendant Console only. On a second (VI: secondary) Direct Trunk Console or on a Switched Loop Attendant Console (V2 only), assign Night Service to a flexible button.

- Assign Night Service Access Code.

Hardware Requirements

TAAS NS requires an associated external alert (such as a bell). Each alert requires a port on a ZTN78 Tip Ring Line or a TN742 Analog Line CP. Refer to the “External Alerts” subsection for detailed information and a connection diagram.

NIGHT SERVICE DELAY ANNOUNCEMENTS

Description

Provides a recorded announcement for incoming trunk calls when the system has Directed Night Service (NS) activated and a call is not answered.

Directed NS calls not answered within a specified number of rings (1-15) may be directed to a recorded announcement. Two different recorded announcements may be assigned.

Note that NS calls to a terminal that are not answered do not receive Station Hunting or Call Coverage treatment (unless the trunk also appears on a station's Personal Line button). After the announcement is played, the call is disconnected.

Considerations:

Night Service Delay Announcements provide the calling party with a message that acknowledges the call and can provide additional information as well. Once an NS call goes to the delay announcement, the call will be disconnected from the system after the announcement has been played.

Interactions

- **Night Service:** Incoming calls receiving TAAS NS treatment will not activate the delay announcements. Only trunks that receive Directed NS will activate these announcements.

Administration Requirements

Special Feature Ports:

- Assign first Night Service delay announcement (code 251)
- Assign second Night Service delay announcement (code 252)
- Assign number of rings before Night Service delay announcement (0-15)

Station Ports

- Assign port circuits (ZTN78 or TN742) for each recorded announcement.

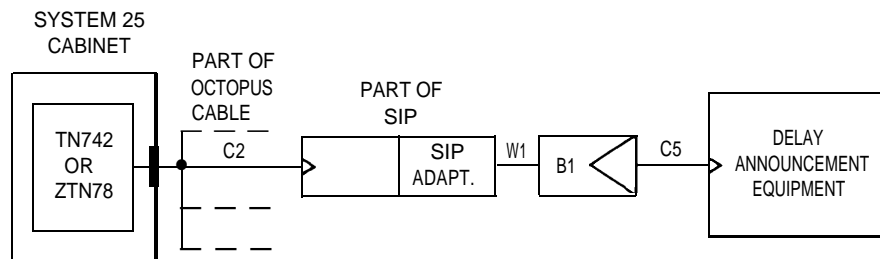
Hardware Requirements

An AT&T Answer-Record 2500 or a Code-A-Phone 2540 or equivalent announcement machine is required for each delay announcement. The equipment requires a port on a ZTN78 Tip Ring Line (or TN742 Analog Line) CP.

The system supports two Directed Night Service delay announcements.

Detailed connection information is provided in Figure 2-32.

Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters, as shown on the figures, are provided under the heading "connectivity" in Section 4.



LEGEND :

- TN742 - ANALOG LINE CP
- ZTN78 - TIP RING CP
- B1- TYPICAL-103A CONNECTING BLOCK*
- C2 - OCTOPUS CABLE (WP90780) PEC 2720-05P
- C5 - MODULAR CORD (D4BU-87)
- W1- 4 PAIR INSIDE WIRING CABLE*
- * - FURNISHED BY INSTALLER

Figure 2-32. Delay Announcement Equipment Connections (FCC Registered)

OFF-PREMISES STATIONS (OPS)

Description

An OPS is a single-line voice terminal that is located in another building and connected to System 25 via arrangements with the local CO. The station has the same features as an on-premises single-line station except that it is counted as an outside party in a conference call. Also, the Message feature will not operate with these sets.

Considerations

This service is sometimes furnished to executives at their residences. It allows them remote access to System 25 features and services.

Interactions

- **Conference:** For conference purposes, an OPS counts as one of the two allowable outside lines.
- **Distinctive Ringing:** For V2 systems, Distinctive Ringing is not provided; OPS stations will always receive standard (that is, single) ringing for calls.

Administration Requirements

Single-Line Voice Terminals (Station Port):

- Assign port on Analog Line (TN742) CP
- Make This an Extended Station (Yes/No) - Default = No. (This is how the system knows the station is an OPS.)

Hardware Requirements

Requires a port interface on a TN742 Analog Line CP.

The OPS must be a FCC registered single-line voice terminal.

Connectivity information is provided in Section 4, "Hardware Description."

OUT-OF-BUILDING STATIONS

Description

Single-line voice terminals and multiline 7300H series terminals may be directly connected to the system even though they are not located in the same building. For 7300H series terminals special In-Range Out of Building (IROB) units are used to protect the switch and its users from lightning, power crosses, etc. Out-Of-Building Stations can access all system features.

Considerations

Single-line voice terminals may be located at distances up to 24000 feet from the system cabinets. Carbon protection devices are required for lightning and power cross protection.

Multiline voice terminals must be located within 2000 feet of the system cabinets (local power required beyond 1000 feet) and must have IROB protection devices. MET sets may not be used for Out-Of-Building service.

Interactions

None

Administration Requirements

None

Hardware Requirements:

Out-Of-Building multiline voice terminals require IROB units. Single-line voice terminals require carbon protection devices and must be connected to ports on the TN742 Analog Line CP. Connectivity information is provided in Section 4, "Hardware Description."

PAGING SYSTEM ACCESS

Description

Provides users with dial access or feature button access to paging equipment.

As many as three paging zones can be provided, each with its own access code. (A zone is the location of paging loudspeakers, for example, conference rooms, warehouses, or storerooms.) In addition, one access code can be provided to activate all zones.

A single-line or multiline voice terminal user (including the attendant) can access paging equipment by dialing the zone access code. Multiline users can press a Flex DSS or DSS button that has been programmed with the paging access code.

A paging zone maybe administered to be dial restricted. This restricts users from accessing the equipment unless they are assigned an access button.

A PagePac* paging system may be used. Some paging systems require only one port assignment to support all zones. Other systems may require separate ports for each zone. Compatible PagePac paging systems include:

- PagePac 20 (Powermate)

This is the smallest PagePac system. The basic system provides a single paging zone with an input source for background music. An Auxiliary Trunk port interface is required for this system.

- PagePac 20 Control Unit and Zone Mate 9 (includes Common Control Unit)

Allows a user to dial one paging access code and then dial a single zone or all zones code (1-9) to access a paging zone. This system is equipped with a Control Unit and connects to a Ground Start or Loop Start trunk port.

PagePac equipment is easy to use. A user simply dials the paging access code and receives Confirmation Tone. If the equipment provides just one zone, the user then makes the announcement. If the equipment provides multiple zone access, the user, after hearing Confirmation Tone, dials a code to access the desired zone(s) before paging.

Considerations

Paging is particularly useful when used in conjunction with the Call Park feature. When a user is away from his or her location and receives a call, the call can be answered and parked by another user. The called party can then be paged and told what extension number to call to retrieve the parked call. The called party can then retrieve the call from any voice terminal.

If PagePac multi-zone equipment is used, only one port assignment is required.

* Trademark of Harris Corporation Dracon Division

Interactions

- **Direct Inward Dialing:** A DID call may access a paging code. This allows the user to dial in and utilize the Paging System Access feature. Dial restricting the paging code will block this interaction.

Administration Requirements

Special Feature Ports (Auxiliary Trunk Interface):

- Assign Paging access code for each paging zone (maximum = 3) to be provided
- Assign All Zones access code
- Dial restrict zone (Yes, No).

Multiline Voice Terminals (Station Port):

- Assign DSS button with paging zone access code as required.

Trunk Ports:

- Requires a port on a ZTN76 Ground Start Trunk or ZTN77 Loop Start Trunk CP for each port interface required. If the paging equipment requires a contact closure, a port on a TN763 Auxiliary Trunk CP is required instead of the ZTN76 or ZTN77.
- Assign Trunk Access Code
- Dial restrict zone (Yes, No)
- Assign other appropriate CO trunk parameters.

Hardware Requirements

Requires a PagePac or other compatible paging system. Also requires Auxiliary or CO trunk ports.

Paging may occur in up to three zones or in all zones at once. The recommended method of interfacing is via CO trunks (either loop or ground start) using a PAGE-PAC 20 equipped with Zone Mate 9; this requires only one trunk port interface.

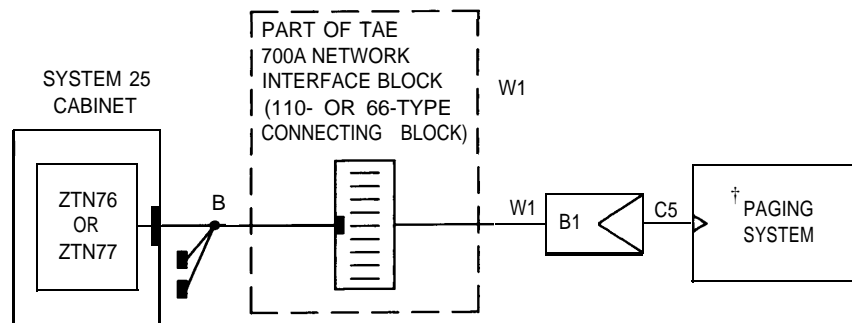
The following PAGE-PAC units may be used:

- PAGE-PAC-Family equipped with Common Control Unit (one zone only)
- PAGE-PAC-Family equipped with Zone Mate 9.

Some customer-provided equipment may require separate ports on the Trunk CPs for each zone. The ZTN76 Ground Start Trunk or ZTN77 Loop Start Trunk CP is the preferred paging equipment interface. If the paging equipment requires a contact closure, a TN763 Auxiliary Trunk CP Pack must be used.

Detailed connection information is provided in Figures 2-33 and 2-34.

Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters, as shown on the figures, are provided under the heading "Connectivity" in Section 4.



LEGEND :

ZTN76 - CO GROUND START TRUNK CP

ZTN77 - CO LOOP START TRUNK CP

B - 3 TO 1 SPLITTER CABLE-CONNECTORIZED (OR6016) PEC 2720-06X

B1 - TYPICAL-103A CONNECTING BLOCK*

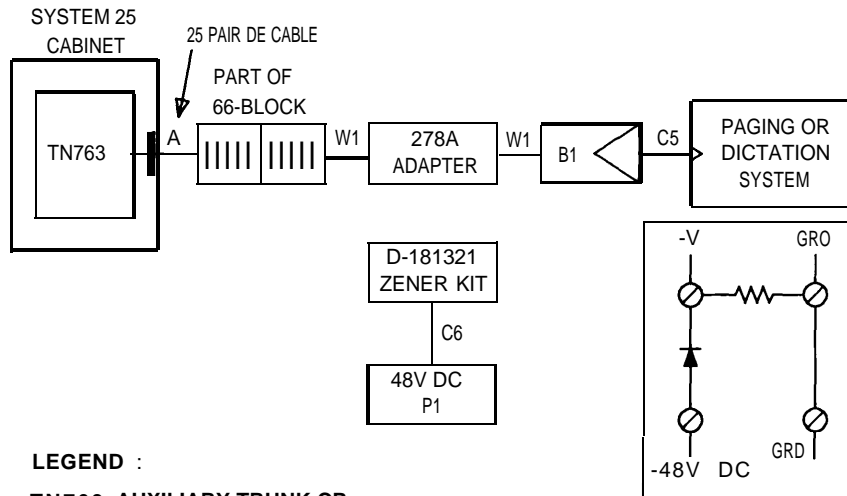
C5 - MODULAR CORD (D4BU-87)

W1 - 4 PAIR INSIDE WIRING CABLE*

* FURNISHED BY INSTALLER

† PAGING SYSTEM - PAGE PAC 20 E/W ZONE MATE 9 - PROVIDES 9 PAGING ZONES, PLUS ALL-ZONE PAGING

Figure 2-33. Paging Equipment Connections Using CO Trunk Ports (FCC Registered)



LEGEND :

TN763- AUXILIARY TRUNK CP

A - SINGLE-ENDED 25 PAIR CONNECTOR CABLE (A25D) (NOTE 1)

B1 - TYPICAL - 103A CONNECTING BLOCK*

C5 - MODULAR CORD (D4BU-87)*

C6 - SINGLE-ENDED MODULAR CORD (DYB4) (NOTE 2)

P1 - KS-22911, L1, POWER SUPPLY, 48 VOLT DC (NOTE 2)

W1 - INSIDE WIRING CABLE (4-PAIR)*

NOTES :

1. APPARATUS CODE D-181523 (PEC 62511) INCLUDES

66E3-25 BLOCK CONNECTOR AND CABLE B25A 15/DE.

2. APPARATUS CODE D-181524 (PEC 62512) INCLUDES C6, P1, 278A

ADAPTER AND ZENER KIT.

* FURNISHED BY INSTALLER

Figure 2-34. Paging Equipment Connection to TN763 CP Using 278A Adapter

PERSONAL DIAL CODE (PDC)

Description

Each station is assigned a PDC. The user may log in the PDC at any other voice terminal, and calls to the PDC will follow the user.

A PDC can be assigned to a convenience voice terminal (i.e., not associated with a particular user) and to data terminals with modems. Digital data endpoints are assigned Data Dial Codes (DDCs).

There are two types of PDCs:

- **PDCs:** Assigned to users with their own voice terminal (referred to as the “home terminal”). To use the PDC at another voice terminal, the PDC may be logged-in there.
- **Floating PDCs (FPDCs):** Assigned to employees who do not have their own voice terminal and to visitors who will be receiving calls. Calls to an FPDC will ring at the terminal where it is logged in, or, optionally, at the Attendant Console if the FPDC is not logged in anywhere.

Up to 200 PDCs and 300 FPDCs can be assigned in a system.

Note: The following call types are station oriented. They do not redirect to an “away terminal.”

- Automatic Intercom Calls
- Directed Night Service Calls.
- DSS Calls (V1 only)
- DGC Group Calls
- Manual Signaling
- Message Waiting
- Personal Line Calls

Calls Placed to A PDC:

If the PDC is either logged in at its home terminal or is not logged in anywhere, a call to the PDC will simply be directed to the home terminal and will receive that terminal’s normal hunting or call coverage treatment.

If the PDC is logged in at another terminal, then that terminal is termed the “away terminal.” There are two cases to consider when a call is placed to this PDC:

- Call not placed from the away terminal (the general case)
- Call placed from the away terminal (a special case).

(1) Call Not Placed from The Away Terminal (the general case): The call will first be directed to the away voice terminal. Ringing will occur at the away terminal if it is an on-hook single-line voice terminal or if it is a multiline voice terminal with an idle System Access button.

A call unanswered at the away terminal will be directed back to the home terminal unless one of the following busy conditions exists at the home terminal: (1) it is a multiline terminal with both System Access buttons busy and with no idle call coverage receiver or (2) it is a single-line voice terminal that is off-hook, has no idle Call Coverage receiver, and has no idle hunt-to station. While either of these busy conditions exists at the home terminal, the call will not return to the home terminal. Instead, it will continue to ring at the away terminal until answered or timed out.

If either of the above two busy conditions ends at the home terminal while the call is waiting at the away terminal, the call will be directed back to the home terminal. If the call is sent back to the home terminal, it can be answered or it can receive the terminal's hunting or call coverage treatment. The call coverage treatment given to calls that are returning from an away terminal differs in two respects from the treatment provided to calls initially directed to the home terminal. First, the home terminal and its coverage station receive ringing simultaneously, rather than having the coverage ringing delayed. Second, coverage terminals will ring for calls returning to the home stations even if the call coverage ring options of the home terminal are "no ring." Both of these call coverage modifications expedite the answering of calls that are returning to the home station from an away station.

Once the call is directed back to the home terminal, it is removed from the away terminal. This is true even if the away terminal was busy but subsequently became idle after the call was sent back to the home terminal.

(2) Call Placed From The Away Terminal: A call to a PDC from the station where it is logged in will be directed to the PDC's home terminal.

Calls Placed to a Floating PDC:

When a valid FPDC is dialed, the call will be directed to the terminal where the FPDC is logged in and will be provided the coverage treatment administered for that terminal. If the FPDC is not logged in and if the attendant position is that FPDC's "home," then the call will be directed to the attendant position. However, if the FPDC call was placed from the attendant position, then it will not be redirected to the attendant but will instead be provided Reorder Tone. Finally, if the FPDC is not logged in, and if the attendant position is not that FPDC's "home," then the calling party will receive Reorder Tone.

For non-DID calls, if an invalid FPDC is dialed, then the calling party will receive Reorder Tone. If a DID call does not match any assigned number in the dialing plan, it will be directed to the attendant or to Reorder Tone, as administered.

Considerations

The Personal Dial Code (PDC) feature provides flexibility for users and visitors. Visitors, once assigned a FPDC, can inform callers and the attendant. Calls can then be directed to the voice terminal where the FPDC is logged in. Calls to FPDCs not logged in may be directed to the attendant for further handling.

Interactions

- **Call Coverage:** Calls to a logged-in FPDC receive the call coverage of that terminal. Unanswered calls to a PDC at an away terminal return to the home terminal and receive the home terminal's call coverage treatment; they do not receive the away terminal's call coverage.
- **Direct Inward Dialing:** In systems with DID service, PDCs, FPDCs, DGC group numbers, DDCs codes, and facility access codes may match the last 2, 3 or 4 digits of DID numbers. For example, the code matching DID number 555-2345 may be 45, 345 or 2345, depending on the system dial plan.
- **Direct Station Selection:** If an attempt is made to program an FPDC (rather than a PDC) on a Flex DSS button, Reorder Tone is received.

Administration Requirements

System:

- Send DID calls to unassigned DID numbers to the Attendant - Default = Yes
- Route calls to not-logged-in FPDCs to the attendant - Default = Yes
- Add/Delete FPDCs.

Hardware Requirements:

None

PERSONAL LINES

Description

Provides a dedicated outside line for multiline voice terminal users.

Unlike pooled facilities, which can be accessed via dial codes, Personal Lines can be accessed only via a dedicated feature button, and provide both incoming and outgoing service. Up to 16 terminals may share a Personal Line (up to 8 in R1V1). Up to four parties may be off-hook on the line at the same time (the line itself is the fifth conferee). When the line is busy, its status LED lights at all terminals on which the line appears. Ringing may be provided optionally to one or more of the terminals sharing the line.

For each Personal Line, one station is administered as the principal (owner). The call coverage of that terminal determines the call coverage of the Personal Line.

Considerations

Personal Lines provide facilities to users who desire direct access to the exchange network. In addition, Personal Line appearances are provided on the Direct Trunk Attendant Console for general use trunks. Appearances of these lines may also be provided at selected multiline voice terminals to ensure call coverage when the attendant is not available. DID trunks cannot be terminated on Personal Line buttons.

Personal Lines provide direct access for callers, bypassing the attendant. In some cases, they may substitute for DID service.

Interactions

- **Attendant Console, Direct Trunk:** On the Direct Trunk Attendant Console (DTAC), trunks are terminated as Personal Lines. The DTAC can accommodate a maximum of 26 Personal Lines (24 is the practical limit).
- **Attendant Console, Switched Loop:** Personal Lines cannot be terminated on a Switched Loop Attendant Console.
- **Call Coverage:** The call coverage of the principal station (owner) determines call coverage for the line.
- **Call Park:** A parked Personal Line is bridgeable by any user with a button appearance of that line. Bridging on to the connection does not unpark the call. The parked call will not return to the parking user.
- **Call Pickup:** After a call is picked up from a Personal Line button, the called terminal can still enter the call.
- **Direct Group Calling:** The same trunk may be used as a Personal Line and also be directed to a DGC group. If an incoming call is not answered by the DGC group after a predetermined number of rings, ringing and LED flashing will be transferred to all button appearances of the line (unless a DGC delay announcement is provided).
- **Hold:** A Personal Line cannot be placed on hold if any other stations that share the line are also off-hook on the line.

- **Line Selection (Prime Line Preference):** Prime Line Preference may be assigned to a Personal Line.
- **Pooled Facility:** A Personal Line may also be a member of a pooled facility group.
- **Toll Restriction (see “Calling Restrictions”):** A call over a Personal Line is subject to the toll restrictions of the station on which the call was placed.

Administration Requirements

Voice Terminal (Station Port):

- Assign Personal Line trunk number
- Assign Personal Line feature button
- Make this the Principal Station (Yes, No)
- Enable Personal Line Ringing (Yes, No).

Hardware Requirements

Requires port assignments for each trunk interface to be provided and a button termination on multiline voice terminals.

POOLED FACILITY - DIAL ACCESS

Description

Allows both multiline and single-line voice terminal users to access a common pool of trunks by dialing a facility access code.

Up to 16 facility access codes can be assigned (one per trunk group). The codes can be one to four digits in length. A group of similar trunks assigned the same access code is referred to as a trunk group. Additional information is provided in the “Trunk Groups” feature description.

After going off-hook on a System Access or Loop button, receiving system dial tone, and dialing a facility access code, the user will be connected to an idle trunk. However, the connection will not be made if the terminal is restricted from dialing this trunk group or if dial access is restricted, in general, to trunks in the group. The LEDs associated with the System Access button will be lighted, and the user may complete the call. Single-line users do not receive LED indications of the status of the pool. An attempt to originate a call on a busy facility will result in Reorder Tone (fast busy).

Considerations

Pooled Facility-Dial Access provides users of single-line terminals, or multiline voice terminals without Direct Facility Access (FACILITY) buttons, access to the system’s pooled facilities.

Interactions

- **Direct Inward Dialing (DID):** Access to pooled facilities via DID is permitted. This includes access to WATS, FX, tie trunks, private lines, dictation equipment, and paging systems. This access is provided by selecting facility access codes so that they will match DID numbers.
- **Outward and Facility Access Restriction (see “Calling Restrictions”):** A terminal can be denied dial access to some or all pooled facilities, or may be totally restricted from making any outside calls.
- **Toll Restriction (see “Calling Restrictions”):** Denies the use of pooled facilities for certain toll calls, but does not block access to the pooled facilities.

Administration Requirements

Voice Terminal (Station Port):

- Restrictions - Refer to “Calling Restrictions” feature description.

Trunk Port:

- Assign facility access codes
- Allow dial access for facility (Yes, No).

Hardware Requirements

None

POOLED FACILITY - DIRECT ACCESS

Description

Allows multiline voice terminal users to access a common pool of trunks via a Direct Facility Access (FACILITY) button.

Upon pressing a FACILITY button and going off-hook, a multiline voice terminal user is connected to a common pool of outside trunks (i.e., CO, FX, WATS, tie). If the Status and I-Use LEDs associated with the button light steadily, the user may complete the call. If no idle trunk is available (facility busy indication), an attempt by the user to originate a call will be denied and the I-Use indicator will be Off. A user requiring access to several different trunk pools must have a separate FACILITY button for each pool.

If all trunks in a pool are busy, the Status LED will be lighted. The user may press FACILITY and remain on-hook to receive the busy-to-idle reminder when a trunk becomes available. The busy-to-idle reminder is a short burst of tone that will be heard when a trunk in the pool becomes available. When Prime Line Preference is assigned to a FACILITY button, the button must be pressed to invoke the busy-to-idle reminder, even though the I-Use LED is lighted.

Refer to the “Trunk Groups” feature description for additional information.

Considerations

Pooled Facility-Direct Access provides easy access to the exchange network for users who make many outside calls. The feature eliminates the need to dial a facility access code. In addition, the associated status LED provides pool busy/idle status and the busy-to-idle reminder.

Interactions

- **Automatic Route Selection (ARS):** Multiline voice terminal users who have pressed FACILITY to activate busy-to-idle reminder must wait until all queued ARS users have been serviced.
- **Facility Access Restriction (see “Calling Restrictions”):** A trunk group may be reserved for a group of users by dial-access restricting the trunks. In this way, only users who have a FACILITY button, a Personal Line appearance, or who use ARS can use the trunks.
- **Line Selection (Prime Line Preference):** Pressing a FACILITY button to invoke the busy-to-idle reminder overrides Prime Line Preference.
- **Toll Restriction (see “Calling Restrictions”):** Toll-restricted voice terminals receive standard toll restriction treatment on all FACILITY buttons.

Administration Requirements

Voice Terminal (Station Port):

- Assign Direct Facility Access (FACILITY) buttons.

Trunks:

- Assign Facility Access Codes.

Hardware Requirements

None

POWER FAILURE TRANSFER (PFT)

Description

Provides service to and from the CO for a limited number of prearranged *single-line* voice terminals during a commercial power failure (or when voltage drops below 90 volts for longer than 250 milliseconds) and during other service interruptions. Any loop start or ground start trunk may be arranged to terminate at a specific station on a one-to-one basis. When a failure occurs, these prearranged connections are made, bypassing the system and connecting terminals directly to the CO trunks. System features and restrictions are not available during this time.

The system supports up to four Emergency Transfer Units (ETUs). Each ETU can provide up to five voice terminals with direct connection to CO trunks.

When the system connects to dial pulse trunks, only rotary sets may be used to support Power Failure Transfer (PFT). When the system interfaces the CO via touch-tone trunks, touch-tone single-line voice terminals are used as PFT stations.

When power is restored, the following will be restored to their previous state:

1. Night Service mode (on or off).
2. User-programmed Flex DSS numbers.
3. PDCs logged in at a "home station" or an "away station" remain logged in there.
4. If a voice terminal has been removed/not removed from a DGC group, the terminal will remain in that state.
5. User-programmed Repertory Dialing numbers.
6. All system/station features programmed through system administration.

Considerations

Power Failure Transfer provides emergency incoming and outgoing telecommunications service to a number of predesignated single-line voice terminals. This is particularly important for organizations providing public services such as fire, police, medical, etc.

Interactions

None

Administration Requirements

None

Hardware Requirements

The 10B Emergency Transfer Unit (ETU) in Figure 2-35 supports up to five Power Failure Transfer (PFT) sets and a DID make-busy function. Up to four ETUs can be supported for a maximum of 20 PFT sets. The sets can be connected to selected Loop Start or Ground Start

trunks. If Ground Start trunks are used, a 55A1 Ground Start Button must be provided at each PFT set.

Only FCC registered single-line voice terminals may be used for PFT stations. Rotary sets must be used for dial pulse PFT trunks; touch-tone sets must be used for touch-tone PFT trunks.

In the event of a Power Failure Transfer (switch has lost power or a major fault has occurred) a contact closure is provided to the Central Office (CO) over a dedicated pair of wires. The CO then makes busy all DID trunks. When power is restored, the closure is removed and the CO restores DID service. External alarm contacts are provided on the front of the ETU for use as required.

Note: It is recommended that customers with Disservice make provisions with their CO to provide this arrangement.

The ETUs are mounted on the cross-connect backboard. Connections are via 25-pair receptacle-ended (CO and SIP) and plug-ended (switch line and trunk) connectors. Modular jacks are provided for the -48V control signal from the CPU (Call Processor Unit) and for additional ETUs. Screw terminals are provided for the connection of external alarms.

When calculating Unit Loading (see Section 5, "Unit Loads"), all ETU loading counts against Cabinet 1.

The 10B ETU is mounted on the cross-connect field as shown in Section 6, "Environmental Requirements."

ETU Power Failure Transfer connections are shown in Figure 2-36. Part (a) on the figure shows a single-line voice terminal that has been connected as a Power Failure Transfer station. In normal operation, the Call Processor CP supplies -48V dc to the ETU. The voice terminal is connected through the ETU to the station port CP and can support all calling activities. The trunk connection through the ETU to the trunk port supports normal trunk calls.

Part (b) on Figure 2-36 shows the ETU connections when a Power Failure Transfer has occurred. The transfer is initiated by the removal of the -48V dc to the ETU. All connections through the system are dropped, and direct connections between PFT stations and CO trunks are established. A contact closure toward the CO makes all DID trunks busy.

When the system is again able to process calls, normal service is automatically restored.

A multiple ETU arrangement is shown in Figure 2-37. As discussed earlier, separate -48V dc control signals from the Call Processor are provided via legs 7 and 8 on Octopus cable C2. The 25-pair cable from the Analog "Line" CP provides connectivity for eight voice terminals at the Line input to the ETU. Since the ETU supports only five PFT stations, three of the voice terminals are wired straight through the ETU and are not switched during service interruptions. A similar condition exists for the 25-pair cable (D) from the CO Trunk CP to the Trunk input of the ETU. Three of the eight trunk port appearances are wired straight through the ETU to the CO and are not switched. Trunk ports connected by legs 2 and 3 of the splitter cable are wired directly to the TAE Block.

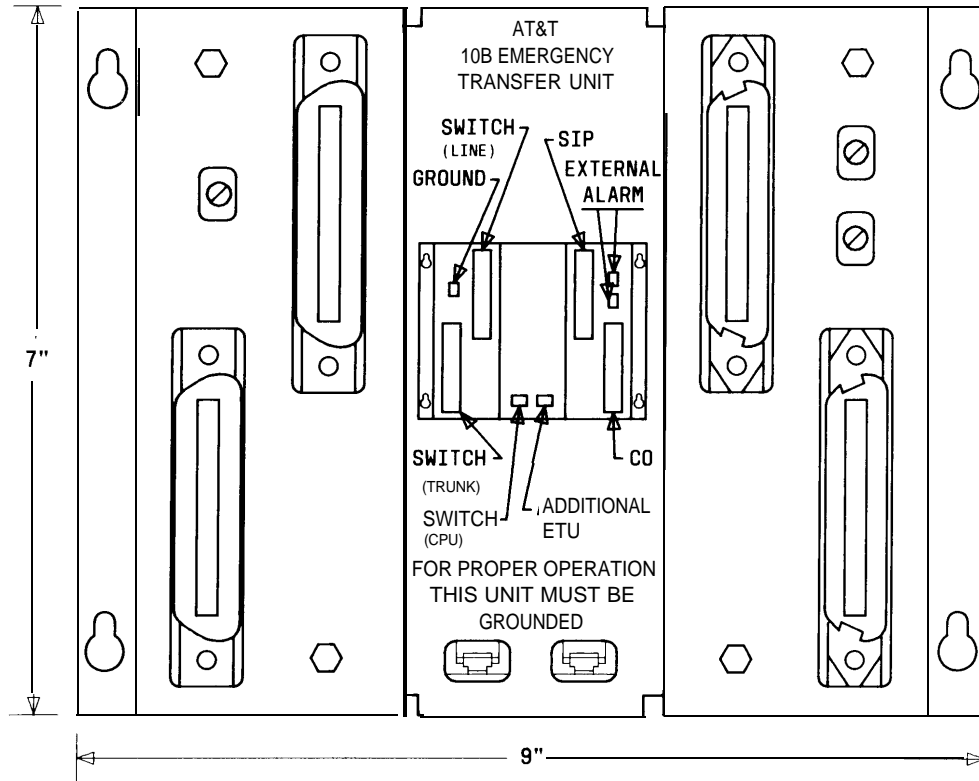
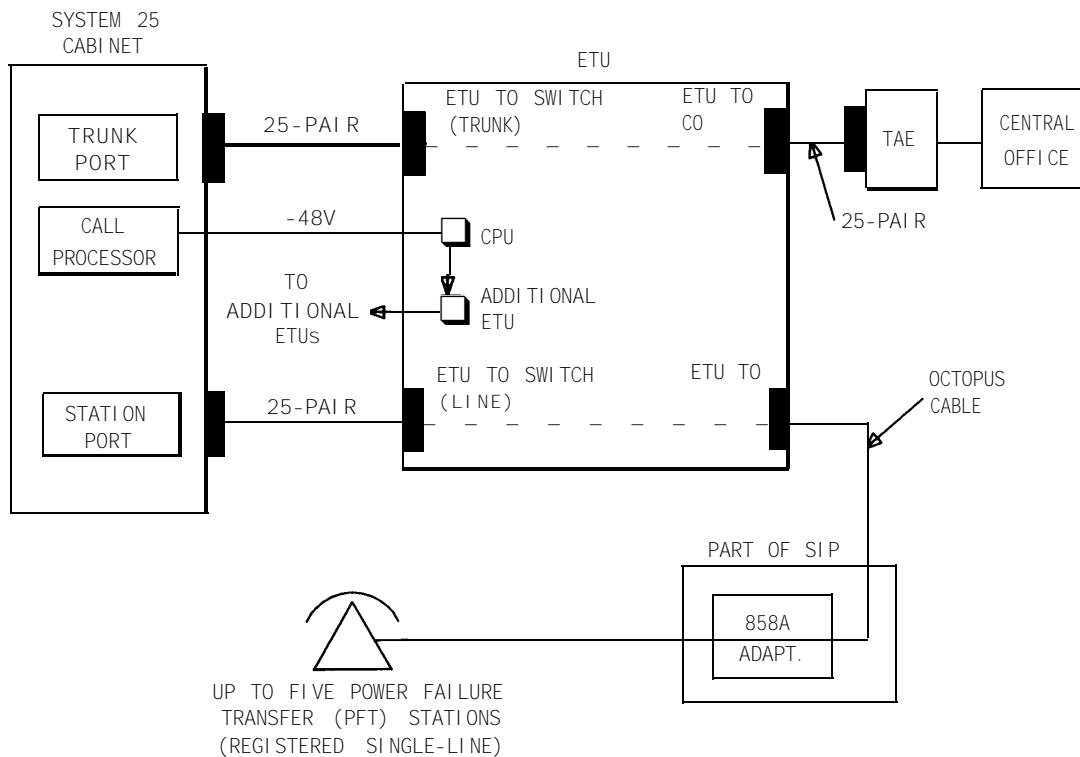
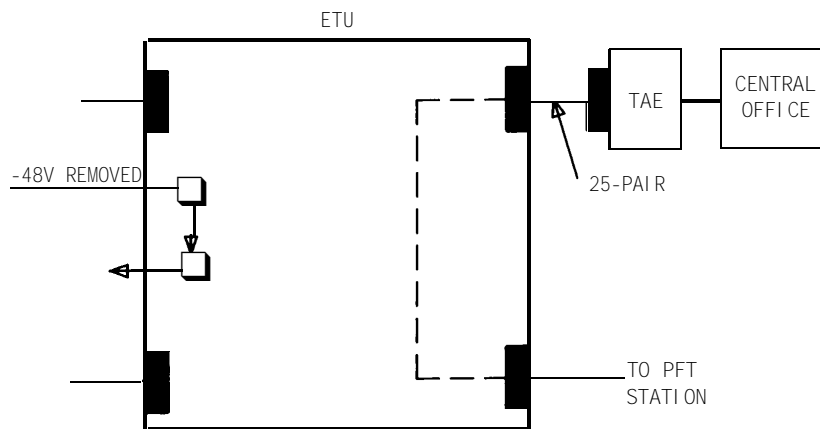


Figure 2-35. 10B Emergency Transfer Unit (ETU)

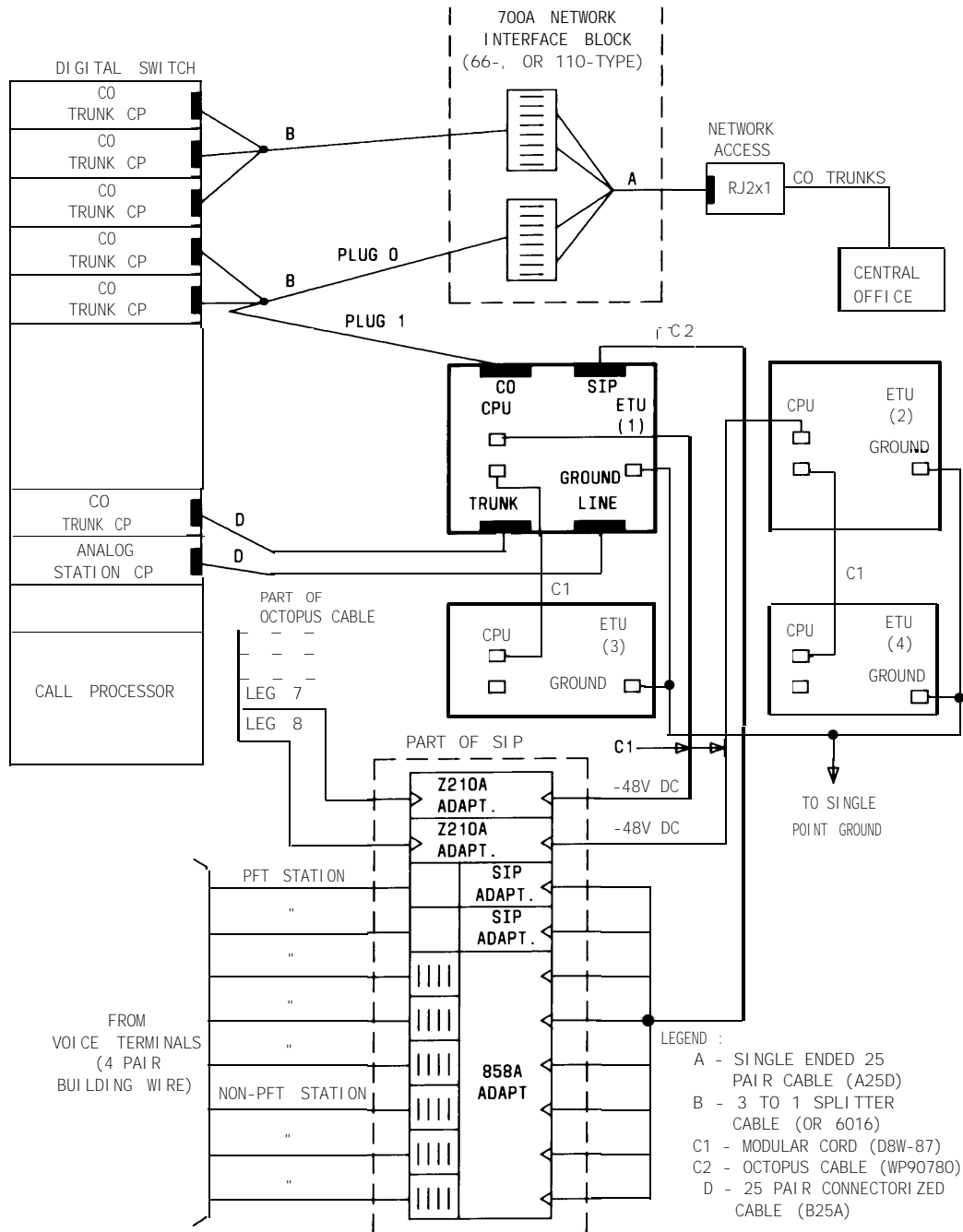


(a) ETU CONNECTIONS (DASHED LINES) NORMAL OPERATION - (NO PFT) (TRUNK SUPPORTS STANDARD CO CALLS)



(b) ETU CONNECTIONS (DASHED LINES) ON PFT

Figure 2-36. Emergency Transfer Unit Connections



NOTES :

1. TRUNK AND STATION CONNECTIONS TO ETU 2-4 ARE SIMILAR TO ETU (1).
2. THREE OF EIGHT STATION LINES (FROM ANALOG STATION CP) AND THREE OF EIGHT TRUNKS (FROM CO TRUNK CP) WHEN USED ARE "FED-THROUGH" ETU, THEY ARE NOT SWITCHED IN THE EVENT OF A POWER FAILURE.
3. MAXIMUM ETU(S) = 4, MAXIMUM PFT STATIONS PER ETU = 5.

Figure 2-37. Multiple ETU Arrangements

PROGRAM

Description

Enables system users to store numbers for access by feature buttons or code dialing. Multiline voice terminal users can program numbers on REP DIAL and FLEX DSS buttons. Both multiline and single-line terminal users can store Personal Speed Dialing numbers. FLEX DSS buttons provide access to inside extension numbers only; REP DIAL buttons and Personal Speed Dialing are used for account codes and outside numbers (maximum of 28 digits and 25 digits, respectively).

Special Characters (VI)

The following special characters may be used in Repertory Dialing and Personal Speed Dialing numbers on R1V1 systems only:

CHAR.	USED IN REPERTORY DIALING NUMBERS
*	(1) In the first character position, transmits an actual "*" (as the first character of certain feature access codes). (2) Otherwise, produces a 1.5 second pause. (Since System 25 does not have a Dial Tone detector, judicious use of the pause character will help to ensure that intermediate Dial Tones are obtained before more digits are sent.)
#1xx	Represents a System Speed Dialing code (where xx = 00-89); # is allowed only in the first character position of the stored number.
#2x	Represents a Personal Speed Dialing code (where x = 0-6); # is allowed <i>only</i> in the first character position of the stored number.

CHAR.	USED IN PERSONAL SPEED DIALING NUMBERS
*	Produces a 1.5 second pause. (Since System 25 does not have a Dial Tone detector, judicious use of the pause character will help to ensure that intermediate Dial Tones are obtained before more digits are sent.)
#1xx	Represents a System Speed Dialing code (where xx = 00-89); # is allowed <i>only</i> in the first character position of the stored number.

Special Characters (V2)

The following special characters may be used in Repertory Dialing and Personal Speed Dialing numbers on R1V2 systems only:

CHAR.	USED IN REPERTORY DIALING NUMBERS
*	Produces a 1.5 second pause. (Since System 25 does not have a Dial Tone detector, judicious use of the pause character will help to ensure that intermediate Dial Tones are obtained before more digits are sent.)
#*	Transmits an actual "*".
##	Transmits an actual "#".
#1xx	Represents a System Speed Dialing code (where xx = 00-89) or a Virtual Facility code (where xx = 90-99). If using a Virtual Facility code, it may appear <i>only</i> at the beginning of the stored number.
#2x	Represents a Personal Speed Dialing code (where x = 0-6).
#3	Marks the beginning of End-to-End Signaling. (System begins transmitting touch-tone signals to the far end switch.)

CHAR.	USED IN PERSONAL SPEED DIALING NUMBERS
*	Produces a 1.5 second pause. (Since System 25 does not have a Dial Tone detector, judicious use of the pause character will help to ensure that intermediate Dial Tones are obtained before more digits are sent.)
#*	Transmits an actual "*".
##	Transmits an actual "#".
#1xx	Represents a System Speed Dialing code (where xx = 00-89) or a Virtual Facility code (where xx = 90-99). If using a Virtual Facility code, it may appear <i>only</i> at the beginning of the stored number.
#3	Marks the beginning of End-to-End Signaling. (System begins transmitting touch-tone signals to the far end switch.)

Programming Procedures

Program mode may be entered either by dialing a code (#4) or by moving the Test/Program switch to P (on voice terminals so equipped). If the code is used to enter program mode, the terminal remains in program mode until the user goes on-hook or a timeout occurs. If the switch is used to enter program mode, the terminal remains in program mode until the switch is returned to the midpoint between P and T; the system will send a single-ring reminder every 60 seconds until the switch is repositioned.

Programming a number always removes the number that was previously stored in the same location. If a user wants to remove an old number and *not* replace it with a new one, “ 0” should be entered in place of the number to be programmed.

To Program a Number by Using a Code:

1. Lift the handset and listen for dial tone.
2. Dial #4.
3. Dial the number you wish to program.
4. Either:
 - press the FLEX DSS or REP DIAL button,
 - or**
 - dial the Personal Speed Dialing code (#20 -#26)to indicate where this number should be stored.
5. Listen for confirmation tone and dial tone.
6. Hang up.

To Program a Number by Using the Switch:

1. Slide the switch on the left side of the voice terminal to P.
2. Lift the handset and dial the number you wish to program.
3. Either:
 - press the FLEX DSS or REP DIAL button,
 - or**
 - dial the Personal Speed Dialing code (#20-#26)to indicate where this number should be stored.
4. Listen for confirmation tone and dial tone.
5. Hang up.
6. Slide the switch back to the midpoint between P and T.

Considerations

The Program feature allows users to assign their own Personal Speed Dial/Flex DSS/Repertory Dial numbers. This is particularly useful where working arrangements or personnel responsibilities change.

Interactions

- Users cannot place or answer calls while in the program mode. Incoming calls will receive Busy Tone.
- Should a number require more digits than are free in the common Personal Speed Dialing/Repertory Dialing memory (approximately 5000 digits), Reorder Tone will be returned after the indication of where this number is to be stored (see Procedures, above).

Administration Requirements

Voice Terminal (Station Port):

- Assign FLEX DSS and REP DIAL buttons
- Allow Personal Speed Dialing on a per-station basis.

Hardware Requirements

None

RECALL

Description

Single-line voice terminal users can obtain System 25 Recall Dial Tone by pressing the RECALL button (not all single-line sets have a RECALL button). Pressing RECALL is equivalent to briefly pressing and releasing the switchhook (switchhook flash).

The RECALL button on a multiline voice terminal can be used, under specialized conditions, to send a switchhook flash to the Central Office (for example, to access Centrex services). However, it can never be used to send a switchhook flash to the System 25.

Considerations

None

Interactions

None

Administration Requirements

None

Hardware Requirements

None

REMOTE ADMINISTRATION INTERFACE

Description

Provides dial-up access to the system's administration port, either for a standard system administration terminal or for a PC running Advanced Administration software.

Both read and write capability is provided with access to all system translation and fault tables. A remote administration terminal can perform the same functions as the on-premises SAT.

Remote Administration allows remote access to the system by maintenance personnel, the System Administrator, and others.

Interactions

Only one System Administration Terminal can be connected at one time.

Administration Requirements

Depends on the connecting arrangements selected (see below).

Hardware Requirements

Requires a remote SAT.

Requires that port #1 of the Call Processor CP be connected to: (1) a dedicated modem and dedicated facility (private line or CO trunk), or (2) a dedicated modem connected to a tip ring station port, or (3) an ADU connected to a data line port. See the subsection titled "System Administration" for additional information. Connectivity information is also provided.

REPERTORY DIALING

Description

Allows multiline voice terminal users to store a telephone number, account code, or feature access code in the system's memory and associate that number with a REP DIAL button. Pressing REP DIAL is equivalent to dialing the stored number. Individual numbers can be up to 28 digits in length.

Programming the number is accomplished from the user's voice terminal. Programming procedures and other information can be found in the "Program" feature description. Should the user attempt to enter more than 28 digits, Reorder Tone will be given.

The user can press REP DIAL under any of the following conditions:

1. When off-hook receiving Dial Tone
2. When off-hook on a call on which more dialed digits are expected
3. When off-hook on a call and connected to an outgoing trunk (End-to-End Signaling might apply in this case)
4. After pressing ACCT ENTRY.

When REP DIAL is pressed, the button's status LED lights briefly and then goes dark.

Considerations

Repertory Dialing simplifies dialing long or frequently called numbers, and allows one-button access to many features.

Interactions

- **Account Code Entry:** An Account Code may be stored on a REP DIAL button. The REP DIAL button should be pressed at the point where the account code would normally be dialed.
- **Calling Restrictions:** A user can not use Repertory Dialing to access a number that he or she is restricted from dialing.
- **Last Number Dialed (V2):** A number called by pressing a REP DIAL button is saved by the Last Number Dialed feature.
- **Speed Dialing:** Numbers already stored as System Speed Dial numbers can also be stored as Repertory Dial numbers. Storing a System Speed Dial code (#100 -#189) on a REP DIAL button saves memory space (compared to storing the whole number again on a REP DIAL button).
- **Virtual Facilities (V2):** Virtual Facility codes can be stored on REP DIAL buttons.

Administration Requirements

Voice Terminal (Station Port):

- Assign Repertory Dial (REP DIAL) buttons.

Hardware Requirements

None

SEND ALL CALLS (V1)

Description

Allows multiline voice terminal users whose calls are covered to temporarily direct some incoming calls to coverage and turn off their ringers to these calls.

This feature is activated by pressing the SEND ALL CALLS button. It is deactivated by pressing the button a second time.

Considerations

Send All Calls is a relevant feature only for stations that have some form of coverage treatment. Activating the feature has no effect at stations with no coverage; it cannot be used simply to create a “do-not-disturb” condition.

When Send All Calls is activated, full (repeated) ringing for incoming calls *must* occur somewhere:

- If there is a coverage station that has an idle COVER button *and* is on-hook, then ringing is sent to the coverage station; the call flashes at the sending station.
- If there is *no* coverage station that has an idle COVER button and is on-hook, the incoming call will remain at the sending station and ring, even though Send All Calls is in effect.

When Send All Calls is in effect at a station, ringing at that station is not necessarily canceled completely. A single-ring reminder for incoming calls is optional, assigned by the System Administrator for each Send All Calls button. Single-ring reminder sounds under the following conditions:

- Send All Calls is activated, and a call goes to coverage.
- A covering receiver is not available, and a call comes in while this station is off-hook; this occurs even if single-ring reminder was not assigned.

The following types of calls always ring at a station, regardless of the status of Send All Calls:

- Automatic Intercom calls.
- Directed Night Service calls.
- Calls to a PDC that is logged in at the station.
- Calls returning to a DTAC on RTN-BUSY or RTN-DA buttons.

Send All Calls must be assigned to a button that has a status light. The light turns on when the feature is activated.

Interactions

- **Call Coverage:** If no coverage station is available (that is, on-hook with an idle COVER button) to accept a redirected call from a station with Send All Calls activated, the call remains at the sending station and rings.

A covered station with Send All Calls activated will ring when called by its individual coverage station if there is only one coverage receiver for this station.

- **Distinctive Ringing:** Normal audible ringing is turned off for most incoming calls when Send All Calls is activated, unless single-ring reminder is administered.
- **Line Status Indications:** The line status LEDs still flash for incoming calls when Send All Calls is in effect, even though normal ringing is cut off. The LED at the sending station stops flashing when a call is answered by a covering station (LED lights steadily) or dropped by the caller (LED goes dark).
- **Personal Line:** Ringing on Personal Lines is turned off by activation of Send All Calls only at the principal station. Calls continue to ring on non-principal appearances and cannot be turned off by Send All Calls.

Administration Requirements

Voice Terminal:

- Assign Send All Calls button
- Assign single-ring reminder if required.

Hardware Requirements

None

SEND ALL CALLS (V2)

Description

Allows multiline voice terminal users to turn off their ringers and invoke a “do not disturb” condition toward incoming calls. In addition, users who have coverage will have those calls directed immediately to coverage, without the normal system ringing delay. Send All Calls also allow-s covering users to temporarily remove their voice terminals from the coverage path.

This feature is activated by pressing the SEND ALL CALLS button. It is deactivated by pressing the button a second time.

Considerations

Send All Calls gives the user the option of having incoming calls sent directly to coverage or making the terminal busy to incoming calls without sending them to coverage. The feature is intended for occasional or temporary use.

Send All Calls must be assigned to a button that has a status light. The light turns on when the feature is in effect.

The following types of calls always ring at a station, regardless of the status of Send All Calls:

- Automatic Intercom calls.
- Directed Night Service calls.
- Calls to a PDC that is logged in at the station.

Note: Calls to an FPDC do not ring when Send All Calls is in effect.

- Calls returning to a DTAC on RTN-BUSY or RTN-DA buttons.

Send All Calls cannot be assigned to a SLAC. The Attendant Position Busy feature provides a similar capability.

When Send All Calls is in effect at a station, and incoming calls are directed to coverage, ringing at the sending station is not necessarily canceled completely. A single-ring reminder for incoming calls is optional, assigned by the System Administrator for each Send All Calls button.

On calls to non-busy stations where Send All Calls has been activated, the callers hear ringing until a covering station answers, or, if the station is not covered, until the call is dropped.

Interactions

- **Call Coverage (General):** Send All Calls works in conjunction with the Call Coverage features at covered and covering stations. At stations not associated with Call Coverage, Send All Calls simply serves to silence the ringer on incoming calls; no redirection occurs.

If a station is translated to *not* send ringing to coverage when calls to this station are unanswered, the Send All Calls feature overrides this instruction (ringing *will* be sent).

If a covering station activates Send All Calls, the station is removed from the coverage path completely. Coverage calls will not be directed to the station.

- **Call Coverage—Group:** If a station with Send All Calls activated has group coverage and all the coverage receivers are busy, a call waits at the station while the system checks for an idle receiver every five seconds. When one becomes available, the call is directed to the covering station.
- **Call Coverage—Individual:** If a station with Send All Calls activated has only individual call coverage and all coverage receivers are busy, a call stays at the station; it does not go to coverage.
- **Distinctive Ringing:** Normal audible ringing is turned off for incoming calls when Send All Calls is activated, unless single-ring reminder is administered.
- **Hands-Free Answer on Intercom:** Activating Send All Calls will cause an active AUTO ANS button to turn off. As long as the Send All Calls feature is in use, AUTO ANS cannot be turned on.
- **Line Status Indications:** The line status lights still flash for incoming calls when Send All Calls is in effect even though normal ringing is cut off. The lights stop flashing when the calls are answered by a covering station or dropped by the caller.
- **Personal Line:** Ringing on Personal Lines is turned off by activation of Send All Calls whether the station is the principal (owner) of the line or not. Personal Line calls follow the coverage arrangements of the principal station. If the principal station is not covered, the call will simply stay at the principal station until dropped (even if other stations with that Personal Line have coverage).

Administration Requirements

Voice Terminal:

- Assign Send All Calls button
- Assign single-ring reminder if required.

Hardware Requirements

None

SPEAKER

Description

Allows 7300 H-series voice terminal users to turn On a built-in speaker. The speaker allows on-hook dialing, group listening, and monitoring of call progress signals. The speaker is turned on by pressing SPEAKER. Pressing the button at an idle terminal has the same effect as lifting the handset: the user is connected to the selected line and hears Dial Tone. An associated LED is lighted when the Speaker is On. Speaker volume may be adjusted by the terminal's volume control located on the left side of the set.

The speaker and associated LED are turned Off by pressing SPEAKER again or by lifting the handset. The latter operation connects the handset to the associated voice channel. When using the handset, pressing SPEAKER will turn On the speaker to support the Group Listen feature; pressing SPEAKER again will turn Off the speaker and associated LED. Note that once the user has lifted the handset, it is not possible to return to "hands-free" operation. Hanging up the handset will terminate the call whether the speaker is On or off.

Note: The built-in speaker provides one-way communication (listen only). The user must pickup the handset to converse.

Considerations

The built-in speaker supports group listening, monitoring of calls (e.g., while waiting on hold), and on-hook dialing.

Interactions

None

Administration Requirements

None

Hardware Requirements

Only 7300H-series (MERLIN) voice terminals with a SPEAKER button support this feature.

SPEAKERPHONE ADJUNCT

Description

The speakerphone adjunct permits users of voice terminals not equipped with built-in speakerphones to place and receive calls without lifting their handsets. The adjunct has an On/Off switch, a switch to temporarily mute the microphone, status lamps, and a volume control (for incoming voice only).

All voice terminal features operate normally with the speakerphone adjunct.

Lifting the handset during speakerphone operation automatically turns off the speakerphone. The speakerphone may be turned on during a call by pressing the On/Off switch and hanging up the handset.

Considerations

Speakerphone operation allows users to perform other activities while carrying on a conversation. Speakerphones also facilitate conference calls.

Interactions

- **Headset Adapter Adjunct:** A voice terminal cannot have both a speakerphone adjunct and a headset adapter adjunct.

Administration Requirements

None

Hardware Requirements

4A Speakerphone System

The 2500SM single-line voice terminal and 2991-type 10-Button MET set require a 4A Speakerphone System. The 4A (Figure 2-38) provides a speaker and associated microphone, indicator lamp and operating controls. The controls include a two position ON OR QUIET/OFF rocker switch and a volume control.

The 4A Speakerphone requires an 85B1 power unit.

S101A/S102A Speakerphone (PEC 3163-HFU)

The S101A Speakerphone (Figure 2-38) can be used with the 12-Button MET Set (7203 M). The S102A Speakerphone can be used with 7300 H-series voice terminals except the 5-Button and HFAI sets.

The S101A/S102A speakerphones are equipped with a 4-foot connecting cord that plugs into the voice terminal. Connecting cords are available in optional lengths of 18 inches and 14 feet.

The unit has a SPEAKERPHONE ON/OFF pushbutton switch and a MICROPHONE ON/OFF pushbutton switch. The former controls the entire unit; the latter turns the microphone on and off for privacy. Each button has an associated green status LED.

The S101A Speakerphone must be powered locally with a 2012D Transformer that plugs into a 115V ac receptacle. Adjunct power supplies are described in Section 4, "Hardware Description." The S102A Speakerphone does not require supplemental power, except when used with a 34-Button Deluxe voice terminal.

Detailed speakerphone adjunct connection information is provided in the following figures:

- Figure 2-39—Speakerphone Connections for 7300H Series Multiline Voice Terminals (Except 34-Button Deluxe)
- Figure 2-40 —Speakerphone Connections for 34-Button Deluxe Voice Terminals
- Figure 2-41 —Speakerphone Connections for 12-Button MET Sets.

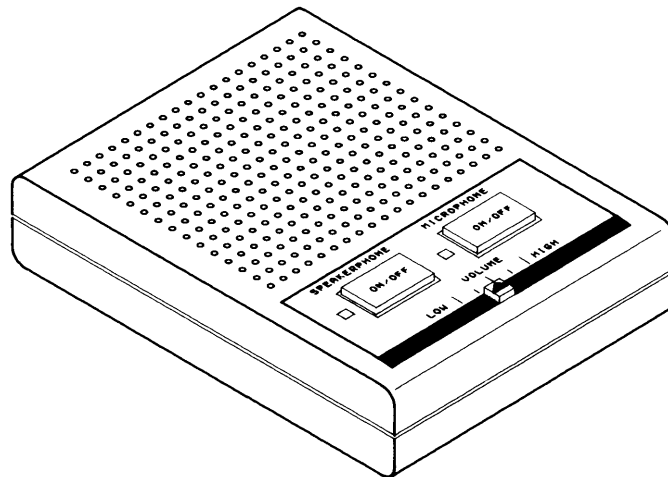
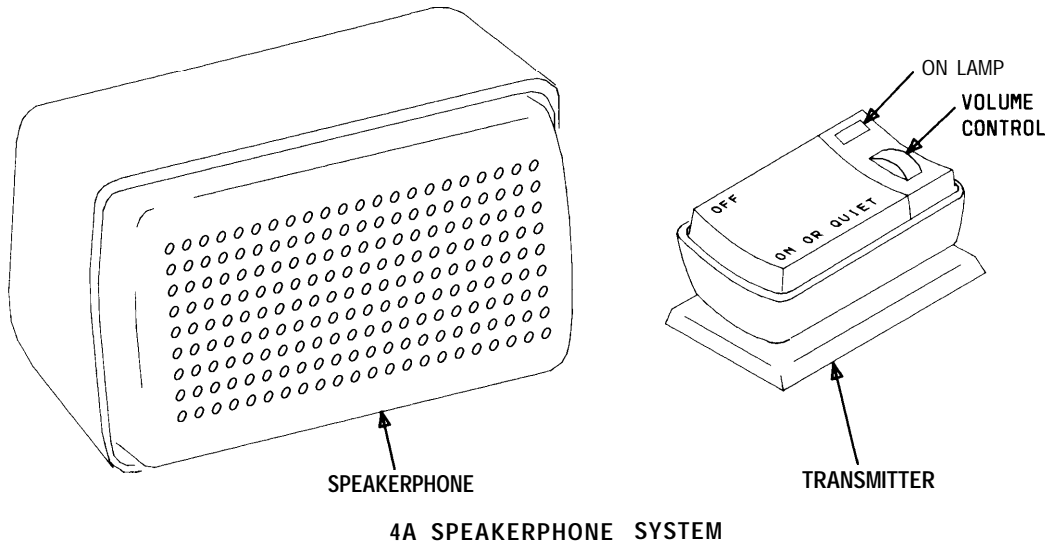
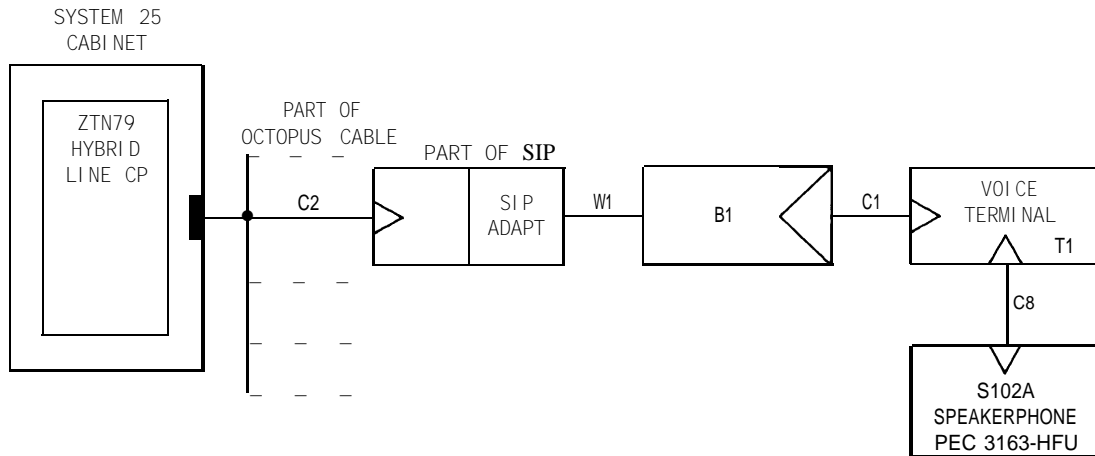


Figure 2-38. Speakerphone Adjuncts

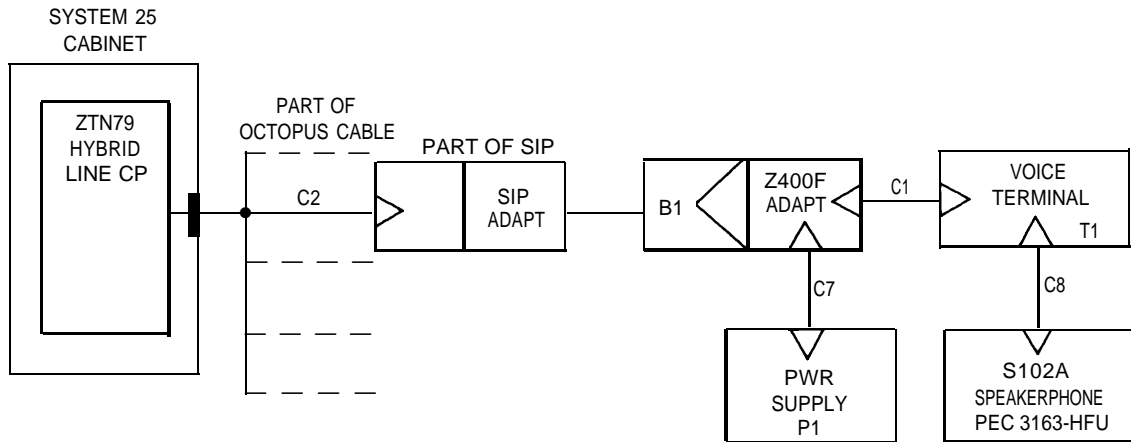


LEGEND :

- B1 - TYPICAL-103A CONNECTING BLOCK***
- C1 - MODULAR CORD (D8W-87) - FURNISHED WITH SET**
- C2 - OCTOPUS CABLE CABLE (WP90780) - PEC 2720-05P**
- C8 - SPECIAL CORD FURNISHED WITH ADJUNCT**
- T1 - 7300H SERIES VOICE TERMINALS EXCEPT 34-BUTTON DELUXE**
- W1 - 4 PAIR INSIDE WIRING CABLE***

* - FURNISHED BY INSTALLER

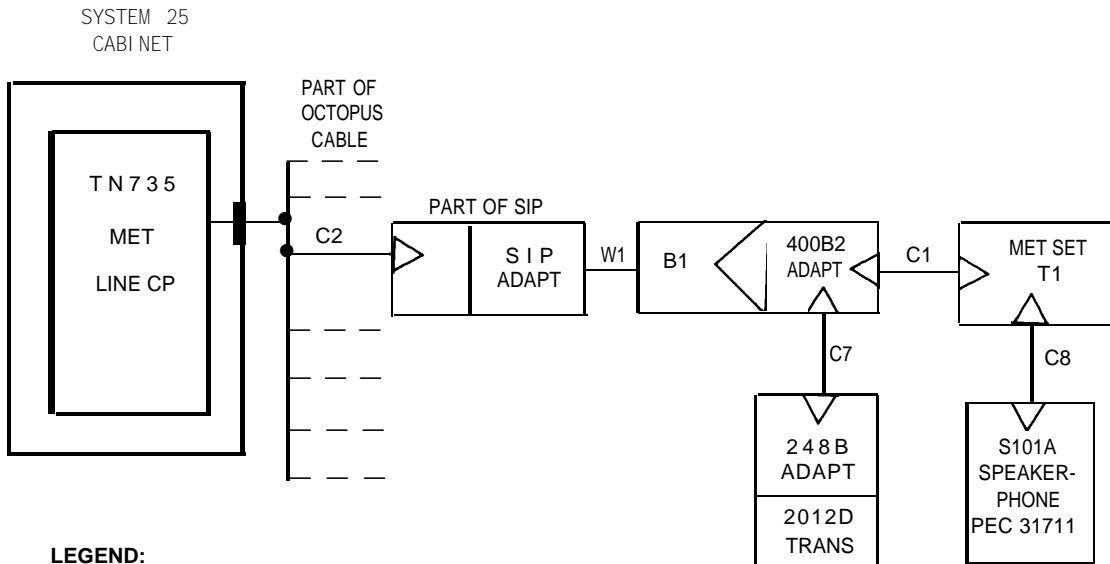
Figure 2-39. Speakerphone Connections For 7300H Series Multiline Voice Terminals (Except 34-Button Deluxe)



LEGEND :

- B1 - TYPICAL-103A CONNECTING BLOCK*
 - C1 - MODULAR CORD (D8W-87) - FURNISHED WITH SET
 - C2 - OCTOPUS CABLE CABLE (WP90780) - PEC 2720-05P
 - C8 - SPECIAL CORD - FURNISHED WITH ADJUNCT
 - T1 - 7305H02B VOICE TERMINAL (34-BUTTON DELUXE)
 - W1 - 4 PAIR INSIDE WIRING CABLE*
 - C7 - MODULAR CORD (D6AP-87)
 - P1 - KS 22911 POWER SUPPLY
 - Z400F - ADAPTER
- } PEC 62510
- * - FURNISHED BY INSTALLER

Figure 2-40. Speakerphone Connections For 34-Button Deluxe Multiline Voice Terminals



LEGEND:

- B1 - TYPICAL-103A CONNECTING BLOCK*
 - C1 - MODULAR CORD (D8W-87) - FURNISHED WITH SET
 - C2 - OCTOPUS CABLE (WP90780) PEC 2720-05P
 - C8 - SPECIAL CORD - FURNISHED WITH ADJUNCT
 - T1 - 7203M SET - 12-BUTTON MET SET
 - W1 - 4 PAIR INSIDE WIRING CABLE*
 - 248B ADAPTER - MODULARIZES 2012D TRANSFORMER
 - 400B2 ADAPTER - POWER ADAPTER
 - 2012D TRANSFORMER - 15-18V AC TRANSFORMER
 - C7 - MODULAR CORD (D6AP-87)
- } PEC 21691
- * - FURNISHED BY INSTALLER

Figure 2-41. Speakerphone Connections For 12-Button MET Sets

SPEED DIALING

Description

There are two types of Speed Dialing: (1) System Speed Dialing, and (2) Personal Speed Dialing.

System Speed Dialing:

Allows the System Administrator to store up to 90 numbers (maximum of 28 characters in length) that are accessible by dialing 3-digit codes from any voice or data terminal.

Examples of typical System Speed Dial ring numbers include frequently-dialed DDD numbers (together with leading facility access codes for WATS, FX etc.) and account codes.

V1 Systems: The following special characters may be used in System Speed Dialing numbers.

CHAR.	FUNCTION
*	Produces a 1.5 second pause. (Since System 25 does not have a Dial Tone detector, judicious use of the pause character will help to ensure that intermediate Dial Tones are obtained before more digits are sent.)

V2 Systems: The following special characters may be used in System Speed Dialing numbers.

CHAR.	FUNCTION
*	Produces a 1.5 second pause. (Since System 25 does not have a Dial Tone detector, judicious use of the pause character will help to ensure that intermediate Dial Tones are obtained before more digits are sent.)
#*	Transmits an actual "*" .
##	Transmits an actual " # " .
#1xx	Represents a Virtual Facility code (where xx = 90-99). This may appear <i>only</i> at the beginning of the stored number.
#3	Marks the beginning of End-to-End Signaling. (System begins transmitting touch-tone signals to the far end switch.)

A user cannot use a Speed Dialing number for which he/she is toll restricted, outward restricted, or facility access restricted.

System Speed Dialing Operation: To place a call using a System Speed Dialing number, the user goes off-hook and presses the # button on the dial pad followed by the 3 digit code assigned to the desired number. The system interprets the associated stored number as if it

were dialed directly. This includes analysis of the number for the various types of restriction.

This feature can also be used when entering account codes. After pressing ACCT ENTRY or flashing the switchhook and dialing “*O”, the user can enter a System Speed Dialing code. The stored number associated with the code (the account code) will be listed in the SMDR report.

Multilane voice terminals users may assign System Speed Dialing codes to Repertory Dialing (REP DIAL) buttons.

Personal Speed Dialing:

Allows users to program up to seven Personal Speed Dialing numbers (maximum of 25 characters in length) that are accessible only from their terminals. The numbers are accessed by dialing associated access codes (#20-#26).

Personal Speed Dialing is authorized on a per-station basis through System Administration. The System Administrator will inform users if they can use this feature.

Refer to the “Program” feature description for more information about programming Personal Speed Dialing numbers.

If enough storage space is available in memory to allow assignment of a Station Speed Dialing number, Confirmation Tone will be returned after each number is programmed. If not, Reorder Tone will be returned.

Note: Personal Speed Dialing is voice terminal oriented, not PDC oriented. A user who logs in at another terminal cannot use his/her Personal Speed Dialing numbers.

Considerations

System Speed Dialing allows users to dial a number by simply dialing #100-#189. The stored number associated with each code is (typically) a common-use phone number and is programmed via System Administration.

Personal Speed Dialing allows users to program up to seven (private) numbers for their personal use; these numbers can only be accessed from the terminal where originally programmed.

The system will compare the restrictions applicable for the voice terminal against the number associated with the Speed Dialing code, then allow or deny the call just as if the number had been dialed directly from the terminal.

Interactions

- **Account Code Entry:** System Speed Dialing codes may be used to store account codes.
- **Calling Restrictions:** A terminal that is restricted from placing a particular call cannot avoid restriction by using the Speed Dialing feature.
- **Data Terminal Dialing:** System Speed Dialing codes can be entered during Data Terminal Dialing.
- **Last Number Dialed (V2):** A number called with a Speed Dialing code is saved by the Last Number Dialed feature.

- **Personal Lines:** The Speed Dialing feature is not accessible from Personal Lines.
- **Repertory Dialing:** Storing a System Speed Dialing code (#100 -#189) on a REP DIAL button saves memory space, compared to storing the whole number again on the REP DIAL button.
- **Speed Dialing:** A Personal Speed Dialing number can include a System Speed Dialing code only as the first four characters (but nowhere else). Personal Speed Dial numbers *cannot* include Personal Speed Dialing codes. System Speed Dialing numbers *cannot* include any Speed Dialing codes.
- **Virtual Facilities (V2):** A Virtual Facility code may be used within Personal or System Speed Dialing numbers. When used, it must appear at the beginning of the stored number (first four characters).

Administration Requirements

System:

- Assign System Speed Dial Numbers.

Voice Terminal (Station Port):

- Allow/Deny Personal Speed Dialing on a per station basis.

Hardware Requirements

None

STARLAN NETWORK ACCESS (V2)

Description

The AT&T STARLAN NETWORK (STARLAN NETWORK) Access feature provides connectivity between System 25 and a *colocated* STARLAN NETWORK. This connectivity is provided by a combination of hardware and software elements.

The STARLAN INTERFACE circuit pack (ZTN84) is the principal hardware element connecting System 25 and the STARLAN NETWORK. One or more of these circuit packs may be mounted in the System 25 cabinet(s). The STARLAN circuit pack (CP) communicates with System 25 call processing over System 25's Time Division Multiplex (TDM) bus. To System 25, this circuit pack functions like a 4-port Data Line circuit pack (DLC). To the STARLAN NETWORK, the STARLAN CP appears as a STARLAN NETWORK workstation.

Communication between STARLAN NETWORK equipment (workstations, servers, hosts) and data terminals, PC6300s, and host computers connected to System 25 is provided by firmware on the STARLAN CP and communications program(s) on the PCs and hosts. Two communications programs are available to users:

• **System 25 STARLAN NETWORK ACCESS (ACCESS)**

ACCESS allows MS-DOS personal computers (PCs) connected (via the PC's serial port) to System 25 to communicate with DOS Servers on the STARLAN NETWORK and to function as client workstations. The interface from the System 25 to the STARLAN NETWORK is the STARLAN CP operating in *bridge mode*. Bridge mode provides a transparent connection between the PC and the STARLAN NETWORK. Personal computer users may access the STARLAN NETWORK just as though they were connected to the STARLAN NETWORK with a Network Access Unit (NAU), although at lower speed. (The NAU is a CP mounted in STARLAN NETWORK workstations that permits access to other workstations and/or servers in the network.) Data transmission through the STARLAN CP is limited to a maximum of 9,600 bps. This is much less than the 1 million bps transmission rate between workstations/servers on a STARLAN NETWORK.

Applications that require frequent and lengthy transfers of data over the Local Area Network (LAN) will appear slow. Applications should be designed/configured to run the executable program locally (on the PC) and to access data from the file server on the LAN. ACCESS is recommended primarily for shared file and printer access. Applications should be copied to the user's (local) disk before they are run.

This program also permits STARLAN NETWORK access for remote PCs if the System 25 is equipped with a Pooled Modem CP (TN758) or external modem pool. Remote PCs can dial the STARLAN CP through a modem using either the Direct Group Calling (DGC) feature or Direct Inward Dialing (DID) trunks to obtain a connection through System 25 to the STARLAN NETWORK.

ACCESS **must** be used in conjunction with the AT&T STARLAN NETWORK Server software (Version 2.0 or later). Installation software furnished with ACCESS requires the STARLAN NETWORK client installation diskette in order to install ACCESS.

- **Communications Access Manager (CAM)**

CAM is an MS-DOS applications program that provides an enhanced calling interface and terminal emulation for PCs connected to System 25 or a STARLAN NETWORK. This connection must be through a DLC or a STARLAN NETWORK that is, in turn, connected to System 25 by a STARLAN CP. Refer to the Communications Access Manager (CAM) Program feature description for a more detailed description of the program.

STARLAN INTERFACE Circuit Pack

The STARLAN INTERFACE CP (ZTN84) requires a single modular connection to the STARLAN NETWORK (see Figure 2-42). It provides an interface between System 25's Time Division Multiplex (TDM) bus and STARLAN NETWORK's packet switched network. The STARLAN CP provides four full-duplex data connections at speeds up to 9,600 bits per second.

The STARLAN CP operates in two modes: Gateway Mode and Bridge Mode. Gateway mode supports connections from System 25 data terminals to STARLAN NETWORK UNIX® system hosts, or from STARLAN NETWORK UNIX system hosts or client workstations to System 25 hosts or modem pools. In Bridge Mode, the STARLAN CP passes the STARLAN NETWORK's Universal Receiver Protocol (URP) through System 25 to a local or remote PC. This is referred to as Bridge Mode and provides a through connection between PCs running ACCESS and a STARLAN NETWORK. The proper mode (Bridge or Gateway) is auto-selected by the system.

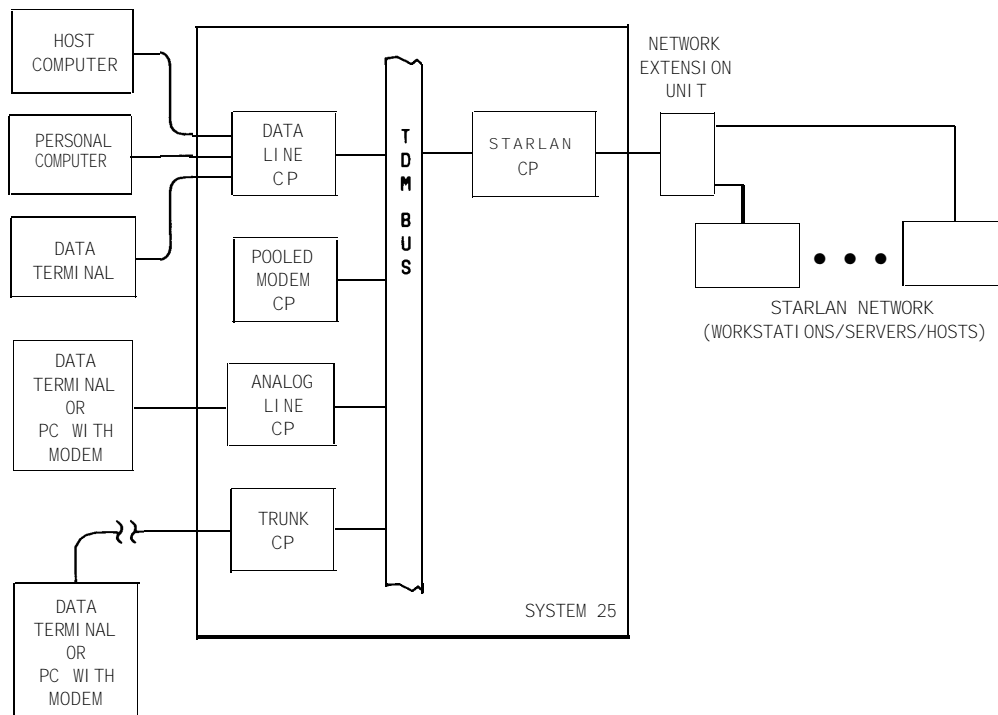


Figure 2-42. STARLAN NETWORK and System 25 Configuration

The STARLAN NETWORK View of System 25

From the STARLAN NETWORK, the STARLAN CP functions like a STARLAN NETWORK workstation equipped with a Network Access Unit (NAU). The NAU enables STARLAN NETWORK workstations and servers to access and exchange data over the network. Plug number 1 of the STARLAN CP octopus cable should be connected to an "IN" jack on the Network Extension Unit (NEU) (see Figures 2-43 and 2-44).

The System 25 View of a STARLAN NETWORK

From System 25, the STARLAN CP looks and functions like a TN726 DLC (with only four ports). The STARLAN CP differs from the DLC in that, when it is dialed, the STARLAN CP auto-answers the call and provides a second dialing prompt for completing the call to a STARLAN NETWORK address. Depending on user's data terminal type, the STARLAN CP automatically selects the operating mode and enables the user to access and exchange data over the network as described below.

Procedures for setting up connections (calls) between STARLAN NETWORK devices and devices connected to System 25 vary, depending on both calling and called device. Generally, a two-stage dialing procedure is used. The scenarios described below cover most situations.

Calls from System 25 to the STARLAN NETWORK

- A. An MS-DOS PC connected to System 25 uses the STARLAN NETWORK ACCESS software to run STARLAN NETWORK applications.

The PC may be connected either to a System 25 DLC port or to an analog station or trunk port. (The latter arrangement uses System 25's Modem Pooling feature.)

A typical call is as follows:

1. The PC user loads ACCESS and is automatically connected to the STARLAN NETWORK. (The STARLAN CP phone number may be entered when ACCESS is installed.)
2. The PC user may now access the STARLAN NETWORK just as if he/she were a client connected to the STARLAN NETWORK through an NAU.

Note: Applications that are to be run frequently or are large (>10K bytes) should be copied to the user's disk before they are run.

- B. A Data terminal user accesses a UNIX system host on the STARLAN NETWORK.

When the user dials the STARLAN CP, the CP answers in Gateway Mode and presents the user with a "STARLAN Address" prompt.

A typical call is as follows:

1. The user dials the STARLAN CP.
2. The STARLAN CP provides the address prompt. The user enters the logical name of the STARLAN NETWORK host (for example, 3B2).
3. The user is connected to the UNIX system host and receives the host login prompt.

Calls from STARLAN NETWORK to System 25

A STARLAN NETWORK workstation accesses a host computer connected to System 25 (either a local host or a remote host that can be reached using the Modem Pooling feature).

1. The STARLAN NETWORK workstation loads CLIENT and NAUCOM and then CAM software (discussed in the Communications Access Manager Program feature description) and selects a directory entry for the host.
2. CAM communicates with the STARLAN CP to place the call.
3. After a connection message is received, CAM automatically switches to terminal emulation (data) mode.
4. The user may now log into and converse with the remote host.
5. To disconnect, the user selects the CAM disconnect command.

Flow Control

Software flow control (XON/XOFF) may be enabled or disabled by System 25 data endpoints. After the "STARLAN Address" prompt is returned to the user, a CONTROL-X may be entered instead of a logical name. The user will be prompted further to enable or disable flow control. After that, the user is again prompted for a STARLAN address. This option also works for calls from the STARLAN NETWORK to System 25.

Data Call Disconnect

Data calls may be disconnected at either endpoint. Connections are dropped through the normal disconnect procedures of each network. If a failure in the established connection occurs, call disconnections are initiated from both sides.

Third-Party Call Setup

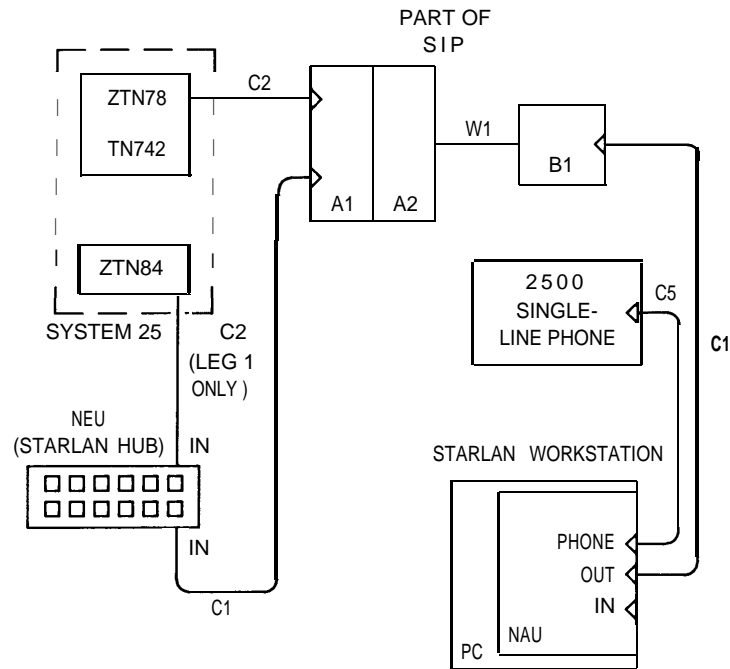
A data terminal (on System 25) or workstation (on the STARLAN NETWORK) can set up a call between two other stations (voice or data) using the Third-Party Call Setup feature. Since voice port/data port associations are not meaningful for STARLAN CP ports, STARLAN NETWORK workstations must always specify the Personal Dial Code of the source voice terminal or the Data Dial Code of the source data terminal. Note that this feature can only be administered for the STARLAN CP ports as a group, and not for individual STARLAN NETWORK workstations.

When placing voice calls using CAM, Third-Party Call Setup is used automatically.

Wiring

The STARLAN NETWORK wiring plan is based on standard 4-pair building wiring. The STARLAN NETWORK uses two pairs of the 4-pair cable, allowing the remaining two pairs to be used for voice service. STARLAN NETWORK data is transmitted over pairs two and three. Figures 2-43 and 2-44 provide typical connection information.

A Y-adaptor may be used to combine/split the pairs at the System 25 cross-connect field. STARLAN NETWORK NAUs provide an RJ11 phone jack that terminates pair 1. Single line sets may be plugged directly into this jack (Figure 2-43). MERLIN Communications System sets require an ATL adaptor and local power (Figure 2-44).

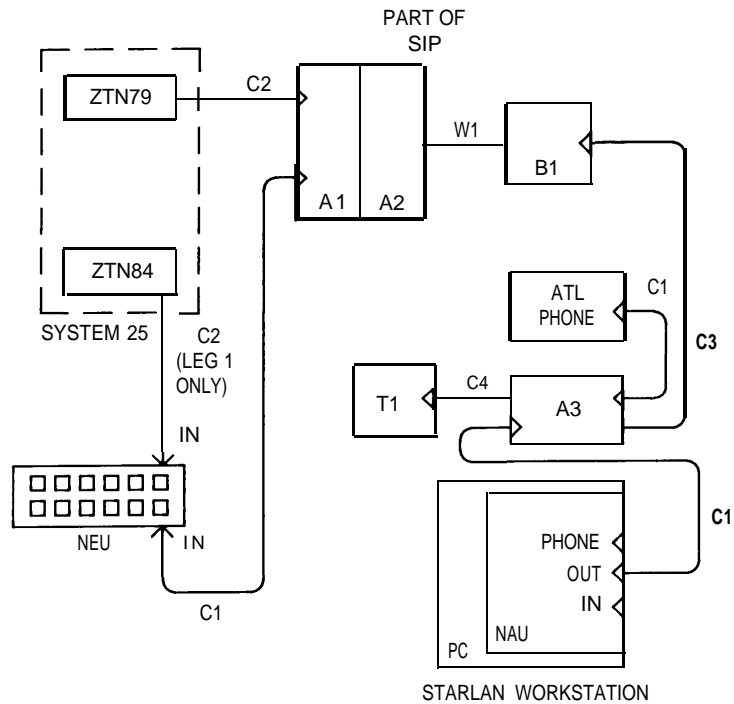


- NAU - NETWORK ACCESS UNIT - PEC 2614-100
- NEU - NETWORK EXTENSION UNIT - PEC 2610-001
- ZTN84 - STARLAN CP PEC 62518
- ZTN78 - TIP/RING LINE CP - PEC 62504
- TN742 - ANALOG LINE CP - PEC 63511
- A1 - WP90851-L1 (Y ADAPTER) - PEC 2750-T05 (NOTE 1)
- A2 - 858A ADAPTER
- B1 - 103A CONNECTING BLOCK*
- C1 - MODULAR CORD D8W-87 (FURNISHED WITH NAU)
- C2 - OCTOPUS CABLE WP90780- PEC 2720-05P (NOTE 1)
- C5 - MODULAR CORD D4BU-87 (FURNISHED WITH PHONE)
- W1 - FOUR PAIR BUILDING WIRING*

NOTE 1: C2 AND A1 ARE NOT REQUIRED IF NO PHONE IS PLUGGED INTO THE NAU.

* FURNISHED BY INSTALLER

Figure 2-43. STARLAN NETWORK Connection to System 25 (With 2500 Single-Line Telephone)



- NAU - NETWORK ACCESS UNIT - PEC 2614-100
- NEU - NETWORK EXTENSION UNIT - PEC 2610-001
- ZTN79 - ATL LINE CP - PEC 62505
- ZTN84 - STARLAN CP - PEC 62518
- A1 - "Y" ADAPTER WP90851-L1- PEC 2750-T05 (NOTE 1)
- A2 - 858A ADAPTER
- A3 - ATL ADAPTER (NOTE 1)
- B1 - CONNECTING BLOCK 103A*
- C1 - MODULAR CORD D8W-87 (FURNISHED WITH NAU AND PHONE)
- C2 - OCTOPUS CABLE WP90780 - PEC 2720-05P
- C3 - 6 INCH MODULAR CORD (PART OF A3)
- C4 - 7 FOOT MODULAR CORD (PART OF A3)
- T1 - 48 VOLT DC POWER SUPPLY KS22911 (NOTE 1)
- W1 - FOUR PAIR BUILDING WIRING*

NOTE 1: PEC 62520 INCLUDES A1, A3, AND T1

* FURNISHED BY INSTALLER

Figure 2-44. STARLAN NETWORK Connection to System 25 (With ATL-Type Telephone)

ATL Adapters

The ATL adapter (KS23475) is a connection block that provides: 48V dc power from T1 (via modular cords C4 and C1) to the ATL phone, data connections from the STARLAN workstation, and phone connections from the ATL phone (via modular cord C1) over a shared common cable (C3) to System 25. The phone and data connections are on separate wire pairs that are split-out at the SIP "Y" adapter (A1). Phone wiring is cabled to the ATL Line CP (ZTN79) by octopus cable C2, and data wiring is cabled to the NEU by modular cord C1.

Administrable Parameters

The STARLAN CP is administered as a type of data port. Some items administered on one port are automatically administered for all four ports on the CP, others are individually administrable.

Individually Administrable (default)

- DDC of port
- DDC to hunt to next (none)
- Display ID

Common Administration (default)

- CO trunk pool dial restriction (no)
- Other trunk pool dial restriction (no)
- Outward restriction (no)
- Toll Restriction Class (none)
- ARS FRL (3)
- Restrict Third-Party Call Setup feature (yes)

Hardware Requirements

Requires a STARLAN INTERFACE CP. Each CP provides four interface ports between the System 25 and the STARLAN NETWORK.

STATION HUNTING

Description

Provides linear, circular, or combinational hunting sequences for calls to busy single-line voice terminals and data terminals.

Calls to a busy terminal may hunt to (only) one other terminal; however, up to five terminals may hunt to the same terminal.

Although hunting is not available to or from multiline terminals, single-line terminals may have their calls *covered* by multiline terminals.

Station Hunting takes precedence over Call Coverage. Calls to a single-line voice terminal that is assigned both Station Hunting and Call Coverage will first hunt. If no hunted-to station is available, the call goes to coverage.

The following are examples of the three types of hunting allowed:

• Linear Hunting Example:

Terminals x, y, and z are arranged for linear hunting as follows: (1) Terminal x hunts to Terminal y, (2) Terminal y hunts to Terminal z, and (3) Terminal z does not hunt.

An incoming call to a busy terminal in the chain will hunt in one direction only. Hunting will be toward the terminal that does not hunt.

• Circular Hunting Example:

Terminals x, y, and z are arranged for circular hunting as follows: (1) Terminal x hunts to Terminal y, (2) Terminal y hunts to Terminal z, and (3) Terminal z hunts to Terminal x.

An incoming call to a busy terminal in the chain hunts in one direction until it finds an idle terminal and then rings at that terminal. Any call coverage options assigned to that terminal will then be invoked. If the hunt finds all terminals busy, it will stop at the called terminal. Any call coverage options assigned to the called terminal will then be invoked.

• Combinational Hunting Example:

Terminals w, x, and y all hunt to Terminal z.

An incoming call to a busy w, x, or y Terminal will ring at Terminal z, and any call coverage options assigned Terminal z will be invoked. If Terminal z is busy, the call remains at the called terminal. Any call coverage options assigned the called terminal will then be invoked.

Considerations

Station Hunting provides several flexible alternatives to ensure that calls do not go unanswered. Note that only calls to *busy* terminals will hunt; once a call begins ringing at a terminal it will remain there unless picked up or covered.

Interactions

- **Attendant Camp-On:** When the attendant extends a call to a busy terminal in a hunt group, the call hunts for an idle terminal. If none is found, the call Camps-On to the called terminal.
- **Call Coverage:** Station Hunting initially overrides all call coverage options. When a call to a voice terminal that has Call Coverage exhausts the terminal hunting possibilities, call coverage is invoked.

Administration Requirements

Voice or Data Terminal (Station Port):

- Assign PDC/DDC of terminal to hunt to next.

Hardware Requirements

None

STATION-TO-STATION MESSAGE WAITING

Description

Allows a multiline voice terminal to turn On a Message Waiting LED located on another multiline voice terminal by assigning a MSG WAIT button on each terminal.

Pressing MSG WAIT causes the LED on the signaling and signaled terminals to light. This feature allows a user to inform another user that they have a message for them. Of course, other arrangements can be made as to the meaning of the signal.

A subsequent MSG WAIT press (at either terminal) turns Off both LEDs.

No talking path is associated with this feature.

Considerations

This feature is functionally similar but separate from the Call Coverage Message Waiting and Attendant Message Waiting features. Note that this feature does not light the Message LED; the status LED next to the assigned MSG WAIT button is lighted.

Interactions

None

Administration Requirements

Voice Terminal (Station Port):

- Assign MSG WAIT on both multiline stations.

Note: The MSG WAIT button must always be assigned to *pairs* of stations, and works just between the two stations. If station “A” wants to signal stations “B” and “C”, station “A” needs a separate MSG WAIT button for each.

Hardware Requirements

None

SYSTEM ADMINISTRATION

Description

The software that controls System 25 operation consists of tables located in system memory. These tables contain data associated with:

- Trunk, Station, and Auxiliary Equipment Ports
- System Parameters
- Direct Group Calling Groups
- Toll Calls Allowed Lists
- Peripheral Equipment Data Communications Parameters
- Automatic Route Selection.

Collectively, these software tables are referred to as translations. The system comes equipped with default translations data; when *full-default cold started*, the default translations are copied into translation memory.

System Administration is the process of managing the translations by making changes to modify system operation to meet customer requirements.

The *System 25 Implementation Manual* describes how a system can be configured to meet specific customer needs. Information about a desired configuration is recorded on a set of forms that are used when entering the initial system translations (i.e., initializing the system). These forms are filed in the Administration Records Binder and provide the basis for on-going record keeping.

Modification of initial assignments can be made to meet changing customer needs.

The system provides an EIA RS-232 interface to a System Administration Terminal (SAT), the primary means of entering and modifying translations.

System 25 administration consists of:

- **Centralized Administration:** Configuration of the system and assignment of feature-related parameters, including assignment of feature buttons on voice terminals. Centralized Administration is performed via the SAT.
- **Advanced Administration:** The Advanced Administration Software (AAS) package is a major improvement in system management. It provides the System Administrator a user-friendly, powerful tool for accurately and quickly making changes in voice/data terminal assignments, call coverage, access codes, and other system functions such as ARS.

The software operates on a PC6300 (with 512K ram) and provides an alternative to use of the SAT input terminal.

A main menu gives the user ready access for these tasks:

- Adding/changing/removing voice station assignments
- Adding/removing users to and from coverage groups
- Saving translations.

A significant advantage of the AAS package is that it can be used either at the same location or can be used remotely via a dial-up connection.

Procedures for performing Centralized and Advanced Administration are provided in *System 25 Administration Manual* and *System 25 Advanced Administration Manual*, respectively.

Considerations

For RIV2, the default system administration password (**systemx5**) can be changed through a new administration item. Note, however, that a system warm start or the password to the default.

Hardware Requirements

The System Administration Terminal (SAT) is a Model 703 Data Terminal (see Figure 2-45). It is a general purpose asynchronous full duplex printing data terminal with an RS-232 interface for data entry and retrieval. It provides a paper record of all transactions. When located within 50 feet of the system cabinets, it can be directly connected to channel 1 on the ZTN82 or ZTN128 Call Processor CP. Either on-premises or off-premises access to the administration port is supported. The terminal operates at a speed of 1200 bps (1200 baud).

The Model 703 requires 115V ac 60-hertz commercial power from a 3-wire grounded outlet.

The terminal should be located on a flat surface such as a desk or table top. It is approximately 12 inches wide, 9 inches long, and 3 inches high.

The Model 703 keyboard generates ASCII codes. The terminal produces two audible tones to indicate the completion of activities.

- Short Tone—A tone of less than one half-second indicates the normal termination of an operation
- Long Tone—A one-second tone indicates that an error or an abnormal operating condition has been detected.

The *Model 703 SAT Supplement* contains a complete set of operating instructions for the Model 703 Data Terminal. This document may be of use to customers who want to use the terminal for other purposes in addition to system administration. All the information needed to use the terminal as a SAT is included in the *System 25 Administration Manual*.

The SAT can be connected to the system cabinets in several different ways:

- A direct connection within 50 feet when sharing the same AC outlet as the system cabinets
- A direct on-premises connection at a distance greater than 50 feet from the system cabinets
- A direct off-premises connection via the Central Office (OPS or CO trunk)
- An on-premises switched connection
- An off-premises switched connection.

Maximum cabling distances from the system cabinets are provided in Section 5, "Technical Specifications."

Installation details are provided in the *System 25 Installation and Test Manual*.

The SAT may also be provided by the customer. It must be a RS-232 compatible terminal that has a 25-pin connector providing signal on DTR (pin 20). In addition, it should have the

following characteristics:

Display: The minimum display size is 16 lines by 80 columns. The port provides both carriage return and line feed characters to position the cursor at the start of the next line. Destructive scrolling is also expected (new lines added at the bottom of the screen and top-most lines disappear). Full duplex operation is required. Alphabetic ASCII characters in both upper-case and lower-case will be sent to the SAT, along with ASCII numerals and some basic ASCII symbols. The device used must be capable of displaying ASCII alphabetic characters when either upper-case or lower-case characters are received. However, upper-to-lower case mapping (or vice-versa) for display is acceptable since no meaning is associated with case.

Keyboard: The administration port requires ASCII alphanumeric characters as well as some symbol characters. If the keyboard generates only upper-case or only lower-case alphabetic characters the administration port will respond appropriately, since upper and lower case input is considered identical. The SAT should be capable of sending the following ASCII Characters:

A-Z or a-z
0-9
* , #
.
?
BACKSPACE
RETURN
“(V2)

The data transfer rate is set when a carriage return character is received by the administration port. There are two supported transfer rates: 1200 bps and 300 bps.

Refer to the *System 25 Administration Manual* for administration procedures and additional information.

SAT Connection Information

Detailed connection information is provided in the following figures:

- Figure 2-46—SAT On-Premises Direct Connections (Sharing Same AC Outlet)
- Figure 2-47—SAT On-Premises Direct Connections (Greater Than 50 Feet from System Cabinet)
- Figure 2-48—SAT On-Premises Switched Connections
- Figure 2-49—SAT Off-Premises Direct Connections
- Figure 2-50—SAT Off-Premises Switched Connections

Descriptions of the SIP (Station Interconnect Panel), TAE (Trunk Access Equipment), and associated cables and adapters, as shown on the figures, are provided under the heading “Connectivity” in Section 4.

Maximum cabling distances from the system cabinets to the SAT are provided in Section 5, “Technical Specifications.”

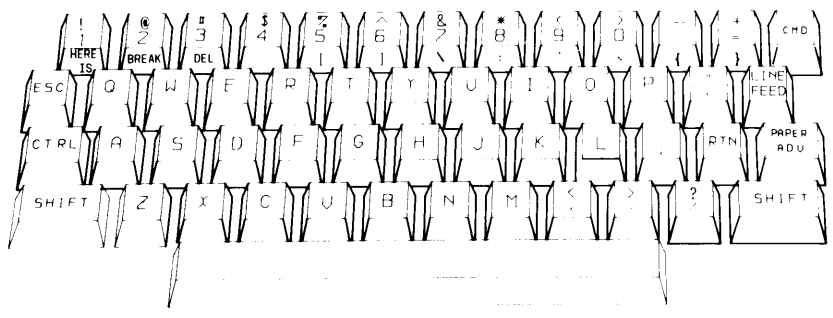
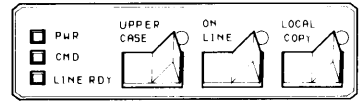
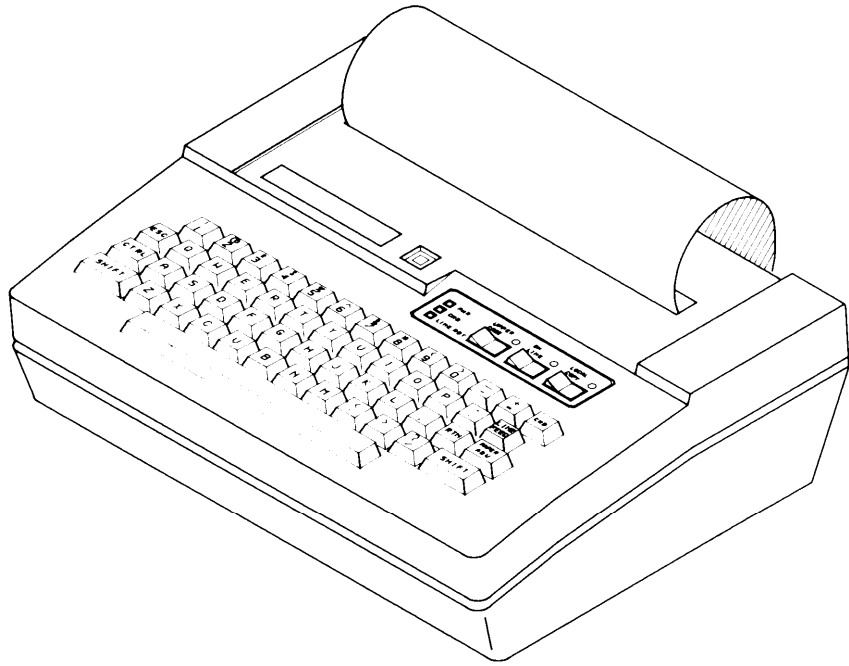
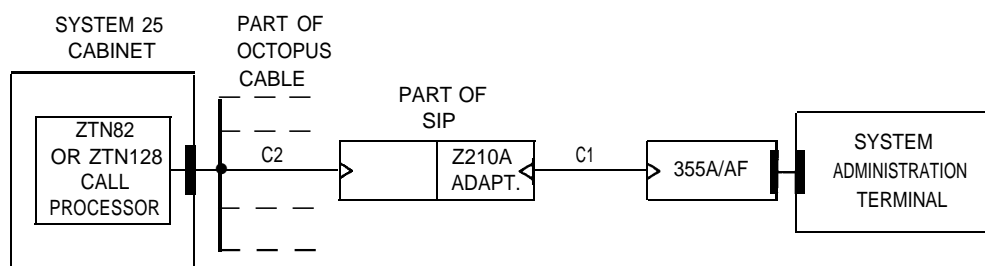


Figure 2-45. Model 703 System Administration Terminal



LEGEND :

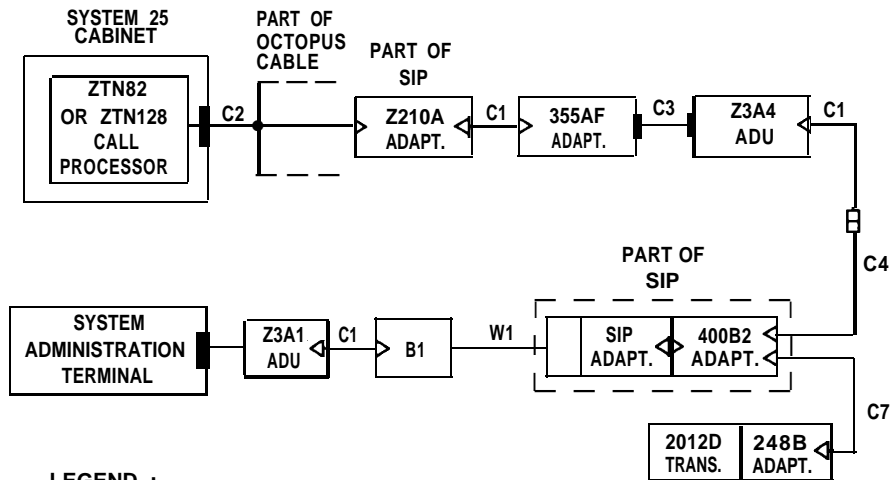
C1 - MODULAR CORD (D8W-87) - PEC 2725-07G

C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P

355A ADAPTER RS 232 PLUG TO MODULAR JACK - PEC 2750-A24

355AF ADAPTER RS 232 RECEPTACLE TO MODULAR JACK - PEC 2750-A25

Figure 2-46. SAT On-Premises Direct Connections (Sharing Same AC Outlet)

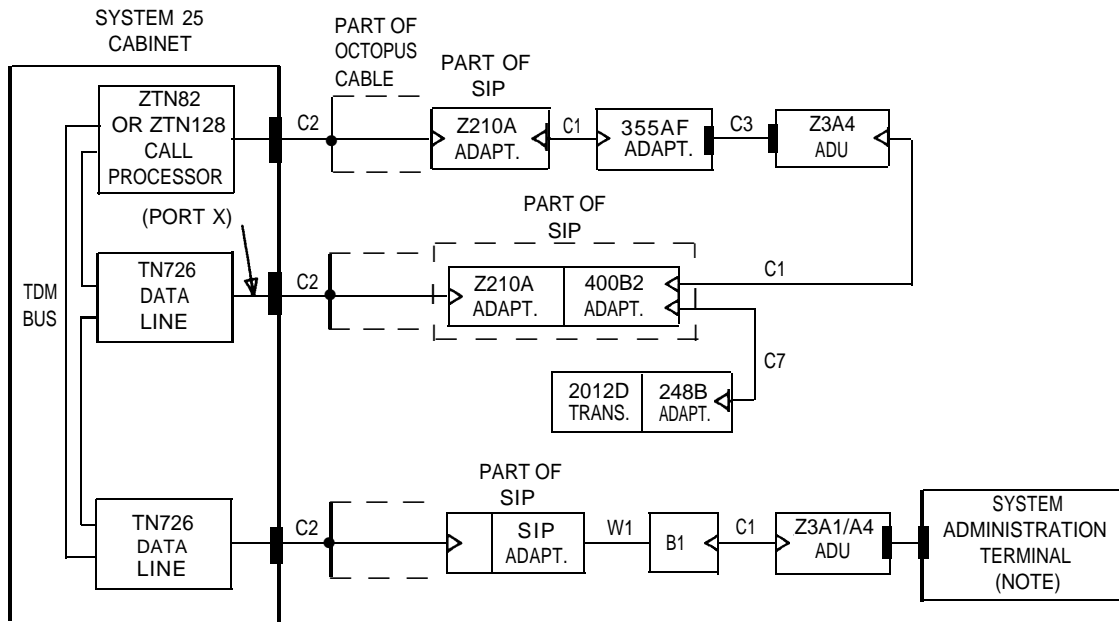


LEGEND :

- B1 - TYPICAL-103A CONNECTION BLOCK*
- C1- MODULAR CORD (D8W-87) - PEC 2725-07G
- C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
- C3 -EIA CROSS OVER CABLE (M7U-87) - PEC 2724-30C
- C4 -ADU CROSS-OVER CABLE (D8AM-87) - PEC 2724-38X
- W1 -4 PAIR INSIDE WIRING CABLE*
- 355AF ADAPTER RS-232 RECEPTACLE TO MODULAR JACK - PEC 2750-A25
- Z3A1 ADU - EQUIPPED WITH A THREE FOOT PLUG-ENDED EIA CORD - PEC 2169-001
- Z3A4 ADU - EQUIPPEO WITH A 3 FOOT RECEPTACLE ENDED EIA CORD
- C7 - MODULAR POWER CORD (D6AP-87)
- 248B ADAPTER - MODULARIZES 2012D TRANSFORMER
PEC 21691
- 400B2 ADAPTER - POWER ADAPTER
- 2021D TRANSFORMER - 15-18V AC TRANSFORMER

* - FURNISHED BY INSTALLER

Figure 2-47. SAT On-Premises Direct Connections (Greater Than 50 Feet From System Cabinet)



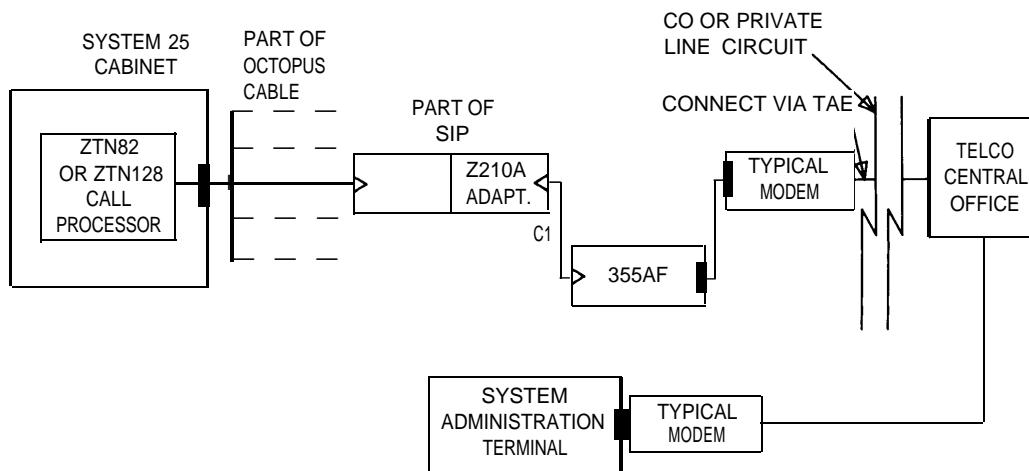
LEGEND:

- B1 - TYPICAL-103A CONNECTING BLOCK*
 - C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
 - C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
 - C3 - EIA CROSS OVER CABLE (M7U-87) - PEC 2724-30C
 - W1 - 4 PAIR INSIDE WIRING CABLE*
 - 355AF - ADAPTER (RS-232 RECEPTACLE TO MODULAR JACK) - PEC 2750-A25
 - Z3A1 ADU - EQUIPPED WITH A 3 FOOT PLUG-ENDED EIA CORD - PEC 2169-001
 - Z3A4 ADU - EQUIPPED WITH 3 FOOT RECEPTACLE ENDED EIA CORD - PEC 2169-004
 - 248B ADAPTER - MODULARIZES 2012D TRANSFORMER
 - 400B2 ADAPTER - POWER ADAPTER
 - 2012D TRANSFORMER - 15-18V AC TRANSFORMER
 - C7 - MODULAR CORD (D6AP-87)
- } PEC 21691

* - FURNISHED BY INSTALLER

NOTE: SYSTEM ADMINISTRATION TERMINAL OR MULTILINE VOICE TERMINAL WITH DATA BUTTON DIALS PORT X DDC TO ESTABLISH DATA CONNECTION TO ZTN-82.

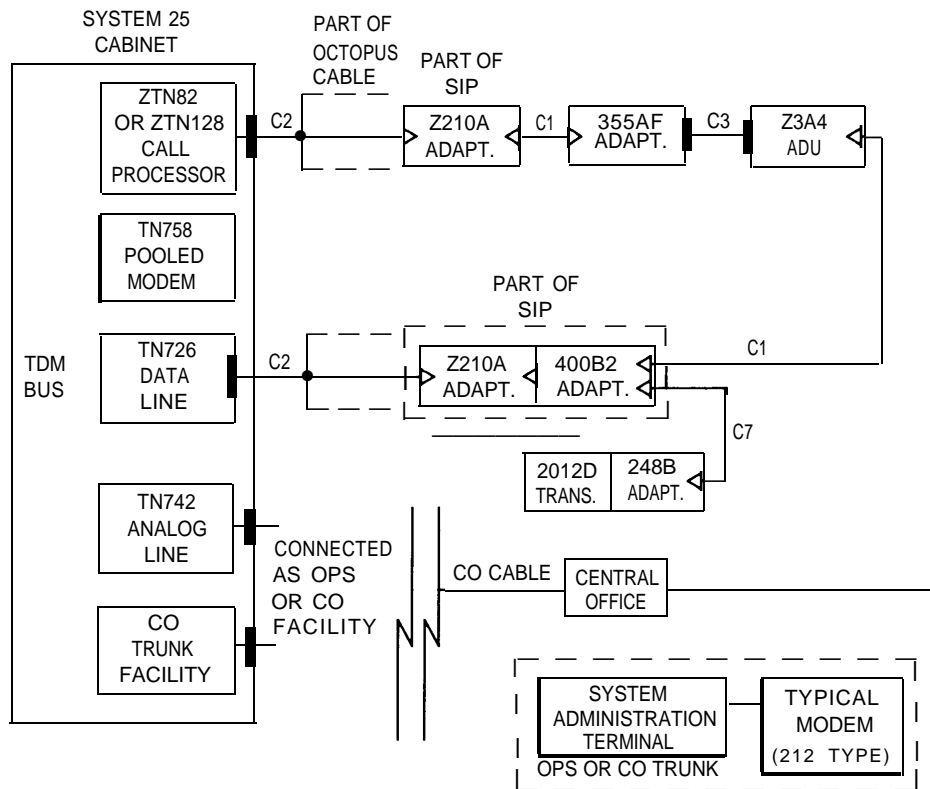
Figure 2-48. SAT On-Premises Switched Connections



LEGEND:

- C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
- C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
- C3 - EIA CROSS OVER CORD (M7U-87) - PEC 2724-30C
- 355AF ADAPTER - (RS-232 RECEPTACLE TO MODULAR JACK) - PEC 2750-A25

Figure 2-49. SAT Off-Premises Direct Connections



LEGEND:

- C1 - MODULAR CORD (D8W-87) - PEC 2725-07G
- C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
- C3 - EIA CROSS OVER CABLE (M7U-87) - PEC 2724-30G
- OPS - OFF PREMISES STATION
- 355AF ADAPTER - RS-23Z RECEPTACLE TO MODULAR JACK PEC 2750-A25
- Z3A4 ADU - EQUIPPED WITH 3 FOOT RECEPTACLE - PEC 2169-004
- 248B ADAPTER - MODULARIZES 2012D TRANSFORMER
- 400B2 ADAPTER - POWER ADAPTER
- 2012D TRANSFORMER - 15-18V AC TRANSFORMER
- C7 - MODULAR POWER CORD (D6AP-87)

Figure 2-50. SAT Off-Premises Switched Connections

SYSTEM MAINTENANCE

Description

The primary objective of System 25 maintenance is to detect, report, and clear troubles as quickly as possible and with minimum disruption to normal service. This goal is supported by periodic automatic diagnostic tests and fault detection hardware. System design allows most troubles to be resolved to the circuit pack level.

System 25 hardware and software are organized as independent units or maintenance objects. Each maintenance object is normally a separately replaceable unit. These units include circuit packs, power units, fans, voice and data terminals, cross-connect hardware, auxiliary, and peripheral equipment.

There are two general categories of system errors: system-detected errors and user-reported problems. The system can automatically detect and log errors without human intervention. For system-detected errors, an Alarm LED on the Attendant Console is lighted if the error qualifies as a Permanent System Alarm (a serious error). Most alarms can be verified by checking the LEDs located on the front edge of the system circuit packs. (At least one Red LED will be on.) User-reported problems are usually detected at individual voice and data terminals and are often related to alarmed conditions.

Alarms may be retired automatically and can also be cleared manually. After a trouble has been cleared, the system retests the previously faulty area. If the fault is no longer present, the error message (and alarm, if applicable) is cleared. It is not necessary for maintenance personnel to retire alarms after a problem has been fixed. However, they may clear error messages and alarms by entering the proper commands at the System Administration Terminal.

System Errors And Alarms

If a maintenance object fails periodic tests, the system automatically generates an error record that is placed in one of three software tables (error logs). The failure may be classified as a Permanent System Alarm or as an unverified failure that never becomes a Permanent System Alarm. A Permanent System Alarm causes the Alarm LED on the Attendant Console to light. This alarm indication is a signal to the attendant to contact maintenance personnel.

System alarms are classified as:

- **Permanent System Alarms:** Failures that cause degradation of service and require immediate attention. These alarms cause the Alarm LED on the Attendant Console to light and an alarm record to be stored in the Permanent System Alarm error log.
- **Transient System Errors:** Potential failures that may cause degradation of service. These do not light the Alarm LED on the Attendant Console. These are errors that have not been verified by system self-tests, and are not yet serious enough to be classified as Permanent System Alarms.

If an error that begins as a Transient System Error is verified or reaches a threshold level of severity, it is reclassified as a Permanent System Alarm.

Transient system errors are stored in the Transient System Error log. The system can store a combined total of 40 Permanent System Alarms and Transient System Errors in the error tables.

- **Most Recent System Errors:** The ten Most Recent System Errors are recorded by the system, regardless of their level of severity. These are stored in the Most Recent System Errors log.

Error Logs

The three error logs can be redisplayed via the System Administration Terminal. The data in the log is useful in diagnosing and analyzing troubles, particularly when the problem has not yet caused an alarm or when alarms cannot be retired by replacement of maintenance objects.

The error log is historical in nature. It lists faults that have not been resolved as well as past alarms, and provides a profile of system maintenance.

Automatic Maintenance Tests

There are two kinds of maintenance testing initiated (only) by the system:

- Periodic
- Demand

Periodic tests are run by the system at fixed intervals. The tests do not affect service.

Demand tests are run by the system when it detects a condition requiring a need for testing. Demand tests are only performed when errors are detected. Maintenance personnel cannot initiate these tests.

For additional information, see *AT&T System 25 Maintenance Manual*.

TANDEM TRUNKING (V2)

Description

Tandem trunking provides an enhanced networking capability for System 25. With this feature, tie trunks can be used to call through System 25 to reach another switching system (CO or PBX). Calls may be completed over on-network or off-network facilities.

To be treated as tandem trunks, tie trunks must be assigned trunk numbers beginning with 9. Incoming calls on these trunks may route out of System 25 over ground start, loop start, or tie trunks. Tandem trunks can gain access to outgoing facilities either indirectly (by the ARS feature) or directly (by dial access).

System 25's Tandem Trunking feature does *not* support traveling class marks or centralized attendant service.

Considerations

The use of tandem trunking with tie trunks provides a cost-effective alternative to toll calling between branches.

Interactions

- **Automatic Route Selection:** Tandem trunk calls that route outbound via ARS receive the same treatment as calls originated by a System 25 station, with one exception. If all facilities in a routing pattern are busy, call queuing is *not* provided. In this case, busy tone is returned to the calling party.

The second digit of the trunk number is used to specify the trunk's "station" Facility Restriction Level (FRL) for use with ARS. FRLs may be specified as follows:

Trunk Number Range	Second Digit	Station FRL
9000-9199	0 or 1	0
9200-9399	2 or 3	1
9400-9599	4 or 5	2
9600-9999	6, 7, 8, or 9	3

To gain access to an ARS routing facility, the tandem trunk's "station" FRL must be equal to or greater than the route's FRL. Thus, a tandem trunk with an FRL of 0 has the least ARS privileges, while an FRL of 3 provides the most privileges. If the restriction level of the tandem trunk is less than all route FRLs, reorder tone is returned to the calling party.

- **Dial Access:** No toll restriction is provided for tandem trunk calls. However, access to outgoing facilities can be controlled via the "allow dial access" option in the outgoing trunk's administration. Tandem Trunk calls receive the following treatment when attempting dial access of System 25 facilities:

- If the requested trunk pool is dial accessible, an outgoing trunk is selected and the call proceeds normally.
- If the requested trunk pool is not dial accessible, reorder tone is returned to the calling party.

Any attempt to dial an outgoing trunk pool by non-tandem tie trunks (that is, tie trunks whose trunk number does not begin with 9) results in reorder tone being returned to the calling party.

- **Station Message Detail Recording (SMDR):** After accessing an outgoing facility, the tandem trunk's Facility Access Code (FAC) will be recorded in the STN field on the call's SMDR record. If the tandem trunk has no FAC, then the 4-digit trunk number (9xxx) will be recorded in the STN field. All other SMDR fields are unaffected.

Administration Requirements

Except for the need to specify the trunk number as described above, administration of tandem trunks is the same as for any other tie trunk.

Hardware Requirements

Only dial-in tie trunks (types 1003-1008) may be used for tandem trucking.

TEST

Description

Provides users of the 7300H series voice terminals the ability to test their terminals.

Placing the Test/Program (T/P) switch in the "T" position causes all red and green LEDs to light alternately. The terminal also rings during the test.

Considerations

Test assures users that all LEDs are working and that the built-in speaker is functional.

The Test switch on some voice terminals is spring loaded; upon release, the switch returns to a normal on-line position. On other terminals, the switch must be manually returned to the center (normal on-line) position.

Interactions

None

Administration Requirements

None

Hardware Requirements

None

THIRD-PARTY CALL SETUP (V2)

Description

The Third-Party Call Setup feature allows a data terminal (the *third party*) in Command Mode to set up a call between an on-premises voice or data terminal (the *source*) and another voice or data terminal (the *destination*; can be on- or off-premises). Once the call has been set up, the *third-party* drops off and is *not* included in the call.

Each *third-party* data terminal may be administered to have a particular source terminal “associated” with it. This association allows an abbreviated form of dialing when activating the Third-Party Call Setup feature. Through further administration, the *third-party* data terminal may be given permission to set up calls for *any* source terminal, for only the *associated* source terminal, or for *no* source terminal (feature disabled).

When the user successfully activates Third-Party Call Setup and has dialed all digits correctly, the following occurs, depending on the *source* terminal type:

- **Voice Terminal (source)**

The *source voice* terminal (if not busy) receives priority ringing. A priority ringing cycle consists of two short bursts followed by one long burst. The *source* terminal’s handset must be picked up within three ringing cycles; the *destination* terminal will then be called. Regardless of the call outcome, the *third-party* data terminal displays the message **CONFIRMED** and **DISCONNECTED** immediately after calling the *destination* terminal.

If the *source* terminal’s handset is not picked up within three priority ringing cycles, the third-party data terminal displays the messages **NO ANSWER** and **DISCONNECTED**. If the *source* phone is busy, the *third-party* data terminal displays **BUSY** and **DISCONNECTED**.

A call to a *source* terminal that has the Hands-Free Answer on Intercom (HFAI) feature activated results in the automatic answering of the *source* end, and the *destination* terminal will be called. The *third-party* data terminal displays **CONFIRMED** and **DISCONNECTED**.

- **Data Terminal (source)**

If the *source* terminal is a data terminal whose speed is set to the highest optioned speed of the data port, the message **REMOTE SETUP** is displayed at the *source* terminal and the **CONFIRMED/DISCONNECTED** message is displayed at the *third-party* data terminal. The *destination* terminal will then be called automatically from the *source* terminal. If the *source* terminal and *destination* terminal are compatible, a data connection is established.

Since the System 25 does not provide call progress tone detection for an off-premises call (can’t detect second dial tone, for example), pause characters should be inserted at appropriate places in the dialed digit string. In addition, Third-Party Call Setup calls are subject to the administered restrictions assigned to the *source voice* or data port. For example, if the *source* terminal is restricted to on-premises calls only, a call to an off-premises *destination* terminal will be blocked.

Setting Up A Third-Party Call

To set up a call from the third-party data terminal, the user selects <Voice call> from the entry-level Command Mode menu (see Figure 2-51). The user then enters the characters as required to call the *destination* terminal. Calls may be completed as follows.

Note: If a character is entered incorrectly, the ASCII character backspace (BS or CTRL-H keys) or underscore (_) may be used to cancel a previously entered character.

Calling a Destination Terminal (source terminal is NOT associated with the third-party data terminal)

After the user has selected <Voice call> from the Command Mode menu, a DIAL: prompt is displayed on the *third-party* data terminal. The user has 15 seconds to begin entering the digits to be dialed before being disconnected. The format of the digits following the DIAL: prompt is shown below:

DIAL: {Destination}F{Source}

The Destination number must include all digits required to call the *destination* terminal and may contain facility access codes, speed dialing codes, and pauses (*). An “F” may be entered immediately following the Destination digits; this character is used to separate the *Destination* number from the *Source* number. The Source number must be a Personal Dial Code when the source is a voice terminal, or a Data Dial Code when the *source* is a data terminal. Floating PDCs are not allowed.

If the user enters the Destination number but not the Source number, the system prompts as indicated below:

FROM: { Source }

The user must then enter the Source number.

Calling a Destination Terminal (source terminal is associated with the third-party data terminal)

Following the DIAL: prompt the user enters the Destination number only. (System 25 will automatically select the associated PDC or DDC as the Source number.)

If the third-party data terminal is permitted to establish calls for any *source* terminal, the format {Destination}F{Source} must be used to set up calls for any terminal except the associated *source* terminal.

Considerations

With this feature, computer-based telemarketing or other calling applications can set up calls for the user.

Interactions

- **Call Coverage**

Third-Party Call Setup calls to the *source* terminal will not be directed to a coverage station. If the *source* terminal is not answered before coverage is invoked, the call is dropped and the **NO ANSWER/DISCONNECTED** messages are displayed at the *third-party* data terminal.

- **Call Following**

If calls to a *source* terminal have been forwarded to another terminal (via Call Following), Third-Party Call Setup calls will be forwarded. Note, however, that the maximum of three ringing cycles (combined cycles at the home and away terminals) still applies before the system drops the call and displays the **NO ANSWER/DISCONNECTED** messages at the *third-party* data terminal.

- **Expert Mode**

Refer to the Expert Mode feature description for additional methods in dialing when using the Third-Party Call Setup feature.

- **Station Message Detailed Recording (SMDR)**

SMDR records will be generated for Third-Party Call Setup calls just as if they were placed by the *source* terminal.

Administration Requirements

- **Data Port**

The user's data terminal may be administered to have a particular source terminal associated with it. This allows the abbreviated form of dialing when activating the Third-Party Call Setup feature.

A *source* terminal may be associated via administration with several *third-party* data terminals. A *third-party* data terminal, however, may be associated with only one *source* terminal (voice or data).

In addition, the *third-party* data terminal can be administered so that calls may be established for:

- Any voice or data *source* terminal
- One associated *source* terminal only
- No *source* terminals (feature disabled; default).

- **AT&T STARLAN NETWORK Access**

Ports administered on the STARLAN INTERFACE circuit pack *may not* have a particular *source* terminal associated with them. Depending on the administration parameters enabled, *third-party* data terminals on the STARLAN INTERFACE circuit pack may establish calls for:

- Any voice or data *source* terminal
- No *source* terminals (feature disabled; default).

Hardware Requirements

None

TIE TRUNKS

Description

Provides a private communications link between System 25 and another PBX.

Incoming tie trunk calls may be directed to the attendant, to a voice terminal, or to a data endpoint. Service may be either automatic, immediate dial, delay dial, or wink start. Dial pulse or touch-tone signalings supported on both incoming and outgoing calls (and maybe different for incoming and outgoing calls).

Considerations

Tie Trunks provide for efficient communications between company employees at different locations. This provides a private network whose control and utilization can be managed.

Tie trunks can be administered for tandem trunking in R1V2. This arrangement enables users to call through an intermediate System 25 to a remote System 25 or other PBX. Refer to the "Tandem Trunking" subsection for more information.

Interactions

- **Conference:** A tie trunk that is part of a conference counts as one of two allowable outside parties.
- **Direct Group Calling:** Only automatic incoming tie trunks can be directed to a DGC group; however, they are not eligible for DGC delay announcement. Dial-in tie trunks can access DGC groups, but are not eligible for queuing (that is, if the DGC group is busy, Busy Tone is returned to the caller).
- **Night Service:** Dial-in tie trunks cannot serve as Night Service trunks.
- **Personal Lines:** When a dial-in tie trunk is assigned as a Personal Line and the line is used for outgoing service at the same time that a call is coming in on the line, the terminal may be connected to the incoming call even though the call is intended for another terminal that shares the line. For this reason, it is recommended that tie trunks not be assigned as Personal Lines.

Administration Requirements

Trunk Port:

- Assign Trunk Type And Number
- Assign Class Of Service Code - (0-15)
- Assign Facility Access Code - Default = 102
- Allow Dial Access (Yes, No) - Default = Yes
- For Auto-in Type Only—Assign To DGC Group (Group Number 1-32, or 0 for none) - Default = 0
- Make This a Directed Night Service Trunk (Yes, No) - Default = Yes

- Assign Night Service Delay Announcement (1, 2, or 0 for none) - Default = 0
- Dial-Inward Capability (Tone or Pulse) - Default = Pulse.

Hardware Requirements

Requires port interfaces on a TN760B Tie Trunk CP.

TOUCH-TONE AND DIAL PULSE SERVICES

Description

All single-line voice terminals and MET sets are equipped with touch-tone dial pads that generate Dual Tone Multifrequency (DTMF) signals when a dial button is pressed. The 7300H series (MERLIN) voice terminals are equipped with touch dial pads that generate digitally coded signals when a dial button is pressed.

Each pool of outside lines and each Personal Line maybe independently arranged for either touch-tone or dial pulse service.

Touch-Tone Dial Pads

On outgoing calls on trunks requiring touch-tone signals, cut-through-dialing is provided. Where the trunk requires dial pulse signals, conversion of the touch-tone signals to dial pulses is provided until an end of dialing signal is detected. Cut-through is then provided, and all subsequent digits are sent as touch-tone signals. See the "End-to-End Signaling" subsection for more information.

Touch Dial Pads

On outgoing calls on trunks requiring touch-tone signals, all dialed digits are converted to touch-tone signals. Where the trunk requires dial pulse signals, the dialed digits are converted to dial pulses until an end of dialing signal is detected. Cut-through is then provided, and all subsequent digits are converted to touch-tone signals. See the "End-to-End Signaling" subsection for more information.

Considerations

None

Interactions

None

Administration Requirements

Trunk Port:

- Assign trunk Class Of Service (COS). (COS includes touch-tone/dial pulse specification.)

Hardware Requirements

None

TRANSFER

Description

Allows a user to transfer any call to another voice terminal.

A user can transfer calls either with or without announcement. A multiline terminal user presses TRANSFER; the party is automatically placed on Special Hold (indicated by a broken-flutter on the status LED of the call appearance button) and may receive Music-On-Hold, if available. The system will automatically select an idle System Access button. The user may dial the desired number or select another facility button and dial the call. The user then can do one of two things: (1) hang up or (2) wait until the called party answers, announce the call, and then hang up. The held call receives music if provided (and so administered) until the transferring station hangs up, after which it receives ringback until the transferred-to station answers. Unanswered transfers will receive the coverage treatment of the transferred-to station.

A Personal Line transferred by a multiline voice terminal user will indicate the Special Hold status at the transferring voice terminal until answered, and may be reentered if the call is not answered. Reentering the call will automatically terminate the transfer attempt.

Single-line voice terminal users may transfer calls by flashing the switchhook, which puts the caller on hold, listening for Recall Dial tone, dialing the second party, and going on-hook either immediately or after announcing the call to the second party. A call may also be transferred by setting up a conference and then hanging up.

Considerations

Transfer provides a convenient way to redirect a call to another voice terminal. Attendant assistance is not required and the caller does not have to redial. While it is possible to transfer a call without announcing it, it is recommended that call transfers always be announced.

Interactions

- Account Code Entry: A user may transfer a call to another user, then, instead of hanging up, enter an account code. Subsequent account code entries will be ignored.
- Attendant Console: In most cases, the attendant should not use this feature to extend incoming calls, but should use the Start button or Selector Console instead. The exception to this rule occurs if a trunk-to-trunk transfer is desired (see below).
- Call Coverage: Coverage treatment of the transferred-to station is provided to transferred calls. When a covering station transfers a covered call to another station, the call will no longer appear at the covering station but will still appear at the covered station. Note that if you attempt to transfer a call to a station that you provide call coverage for, and that station does not answer, call coverage might *not* be invoked. (This is one of the reasons why announced transfer is recommended.)
- Call Pickup: A transferred call may be answered via Call Pickup.
- Calling Restrictions: A non-restricted user (typically the attendant) can transfer a CO trunk to an outward restricted or toll restricted station, giving the station outward service. The toll restriction class of the *transferring* station will apply for

calls over a transferred trunk.

- **Hold:** An outside call placed on hold during call transfer receives music-on-hold, if available. A user attempting to return to a held internal call that has been abandoned will hear nothing. A user attempting to return to a held CO trunk call that has been abandoned hears CO dial tone or receives CO intercept treatment until the CO disconnects.
- **Music-On-Hold:** For V2, a new administration item allows Music-On-Hold to be enabled or disabled for “Special Hold.”
- **Trunk-To-Trunk Transfer:** A trunk call may be transferred to another trunk. Refer to the subsection on “Trunk-To-Trunk Transfer” for additional information.

Administration Requirements

None

Hardware Requirements

None

TRANSFER TO DATA

Description

Allows multiline voice terminal users to originate or answer a call from their voice terminals and then establish a data connection by transferring the call to a data terminal. This feature was formerly called One-Button Transfer to Data (V1).

(Refer also to the overview of the system's data features provided in the "Data Services Overview" description.)

Data terminal calls can be set up from a multiline voice terminal with a DATA button. The DATA button is associated by Data Dial Code (DDC) with a digital data endpoint. A separate DATA button must be provided for each data terminal that the voice terminal can transfer calls to.

The DATA button status LED provides status indications for the data endpoint:

- Dark–Data endpoint is idle
- Winking–Data endpoint is reserved
- Flashing–Data endpoint is being alerted to an incoming call
- Steadily Lighted–Data endpoint off-hook (busy).

The DATA button status LED will wink only when a voice terminal reserves a data endpoint by Data Call Preindication.

Data Call Origination Using Transfer to Data

A voice terminal user, after calling a DDC or a PDC (to reach an analog data endpoint) receives either answer tone or called party answer, respectively. The user then transfers the call to the associated data terminal by pressing DATA and hanging up. The called party may also use Transfer To Data to transfer the call to a data terminal.

An inside call cannot be transferred via Transfer To Data until the far end answers.

If a handshake failure occurs after Transfer To Data, the data call will be disconnected and the data terminal left in the idle (on-hook) state.

Note: Even if the associated data port is optioned for autobaud, the call will be set up at the highest common speed that the calling and called data terminals are administered for, independent of the current *data terminal* settings.

Data Call Preindication

A multiline voice terminal user, by going off-hook and pressing an idle DATA button, may indicate that a data call will be attempted. This reserves the associated data port and a modem pool conversion resource. This procedure is recommended when the data call is a trunk call. The data port reservation is acknowledged by a winking status LED at the DATA button. Subsequently, invoking Transfer To Data transfers the call to the associated data terminal.

Preindication is canceled:

- If the user goes on-hook before transferring the call to data
- If the user preindicates on a second DATA button
- If, after dialing is complete, a second DATA button is pressed. Preindication is canceled for the first data terminal and the data call is transferred to the second data terminal.

When Preindication is canceled, the associated pooled modem conversion resource reservation is canceled.

Interactions

- **Modem Pooling:** If a conversion resource is required on an external call, invoking Data Call Preindication will cause a pooled modem conversion resource to be reserved. If none is available (e.g., the system has no Pooled Modem CP), Reorder Tone is provided. (This will occur whether a conversion resource is actually required or not.)

Administration Requirements

- Assign DATA buttons on multiline voice terminals.

Hardware Requirements

None

TRUNK GROUPS

Description

Allows each trunk in the system to be reassigned to one of up to 16 trunk groups.

Trunks link two switching systems, such as System 25 and the local CO or System 25 and another PBX. Although not required, trunks can be grouped together in trunk groups (sometimes referred to as pooled facilities) when all the trunks in the group perform the same function. This grouping provides resource pooling that results in better service with a given number of trunks. It also simplifies administration and calling. Calls are routed to the appropriate trunk group; an idle trunk, if available, is selected from the group. Up to 16 trunk groups (pooled facilities) may be assigned in the system.

Several different kinds of trunk groups can be assigned in System 25:

- Central Office (CO)—Provides a link with the local CO for calls except Direct Inward Dial (DID) calls. Trunks classed as “CO” have a number of special characteristics.
- Foreign Exchange (FX)—CO trunks that connect to a CO other than the local CO.
- Wide Area Telecommunications Service (WATS)—CO trunks that connect to an Outward WATS office or a dial 800 (in-WATS) Service office.
- Direct Inward Dial (DID)—Provides incoming (only) service from the local CO. These calls go directly to voice terminals instead of through the attendant.
- Tie—Provides a link with another private switching system or network.

Trunk groups can be one-way (incoming) or two-way. Selection of the trunk group to be used for a given call is determined by the initial digits of a dialed number (or by the ARS feature). These digits are referred to as the facility access code. Each trunk group is assigned a unique code. Assuming an idle trunk in the selected group is found, a seizure signal (service request) is sent to the distant switch. If the distant switch requires dialed digits (as all but some tie trunks do), a signal (Dial Tone) is returned to System 25, indicating readiness to accept dialed digits.

Trunk type refers to the physical design of a trunk circuit. The trunk types supported and a brief description of each are given below. Refer to Section 3, “Functional Description” and Section 9, “Glossary” for additional information.

- Loop Start—A closure signal is sent through the loop formed by the trunk leads.
- Ground Start—Similar to loop start but enhanced with ground signals.
- Immediate Start—No start dial signals are used. On outgoing calls, the system waits at least 80 ms after sending the seizure signal before sending the digits required by the distant switch. This allows the distant switch enough time to attach a digit receiver to the trunk. (Tie and DID trunks.)
- Wink Start—A momentary signal (wink) is sent to the distant switch. (Tie and DID trunks.)
- Delay Dial—A steady signal is sent to the distant switch and is removed when ready to receive digits. (Tie trunks only.)
- Automatic—No start dial signals are used. The seizure signal sent or received is sufficient to route the call. (Tie trunks only.)

Trunk groups connecting with a local CO, WATS office, or FX office can be ground or loop start. DID trunk groups can be immediate or wink start. Tie trunks groups can be automatic, wink start, immediate start, or delay dial.

Dual Tone Multifrequency (DTMF) signaling (touch-tone) or dial pulse signaling can be used between the System 25 and the far end switch. System 25 can send or receive either type of signaling required by the distant switch (DID trunks can only receive dial pulse signals). The type to be used is specified when the associated trunk is administered.

An incoming call can be connected to another trunk, a voice terminal, a data endpoint, an attendant console, or an announcement. When the call is answered, an off-hook indication is sent to the serving office. This signal may be used to initiate the recording of call details normally used for billing.

Trunks in a two-way trunk group should be translated (at the SAT) in the same order that the serving office hunts when searching for an idle trunk. System 25 will then hunt in reverse order. This reduces the probability that both switches will attempt to seize the same trunk at the same time.

Considerations

Trunks of the same type and Class Of Service may be assigned a (Pooled) Facility Access Code. This provides users with dial or button access to the trunk pool. Trunks may be dial access restricted to reserve them for AILS and button access only.

Refer to “Recommended Central Office Trunk Facilities” (Section 5) for an estimate of CO trunk requirements based on traffic considerations. See the “Pooled Facility-Button Access” and “Pooled Facility-Dial Access feature descriptions for additional information.

Trunks may be reserved for incoming calls (e.g., sales or service department calls) by specifying this in the (administered) Class Of Service code.

Interactions

- **Direct Group Calling:** Most trunks may be administered so that incoming calls are directed to a specified DGC group. For tie trunks, only the automatic-in type may be so administered.
- **Facility Access Restriction (see “Calling Restrictions”):** Stations may be restricted from dialing the CO trunk pool and/or all (fifteen) other trunk groups (as a whole). Stations so restricted may still dial out if they are transferred to a trunk by another station not so restricted.
- **Tie Trunks:** Refer to the Tie Trunk description.
- **Toll Restriction (see “Calling Restrictions”):** When toll restricted stations access FX, WATS, or Tie trunks, *they are not toll restricted* (i.e., toll restriction applies to CO trunks only).

Administration Requirements

Trunk Port:

- Assign Trunk Type And Number
- Assign Class Of Service Code [DID - (1-4); Other - (0-15)]
- Assign Facility Access Code - Default (See Note)
- Allow Dial Access (Yes, No) - Default = Yes
- Assign To DGC Group (Group Number 1-32, or 0 for none) - Default = /
- Make This a Directed Night Service Trunk (Yes, No) - Default = Yes
- Assign Night Service Delay Announcement (Announcement 1 or 2 or 0 for none) - Default = 0
- Dial-Inward Capability (Tone or Pulse) - Default = Pulse (Tie trunks only).

Note: Default Facility Access Codes are based on the CPs in a system. Defaults are assigned as follows:

- Loop Start Trunks - 100
- Ground Start Trunks - 101
- Tie Trunks - 102.

Hardware Requirements

Associated trunk port interfaces.

TRUNK-TO-TRUNK TRANSFER

Description

Allows users to connect incoming trunk calls to other outside trunks and then hang up (under certain conditions).

Incoming trunk calls may be transferred to another trunk, or conference with another trunk. In all cases and at all times, either a System 25 station must remain in the conference or one of the calls must be an incoming call on a ground start, DID or tie trunk. The other call may be on any type of trunk and may be incoming or outgoing.

Considerations

Trunk-to-trunk transfer is particularly useful when an outside caller requests a transfer to another outside number. For example, an employee can call in and have their call transferred elsewhere. Note that as long as an inside station stays on the call (even if a *multiline* station puts the call on hold and hangs up) *any* two trunks may be conference. If the station drops out of the call, the trunk conference will be torn down unless the above conditions are met.

If a System 25 station enters a trunk-to-trunk transfer call via a line appearance button for one of the conference trunks, the call will still be broken down when one of the outside parties hangs up.

A Direct Group Calling call that comes in on a ground start trunk and is answered at a single-line set is not eligible for trunk-to-trunk transfer.

Interactions

- **Conference:** Trunk-To-Trunk transfers may be set up using the Conference feature. The conference must include an incoming trunk call on either a ground start, DID, or tie trunk if it is to continue after all inside stations have dropped off.
- **Direct Group Calling:** A DGC call that comes in on a ground start trunk and is answered at a single-line set is not eligible for trunk-to-trunk transfer.

Administration Requirements

None

Hardware Requirements

None

USER CHANGEABLE OPTIONS (V2)

Description

User Changeable Options allows a data terminal user who is in the Command Mode to view and change the settings of certain data port options. This feature is available to users of Data Line CP ports; users of STARLAN Interface CP ports cannot change their port options. Table 2-J contains brief descriptions of the user changeable options.

TABLE 2-J. User Changeable Options

Option	Definition
Speed	low, 300, 1200, 2400, 4800, 9600, 19200, autobaud
Parity	odd, even
Permit Mismatch	Allows two data endpoints to communicate at different rates.
Local Echo	Determines whether characters from the data equipment will be echoed by System 25 during Command Mode.
Answer Text	Enables call progress messages to be displayed at the called data endpoint.
Connection Indication	Determines whether users who have Command Mode enabled will receive the "CONNECTED" message when a connection has been made.
Recall Sequence (disconnect)	Two short breaks or one long break; the sequence used to disconnect a data call.

Note: The System Administrator may, under data port administration, deny permission for users of specific data ports to self-administer these options.

The user selects the Options menu from the Command Mode entry level menu. (Figure 2-51 illustrates all available Command Mode menus.) The user now has the choice of viewing options, changing options, or exiting the Options menu.

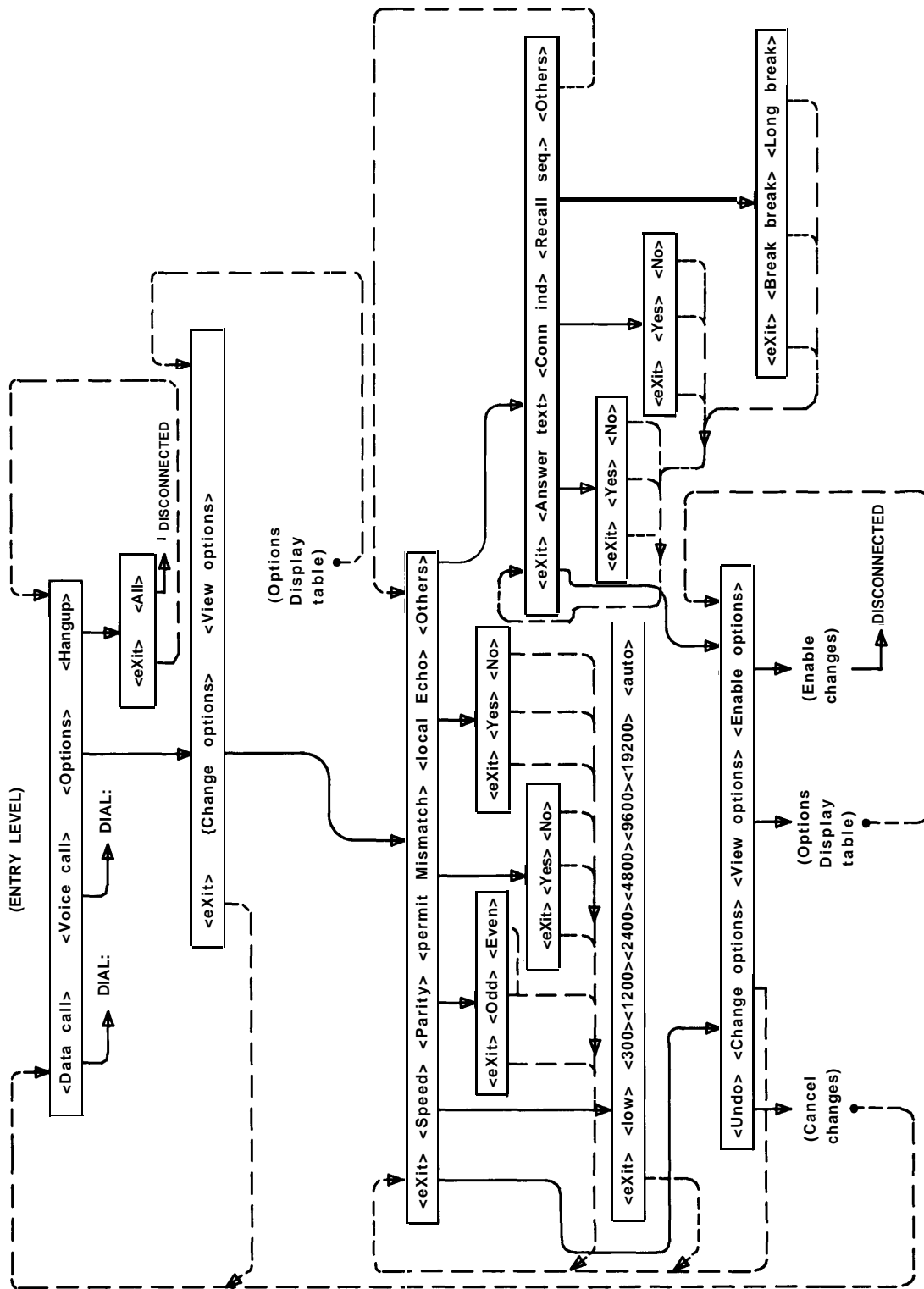


Figure 2-51. Command Mode Menu Tree

Viewing Options

When **<View options>** is selected, current and default values for the various data port options are displayed, as shown below:

<u>OPTIONS</u>	<u>CURRENT</u>	<u>DEFAULT</u>
Speed (highest)	9600	19200 (Auto)
Parity	Even	Even
Mismatch	Yes	No
Local Echo	Yes	Yes
Answer Text	Yes	Yes
Connect Indicat.	Yes	Yes
Recall Sequence	Br - Br	Br-Br

<eXit> **<Change options>** **<View options>**

At this point the user can exit from the View Options menu, Change options, or View options again (redispays the Options table). If the user elects to exit, the terminal returns to the Command Mode entry level menu.

Note: Typing the capital letter found within a menu will select that item and move the user up or down the menu tree. For example, the user simply enters X or x (lower-case) to **<eXit>** the Options menu shown above and return to the entry level menu.

Changing Options—General

When on the Options Menu, the user selects **<Change options>** either by moving the cursor (with the space bar) beneath **<Change options>** and pressing RETURN, or by typing the single letter code (c) associated with that item.

If the user selects **<Change options>** from the Options menu, the first half of the Change Options menu is displayed, as shown below and on Figure 2-51.

<eXit> **<Speed>** **<Parity>** **<permit Mismatch>** **<local Echo>** **<Others>**

If the user selects **<Others>**, the second half of the Change Options menu is displayed:

<eXit> **<Answer text>** **<Corm ind>** **<Recall seq. >** **<Others>**

If the user selects **<Others>** from the second half of the menu, the first half of the Change Options menu is redisplayed. In this way, users can “toggle” back and forth between the first and second halves of this menu.

Since these two lines are actually two halves of a single menu, users may select a particular menu item while active on either half of the menu. For example, users who are active on the first half of the menu may select **<Answer text>** by typing “a”.

Once a user has selected an option to be changed, a menu of valid settings for this option is displayed (**<Yes>**, **<No>**, etc.). An “X” is displayed beneath the current setting of the options, or beneath an option that may have been changed but not yet enabled. For all options except **<Speed>** (see below), settings may be selected either by moving the cursor (using the space bar) beneath the item desired and then pressing RETURN, or by typing the single-letter code associated with that setting. The user is then returned to the Change Options menu to make additional changes if required.

Changing Data Port Speed

The procedure for changing Speed settings is different from the procedure for changing the settings of other options. Within the Speed menu, the user may find that several values are marked with Xs. To change a speed, move the cursor beneath each value to be changed and type "+" to add the value or "-" to delete it. Once the new settings have been marked, press RETURN to translate the plus and minus signs to their proper "X" values and then type "x" to <eXit> from the Speed menu and return to the Change Options menu. For example:

The user enters the Speed menu and finds the following settings active:

```
<eXit>  <low>  <300>  <1200>  <2400>  <4800>  <9600>  <19200>  <auto>
                X      X      X      X      X
```

To remove 9600 baud and activate autobaud, enter - under <9600> and + under <auto>, as shown below:

```
<eXit>  <low>  <300>  <1200>  <2400>  <4800>  <9600>  <19200>  <auto>
                X      X      X      X      X-      +
```

With the cursor under any item except <eXit>, pressing RETURN provides the following:

```
<eXit>  <low>  <300>  <1200>  <2400>  <4800>  <9600>  <19200>  <auto>
                X      X      X      X                        X
```

Entering "x" now allows the user to <eXit> the Speed menu and return to the Change Options menu. The user can make additional changes, as required.

When all of the changes have been made, the user should <eXit> the Change Options menu. The following menu is then displayed:

```
<Undo>  <Change options>  <View options>  <Enable options>
```

From the above menu:

- If the user selects <Undo>, the user is returned to the Command Mode entry level menu, deleting any option-change requests.
- If the user selects <Change options>, the Change Options menu is displayed and the user can make additional changes as required.
- If the user selects <View options>, the following menu is displayed:

OPTIONS	CURRENT	REQUESTED
Speed (highest)	9600	4800 (AUTO)
Parity	Even	Even
Mismatch	Yes	Yes
Local Echo	Yes	Yes
Answer Text	Yes	Yes
Connect Indicat.	Yes	Yes
Recall Sequence	Br-Br	Br - Br

The value in the CURRENT column indicates the current (active) status of the option. The value in the REQUESTED column indicates the most recently entered value (not yet enabled).

- If the user selects <**Enable options**>, the system incorporates the changes requested and displays the message **DISCONNECTED**. If Autobaud is off, the user must now press BREAK to return to Command Mode. If Autobaud is on, the user must press BREAK and RETURN to return to Command Mode.

Note: If a user attempts to enable options during a “SAVE” operation by the system administrator, the message “options changed FAILED” will be displayed. The user will be returned to the top level of the Command Mode menu, and all change requests discarded. After waiting a few minutes, the user may try again to change his/her data port parameters.

Considerations

For those cases where a data terminal user accesses various data endpoints, each requiring option changes, the User Changeable Options feature simplifies the process of administering the data port to allow data call setup. A system administrator is not required to enter each change.

Interactions

- **Expert Mode:** See the Expert Mode feature description for an abbreviated method of accessing Command Mode menus.

Administration Requirements

- **Data Port**

A data port must be administered to allow the terminal user to change options from the Command Mode menu. If permission is denied, the user may view the current option settings but not change them.

- **AT&T STARLAN NETWORK Access**

The User Changeable Options feature is not available for ports administered as STARLAN INTERFACE ports. Refer to the “STARLAN NETWORK Access” feature description for additional information.

Hardware Requirements

None.

VIRTUAL FACILITIES (V2)

Description

A virtual facility (VF) is a call routing facility that is not defined by the physical facility (trunk) over which calls are routed, but is instead defined by a combination of access codes, authorization codes, and coded characters that allow special handling of the destination telephone numbers. VFs can be used to automatically route calls via other carrier networks, private networks, or tie trunks.

Up to ten virtual facility numbers (VFNs) may be administered. Each stored number may be up to 28 characters in length and is associated with a code in the range of #190 to #199. The first digits in a stored VFN must be the facility access code (FAC) for a physical trunk group over which the call is to be routed. A series of digits and special characters are stored following the FAC to define additional routes, Inter-Exchange Carrier (IXC) codes, identification codes, or instructions concerning special handling of the destination telephone number. When a VF has been defined using a particular trunk group, it has full access to all trunks in that group. It is considered “busy” only when the physical trunk group is busy. When a VF is dial accessed by a system user, calling restriction is based on the station’s administered calling restrictions.

A system user may gain access to a VF by:

- Dialing the VF code (#190-#199).
- Storing the VF code as the first digits on a REP DIAL button. Other digits (for example, the destination telephone number) may be stored following the VF code.
- Dialing a System Speed Dialing code (#100-#189) that contains a VF as part of the stored number. The VF code must be the first digits stored in the speed dialing number.
- Dialing a Personal Speed Dialing code (#20 -#26) that contains a VF as part of the stored number. The VF code must be the first digits stored in the speed dialing number.
- Using Automatic Route Selection (ARS) and having a VF as the route selected by ARS.

Note: The system can be administered to allow or restrict dial access for each VF code. If dial access is restricted (system default), a VF may be accessed only when used in an ARS routing pattern.

When virtual facilities are used in ARS patterns, they assume the same capabilities and restrictions as physical facilities. For example:

- Each VF may have a digit deletion and insertion scheme associated with it.
- Selective restriction of a VF may be accomplished by assignment of Facility Restriction Levels (FRLs).

If the VF is used in the first position of a routing pattern, calls may queue on it if all of the routes are busy.

Whenever a VF is used to complete a call (either by dial access or through ARS), call processing treats the number as a physical facility for Station Message Detail Recording (SMDR) purposes. Thus, if VF code #190 is used to complete a call, the SMDR call record will show "#190" as the facility used.

A VFN may contain up to 28 characters. The pound sign (#) is used as an escape character within the digit string and indicates that the character following the pound sign requires special interpretation. The following table defines the special characters that may be included in a VFN.

CHARACTER	FUNCTION
*	1.5 second pause
#*	Transmit *
##	Transmit #
#3	Begin transmission of End-to-End Signaling (system begins transmitting touch-tone signals to the far end switch).
#5	Insert dialed digits (destination telephone number) here. The destination telephone number may be up to 16 digits in length (21 if ARS digit translations have occurred).
	If #5 is used, it must be placed within the last nine digits of the VFN. If #5 is <i>not</i> used within a VFN, the dialed digits are appended to the end of the VFN.

Examples of Virtual Facility Numbers

The use of VFS can be demonstrated with the following examples:

- **Example 1:** VF Code = #191, VFN = 100 10288
 - The first three digits (100) represent the FAC for a CO trunk group.
 - 10288 represents an access code for a non-primary IXC.
 - The destination telephone number (dialed by the user) will be transmitted after this IXC access code, since "#5" was not used within the VF number.
- **Example 2:** VF Code = #193, VFN = 2219* 5554343 *#5#*12345
 - The first three digits (221) represent the FAC for a tie trunk to a remote PBX.
 - The "9" is used to access the remote PBX's ARS.
 - The "*" represents a pause of 1.5 seconds (allows time for dial tone to occur).
 - The "555 4343" defines the local address of a private network and its internal routing table.
 - The "*" represents a 1.5 second pause.

The “#5” indicates that the destination number should be inserted here, rather than at the end of the VF translation.

- The “#*” indicates that the system should transmit a “*” symbol as the first character of an identification code.
- The “12345” represents the remaining characters in the identification code.

● **Example 3:** VF Code = #195, VFN = 104 5554567 *****1234*9

This example demonstrates how a VF might be used in place of a tie trunk group connecting two local PBXs, when you have permission to access the other PBX’s facilities to complete calls.

- The first three digits (104) represent the FAC for a CO trunk group.
- The “5554567” represents the number for the other PBX.
- The “*****” represents a 6 second pause (allows time for the other PBX to answer and return new dial tone).
- The “1234” represents a “barrier” (security) code required to access the other PBX’s facilities.
- The * represents a 1.5 second pause.
- The “9” represents an ARS access code for the other PBX.
- Since “#5” was not used within the VFN, the destination telephone number (dialed digits) will be transmitted after this ARS access code.

Accessing a Virtual Facility

Dial access is provided by dialing the VF code (#190-#199, including the “#”), followed by the destination telephone number.

● **Example:**

- The user dials the following VF code and associated destination telephone number.
#192 12125551643.
- The stored VFN associated with VF code #192 is defined as 2222*333*444.
- The first four digits (2222) represent the FAC for a tie trunk group to a remote PBX.
- The * represents a 1.5 second pause.
- The next three digits (333) represent the security code required by the remote PBX, indicating that you have permission to access their facilities.
- The * represents a 1.5 second pause, as the remote PBX checks the validity of your security code.
- The final three digits (444) represent the FAC required by the remote PBX to access their Band 5 WATS trunks.

- Since “#5” was not embedded within this VFN, the destination telephone number (12125551643) will be transmitted after the WATS access code.

VF codes may be included in numbers stored in REP DIAL buttons, System Speed Dialing codes, and Personal Speed Dialing codes **if** the VF code is used at the beginning of these numbers. VF codes may not be assigned to FACILITY buttons and may not be embedded in other virtual facility numbers.

VFs may be used in ARS routing patterns just as if they were physical facilities.

Considerations

VFs enhance the Automatic Route Selection feature by increasing the number of facility types available for use in routing patterns. Using ARS ensures that the least-cost facility is used to complete each call. User intervention is minimized and associated user dialing errors are essentially eliminated.

In addition, in those systems where users are permitted dial access to VFs, user dialing of long digit strings is minimized, as are the associated dialing errors.

Interactions

When using a VF through dial access, calling restrictions will be based on the station's class of service. Dial access VF calls will be completed only if:

- The VF code is valid and not dial restricted.
- The station is not outward restricted.
- The station has dial access permission for the physical facility embedded within the VFN.
- The destination telephone number is valid and allowed for the station's toll restriction class.

VFs cannot be assigned to Facility buttons. Button access is provided by programming REP DIAL buttons only.

Administration Requirements

A VF must be programmed via System 25 administration. The following items are administrable:

- Specify a virtual facility code (#190-# 199).
- Assign a virtual facility number to this code.
- Allow dial access to this virtual facility (I = Y/0 = N).

Hardware Requirements

None.

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SECTION 3—FUNCTIONAL DESCRIPTION

This section describes how the digital switch and the software of System 25 provide control and switching.

Digital Switch

Figure 3-1 shows a schematic diagram of the System 25 digital switch.

The basic switch hardware consists of the following:

- Common Control
 - Memory Bus
 - Call Processor Circuit Pack (CP)
 - Memory CP.
- Switching Network
 - Time Division Multiplex (TDM) Bus
 - Port Circuits
 - System Resource Circuits: Service Circuit, Tone Detector, and Pooled Modem.

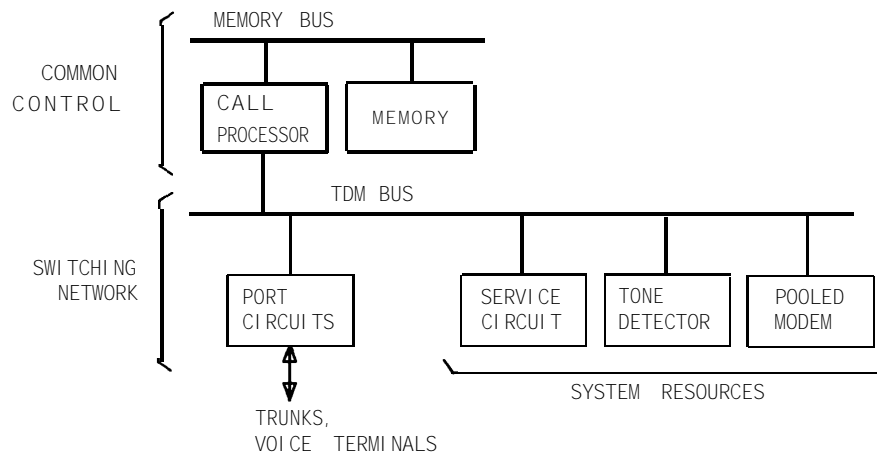


Figure 3-1. System 25 Digital Switch

Common Control

The Common Control circuitry consists of the Call Processor (ZTN82 in R1V1 or ZTN128 in R1V2) and Memory (ZTN81 in R1V1 or ZTN127 in R1V2) Circuit Packs and associated memory bus which is a 60-wire (including grounds), 39-bit (16-data, 23-address.), 6 MHz frontplane flat ribbon cable.

Call Processor Circuit Pack (ZTN82 or ZTN128)

The Call Processor runs the system feature code. It is powered from the backplane by +5 and -5 volts. It also draws -48 volts from the backplane to drive the Emergency Transfer Unit. Each system must include one Call Processor Circuit Pack. The Call Processor circuitry, as shown in Figure 3-2 includes:

- Microprocessor
- Memory management
- On-board memory
- EIA channels
- Network controller
- Clock
- Front plane interface
- Reset circuitry
- Bus error circuitry
- Interrupt circuitry
- Emergency Transfer Unit Control.

Microprocessor: A 68010 16-bit microprocessor that executes call processing and data processing features. This includes all maintenance, administration, testing, and reporting software.

Memory Management: Memory management separates the on-board Random Access Memory (RAM) into 1024 memory pages of 256 bytes each. Each page is read and write protected, generates bus errors when violated, and each is recappable allowing data areas to remain contiguous.

On-Board Memory: On-board memory includes 64k bytes of Read Only Memory (ROM) containing the power-up tests and the switch operating system. In addition, there is 80k bytes of protected RAM containing writable data storage for call processing. The RAM is backed up by an on-board trickle-charge battery that maintains memory contents for up to two months. Of the 80k RAM, 24k is dedicated to translation data. The remainder is dedicated to call status data and the operating system message queues.

EIA Channels: Four asynchronous RS-232 EIA ports (1-4) are included to permit communication with an administration terminal, a Station Message Detail Recording (SMDR) device, and a digital tape unit. (The fourth port is reserved for future use.) Each port can support 300, 1200,4800, or 9600 baud rates.

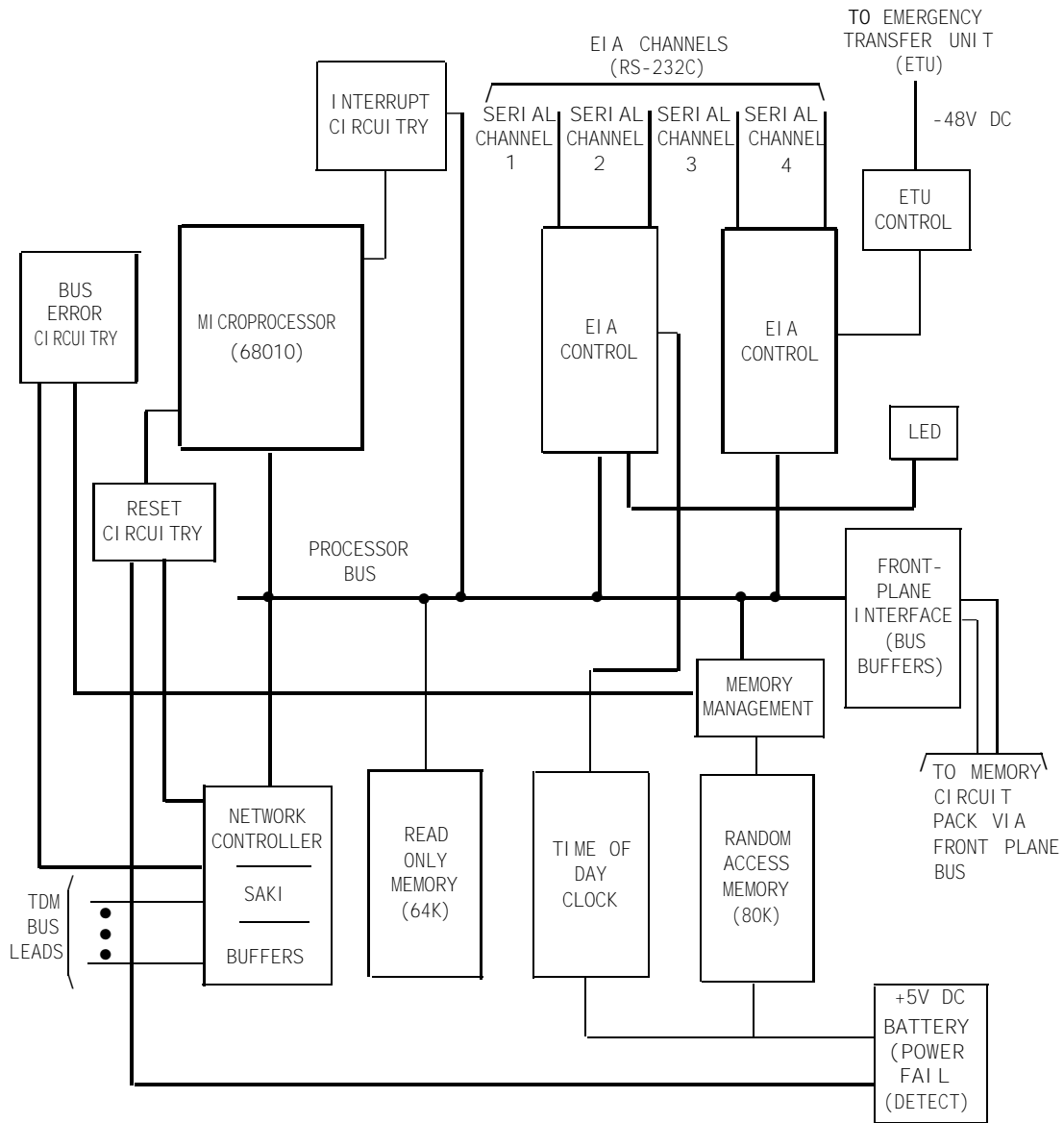


Figure 3-2. Call Processor (ZTN82 or ZTN128) Circuitry

Network Controller: The network controller transmits control channel messages between the Call Processor and the port circuits over the TDM bus. The controller also monitors system clocks.

The controller includes an 8-bit microprocessor that acts as a throttle passing messages between the Call Processor and the port board microprocessors.

All uplink messages from the port circuits are checked for consistency and passed to the Common Control. The controller is the distribution control point for all downlink control messages. It continuously scans, over the TDM bus, the port circuit microprocessors for sanity and activity. External RAM associated with this microprocessor- stores control channel information and port related information.

The controller consists of bus buffers and a Sanity and Control Interface (SAKI). The bus buffers provide the interface between the TDM bus and the on-board data buses to the SAKI. The SAKI receives and transmits control messages on the first five time slots on the TDM bus. The microprocessor communicates with the SAKI and external RAM over the address and data bus.

Clock: A clock provides both time-of-day information (in seconds, minutes, and hours), and the date to the 68010. The clock automatically adjusts for leap years. An on-board battery backs up the clock, so that accurate time is maintained even when the system power is off.

Front Plane Interface: Dedicated buffers provide an interface to the front plane, which is the communication path to the Memory Circuit Pack.

Reset Circuitry: The processor is automatically reset when power is turned on, when the +5 volt power supply drops below 4.5 volts (after it returns to +5 volts), or when the network controller determines that the processor is not functioning correctly. The processor can also reset the network controller when it determines that the network controller is not functioning correctly.

Bus Error Circuitry: Bus errors suspend the processor from executing code. Bus errors are generated when memory management detects illegal reads or writes to RAM, when the processor attempts to access circuit packs or chips not physically present, or when the network controller determines that the processor is not functioning correctly.

Interrupt Circuitry: Interrupts are prioritized into seven levels, of which the highest (level 7) is nonmaskable. The interrupts are:

<i>Interrupt</i>	<i>Level</i>
AC Fail	7
Work cycle	6
Off board	5
EIA ports 3 and 4	4
EIA ports 1 and 2	3
Off board	2
Off board	1

Emergency Transfer Unit (ETU) Control: Removes -48V de power from the ETUs of the system when the system loses power or a major system malfunction occurs.

Memory Circuit Pack (ZTN81 or ZTN127)

The Memory Circuit Pack provides for the storage of software associated with system operation. This software includes call and administration processing, and other related programs. The circuit pack is powered from the backplane by +5 volts. Each system must include one Memory Circuit Pack. The Memory Circuit Pack circuitry (Figure 3-3) includes:

- Address and data buffers
- ROM array
- ROM select
- Timing and control logic
- Built-in TDM bus termination resistors.

Address and Data Buffers: The address and data buffers interface the Memory Circuit Pack to the address and data lines on the front plane.

ROM Array: The memory array consists of 16 ROM devices of 32K, 8 bit bytes each, for a total capacity of 512K ROM. The ROMs are organized into pairs allowing the Call Processor to access 16 bit words.

ROM Select: The memory selects the proper pair of ROMs according to address information.

Timing and Control Logic Circuit: Controls the access speed of the ROM (no wait states) by returning a Data Transfer Acknowledge signal at the proper time.

Bus Terminators: These resistors are required for proper operation of the TDM bus. The Memory Circuit Pack provides the proper termination for one end of the bus, and a plug-in TDM bus termination circuit card (plugs into cabinet backplane) is used to terminate the other end. For this reason, the Memory Circuit Pack must always be located in slot #1 of Cabinet 1.

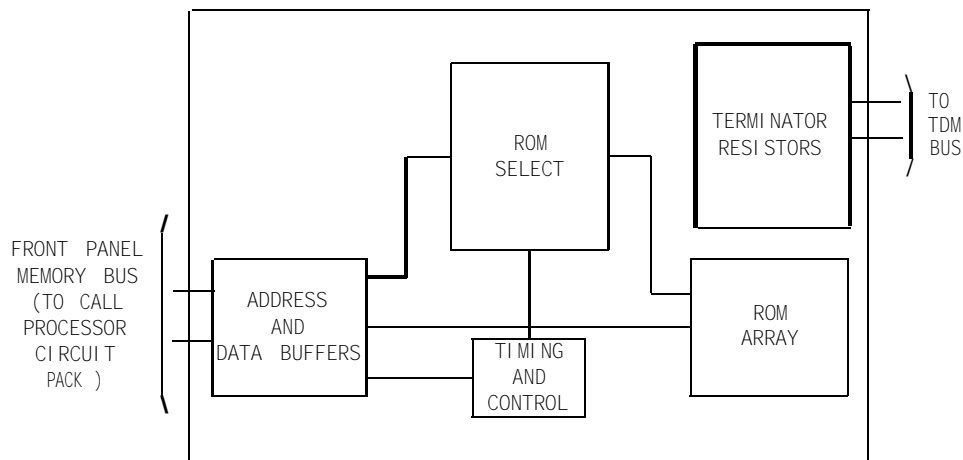


Figure 3-3. Memory (ZTN81 or ZTN127) Circuitry

Switching Network

System 25 *uses* distributed processing techniques to provide switched voice and data services. The switch operates at 64 Kbps. The switching network consists of the following:

- Time Division Multiplex (TDM) bus
- Port Circuits
- System Resource Circuits.

The TDM bus connects the intelligent ports to the Common Control circuit packs and other ports through the network control circuit. The system resource circuits provide tone sources, receivers, detectors, and pooled modems. The intelligent ports connect external communications facilities to the TDM bus.

TDM BUS

The TDM bus consists of two groups of eight signal leads and five control leads, each with matching grounds. The port circuit packs place digitized voice [pulse code modulated (PCM)] signals on the bus.

The bus operates at 2.048 MHz. The framing pulse rate is 8 kHz. This provides 256 time slots (0-255) on the bus. The time slots are 488 ns wide. Time slots are generated as shown in Figure 3-4. The first five time slots are used for communications between the Common Control, the intelligent port, and resource circuit packs.

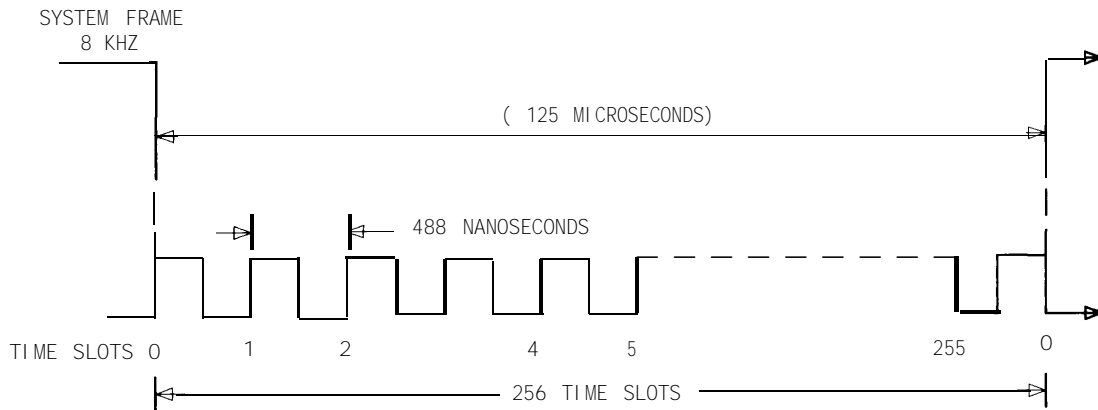


Figure 3-4. TDM Bus Time Slot Generation (Not A Timing Diagram)

Two time slots are required for each 2-party conversation. Each party transmits (talks) on one time slot and receives (listens) on another. Only five parties are allowed in a conference. During a conference connection, each member of the conference transmits on an individual time slot while receiving on as many as four other time slots. The actual switch capacity is 115 simultaneous 2-party conversations).

Table 3-A shows the allocation of the 256 time slots. Five are used for system control, 15 for tones, 235 for call processing, and one is not used.

Physical Characteristics

The TDM bus is an 8-bit bus. The bus snakes continuously between cabinets in a multicabinet system as shown in Figure 3-5. The total length is about 9 feet for a three cabinet system. The bus is driven from any of the circuit packs in the cabinets. Similarly, a signal on the bus can be received by any circuit pack.

Within a cabinet, the bus is printed on one side of the circuit pack carrier backplane while the other side is solid ground. Ribbon cables are used to cable the TDM bus between cabinets in a multi-cabinet system.

Electrical Characteristics

The TDM bus is an unbalanced, low characteristic impedance transmission line. Paths printed over a ground plane on the carriers and the flat ribbon cables between carriers maintain this impedance level over the full length of the bus.

One end of the bus is terminated to ground with a bus termination circuit card and the other end is terminated by a network on the ZTN81B or ZTN127 Memory CP. Each circuit pack connects to the bus through a custom bus driver device. The bus driver is a switchable constant current source so that even in the "high" output state there is no bus loading to cause reflections. The current output of the drivers is adjusted so that logic "high" is 1.5 volts compared to a "low" of 0 volts.

TABLE 3-A. TDM BUS Time Slots

Time Slot No.	Function
00 thru 04	Control (5)
05 06 07 08 09 10 11 12 13 14 15 16 17 18 19	-Tones- Dial Tone Busy Tone Re-order Tone Ringback Tone Data-Null Voice-Null Music 697 Hz* 770 Hz* 852 Hz* 941 Hz* 1209 Hz* 1336 Hz* 1447 Hz* 1637 Hz* (15)
20 thru 254	Call Processing (235)
255	Not Used (1)

* These tones are used to generate touch-tone signals.

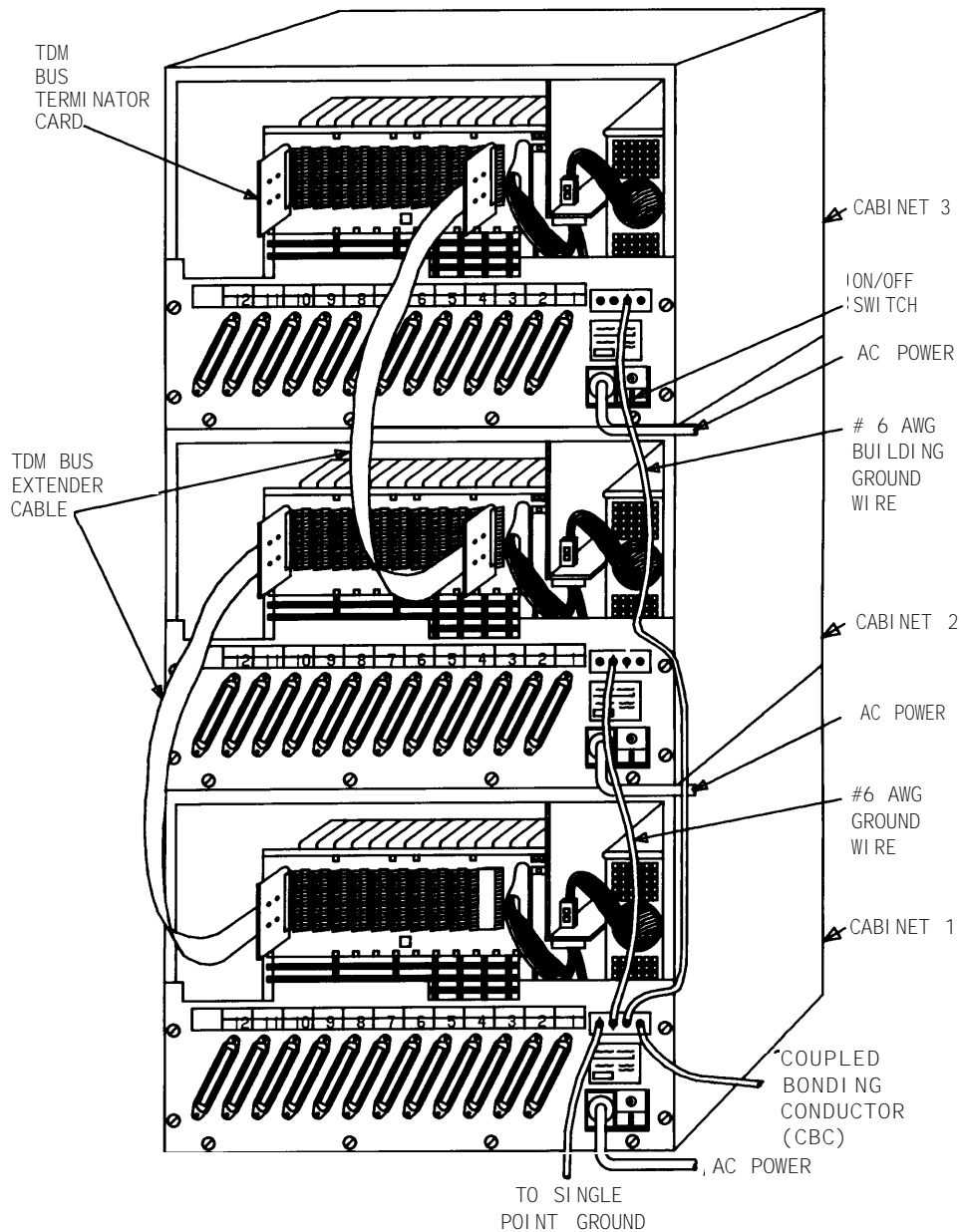


Figure 3-5. TDM Bus Diagram -Three Cabinet System

Port Circuits

The following port circuit packs provide the link between trunks and external equipment and the TDM bus:

- Analog Line (TN742)
- ATL Line (ZTN79)
- Auxiliary Trunk (TN763)
- Data Line (TN726)
- DID Trunk (TN753)
- Ground Start Trunk (ZTN76)
- Loop Start Trunk (ZTN77)
- MET Line (TN735)
- STARLAN Interface (ZTN84)
- Tie Trunk (TN760B)
- Tip Ring Line (ZTN78).

Figure 3-6 shows the equipment types that can be connected to the digital switch by the Call Processor and port circuit packs.

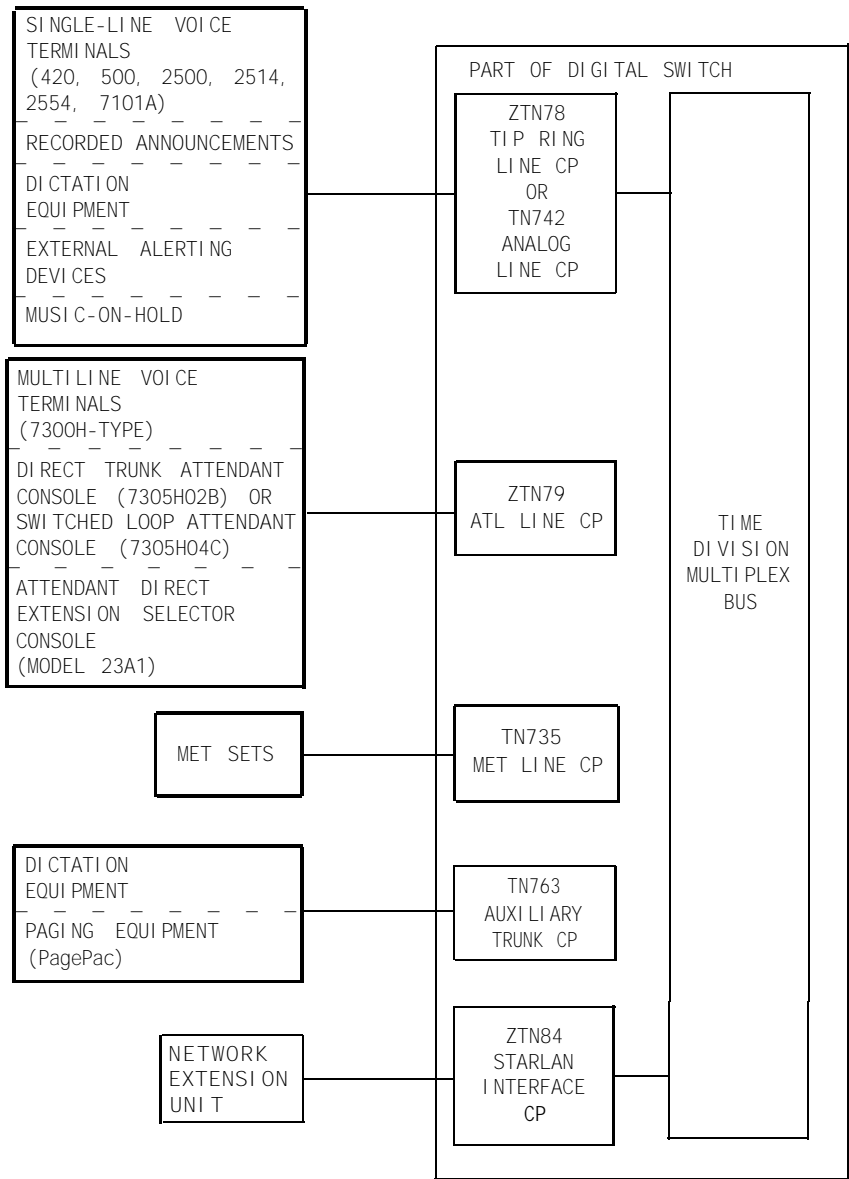


Figure 3-6. Equipment Connected to System 25 Via the Call Processor and Port Circuit Packs (Sheet 1 of 3)

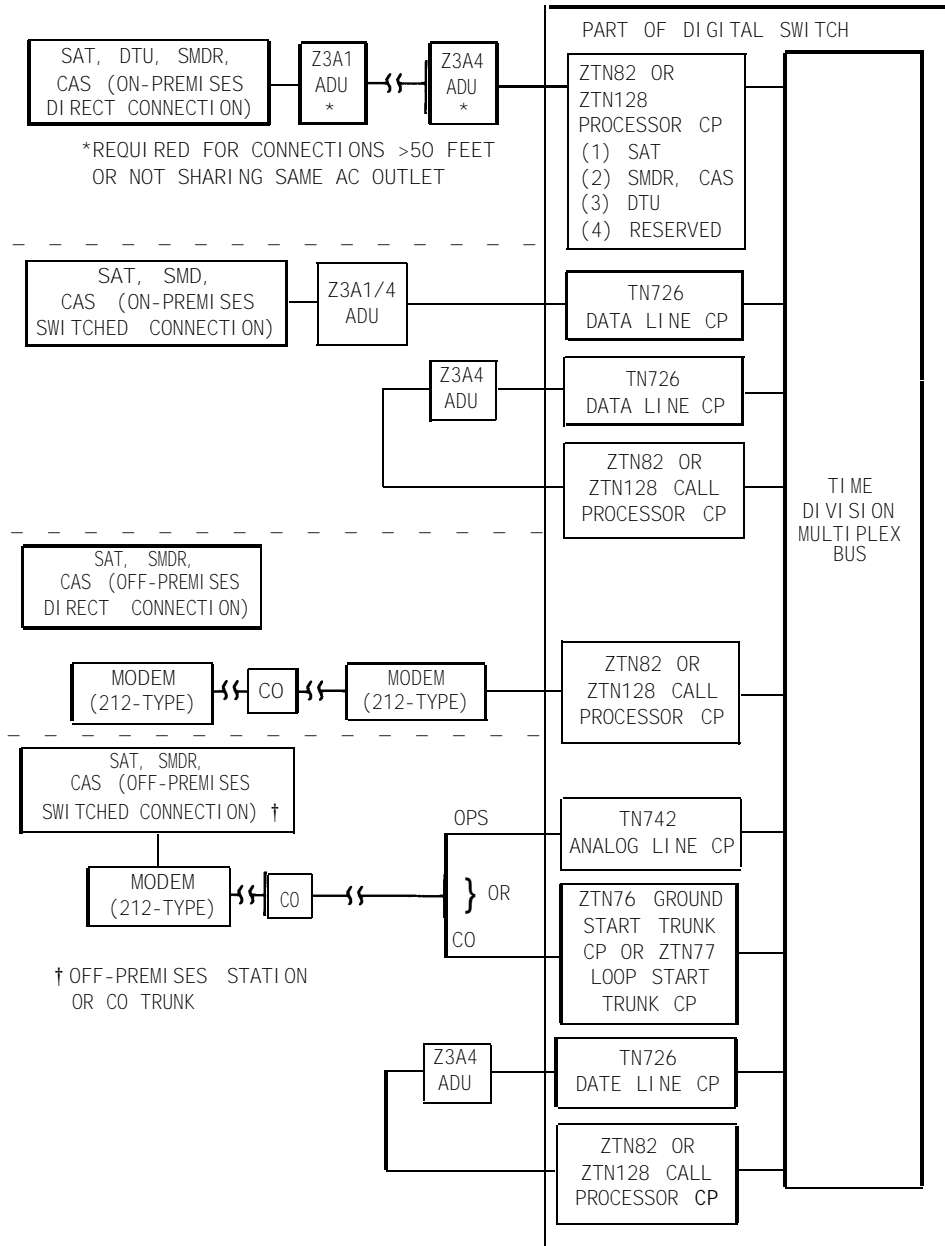


Figure 3-6. Equipment Connected to System 25 Via the Call Processor and Port Circuit Packs (Sheet 2 of 3)

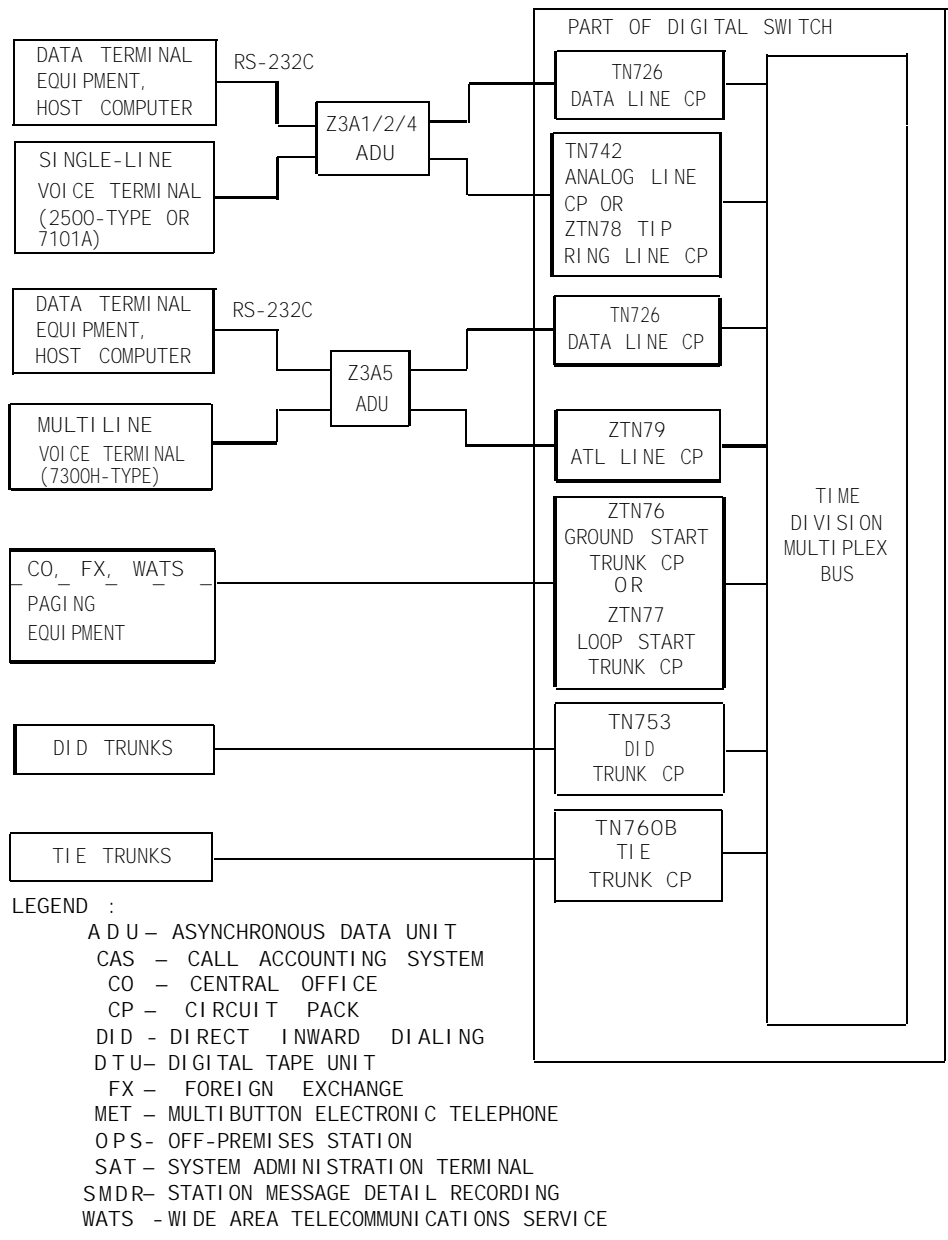


Figure 3-6. Equipment Connected to System 25 Via the Call Processor And Port Circuit Packs (Sheet 3 of 3)

Eight port circuits are provided on most port circuit packs. The Multibutton Electronic Telephone (MET) Line, Tie Trunk, and Auxiliary Trunk Circuit Packs each contain four port circuits. The port circuits provide an interface between terminals/trunks and the TDM bus.

The number of port circuit packs required varies according to customer requirements and equipment configuration.

Each of the System 25 port circuit packs contain a number of common elements (see Figure 3-7) as well as the unique port circuits. The common elements are as follows:

- Bus buffers
- Sanity And Control Interface (SAKI)
- On-board microprocessor with external Random Access Memory (RAM)
- One or more Network Processing Elements (NPEs)
- Circuit Pack Address Leads.

Bus Buffers: The bus buffers are the digital interface between the backplane TDM bus wires (system bus) and the on-board circuitry (data bus). They also receive and distribute clock and frame signals.

SAKI (Sanity And Control Interface): The SAKI is the control interface between the Common Control that sends information via the network control circuit down the TDM buses and the on-board circuitry controlled by the on-board microprocessor. The SAKI receives control information (down-link messages) on the first five time slots and, as requested by the on-board microprocessor, transmits control information (up-link messages) on these same time slots.

The SAKI also does the following functions:

- Identifies the circuit pack to the Common Control (location and vintage)
- Controls status indicator Light-Emitting Diodes (LEDs) - red (failure), green (translated), and yellow (circuit busy)
- Initiates power-on startup procedures
- Checks the on-board microprocessor for sanity and causes reinitialization in case of problems
- Takes NPEs out of service under control of the on-board microprocessor
- Resets the protocol handler on the ATL Line Circuit Pack
- Takes the whole circuit pack out of service on command from the Common Control or when it determines that on-board interference is present in the control time slots.

On-Board Microprocessor With External RAM: The on-board processor does all low level functions such as scanning for changes and relay operations. In general, it carries out commands received from the Common Control and reports status changes to it. The external RAM stores control channel information and port-related information.

NPEs (Network Processing Element): Each port circuit pack contains one or two NPEs. The Analog Line, ATL Line, Tip Ring, Data Line, Ground Start, Loop Start, and DID Trunk circuit packs contain two NPEs. The MET Line, Auxiliary Trunk, and Tie Trunk Circuit Packs contain one NPE.

The NPEs do switching network functions for the port circuits. Under control of the on-board microprocessor, an NPE can connect a port circuit to any one of the TDM bus time slots. More specifically, it allows a port circuit to talk on one time slot and listen to the same time slot (NPE sidetone) and on up to four other time slots at the same time. In 2-wire circuits that provide their own sidetone, the NPE sidetone is not used.

Circuit Pack Address Leads: Seven leads (BA0-BA6) are tied to Corresponding logic levels to uniquely identify each CP slot in the system, including multiple cabinet systems. The logic values on leads BA4 and BA5 are used to identify the cabinet (Cabinet 1, 2, or 3) and are tied via the cabinet address plugs to either +5V de or ground, as appropriate. Lead BA6 is tied to ground.

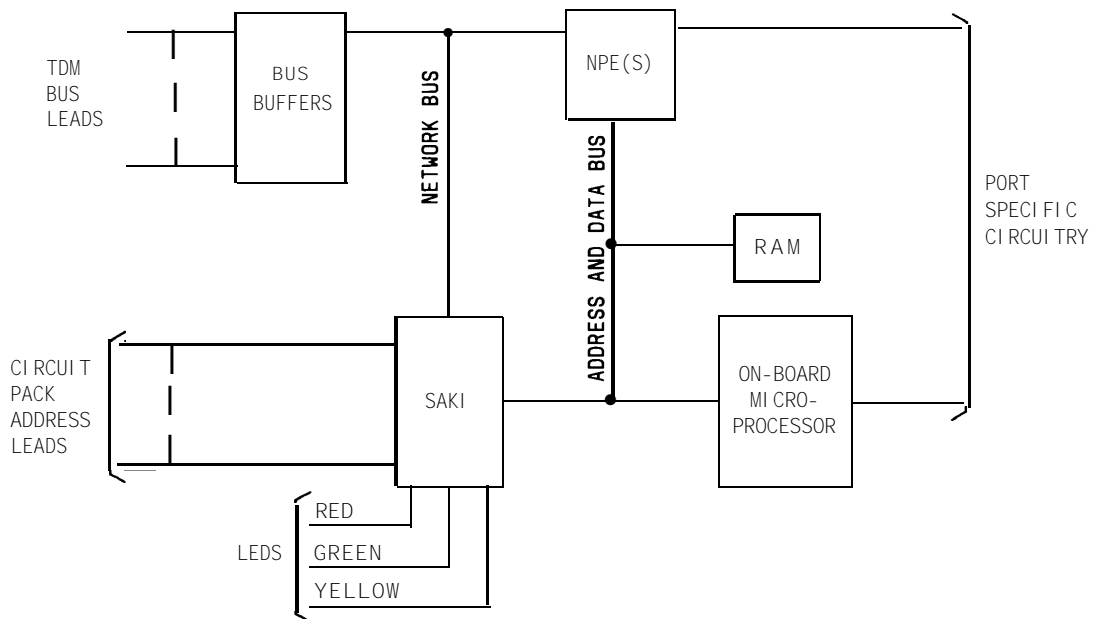


Figure 3-7. Port Circuit Pack Common Circuitry

Ground Start Trunk (ZTN76)

The Ground Start Trunk Circuit Pack interfaces eight central office trunks and the TDM bus. Figure 3-8 shows the following Ground Start Trunk unique circuitry:

- Ground detector circuit
- Port Input/Output (I/O) circuit
- Eight port circuits.

Ground Detector Circuit: The ground detector circuit determines if ground has been applied to the tip lead for incoming seizure. It also senses tip ground on outgoing seizure indicating dial tone is present. One ground sensor is used for each port circuit. Input for the ground sensor comes from the port circuit as an analog current to the -48 volt dc supply. The output of the ground sensor is a port control point to the port I/O circuit.

Port I/O Circuit: This circuit consists of bus expanders for communication between the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the on-board microprocessor.

Port Circuits: The eight port circuits are identical. Each port circuit consists of a coder/decoder (codec), hybrid circuit, line transformer, relay driver, and surge protection circuit.

The codec is a 4-wire circuit that converts the NPEs digital output to an analog signal. Likewise, it converts the analog signal from a central office trunk to a Pulse Code Modulated (PCM) data signal to the NPE. The hybrid circuit converts the codec 4-wire analog signal to a 2-wire analog signal that is connected to the central office trunk by the line transformer.

The relay driver buffers and inverts the relay drive signals from the port I/O circuit so that a logic high input operates the appropriate relay. The relays control circuitry provides the proper signaling for ground start trunks. The trunks support touch-tone dialing. The surge protection circuit provides overvoltage lightning surge protection for the circuit pack.

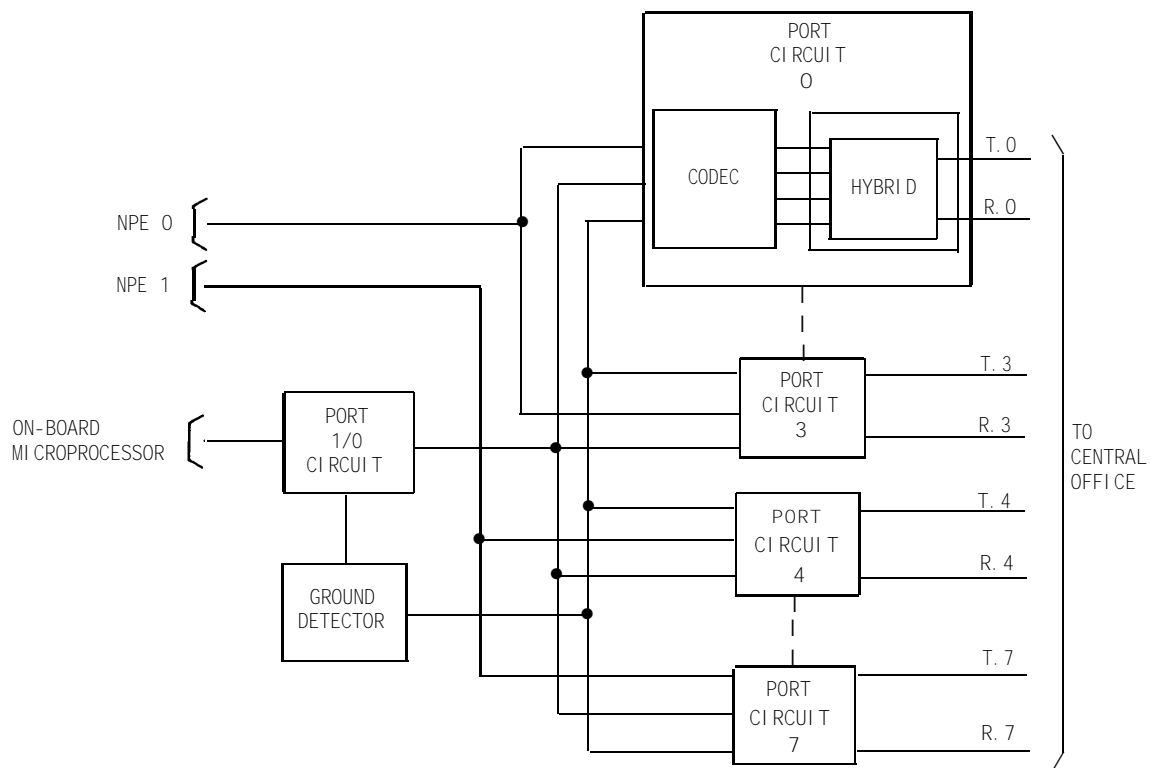


Figure 3-8. Unique Ground Start Trunk (ZTN76) Circuitry

Loop Start Trunk (ZTN77)

The Loop Start Trunk Circuit Pack interfaces eight central office loop start trunks and the TDM bus.

Figure 3-9 shows the following Loop Start Trunk unique circuitry:

- Port Input/Output (I/O) circuit
- Eight port circuits.

Port I/O Circuit: This circuit consists of bus expanders for communication between the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the on-board microprocessor.

Port Circuits: The eight port circuits are identical. Each port circuit consists of a codec, hybrid circuit, line transformer, relay driver, and surge protection circuit.

The codec is a 4-wire circuit that converts the NPEs output to an analog signal. Likewise, it converts the analog signal from a central office trunk to a PCM data signal to the NPE. The hybrid circuit converts the codec 4-wire analog signal to a 2-wire analog signal that is connected to the central office trunk by the line transformer.

The relay driver buffers and inverts the relay drive signals from the port I/O circuit so that a logic high input operates the appropriate relay. The relays control circuitry provides the proper signaling for loop start trunks. The trunks support touch-tone dialing and dial pulse signaling. The surge protection circuit provides overvoltage lightning surge protection for the circuit pack.

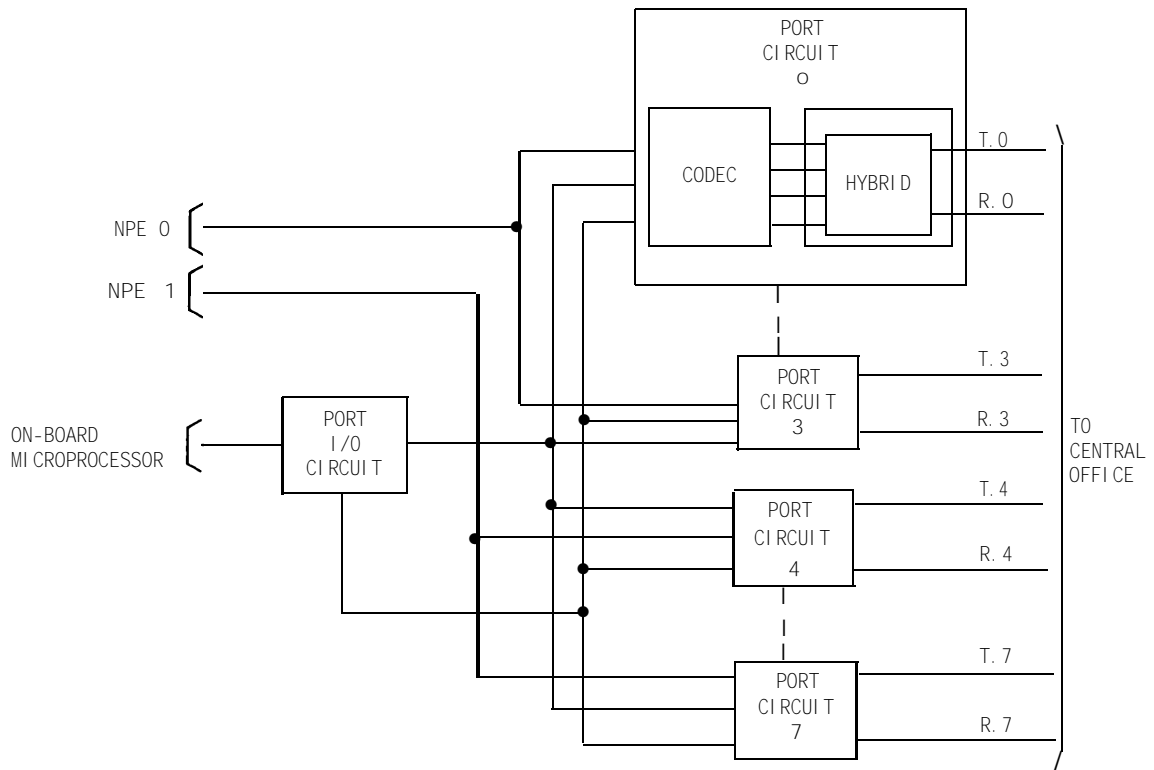


Figure 3-9. Unique Loop Start Trunk (ZTN77) Circuitry

Tip Ring Line (ZTN78)

The Tip Ring Line Circuit Pack interfaces eight analog tip and ring voice terminal lines (single-line voice terminals) and the TDM bus. Figure 3-10 shows the following Tip and Ring Line unique circuitry:

- Ringing application circuit
- Port Input/Output (I/O) circuit
- Eight port circuits.

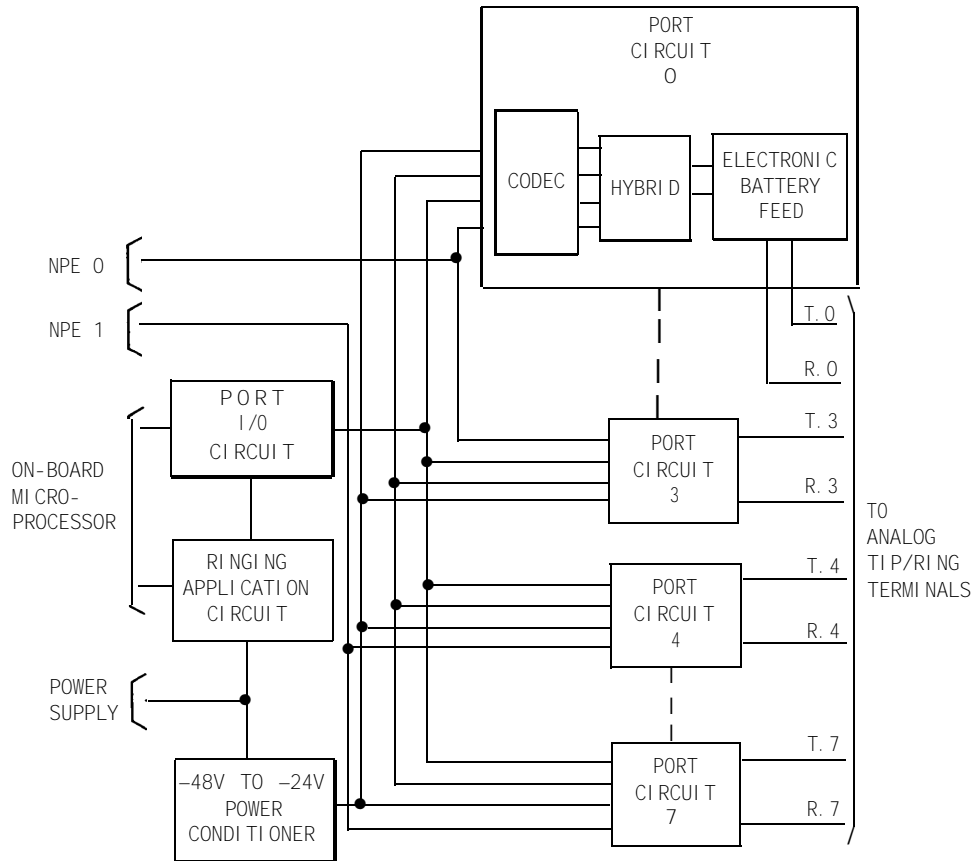


Figure 3-10. Unique Tip Ring Line (ZTN78) Circuitry

Ring Application Circuit: This circuit receives ringing voltage from the power supply. It monitors ringing voltage and current and generates signals to the on-board microprocessor indicating zero ringing voltage and current. It also detects when a terminal user has lifted the receiver during ringing preventing the application of ringing to the terminal's handset receiver.

Port I/O Circuit: This circuit includes bus expanders connecting the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the on-board microprocessor.

-48 V To -24 V Power Conditioner: This circuit converts -48 V power from the power supply into a conditioned source of -24 V power for the electronic battery feed circuits.

Port Circuits: Each port circuit is identical. A port circuit consists of a coder/decoder (codec), hybrid circuit, battery feed circuit, and ring relay.

The codec is a 4-wire circuit that converts the NPEs output to an analog signal. Likewise, it converts the analog signal from a central office trunk to a PCM data signal to the NPE. The hybrid circuit converts the codec 4-wire analog signal to a 2-wire analog signal that is connected to the central office trunk by the line transformer.

The battery feed circuit provides talking battery to the voice terminal. It also detects when a receiver is lifted, and provides the message waiting signal by periodically reducing the feed voltage to zero.

The ring relay provides the interface between the ringing application circuit and the port circuit. It causes ringing to turn on and off.

Note: The TN742 can be used instead of the ZTN78 Tip Ring CP. The TN742 supports up to five bridged single-line voice terminals; however, only two can be off-hook at one time. The ZTN78 does not support bridged terminals. In addition, the TN742 supports out-of-building, extended, and off-premises stations, while the ZTN78 does not. The ZTN78 supports only a 1.2 Ringer Equivalency Number (REN).

ATL Line (ZTN79)

The ATL Line Circuit Pack interfaces eight hybrid voice terminal (7300H series) lines and the TDM bus. It terminates three pairs of wires from each terminal: analog voice pair, digital control pair, and power pair. Figure 3-11 shows the following ATL Line unique circuitry:

- Protocol handler
- Eight port circuit.

Protocol Handler: The 8-bit on-board microprocessor translates the control information in Control Channel Message Set (CCMS) message format to the control information message format used by the 7300H series voice terminals. The protocol handler sends the messages to the terminals via transceivers located in the port circuits.

Port Circuits: Each port circuit is identical. A port circuit consists of an analog port, one-half of a transceiver, and an electronic power feed device.

The analog port circuit consists of a codec, a hybrid circuit, an isolation transformer, and associated power filtering circuitry. The codec and hybrid circuit perform the same function as the codec and hybrid circuit in the Analog Line Circuit Pack (TN742). The output of the hybrid circuit is connected to the primary of the isolation transformer. The secondary of the transformer is connected to the analog voice pair.

The transceiver interfaces the voice terminal pair to the protocol handler. The electronic power feed device provides -24 volts dc on the power pair to the voice terminal. The device is polled by the on-board microprocessor, periodically and on demand, to test for an overcurrent or no-current condition.

Each Electronic Power Feed (EPF) circuit supports two ports. If one of the associated lines becomes overloaded, the associated pair of lines will also be out of service. One EPF supports Ports 0 and 1, one Ports 2 and 3, one Ports 4 and 5, and one Ports 6 and 7. The on/off state of the device is controlled by the on-board microprocessor.

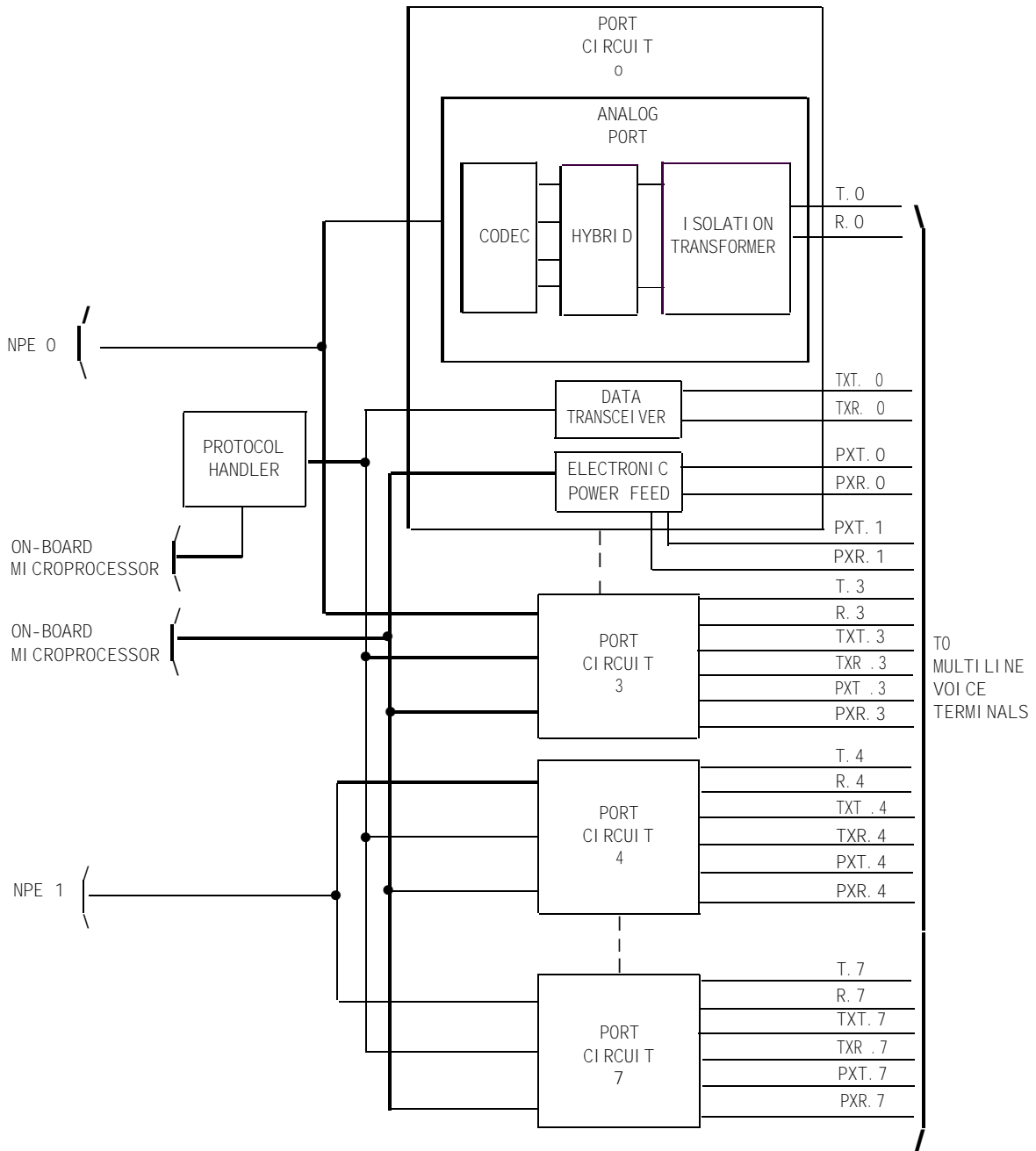


Figure 3-11. Unique ATL Line (ZTN79) Circuitry

Data Line (TN726)

The Data Line Circuit Pack interfaces eight Asynchronous Data Units (ADUs) data devices and the TDM bus. The ADUs are typically, in turn, connected to RS-232 type devices. Figure 3-12 shows the Data Line unique circuitry that includes:

- A bit clock
- Bus isolation
- Eight port circuit.

Bit Clock: The bit clock circuitry is used to provide the Octal Asynchronous Terminal Mode Two EIA Asynchronous LSIs (OATMEALS) with a clock frequency that is a multiple of each baud rate. In addition, the clock rate is divided down to 160 kHz. The 160 kHz is then compared to the 160 kHz data clock of the system, and is phase-locked to the system clock. The phase-locked circuit is required for low speed operation.

Bus Isolation: This portion of the circuit pack is used to isolate the microprocessor bus. Isolation is required because the realized bus load exceeds the maximum limit specified for this device, due to the large number of devices controlled by the NPE. The OATMEALS are isolated from the common bus structure.

Port Circuits. Each of the eight identical port circuits allows the connection of interface equipment having an RS-232 compatible serial interface to the switch. The circuit provides asynchronous full duplex data transport at standard speeds from 300 to 19,200 bps and a low data rate (<300 bps). Each port includes an Asynchronous Data Unit (ADU) to extend the serial communications link length and provide safe isolation. The ADU terminates to another ADU at the Customer Provided Equipment (CPE). The distance between the digital switch and CPE is inversely proportional to the speed at which the link is run.

Throughout the circuit, various gates are used to provide a means of isolating devices for automated circuit pack testing. Typically, these devices are crystal oscillators or memory components attached to the microprocessor bus.

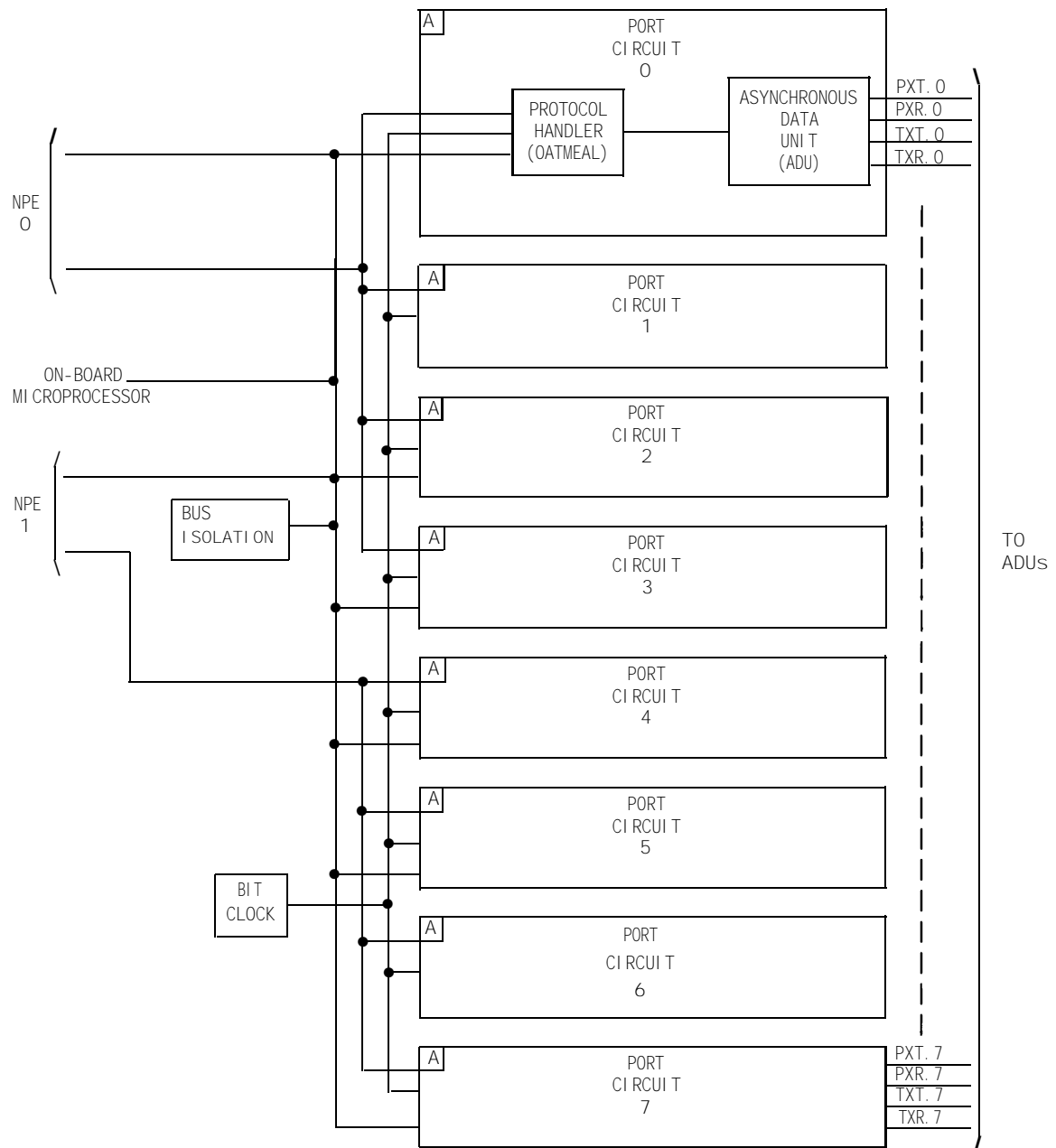


Figure 3-12. Unique Data Line (TN726) Circuitry

MET Line (TN735)

The MET Line Circuit Pack interfaces four Multibutton Electronic Telephone (MET) lines and the TDM bus. The MET Line unique circuitry consists of four port circuits as shown in Figure 3-13.

Port Circuits: The four port circuits are identical. Each port circuit consists of an analog port, a digital port, and an electronic power feed device.

The analog port circuit consists of a codec, a hybrid circuit, an electronic battery feed, and a power filter. The codec, hybrid circuit, and power filter perform the same function as in the Analog Line Circuit Pack (TN742). The electronic battery feed provides talking battery to the MET set. The electronic battery feed produces a controlled de battery feed current for short and long loops and detects when a MET set user lifts a receiver.

The digital port circuit provides a full duplex channel over two 2-wire pairs. All outgoing lamp (LT, I,R) and incoming button depression (BT, BR) information is carried on these channels. Ringing and switchhook information is also sent over these channels.

The electronic power feed device provides phantom -48 volt dc power for the MET terminals over the data channels. The electronic power feed device is a “smart” circuit breaker. When it senses an overcurrent condition, it indicates the condition on an output lead and goes into thermal shutdown if not turned off by the on-board microprocessor. When the overcurrent condition disappears, the circuit breaker can be turned on by the on-board microprocessor.

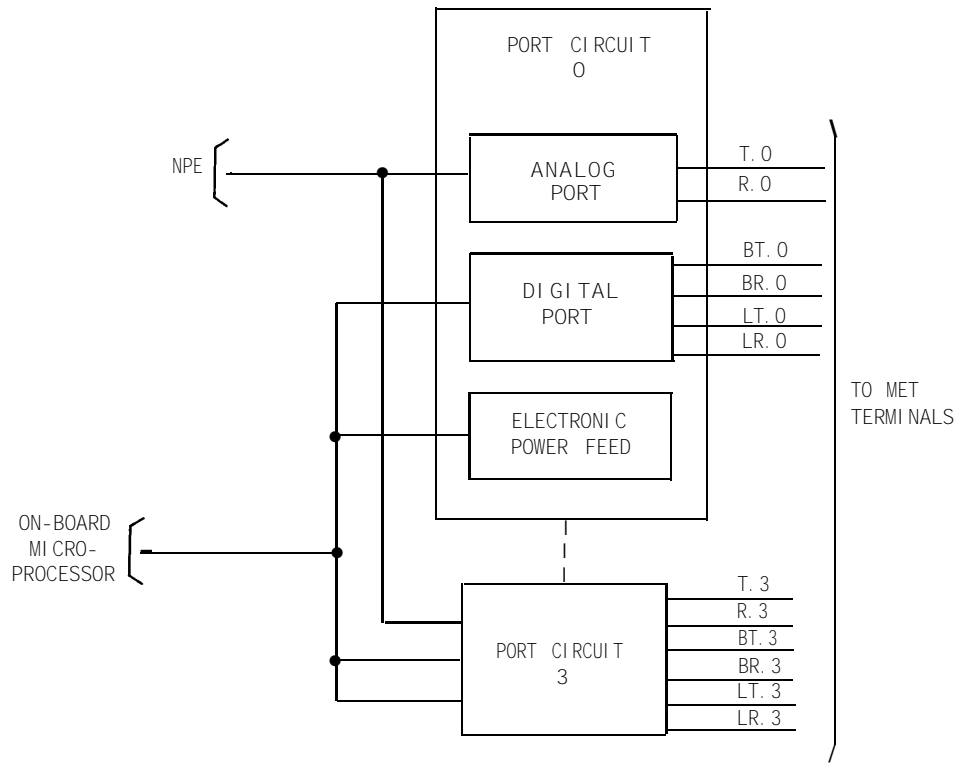


Figure 3-13. Unique MET Line (TN735) Circuitry

Analog Line (TN742)

The Analog Line Circuit Pack interfaces eight analog voice terminal lines and the TDM bus. Figure 3-14 shows the following Analog Line unique circuitry:

- Ringing application circuit
- Port Input/Output (I/O) circuit
- Eight port circuits.

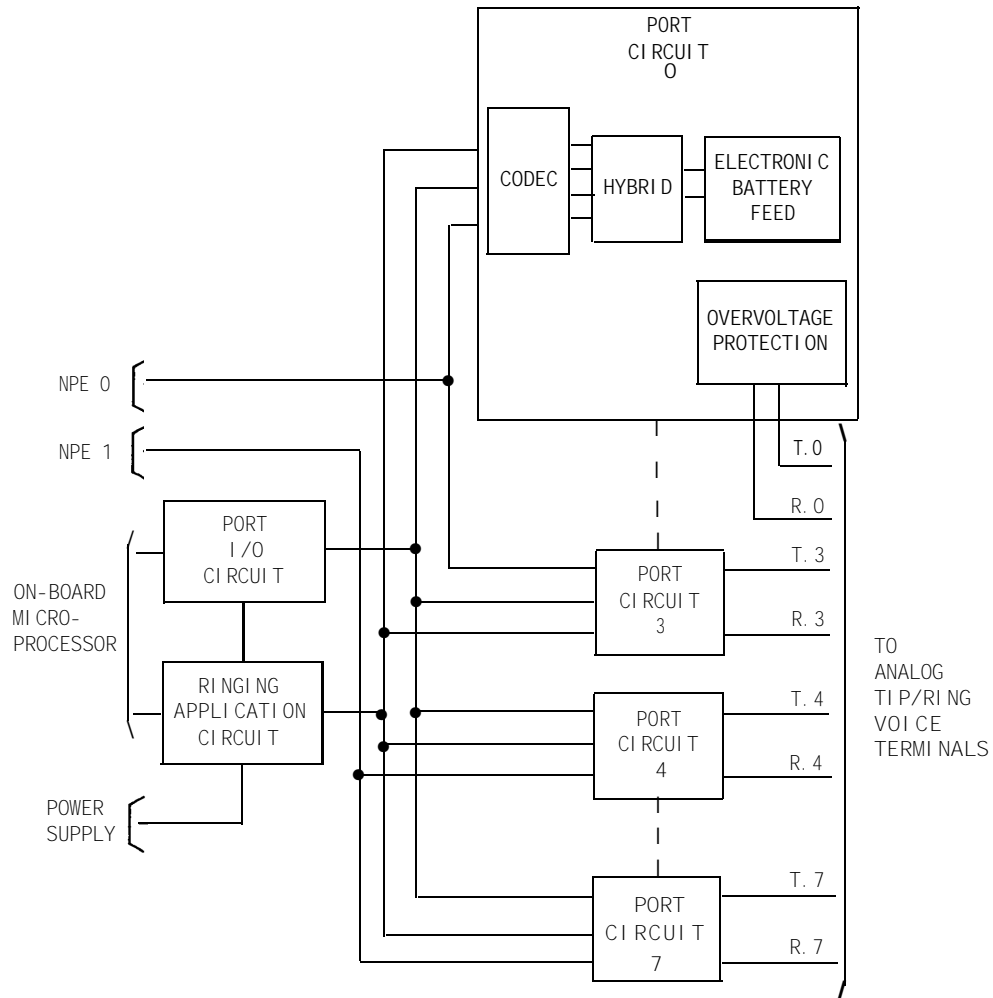


Figure 3-14. Unique Analog Line (TN742) Circuitry

Ring Application Circuit: This circuit receives ringing voltage from the power supply. It monitors ringing voltage and current, generates signals to the on-board microprocessor indicating zero ringing voltage and current, and detects a terminal user lifting the receiver during ringing. This prevents the application of ringing to the port circuit when a terminal user lifts the receiver during the ringing phase. Maintenance circuitry is also included. The maintenance circuitry detects when a terminal is connected to the port circuitry and checks for faults in the ringing application circuitry.

Port I/O Circuit: This circuit consists of bus expanders connecting the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the on-board microprocessor.

Port Circuits: The eight port circuits are identical. Each port circuit consists of a coder/decoder (codec), hybrid circuit, electronic battery feed circuit, ring relay, and overvoltage surge protection circuit.

The codec is a 4-wire circuit that converts the analog signal from a voice terminal to a PCM data signal. It converts an incoming PCM data signal from the NPEs to an analog signal. The hybrid circuit converts the 4-wire analog signal from the codec to a 2-wire analog signal that is connected to the analog line. Filtered power is provided for the codec and hybrid circuits.

The electronic battery feed circuit provides talking battery to the voice terminal. It also produces a controlled de battery feed for short and long loops, detects when a receiver is lifted, and provides the message waiting signal by periodically turning off the feed voltage.

The ring relay provides the interface between the ringing application circuit and the port circuit. It causes ringing turn on and turn off.

The overvoltage surge protection circuit provides lightning surge and power line cross protection for the circuit pack.

Note: The TN742 can be used instead of the ZTN78 Tip Ring CP. The TN742 supports up to five bridged single-line voice terminals, however, only two can be off hook at one time. The ZTN78 CP does not support bridged terminals. In addition, the TN742 supports out-of-building, extended, and off-premises stations, the ZTN78 does not.

DID Trunk (TN753)

The DID Trunk Circuit Pack interfaces eight central office trunks arranged for Direct Inward Dialing ((DID) and the TDM bus. Figure 3-15 shows the following DID Trunk unique circuitry:

- Port Input/output (I/O) circuit
- Eight port circuits.

Port I/O Circuit: This circuit consists of bus expanders for communication between the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the on-board microprocessor.

Port Circuits: The eight port circuits are identical. Each port circuit consists of a codec, balance network, trunk interface unit, and loop termination circuit.

The codec is a 4-wire circuit that converts the NPEs output to an analog signal. Likewise, it converts the analog signal from the Central Office (CO) to a PCM signal to the NPE.

The trunk interface unit contains a hybrid, a 2-wire interface circuit, and control circuitry. The hybrid circuit converts the 4-wire analog signal from the codec to a 2-wire analog signal that is connected to the analog line by the 2-wire interface circuit. The control circuitry controls loop current, internal signal gain, terminating resistance, battery feed shutdown, and battery reversal. The circuit pack accepts both dial pulse and touch-tone signaling.

The loop termination circuit provides a fixed impedance to the DID trunk.

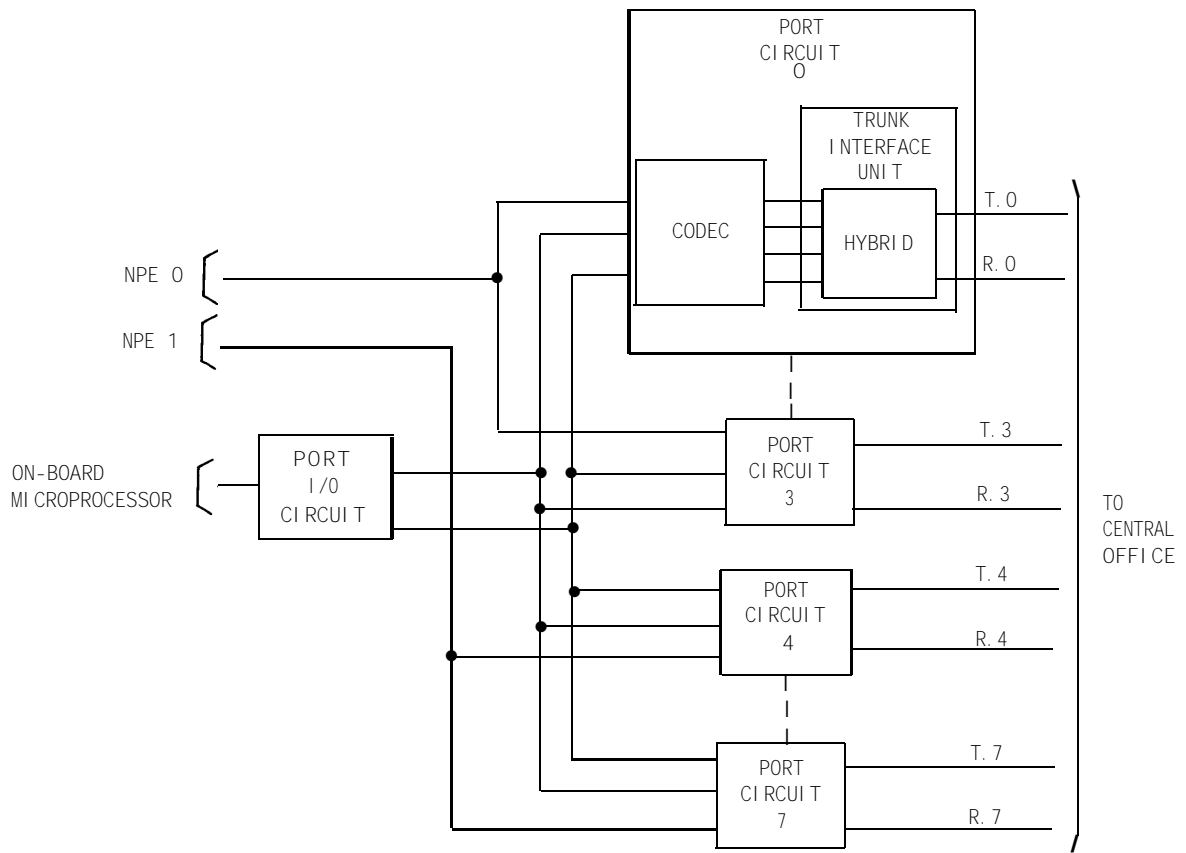


Figure 3-15. Unique DID Trunk (TN753) Circuitry

Tie Trunk (TN760B)

The Tie Trunk Circuit Pack interfaces four 6-wire tie trunks and the TDM bus. Two tip and ring pairs form a 4-wire analog transmission line. An E and M pair are used for signaling. The T and R pair transmit analog signals from the circuit pack. The T1 and R1 pair receive analog signals from the tie trunk. The E and M pair are dc signaling leads used for call setup handshaking. The E lead receives signals from the tie trunk and the M lead provides signals from the circuit pack. The TN760Bs four port circuits support Type I, Type I Compatible, or Type V signaling. Incoming and outgoing trunks can be either automatic, immediate start, wink start, or delay dial. Figure 3-16 shows the following Tie Trunk unique circuitry:

- Ground detector circuit
- Port Input/Output (I/O) circuit
- Four port circuits.

Ground Detector Circuit: This circuit determines if a ground has been applied to the E lead. Ground detector inputs come from the port circuits as an analog current to the -48 volt dc supply. Its output is a port control point to the port I/O circuit.

Port I/O Circuit: This circuit consists of bus expanders for communication between the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the on-board microprocessor.

Port Circuits: The port circuits are identical, except for port 3 where part of the E-lead maintenance circuit is located. Each port circuit consists of a codec with associated input and output line transformers, analog operational amplifiers, a power filter, loop-around transistors, port control comparators, a relay driver, an electronic power feed device, an E-lead test maintenance circuit, and surge protection circuits.

The codec converts the incoming 4-wire analog signal from the tie trunk to a PCM data signal. The codec converts the incoming PCM data signal from the NPE to an analog signal. Outgoing and incoming line transformers provide de isolation to the tip and ring leads. Analog operational amplifiers provide amplification and buffering for the codec and network and loop-around gain compensation. Filtered power is provided to the codec and amplifiers.

The loop-around transistors are under control of the port control comparators and provide a loop-around path for the signal for testing purposes. The relay driver buffers and inverts the relay drive signals from the port I/O circuit so that a logic high input operates the appropriate relay. The relays and electronic power feed device control the M-lead circuitry to provide the proper signaling handshake for call progress tones and dial pulse dialing.

The electronic feed device provides a -48 volt dc current to the M-lead circuits. It also tests the M-lead circuits for opens or shorts and prevents uncontrolled operation during power-up. The E-lead test circuit provides a ground to the ground detector circuit for testing purposes. The surge protection circuitry provides lightning surge and power cross protection for the circuit pack. For each port circuit, E&M/Simplex and surge protection are selected by switch settings as shown on Figure 3-17.

The signaling type is administrable for each port. Table 3-B summarizes the conditions present as the transmit and receive control signals for each signaling type. Table 3-C lists the preferred TN760B tie trunk signaling format to be used in the likely-to-be-encountered installation situations.

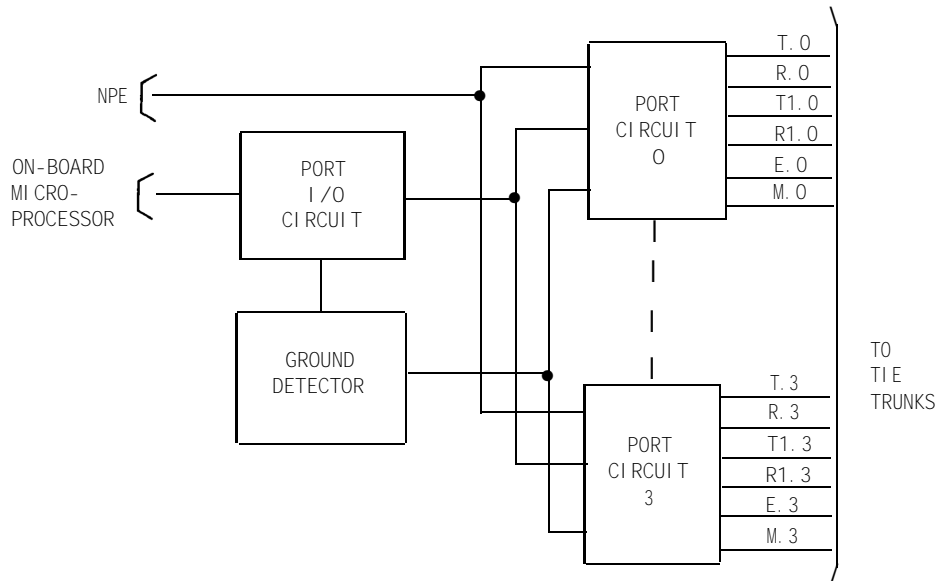


Figure 3-16. Unique Tie Trunk (TN760B) Circuitry

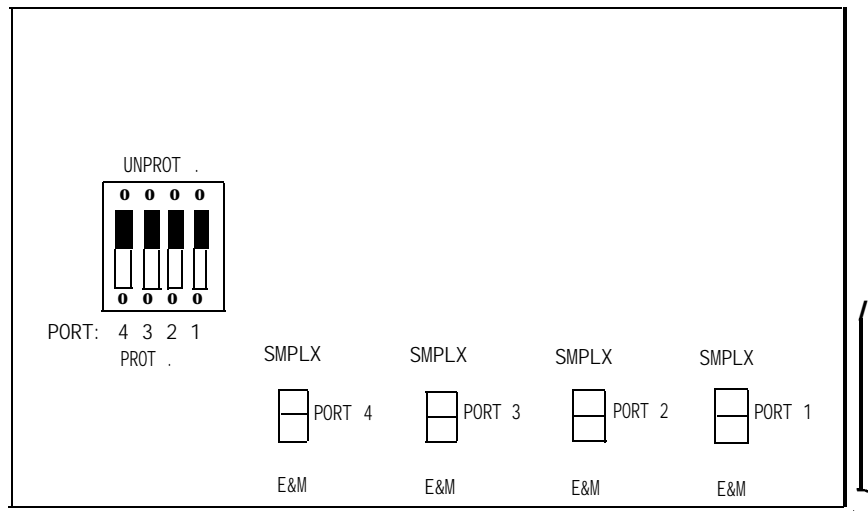


Figure 3-17. Tie Trunk (TN760B) Circuit Pack Option Switches

TABLE 3-B. Signaling Type Summary

SIGNALING TYPE	TRANSMIT		RECEIVE	
	ON-HOOK	OFF-HOOK	ON-HOOK	OFF-HOOK
I Std.	grd	bat	open/bat (*)	grd
I Compat.	open/bat (*)	grd	grd	open/bat (*)
V	open	grd	open	grd

* An open circuit is preferred over voltage.

TABLE 3-C. TN760B Tie Trunk Preferred Signaling Formats

INSTALLATION SITUATION			PREFERRED SIGNALING FORMAT					
FROM	CIRCUMSTANCES	TO	SYSTEM 25			FAR END		
			SIMPLEX OR E&M	SIGNAL TYPE	PROTECTED OR UNPROTECTED	SIMPLEX OR E&M	SIGNAL TYPE	PROTECTED OR UNPROTECTED
S25	Co-Located	S25/S75	Simplex	Type V	(Either)	Simplex	Type V	(Either)
S25	Inter-Building	S25/S75	Simplex	Type V	(Either)	Simplex	Type V	(Either)
S25	Co-Located	S85	Simplex	Type V	(Either)	Simplex	Type V	(Either)
S25	Inter-Building	S85	Simplex	Type V	(Either)	Simplex	Type V	(Either)
S25	Co-Building	Dim.	E&M	Type I Compatible	Unprotected	E&M	Type I Standard	Unprotected
S25	Inter-Building	Dim.	E&M	Type I Compatible	Protected	E&M	Type I Standard	Protected
S25	Co-Located	Other	E&M	Type I Compatible	Unprotected	E&M	Type I Standard	Unprotected
S25	Inter-Building	Other	E&M	Type I Compatible	Protected	E&M	Type I Standard	*
S25	Co-Located	Network Interface	E&M	Type I Standard	Unprotected	(Don't Care)	(Don't Care)	(Don't Care)

* Requires a protection unit.

Auxiliary Trunk (TN763)

The Auxiliary Trunk Circuit Pack interfaces four ports provided for client-provided equipment (CPE) and the TDM bus. It is connected to the CPE by up to three pairs of wires. The transmission pair (T and R) carry voice signals and touch-tone control signals. The T and R also provide a loop start seizure indication to the CPE. The seizure pair (SZ and SZ1) provide seizure indication to the CPE. The signal pair (S and S1) provide answer supervision and/or make-busy information from the CPE. Depending on the application, either the transmission pair only or all three pairs are connected to the CPE.

Figure 3-18 shows the following Auxiliary Trunk unique circuitry:

- Ground detector circuit
- Port Input/Output (I/O) circuit
- Four port circuits.

Ground Detector Circuit: This circuit determines if an answer-supervision or make-busy signal from the CPE is present. The inputs of the ground detector come from the port circuits as an analog current to the -48 volt de supply. Its output is a port control point to the port I/O circuit.

Port I/O Circuit: This circuit consists of bus expanders for communication between the on-board microprocessor and the port circuits. It receives commands from the on-board microprocessor and distributes them to the individual port circuits. It also accesses the port circuit scan points and passes the information to the on-board microprocessor.

Port Circuits: The four port circuits are identical. Each port circuit consists of a codec, hybrid circuit, line transformer, relay driver, battery polarity sensor, and surge protection circuit.

The codec is a 4-wire circuit that converts the analog signal from the CPE to a PCM data signal. It converts an incoming PCM data signal from the NPE to an analog signal. The hybrid circuit converts the 4-wire analog signal from the codec to a 2-wire analog signal that is connected to the CPE by a line transformer.

The relay driver buffers and inverts the relay drive signals from the port I/O circuit so that a logic high input operates the appropriate relay. The relays control circuitry that provide the proper interfaces for CPE.

The surge protection circuit provides lightning surge protection for the circuit pack.

The circuit pack supports both touch-tone and dial pulse signaling. Longitudinal surges are isolated from the hybrid and codec by the line transformer.

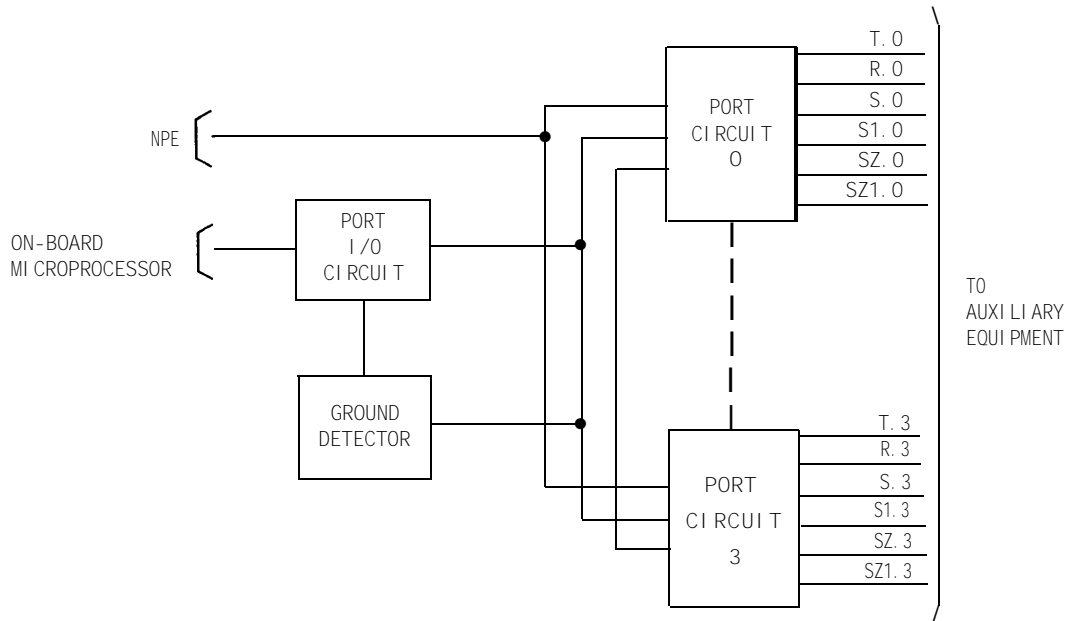


Figure 3-18. Unique Auxiliary Trunk (TN763) Circuitry

STARLAN Interface (ZTN84) (V2)

The STARLAN Interface (ZTN84) is a System 25 circuit pack (CP) designed to function as either a gateway or a bridge between the PBX and the AT&T STARLAN NETWORK. The ZTN84 contains much of the circuitry common to the other CPS in the system, that is a Sanity and Control Interface (SAKI), a Network Processing Element (NPE), and a 8031 microprocessor. The CP also contains the circuitry required to perform the protocol conversion on the data as it travels from one system to the other. These devices include a 80186 microprocessor, 82586 coprocessor, four Octal Asynchronous Terminal Mode 2 to EIA Asynchronous LSI (OATMEAL) devices, and a logic sequencer. The 80186, the 82586, and the logic sequencer (PLS105N) work together to add and delete the protocol used by the Local Area Network (LAN), while the 80186 and the OATMEALS work together to add and delete the protocol used by the PBX.

The ZTN84 can support up to four circuit switch connections between the Private Branch Exchange (PBX) and the Local Area Network (LAN); this capability is provided by the four OATMEALS and the NPE, the latter being a four channel device. In providing a connection between the PBX and the LAN, capabilities such as file sharing, printer services, connections to hosts, and modem pooling may be accessible across systems.

The OATMEAL devices on the ZTN84 are used in such a way as to support asynchronous data communication at any of the standard rates ranging from 300 bps to 19.2 Kbps. The asynchronous protocol that is used is a subset of Digital Communications Protocol (DCP) Mode 2, as only "I" channel information is transmitted, where the data is formatted in High-Level Data Link Control (HDLC) frames.

The ZTN84 has been designed with a hardware interface that allows the CP to be connected to a STARLAN NETWORK as an OUT connection. This can be connected to a STARLAN NETWORK Extension Unit (NEU) IN connection, in a star configuration.

The design of the ZTN84 is not fully compatible with the daisy-chain arrangement of the STARLAN NETWORK, since much of the daisy-chain circuitry was left off of the card. For testing purposes, the card can be used in a limited daisy-chain arrangement, where the ZTN84 is connected to a personal computer (PC) that possesses an Network Access Unit (NAU). The ZTN84 and the PC should be the only two devices forming the LAN. The daisy-chain circuitry was omitted in order to reduce cost and save board space. It is also the architectural design of the system that the PBX be connected to the LAN by a NEU. The NEU can either be local, in the telephone room with the switch, or in a remote office.

System Resources

The System Resource Circuit Packs (CP) are as follows:

- Service Circuit (ZTN85)
- Tone Detector (TN748)
- Pooled Modem (TN758).

Service Circuit (ZTN85)

The Service Circuit CP provides the clock signals of the system. It also generates and receives tones. The Service Circuit CP (Figure 3-19) consists of the following:

- Bus buffers
- Sanity and Control Interface (SAKI)
- On-board microprocessor with external RAM
- Clock circuit
- Tone generator
- Time slot table and counter
- Tone detector ports
- Port I/O and Sanity Check circuit.

The ZTN85 provides four touch-tone receivers, generates all tones for the system, and supplies the system clocks. The ZTN85 can support up to 75 Dual Tone Multifrequency (DTMF) dialers depending on call traffic; the TN748s might be required in heavy traffic situations, even with less than 75 DTMF dialers. Each System 25 must contain one Service Circuit CP. Power for the circuit pack (+5 volts dc) is provided on the backplane.

Bus Buffers: There are four bus buffers on the circuit pack. The clock driver and receive buffers interface three system clock signals (2.048 MHz, 8 kHz, and 160 kHz) to the TDM bus. Two buffers interface the system tones (see Table 3-A) between the TDM bus and the Service Circuit CP. Music is not provided by the Service Circuit but can be provided via a port interface on a Tip Ring Line CP (ZTN78).

SAKI: This circuit functions the same as in the SAKI in the common circuitry for the intelligent port circuits.

On-Board Microprocessor With External RAM: This circuit functions the same as the microprocessor in the common circuitry for the intelligent port circuits. In addition, it tells the dual-port RAM in the time slot table circuit the appropriate time slots in which to place a tone. The external RAM also has work space for complex tones (i.e., those tones that vary with time).

Clock Circuit: The clock circuit consists of a 20.48-MHz oscillator, various dividers, and shift registers. The clock circuit runs independently from the rest of the Service Circuit circuitry. The clock circuits start running when the circuit pack is first powered up and is not controlled by the on-board microprocessor.

The output of the 20.48-MHz oscillator is fed to the clock divider. The divider divides by 10, 2560, and 128. These circuits produce the 2.048-MHz, 8-kHz, and 160-kHz clock signals, respectively. The clock generator feeds these signals to the clock driver/receiver bus buffer and the tone clock. The tone clock uses these signals to synchronize the counters in the tone

generator and time slot table circuits with the TDM bus.

Tone Generator: The tone generator consists of a digital signal processor (DSP), a counter, and a dual-port tone RAM. The DSP operates at 10 MHz and produces 24 different tones. The dual-port tone RAM stores these tones in 24 different addresses. The counter under control of the tone clock causes the DSP to transmit one sample of each tone every 8-kHz. The counter is synchronized to the TDM bus and is offset to provide delay needed for access time.

Time Slot Table and Counter: The time slot table consists of a dual-port time slot table RAM and a counter. The dual-port RAM (DPRAM) contains 256 different addresses. These addresses correspond to the time slots on the TDM bus. The counter sequences through the time slot table addresses in the dual-port RAM and causes the proper tone(s) to be output by the dual-port tone RAM on TDM bus time slots.

Tone Detector Ports: The Service Circuit CP provides four Dual Tone Multifrequency (DTMF) detector port circuit interfaces via the TDM bus. Each port circuit is connected to an NPE serial input and output. Ports 0, 1, 2, and 3 are DTMF tone detectors with NPE loop-around paths.

The four port circuits contain a DSP, NPE to DSP interface circuitry, a DSP restart circuit, and an interrupt generator. One DSP implements two tone receivers.

The TDM bus signals are connected to the DSP in serial form from the NPEs by the DSP interface circuit. The DSP controls the output clocking of the NPE. The system framing signal is synchronized and connects to the DSP.

Port I/O and Sanity Check Circuit: This circuit interfaces the on-board microprocessor to the port circuits and checks the sanity status of the DSPs of the port circuit.

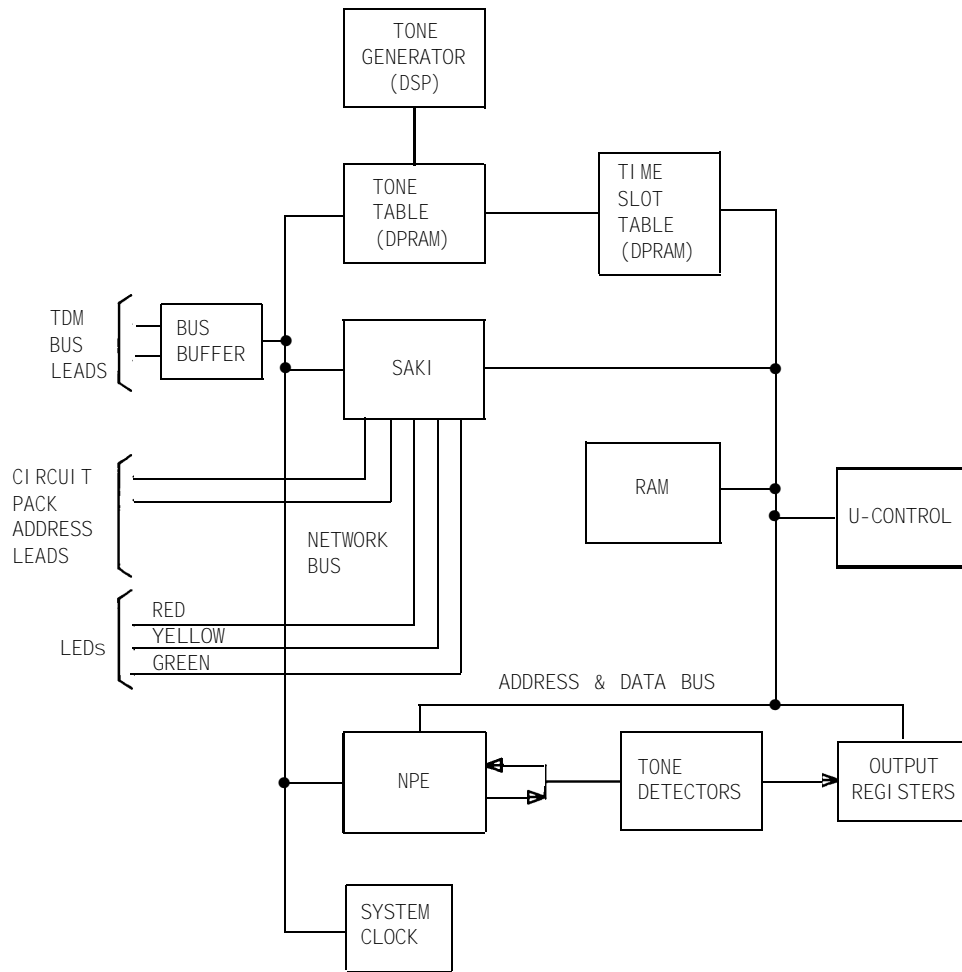


Figure 3-19. Service Circuit (ZTN85)

Tone Detector (TN748)

The Tone Detector Circuit Pack provides four touch-tone receivers and two general purpose tone receivers that detect appropriate system and network tones on the TDM bus.

The Tone Detector CP consists of the same common circuitry as the intelligent port circuits and the following unique circuits (see Figure 3-20):

- Port I/O circuit
- Port or DSP Sanity check circuit
- Four touch-tone port circuits
- Two general purpose tone detector ports
- Two NPE loop-around test ports.

Up to a maximum of two Tone Detector CPs can be provided in the system.

Port I/O and Sanity Check Circuit: This circuit interfaces the on-board microprocessor to the port circuits and checks the sanity status of the port circuits Digital Signal Processors (DSPs).

Port Circuits: There are eight port circuits. Six port circuits are connected to Network Processing Elements (NPEs). Port circuits 0,1, 4, and 5 are DTMF tone detector ports. Each of the six port circuits has an associated Digital Signal Processor (DSP), NPE to DSP interface circuitry, a DSP restart circuit and an interrupt filter. Port circuits 2 and 6 are general purpose tone detector ports. Port circuits 3 and 7 provide digital loop-back testing of each NPE on the circuit pack.

The NPE serializes TDM bus signals that are connected to the DSP in serial form from the NPEs by the DSP interface circuit. Serial clock and data signals connect directly from the NPE to the DSP. The system framing signal is synchronized and connects to the DSP.

The DSP restart circuit controls the DSPs. When the on-board microprocessor is not functioning properly, the DSP restart circuit takes all of the DSPs out of service. It restarts each individual DSP under control of the port I/O and sanity check circuit.

The touch-tone DSPs, under control of the on-board microprocessor, write data synchronously to the NPEs. The interrupt filter detects valid touch-tone signals and allows end-to-end transmission while blocking end-to-end touch-tone signaling.

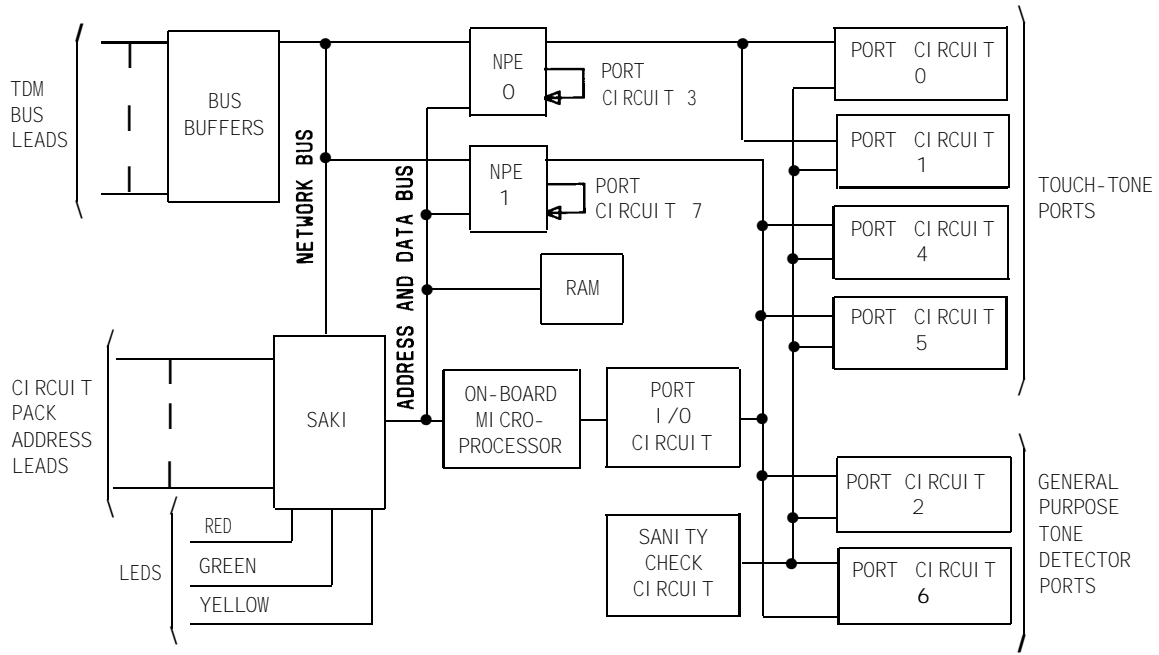


Figure 3-20. Tone Detector (TN748) Circuit

Pooled Modem (TN758)

The Pooled Modem Circuit Pack supports 0-300 and 1200 bits per second (bps) data speeds and provides the following:

- Circuitry to provide a signal compatible with the modulation formats of the 212-series modems
- Modem emulation (see below)

Capability	Data Module Mode
0-300 Asynchronous	Low
300 Asynchronous	300 Asynchronous
1200 Asynchronous	1200 Asynchronous

- Modem control functions corresponding to 212A-series modem operations.

A maximum of two Pooled Modem CPs are allowed in a single cabinet (six in a 3-cabinet system).

The Pooled Modem CP (Figure 3-21) consists of common circuitry and two conversion resources. The conversion resource (port) allows communications between two dissimilar endpoints. For example, the Pooled Modem CP enables a digital data endpoint linked to an ADU connected to a port on the Data Line CP (TN726) to communicate with either a local analog data endpoint, such as a personal computer with a modem, or a remote host via a CO trunk connection. Each port has two connections to the TDM bus: one to the digital data endpoint via an ADU data module, and the other to an analog endpoint.

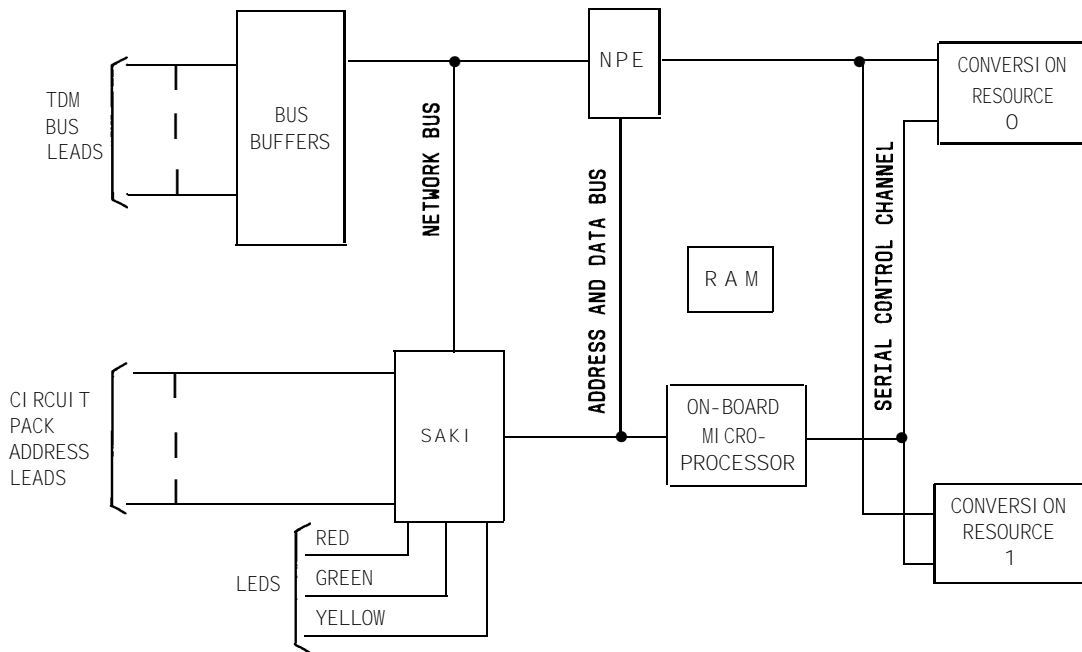


Figure 3-21. Pooled Modem (TN758) Circuit

Common Circuitry: The Pooled Modem common circuitry that includes all circuitry shown on Figure 3-21 except the Conversion Resource circuitry provides the same general function as the intelligent port common circuitry.

Conversion Resources: The two conversion resources (port circuits) are identical and each contain the following:

- Microprocessor
- Transmit and Receive I-channel Controller (TRIC)
- Universal Synchronous/Asynchronous Receiver and Transmitter (USART)
- Data USART Clock (DUCK)
- Digital Signal Processor (DSP).

The microprocessor controls an on-board data module and modem. This microprocessor communicates with the port circuit microprocessor over a serial control channel. This channel allows the on-board microprocessor to send messages to the port circuit microprocessor specifying call startup information, option settings, information requests, various test modes, and call termination information. It also allows the port circuit microprocessor to inform the on-board microprocessor of various port circuit status information.

The DUCK and TRIC interface I-channel information between the port circuit and the remote data module. The microprocessor controls the operation of the DUCK and the TRIC by programming their internal registers. The DUCK and TRIC together recreate the clock and serial data stream from the remote data module, and process an on-board clock and serial data stream for delivery to the remote data module. Control information, handshaking, and RS-232 control leads is passed between the port circuit microprocessor and the remote data module by the TRIC.

The USART interfaces the serial data stream of the DUCK to the conversion microprocessor. The USART can be programmed by the microprocessor to operate synchronously or asynchronously. The USART also does the following tasks for the port circuit microprocessor:

- Appends start and stop bits to parallel data received from the microprocessor in the asynchronous mode
- Converts serial data received from the DUCK to parallel data
- Buffers data in both directions
- Detects and generates break characters.

The DSP provides modem emulation. It interfaces the port circuit signal and the remote modem. The microprocessor directs the DSP to execute one of many programs. The DSP produces data, carrier detection, and timing information for the port circuit microprocessor.

Software

The System software consists of switched services, administrative, and maintenance software. This software runs on top of the real-time operating system software.

Switched Services Software

The switched services software provides voice and data call processing. This software resides in the Call Processor and Memory Circuit Packs (that are collectively referred to as the Common Control circuitry), and in the 8-bit on-board microprocessors located in the port and service circuits.

The switched services software uses the operating system to provide a process based, message passing, execution environment. The operating system scheduler provides scheduling for the software according to process priority.

Administrative Software

The administrative software provides the control for system rearrangement and change via the System Administration Terminal (SAT). This software resides in the Memory Circuit Pack and does the following functions:

- Organizes the translation data for administrable entities in the system in a form that can be viewed and changed at the System Administration Terminal.
- Tests entered data for consistency with data previously entered in order to avoid such errors as the assignment of the same extension number to two voice terminals. An erroneous or inconsistent data entry is disallowed and an error message is provided.
- Causes the translation data to be downloaded, on command, to an optional Digital Tape Unit (DTU).

Maintenance Software

The maintenance software provides automatic periodic testing of maintenance objects within the system as well as consistency tests among the call status tables within the system. In addition, demand testing is initiated when the system detects a condition requiring a need for testing. Software tables are provided for storing error records. The records can be accessed by maintenance personnel via the SAT. A Permanent System Alarm (a serious error) causes an alarm indicator on the attendant console to light and an error record to be stored in the error table.

Memory Allocation

The system software, like the hardware, is identified by release and version number. Each version identifies a particular memory configuration for the release number. Main memory is located in the Common Control circuitry. The operating system and error log software resides on the Call Processor circuit pack and the remaining administration and call processing software is on the Memory circuit pack.

Real-Time Constraints

Real-time constraints are a function of the speed of the common control circuitry and the traffic load. The switch is designed so that many time-consuming and repetitious functions are performed by processors in the port and service circuit packs, thus relieving the common control circuits.

Traffic load, defined as the sum of static and dynamic loads, is a function of the number of features that are executed, the frequency with which they are executed, the system configuration, and the instantaneous (peak) call processing load. The configuration contribution to load is known as dynamic load. The static load consists of maintenance and audit routines.

Software Partitioning

System 25 software is comprised of various modules, each supporting a particular process. Typical modules (referred to as tasks) include the following:

- Administration
- Station Call Processing
- Station Message Detail Recording (SMDR) Call Record Processing
- Trunk Call Processing
- Dial Plan Manager
- Event Timer
- Save/Restore (Administration function)
- Maintenance and Audit Functions.

As shown on Figure 3-22, software tasks associated with the Memory Circuit Pack are Administration and Feature Code Modules, which includes Station Call Processing. Each task controls the storage and movement of data and messages between associated elements within the system.

Memory Circuit Pack

Administration: Provides for administration of station and system features. This software also supports maintenance procedures related to error checking and diagnosing trouble.

Feature Code Modules: Includes the software that receives and sends data to/from the Operating System, as well as control of all voice and data features supported by the system. Station Call Processing includes the processing of messages and data associated with voice terminal on-hook/off-hook indications, associated port identifications, button and LED operations, etc. The SMDR software generates SMDR records associated with a particular call. The records are then sent to the System RAM for storage and then to the SMDR output channel.

Call Processor Circuit Pack

System RAM: Provides for the storage of the following:

- Variables for the various software tasks
- System translations
- Error Records
- Feature Code Data
- Stack.

Error Logger: Prioritizes and stores system errors. The errors are stored in three error records (located in System RAM), that are:

- Permanent System Alarms
- Transient System Errors
- Most Recent System Errors.

The Error Logger lights the Alarm LED (located on the Attendant Console) when a serious error is detected.

Operating System (OS): Controls all message and data flow to/from the Memory Circuit Pack, the Arch Angel Driver Interface, to the microprocessors on the port circuit packs, and to the RS-232 driver interfaces. Messages destined for a particular task are queued until the associated task can receive them. When a task has completed a particular process, the next message is obtained from the message queue of the tasks. The OS provides an interval timer that is used to time tasks. Processes that exceed the set interval (about 60 seconds) are terminated by the OS.

Arch Angel Driver Interface: Provides an interface between the OS and Network Control.

RS-232 Driver Interface: Handles the flow of information between the Call Processor Circuit Pack and the peripheral equipment of the system (i.e., System Administration Terminator Advanced Administration PC, Digital Tape Unit, and SMDR Output Device.)

TDM Bus

Provides an electronic link between the system port circuits (including System Resources) and between the Call Processor Circuit Pack and port circuits.

Port Circuit Packs

Each port circuit pack has on-board software that provides for the sending/receiving of Network Control messages and data. Circuit pack status messages are also sent to the Network Control software.

Step-By-Step Call Description

The following is a description of a call originated between two multiline voice terminals.

1. A microprocessor on a station port circuit pack (port controller) continually monitors associated port circuits for switchhook status/change and button presses.
2. When a user goes off-hook, the port controller detects the change.
3. The port controller sends an off-hook up-link message along with port identification to the Call Processor Network Controller (CPNC) via the TDM bus.
4. The CPNC accepts the message and forwards it to the Operating System (OS) via the Arch Angel Driver Interface.
5. The OS checks a message directory to determine which task (i.e., software module) is to receive the message. A function of the OS referred to as the "transformer", determines it has a message for the Station Call Processing task and queues the message in RAM.

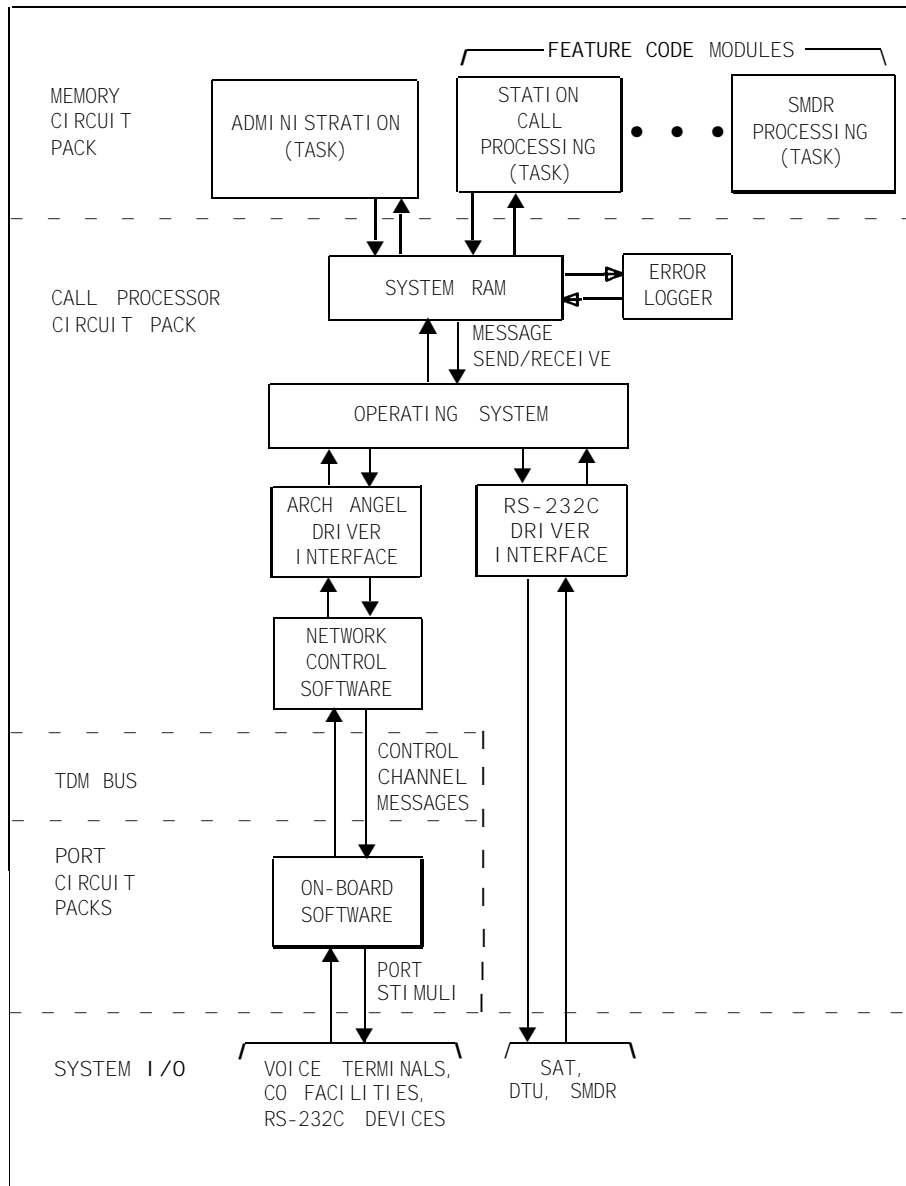


Figure 3-22. System Software Partitioning

6. The Station Call Processing task retrieves its message and interprets it as a call origination. The task determines whether there is an idle call appearance button (System Access button) on the called voice terminal. If so, two available time slots are reserved for the connection.
7. The task sends downlink messages to the port circuit via the OS. The messages instruct the port circuit to listen for dial tone on a specified time slot and to light the call appearance status LED on the terminal.
8. When the user dials the first digit, the port circuit determines the digit dialed. It then listens to appropriate time slots on the TDM bus for the two tones used to generate an equivalent DTMF signal. It then removes dial tone and feeds the DTMF signal back to the user until the user releases the button.
9. The port circuit sends an up-link message with each digit dialed to the OS that routes them to the Dial Plan Manager (DPM).
10. The DPM collects the dialed digits and determines that the call is a station-to-station call.
11. When the DPM collects enough digits to identify an extension number it stops collecting digits.

Note: If the extension number dialed is invalid, the DPM sends a down-link message to the port circuit instructing it to listen to time slot 07 (Reorder Tone) that is then heard by the user. Go to Step 18.

12. A down-link message is sent to the originating port instructing it to listen to time slot 06 (busy) or 08 (ringing), as appropriate. Go to Step 18 for Busy Tone or an unanswered call.
13. Station Call Processing sends a down-link message to the station port circuit pack associated with the called extension to turn on the ringer of the terminal, and to flash the call appearance LED.
14. When the called party lifts the receiver, the associated port circuit pack controller sends a off-hook message to the OS as before.
15. The Station Call Processing task, when it receives the message interprets the off-hook message as an answer.
16. The task sends a down-link message to the called port circuit to turn off the ringer and to change the flashing LED to steadily lighted.
17. Down-link messages are sent to the port circuits assigning talk and listen time slots for the connection.
18. When either of the parties hangs up, the associated port circuit controller sends an up-link message to the Station Call Processing task.
19. Station Call Processing interprets the on-hook message as the end of the call
20. The task then sends a down-link message to the port circuit pack controllers to disconnect the time slot connections and turn off the LEDs associated with the calls.

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SECTION 4—HARDWARE DESCRIPTION

This Section provides descriptions of System 25 hardware components and their functions. The hardware is covered under the following major headings:

- **System Cabinets:** Includes Circuit Pack (CP) carriers and CPs.
- **Terminal Equipment:** Equipment that can be connected to voice or data station ports.
- **Peripheral Equipment:** Equipment that can be connected to the Call Processor CP.
- **Auxiliary Equipment:** Service- and feature-related supporting equipment.
- **Connectivity:** Equipment and arrangements for interconnecting the various elements of System 25 hardware.

Note: Equipment that indirectly associated with a specific feature or service covered in Section 2 is also described there; such equipment is noted in this Hardware Description section, with a reference to the appropriate heading in Section 2.

All system hardware, except Cabinet 1 equipped with Call Processor, Memory, and Service Circuit CPs, and associated cables is optional.

A listing of Product Element Codes (PECs), Apparatus Codes, and Comcodes is provided in Section 5, “Parts Information.”

System Cabinets (J58901A1)

The system can consist of one, two, or three cabinets (Figure 4-1). Each cabinet contains its own power supply and cooling system. A CP carrier frame is integrated into each cabinet. Depending on the circuit pack complement, the cabinet/circuit pack configuration is as follows:

- Cabinet 1 (always required)—Contains the Call Processor, Memory, and Service CPs of the system and can also contain up to nine port CPs.
- Cabinet 2 or 3 (optional)—Provides mounting for up to 12 port CPs each.

The CPs receive power, control signals, and data via the backplane bus of the carrier and associated 25-pair connector interfaces. In multiple cabinet systems, the backplane buses are linked with a bus extender cable (J58901A1, L3).

The Call Processor and Memory CPs are also connected via a ribbon connector cable referred to as the “Front Plane Bus.” This cable connects the front edges of the two CPs.

The cabinets have a brown front cover with beige top and sides. The front cover has a system identification stripe across the top.

Each cabinet is constructed of sheet metal and is 13 inches high, 17 inches wide, and 21 inches deep and weighs about 75 pounds fully loaded. A three cabinet system occupies a vertical space of about 40 inches. It is recommended that the cabinets be placed on a desk- or table-top. They *must not* be placed on a floor where cleaning solutions and dirt can get into them. Refer to Section 6, “Environmental Requirements” for equipment area considerations and associated floor plan recommendations.

The front cover of the cabinet is secured by four screws. These screws must be loosened slightly before the cover can be removed. When removed, access is provided to the CPs, a replaceable air filter mounted just under the CP carrier frame, and two cooling fans. The cooling fans are mounted on an assembly that, when unscrewed, provides access to the power supply. Air intake is at the bottom of the cabinet and exhaust is vented at the left side of the front cover.

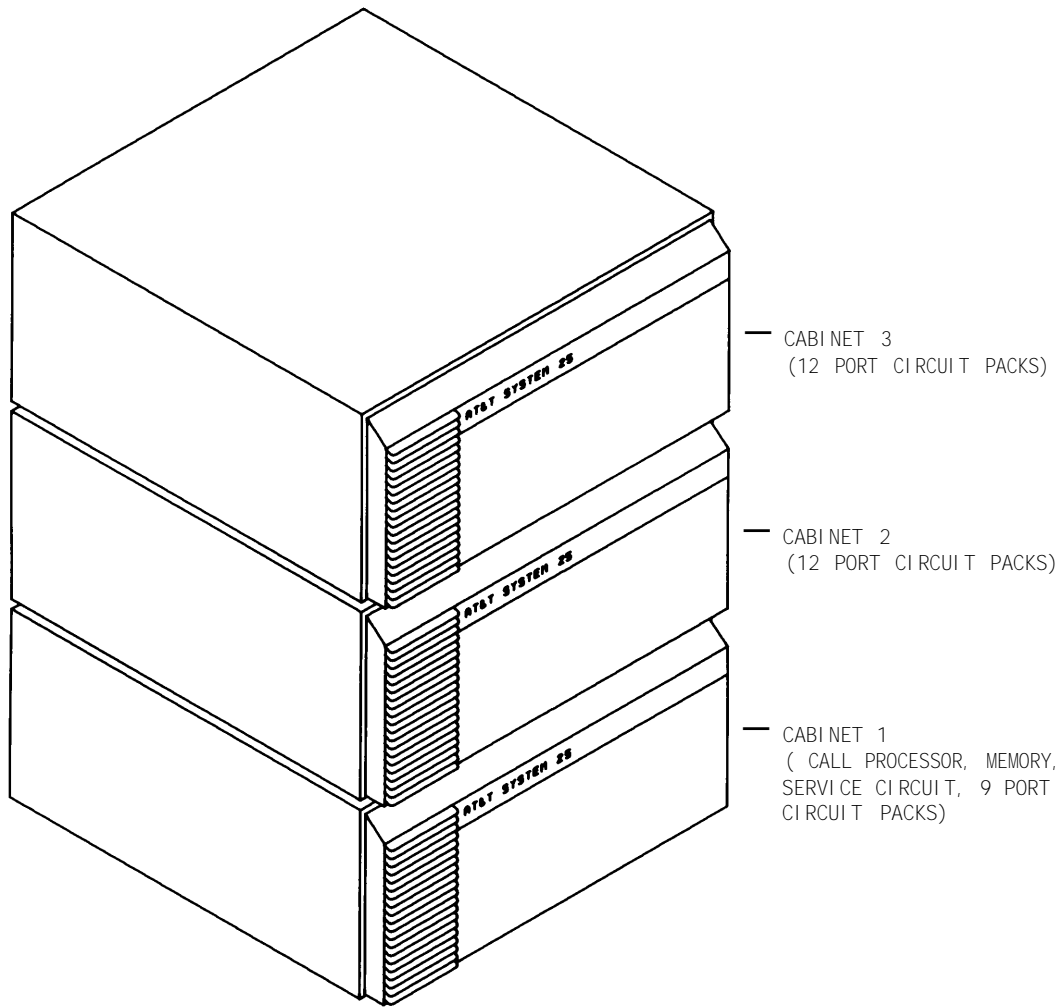


Figure 4-1. System 25 Cabinets (J58901A)—Three Cabinet System

Each cabinet has its own power supply mounted to the left of the CP carrier. The power supply is 3 inches wide and weighs about 9 pounds. Voltage and current supplied to the carrier are: +5 V dc at 35A, -5 V dc at 3A, -48 V dc at 3A, and 90 V ac at 0.16A.

On the front of the supply is a green Light Emitting Diode (LED) that, when lighted, indicates that the +5 V dc is available and within limits. The LED can be viewed through the slotted area on the front cover, and is just behind the fan located at the top left edge of the cabinet.

Mounted on the back of the cabinet (Figure 4-2) is the copper grounding block with four terminating positions, an AC input power receptacle, a power On/Off switch [(1)=ON, (0)=OFF] and twelve 25-pair connectors. The ground block is connected to DC ground on the carrier backplane at a location near the power supply. The 25-pair connectors provide an interface between cross-connect wiring and the CPs immediately behind each connector. Two slots are provided in the rear cover just above the 25-pair connectors for the Time Division Multiplex (TDM) bus extender cable. The TDM cable is used to connect 2- or 3-cabinet systems together in a daisy-chain configuration and provides control and data signals between Cabinets. The Cabinet 1 ground block is connected to the single point ground of the system using 6 AWG wire. Separate 6 AWG wires are then connected from the Cabinet 1 ground block to Cabinet 2 and 3 ground blocks. The Cabinet 1 ground block is also connected to the Coupled Bonding Conductor. An information label is provided across the top portion of the rear panel on each cabinet. The label provides cabinet identification, input electrical requirements, caution and warning notes, and FCC and UL labels.

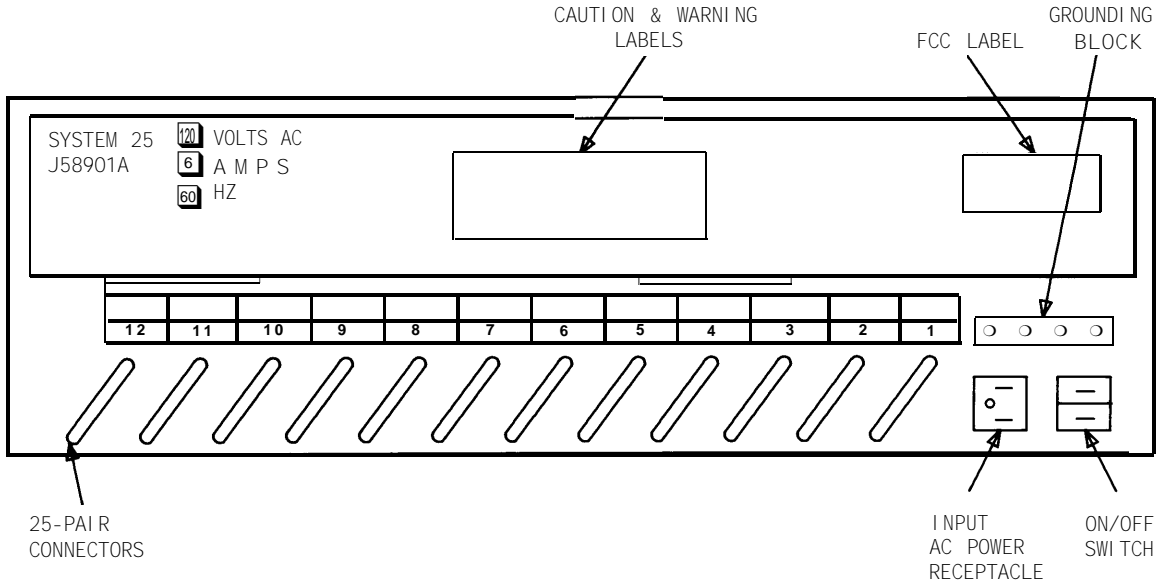


Figure 4-2. System Cabinet (J58901A)—Rear View

Cabinet 1 (Control and Port Circuits)

Cabinet 1 (Figure 4-3) is always required. It provides mounting space for 12 CPs and can support a small telecommunications system (eg., 50 to 60 stations and 10 to 15 trunks). It contains a Memory and Call Processor that together are referred to as the Common Control (CC), a Service Circuit, and up to 9 port CPs. The Memory, Call Processor, and Service Circuit must be mounted in CP slots 1, 2, and 3, respectively. Slots 4 through 12 (9 total) provide mounting for the various port CPs that can be used. Any port CP can be mounted in any of these 9 slots. The Memory and Call Processor are electrically linked by a ribbon cable (Front Plane BUS) that loops between their front edges. The Tone Detector and Pooled Modem CPs of the system (referred to as System Resource CI's) can also be mounted in the port CP slots. Circuit packs are described in this Section under the heading "Circuit Packs".

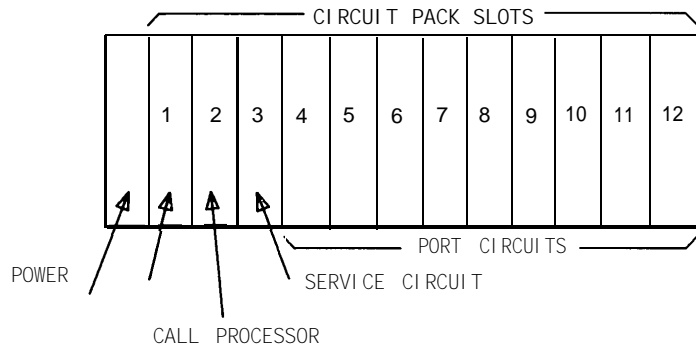
Cabinet Address Plug

An address plug is provided on the middle of the backplane of each cabinet (accessible after removing the top rear cover) and is used to designate the cabinet number to the software. When plugged into the designated area at CP slot 5, the cabinet is identified as Cabinet 1; at slot 6 as Cabinet 2, and at slot 7 as Cabinet 3.

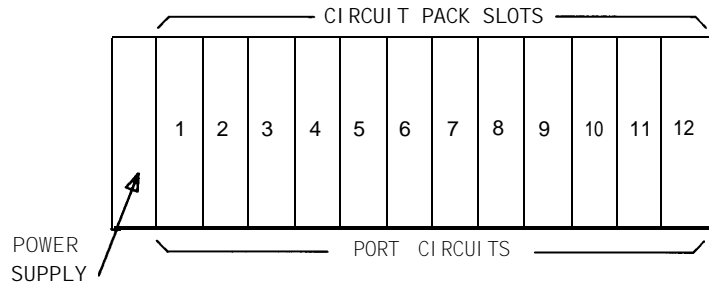
Cabinets 2 and 3 (Port Circuits)

Cabinet 2 and Cabinet 3 (Figure 4-3) can be provided. The cabinets provide mounting space for additional port CPs (12 maximum each) required for larger systems. The Tone Detector and Pooled Modem CPs can also be mounted in these cabinets. These cabinets are simply stacked on top of Cabinet 1.

Table 4-A summarizes port CP capacity of 1-, 2-, or 3-cabinet systems.



(a) CABINET 1 - (MOUNTING FOR 3 CONTROL & 9 PORT CIRCUIT PACKS)



(b) CABINET 2 OR 3 - (MOUNTING FOR 12 PORT CIRCUIT PACKS)

NOTES :

1. REFER TO TECHNICAL SPECIFICATIONS, SECTION 8 FOR CIRCUIT PACK UNIT LOAD INFORMATION.
2. DIVIDE THE TOTAL NUMBER OF VOICE TERMINAL AND TRUNK CIRCUIT PACKS BETWEEN THE CABINETS USED.
3. MOUNT VOICE TERMINAL CIRCUIT PACKS FROM THE RIGHT, TRUNK CIRCUIT PACKS FROM THE LEFT.
4. COMMON CONTROL CIRCUIT PACKS *
 - MEMORY (ZTN81 IN R1V1)
(ZTN127 IN R1V2)
 - CALL PROCESSOR (ZTN82 IN R1V1)
(ZTN128 IN R1V2)
5. SYSTEM RESOURCE CIRCUIT PACKS #
 - SERVICE CIRCUIT (ZTN85)
 - POOLED MODEM (TN758)
(MAXIMUM 2-PER CABINET)
 - TONE DETECTOR (TN748)
6. PORT CIRCUIT PACKS †
 - TIP RING LINE (ZTN78)
 - DATA LINE (TN726)
 - ATL LINE (ZTN79)
 - GROUND START TRUNK (ZTN76)
 - MET LINE (TN735)
 - LOOP START TRUNK (ZTN77)
 - ANALOG LINE (TN742)
 - DID TRUNK (TN753)
 - AUXILIARY TRUNK (TN763)
 - TIE TRUNK (TN760B)
 - STARLAN INTERFACE (ZTN84)

* CIRCUIT PACKS MUST BE MOUNTED IN CABINET 1, IN THE SLOTS INDICATED
 † UNIVERSAL PORT CIRCUIT PACKS CAN BE MOUNTED IN ANY AVAILABLE PORT SLOT
 # SERVICE CIRCUIT MUST BE MOUNTED IN SLOT 3 AS SHOWN. POOLED MODEM AND TONE DETECTOR MAY BE MOUNTED IN ANY PORT CIRCUIT SLOT.

Figure 4-3. System Circuit Pack Configurations

TABLE 4-A. Total Port Circuit Packs Per System

NUMBER OF CABINETS	CABINET NUMBER *	PORT CIRCUIT PACKS	TOTAL PORT CIRCUIT PACKS †
1	Cabinet 1	9	9
2	Cabinet 1	9	21
	Cabinet 2	12	
3	Cabinet 1	9	33
	Cabinet 2	12	
	Cabinet 3	12	

* Cabinet 1 (always required)—Provides mounting for Memory, Call Processor, Service Circuit, and Port CPs including Tone Detectors and Pooled Modems.

Cabinet 2 and 3 (Optional)—Provides mounting for Port CPs including Tone Detectors and Pooled Modems.

† The Number of Ports per CP is specified in the CP descriptions.

Circuit Packs

Required Circuit Packs:

The following CPs are provided with all systems and must be mounted in Cabinet 1:

- ZTN82 (V1) or ZTN128 (V2) Call Processor

The ZTN82 or ZTN128 (one per system) provides a central processing unit, Random-Access Memory (RAM) (memory) for call and feature processing, interrupt controller, programmable timers, real time clock, status display, processor bus interface, and four interface ports. The ports provide the following interfaces:

- Port 1—System Administration Terminal (SAT)
- Port 2—Station Message Detail Recording (SMDR) equipment
- Port 3—Digital Tape Unit
- Port 4—Reserved for future use.

The Call Processor also provides -48 V de control on ports 7 and 8 for Emergency Transfer Units.

The Call Processor CP must be mounted in slot 2 of Cabinet 1.

- ZTN81 (V1) or ZTN127 (V2) Memory

The ZTN81 or ZTN127 (one per system) provides 512K of read-only memory. The Memory CP provides for the software associated with system operation, including call processing, administration, and maintenance.

The ZTN81 or ZTN127 provides a built-in TDM bus terminator; an earlier version (ZTN81) did not.

The Memory CP must be mounted in slot 1 of Cabinet 1.

- ZTN85 Service Circuit

The ZTN85 (one per system) provides four touch-tone receivers, generates all system tones, and supplies the system clocks. The ZTN85 can support up to 75 Dual Tone Multifrequency (DTMF) dialers such as 2500-type voice terminals and touch-tone (incoming) tie trunks.

The Service Circuit CP must be mounted in slot 3 of Cabinet 1.

Optional Circuit Packs:

The following CPs are optional and can be mounted in any other CP slot.

System Resource Circuit Packs

- TN748 Tone Detector

Provides four touch-tone receivers. The TN748 is required in addition to the ZTN85 Service Circuit when more than 75 Dual Tone Multifrequency (DTMF) dialers are to be provided in a system. It might be required in high traffic situations if a system has less than 75 DTMF dialers. Up to two TN748 Tone Detectors can be provided in a system.

- TN758 Pooled Modern

Provides two integrated 212-modem compatible conversion resources for switched connections between analog endpoints (modems), or a digital endpoint and an analog endpoint. A maximum of two TN758s (4 conversion resources) is permitted in each cabinet.

Station Port Circuit Packs

- TN726 Data Line

Provides eight ports for Asynchronous Data Units (ADUs). Used for in-building service within 2000 feet of the system cabinets. Data speeds from 300 bps to 19.2 Kbps are supported. Service beyond 2000 feet at less than 19.2 Kbps is supported; see Section 5 “Technical Specifications.” Extends a serial communications link from the system to data equipment over standard station wiring.

- TN735 MET Line

Provides four ports for Multibutton Electronic Telephone (MET) sets. Used for in-building service within 1000 feet of the system cabinets.

- TN742 Analog Line

Provides eight ports for single-line voice terminals with or without a message waiting lamp. Also supports Off Premises Stations (OPS) and out-of-building service. Auxiliary equipment interfaces are also supported. Used for service within 24,000 feet of the system cabinets. Five voice terminals can be bridged onto each port. Only two terminals can be off-hook simultaneously on each port, otherwise transmission can be degraded.

Note: The Off-Premises-Stations must be FCC registered.

- ZTN78 Tip Ring Line

Provides eight ports for single-line sets with or without message waiting lamps. Used for in-building nonbridged voice terminal service within 2000 feet of the system cabinets.

Note: Equipment connected to the ZTN78 Tip Ring Line CP must meet the following requirements:

- AC impedance: 600 ohms
- DC current: Less than 30ma at 48 volts
- Ringer Equivalence Number (REN): Less than 1.15 (Set plus adjuncts)

- ZTN79 ATL Line

Provides eight ports for MERLIN® voice terminals. Used for service within 1000 feet (305 m) of the system cabinets. Off-premises extensions are not supported. Out-Of-Building stations require In-Building and In-Range Out-of-Building (IROB) units.

- ZTN84 STARLAN Interface (V2)

Provides four ports for interface with AT&T STARLAN NETWORKs. It supports four connections between data endpoints connected to the PBX and data endpoints connected to the STARLAN NETWORK. The Network Extension Unit must be collocated with the System 25 cabinets.

Trunk Port Circuit Packs

- TN753 DID Trunks

Provides eight ports for immediate-start or wink-start Direct Inward Dialing (DID) trunks.

- TN760B Tie Trunks

Provides four ports for Type 1 E&M, Type 1 E&M Compatible, or Type 5 Simplex tie trunks. Operating protocols include automatic, immediate-start, wink-start, or delay dial. The TN760B contains option switches for supporting the following signaling formats:

- Type 1 E&M Standard (Unprotected)
- Type 1 E&M Compatible (Unprotected)
- Type 1 E&M Compatible (Protected)
- Type 5 Simplex

- TN763 Auxiliary Trunk

Provides four ports for on-premises auxiliary equipment (paging systems and dictation systems).

- ZTN76 Ground Start Trunk

Provides eight ports for ground start Central Office (CO), Foreign Exchange (FX), or Wide Area Telephone Service (WATS) trunks

- ZTN77 Loop Start Trunk

Provides eight ports for loop-start CO, FX, or WATS trunks.

Refer to Section 9, “Glossary” for Ground Start and Loop Start definitions. Ground Start trunks are recommended for use where possible.

Trunk specifications are provided in Section 5, “Technical Specifications.”

Circuit Pack Compatibility

The following System 75 CPs can be used in System 25, if required:

- The TN742 Analog Line can be used instead of the ZTN78 Tip Ring CP. The TN742 supports bridged stations and out-of-building or Off-Premises Stations (OPS), the ZTN78 does not.
- The TN762B Hybrid Line (Version 4 or later) can be used instead of the ZTN79.
- The TN747 CO Trunk can be used instead of the ZTN76 (Ground Start Trunk) or the ZTN77 (Loop Start Trunk).

Circuit Pack Features

All system CPs have the following features:

- Solid-state circuitry mounted on 7.6 by 14.1-inch printed wiring board (TN-type)
- Color coded face plate labels identify the CP type and function (White = Control, Purple = Port or System Resource)
- Individual circuit functions all contained on one CP
- Metal tab for grounding
- Locking tab-type handle provides easy insertion or removal of a CP
- Port CPs can be inserted or removed with power “On” and the system processing calls. Only the calls utilizing circuits on a removed CP will be affected.

Note: Power must be turned off when replacing the following CPs:

- Memory (ZTN81 or ZTN127)
- Call Processor (ZTN82 or ZTN128)
- Service Circuit (ZTN85)
- Status LEDs
 - Port CPs:
 - Red—“On” several seconds during power up and test, “Off” with test pass. After test pass, “On” if fault in CP is subsequently detected.
 - Green—“On” indicates resource available (port is translated)
 - Yellow—“On” indicates a call in progress. “Off” when not in use.
 - All LEDs “Off”—CP is not translated.
 - Common Control CPs:
 - Memory CP: Red status LED only.
“On” several seconds during power up and test, “Off” with test pass. After test pass, “On” if fault in CP is detected.
 - Call Processor CP: Green Status LED only.
“Off” for several seconds during power up and test, then lamp flashes to indicate an “OK” state. Steady “Off” or “On” indicates a problem.
 - System Resource CPs:
 - Service Circuit CP:
Similar to port CPs except yellow LED flashes to show system clock is active and is steadily “On” when a tone receiver is in use. “Off” indicates a problem.
 - Modem Pool and Tone Detector CPs:
Same as Port CPs.

Table 4-B lists CPs that can be used with System 25. A description of each CP is provided in Section 3, Functional Description.

TABLE 4-B. System Circuit Packs

CIRCUIT PACK	TITLE	CIRCUIT PACK TYPE *	NUMBER OF PORTS
TN726	Data Line	P	8
TN735	MET Line	P	4
TN742	Analog Line	P	8
TN748	Tone Detector †	R	4
TN753	DID Trunk	P	8
TN758	Pooled Modem	R	2
TN760B	Tie Trunk	P	4
TN763	Auxiliary Trunk	P	4
ZTN76	Ground Start Trunk	P	8
ZTN77	Loop Start Trunk	P	8
ZTN78	Tip Ring Line	P	8
ZTN79	ATL Line	P	8
ZTN81 or ZTN127	Memory	C	
ZTN82 or ZTN128	Call Processor ‡	C	
ZTN84	STARLAN Interface	P	4
ZTN85	Service Circuit	R	4

* P = Port, C = Control, R = System Resource.

† System Resource Circuits (Tone Detector, Pooled Modem, Service Circuit) ports are internal to the system. These ports are not connected to external equipment via 25-pair connectors.

‡ Provides four channels for the peripheral equipment of the system:

- (1) System Administration Terminal
- (2) SMDR Output Device
- (3) Digital Tape Unit
- (4) reserved for future use.

Terminal Equipment

Terminal equipment is connected to System 25 station (voice or data) ports. It is made up of the following groups:

- Voice Terminals
 - Single-Line
 - Multiline (MERLIN sets and MET sets)
- Voice Terminal Adjuncts
- Attendant Consoles
- Asynchronous Data Units (for interface with data terminals).

This subsection provides information on all components in each group or contains references to the Section where information can be found.

Voice Terminals

System 25 supports a wide range of voice terminals, including industry standard touch-tone single-line sets and MERLIN multiline sets.

In addition to providing basic telephone service (placing and answering calls), voice terminals can also be used to activate many system features. The voice terminals supported by System 25 are listed in Table 4-C and described in individual subsections.

Single-Line Voice Terminals

Single-line voice terminals can have only one call appearing at the terminal at a time.

All information (voice and control signals) transmitted to and from a single-line voice terminal is in analog form over a single pair of wires (called tip and ring). Power for these voice terminals is also provided over this pair. The ZTN78 Tip Ring Line CP or TN742 Analog Line CP converts the analog signals to digital format before placing them on the TDM bus. The dial pad on the single-line voice terminals is a touch-tone pad that generates Dual Tone Multifrequency (DTMF) signals.

The following subsections provide descriptions and illustrations of the single-line voice terminals supported by System 25.

Model 500 Series (V2)

The Model 500 Series consists of conventional rotary dial telephones. They are recommended for use as a Power Failure Transfer (PFT) stations if the PFT trunk does not support touch-tone dialing. A 55A1 Ground Start button must be used with these sets if the PFT trunk is ground start. Rotary set users cannot do any procedures that require pressing the * or # buttons. The following 500 Series sets are supported by System 25; these sets are similar in appearance to the 2500 sets shown in Figure 4-4 except for their rotary dials.

- Model 500MM—Basic desk set
- Model 500SM—Desk set with 4A Speakerphone compatibility
- Model 554BMPA—Basic wall set.

TABLE 4-C. Summary of Voice Terminals

TERMINAL TYPE *	MODEL	DESCRIPTION	CIRCUIT PACK INTERFACE
Single-Line Tip Ring (Analog) †	420	Memory Set W/Built-In Speakerphone	ZTN78
	500MM	Rotary Desk Set	
	500SM	Rotary Set Compatible with 4A Speakerphone	TN742
	554BMPA	Rotary Wall Set	
	2500MMGB	Basic Touch-Tone Desk Set	
	2500DMGC	Basic Desk with Message Waiting	
	2500SM	Basic Desk Compatible with 4A Speakerphone	
2514 BMW	Basic Desk with Headset Jack		
2554BM	Basic Wall		
7101A	Desk or Wall		
MERLIN Multiline 7300H Series (Hybrid)	7302H01D	5-Button	ZTN79
	7303H01D	10-Button	
	7305H01D	34-Button	
	7305H02D	34-Button Deluxe	
	7305H03B	BIS Set (Built-In Speakerphone)	
	7305H04C	BIS Set with Display	
	7309H01B	HFAI Set (Hands-Free-Answer on Intercom)	
MET ‡ Multiline (Hybrid)	2991C05	10-Button (Desk)	TN735
	2991D05	10-Button (Wall)	
	2993C04	10-Button with BIS	
	7302M	12-Button (Desk)	

* System 25 supports several voice terminals that are no longer orderable. These include MET sets and the 34-button (basic) MERLIN set.

† The system supports equivalent industry standard touch-tone single-line sets. Voice terminals connected via the ZTN78 Tip Ring Line CP must have a REN less than or equal to 1.20 A/B.

‡ The 2991C04 set [with Busy Lamp Field (BLF)] will not operate with System 25 unless specially modified.

Model 2500 Series

Each of the following Model 2500 voice terminals come equipped with:

- Handset
- Touch-Tone Dial (Dual Tone Multifrequency - DTMF)
- Ringer Volume Control.

Several 2500 series voice terminals are shown in Figure 4-4. Supported 2500 series sets include the following:

- Model 2500 DMGC—Desk Set with message waiting indicator and Recall button for timed switchhook flash
- Model 2500MMGB—Basic desk set
- Model 2500 SM—Basic desk set that can be used with a 4A Speakerphone.
- Model 2514 BMW—Basic desk set equipped with built-in headset speaker jack.
- Model 2554 BM—Basic wall-mounted set.

Adjuncts: Refer to Table 4-D.

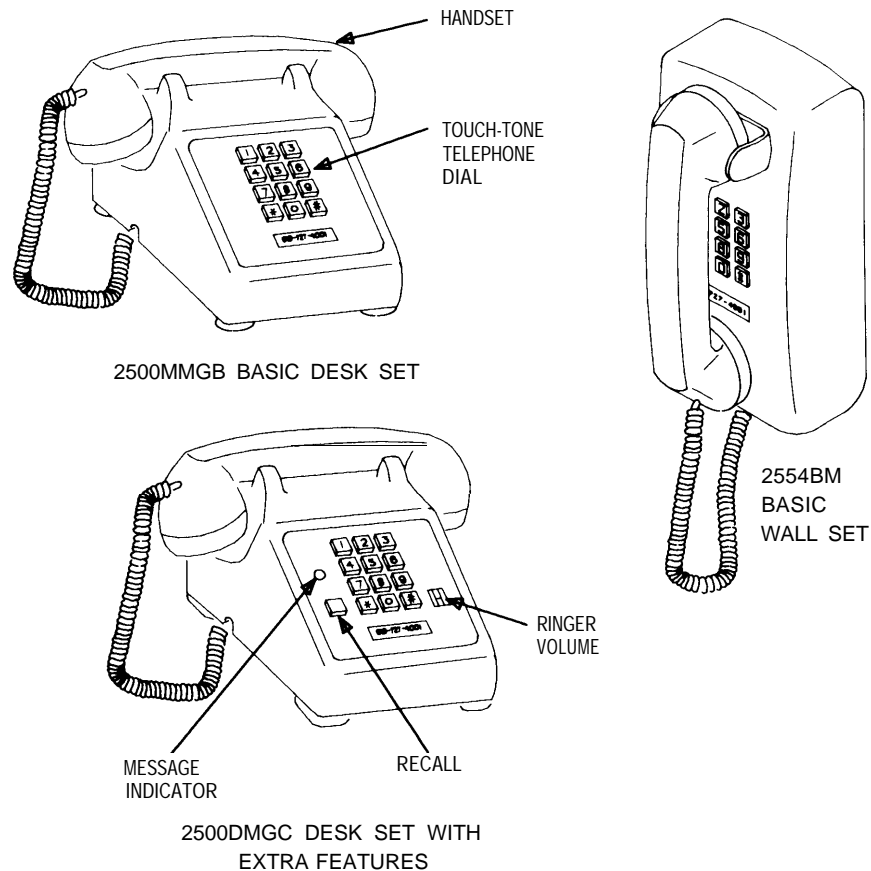


Figure 4-4. Model 2500 Series (Analog) Voice Terminals

Model 7101A (MD) (PEC 3170-00M)

The Model 7101A (Figure 4-5) is about 2-3/4 inches wide, 3-1/2 inches high, and 8-1/2 inches long. The set comes equipped with the following:

- Handset
- Touch-Tone Dial
- Message Indicator
- Tone Ringer with Volume Control
- Two Fixed Feature Buttons
 - Recall—Used to place a call on hold and to obtain recall dial tone for Conference, Transfer, and other features accessible by feature access code.
 - Disconnect—Used to disconnect one call and immediately obtain dial tone for another call.

Adjuncts: None

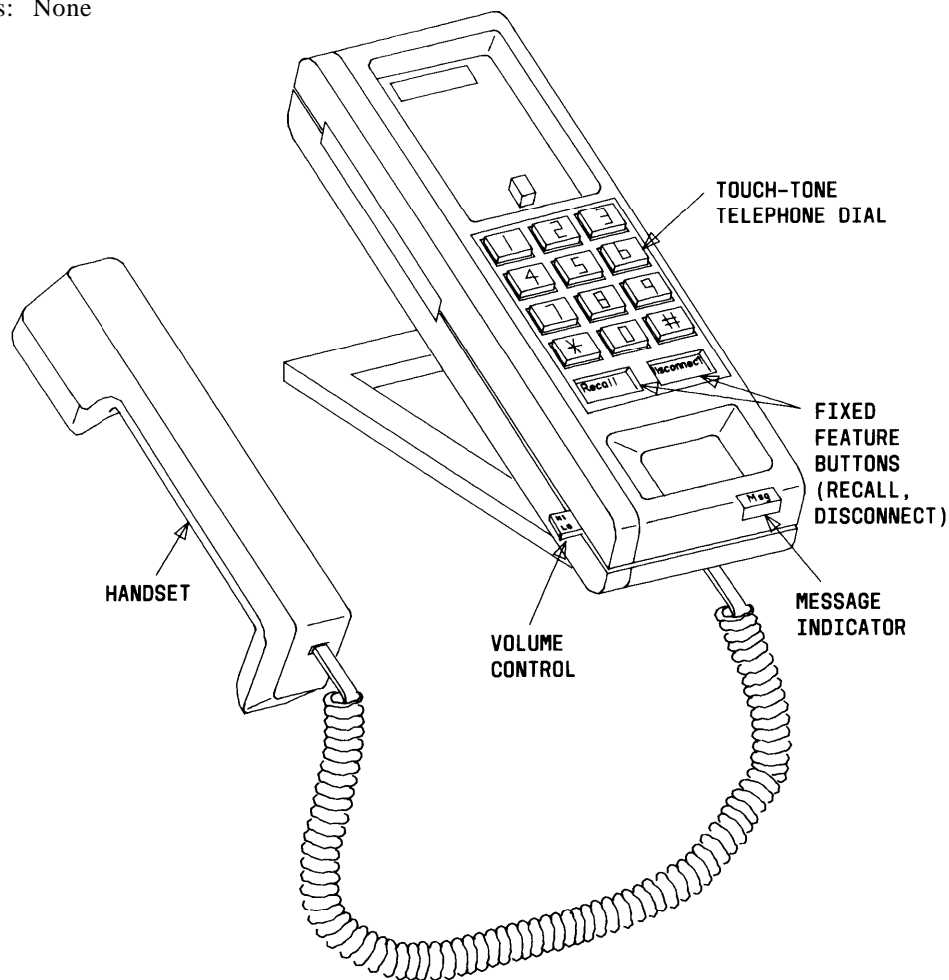


Figure 4-5. Model 7101A (Analog) Voice Terminal

420 Speakerphone (V2)

The 420 Speakerphone (Figure 4-6) is a single-line voice terminal that can be desk or wall mounted. This set comes equipped with the following:

- Handset
- Touch-Tone Dial
- Built-In Speakerphone
- Twelve Memory Buttons—Where emergency numbers and frequently called numbers can be stored for quick calling.
- Six Fixed Feature Buttons
 - Program—For entering the memory button programming mode.
 - Redial—For recalling the last number dialed.
 - Flash—For generating a timed switchhook flash.
 - Mute—For turning off the speakerphone microphone temporarily for privacy
 - Hold (with status lamp)—For putting calls on hold.
 - Speaker (with status lamp)—For making speakerphone calls and for turning on the speaker during handset calls.
- Tone Ringer
- Three Volume Controls
 - Tone ringer
 - Speaker
 - Handset receiver.

Adjuncts: None

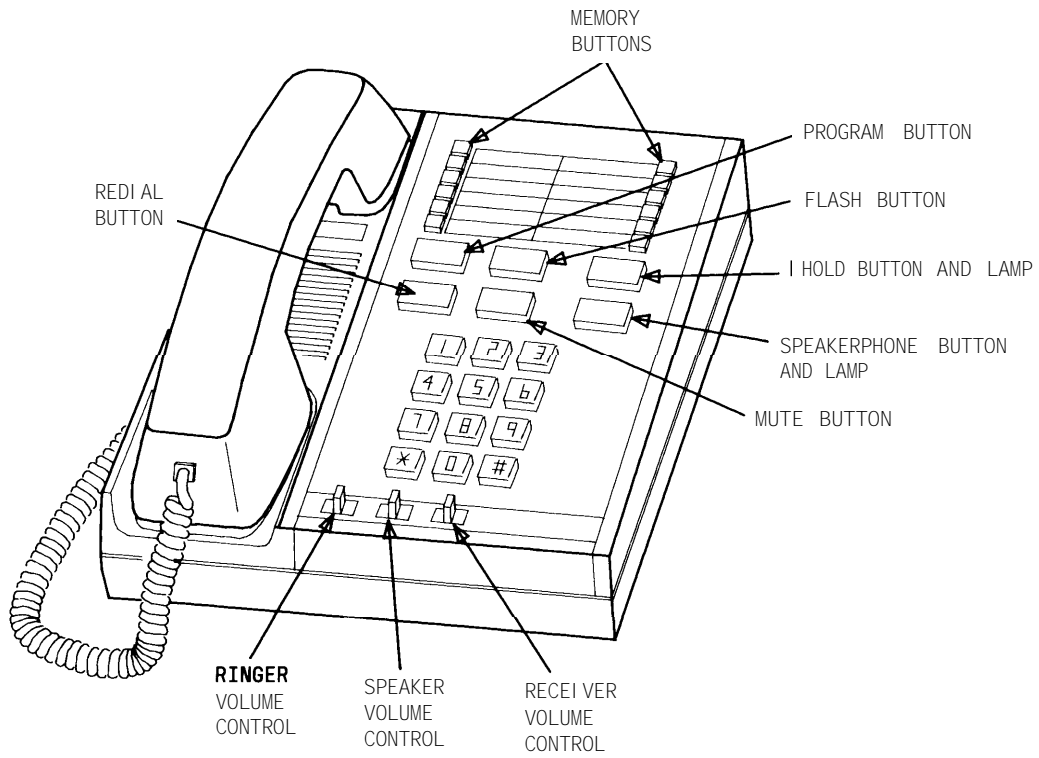


Figure 4-6. 420 Speakerphone Voice Terminal

TABLE 4-D. Model 2500 Series Voice Terminal Adjuncts

Adjunct	2500DMGC	2500MMGB	2500SM	2514 BMW	2554BM
4A Speakerphone			x		
55A1 Ground Start Key	x	x	x	x	x
AT&T Answer-Record 2500 or Code-A-Phone 2540 (Answering Machine)	x	x	x	x	x
Z34A Message Waiting Indicator		x	x	x	x

Single-Line Voice Terminal Connection Information

Single-line voice terminal connection information is provided in the following figures:

- Figure 4-21—On-Premises Single-Line Voice Terminal Connections
- Figure 4-22—Out-Of-Building Single-Line Voice Terminal Connections
- Figure 4-23—Off-Premises Station Single-Line Voice Terminal Connections.

Descriptions of the Station Interconnect Panel (SIP), Trunk Access Equipment (TAE), and associated cables and adapters, as shown on the figures, are provided under the heading “connectivity” later in this Section.

Maximum cabling distances from the system cabinets to single-line voice terminals is provided in Section 5, “Technical Specifications.”

Feature Operations

Refer to *Single-Line Terminal User Guide* (555-520-702) for information about feature operation.

Multiline Voice Terminals

The system supports MET (10- and 12-button only) and MERLIN (7300H Series) multiline voice terminals.

Multiline voice terminals have two LEDs located beside each assignable button (except for the 34-button basic set). The LEDs are referred to as I-Use (red) and Status (green) LEDs. Additional information on the LEDs is provided in the “Line Status and I-Use Indications” feature description in Section 2.

Most multiline voice terminals support adjuncts. The supported adjuncts are described following the voice terminal descriptions.

Multiline voice terminals can have more than one call appearing at the terminal at one time. Each multiline terminal has two System Access buttons on which calls can be made or received. System Access buttons are, essentially, inside line buttons.

Multiline voice terminals transmit voice signals in analog form and control signals in digital form. The terminals operate over Premises Distribution System 4-pair wiring.

7300H Series Voice Terminals: These voice terminals are the same as those used with MERLIN Communications Systems. They connect to ports on the Analog Terminal Link (ATL) Line (ZTN79) CP. The ATL Line CP converts the analog voice signals to digital form before placing them on the TDM bus. Three-wire pairs connect these sets to the port CPs; see “Port Specifications” in Section 5 for details.

The dial pad on 7300H series terminals is not a touch-tone pad in that the signals generated are digital, not DTMF signals. A digital signal is sent to the switch with each button press. Reference to the dial pad on these sets throughout this manual is in the form, “touch dial pad.”

MET Sets: The MET Sets are not orderable as part of the system equipment but can be used in System 25 installations, where appropriate (i.e., in reuse situations). Three-wire pairs connect these sets to the port CPs; see “Port Specifications” in Section 5 for details.

The TN735 MET Line CP converts the analog voice signals to digital form before placing them on the TDM bus. The touch-tone dial pad on MET voice terminals generates DTMF signals.

The following subsections provide descriptions and illustrations of the multiline voice terminals supported by System 25.

5-Button Voice Terminal (7302H01D) PEC 3160-111

The 5-Button Terminal (Figure 4-7) can be desk or wall mounted and is about 5-3/4 inches wide, 5-1/4 inches high, and 8-1/2 inches long. The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- I-Use and Status LEDs
- Test/Program Switch
 - T—Used to test the operation of LEDs and ringer
 - P—Used to program features
- Built-in Speaker
- Speaker/Ringer Volume Control
- Six Predefine Buttons
 - Conference
 - Drop
 - Transfer
 - Hold
 - Speaker
 - Recall
- Five Feature Buttons (each equipped with I-Use and Status LEDs)
 - Two predefine as System Access
 - Three programmable (default assignments for V1 are all Repertory Dial; in V2, one Repertory Dial button is replaced by Last Number Dialed).

Note: This set does not have a Message button or Message LED.

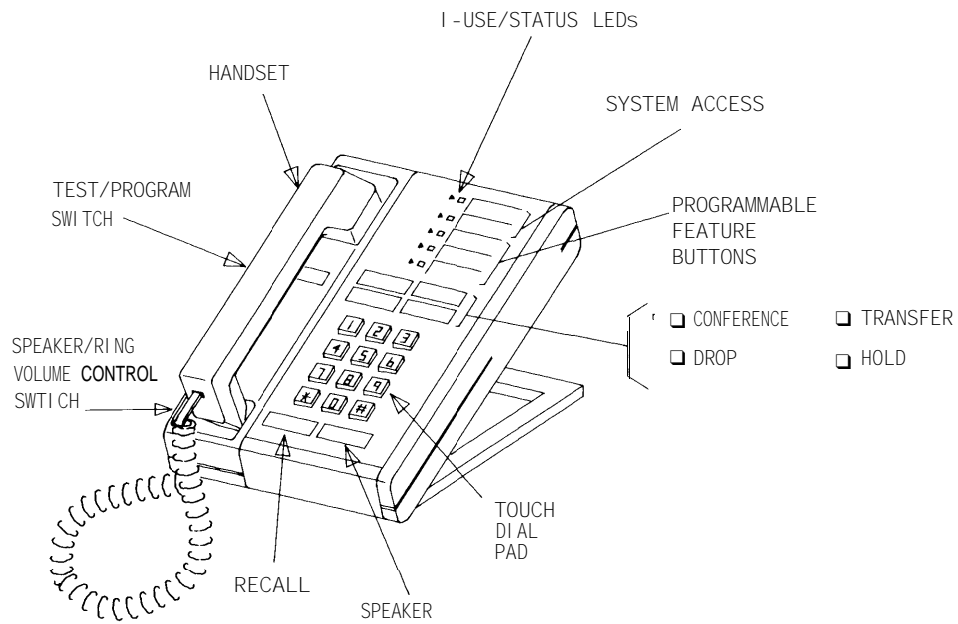


Figure 4-7. 5-Button Voice Terminal (7302 HOLD)

10-Button Voice Terminal (7303H01D) PEC 3161-172

The 10-Button Terminal (Figure 4-8) can be desk or wall mounted and is about 7 inches wide, 5-1/4 inches high, and 8-1/2 inches long. The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- I-Use and Status LEDs
- Test/Program Switch (two positions)
 - T—Used to test the operation of LEDs and ringer
 - P—Used to program features
- Built-in Speaker
- Speaker/Ringer Volume Control
- Seven Predefined Buttons
 - Conference
 - Drop
 - Transfer
 - Hold
 - Speaker
 - Message
 - Recall
- Ten Feature Buttons (each with I-Use and Status LEDs)
 - Two predefined as System Access
 - Eight programmable (default assignments for V1 are Repertory Dial [3], Flex DSS [3], Send All Calls [1], and Account Code Entry [1]; in V2, one Repertory Dial button is replaced by Last Number Dialed).

Adjuncts:

- 502A Headset Adapter (PEC 3164-HFA)
- S102A Speakerphone (PEC 3163-HFU).

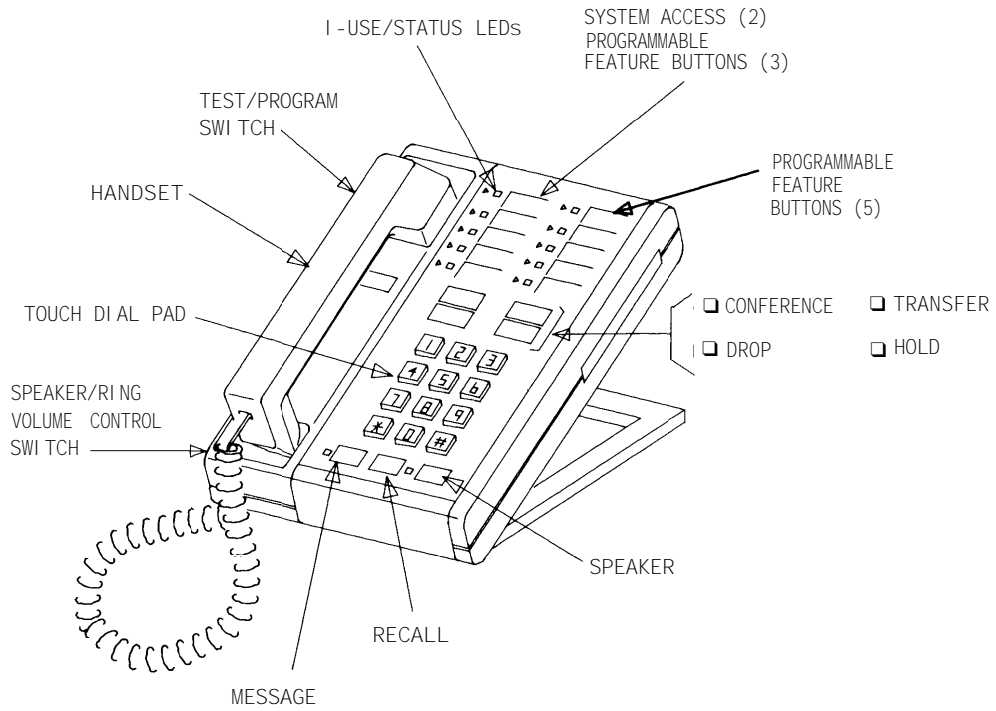


Figure 4-8. 10-Button Voice Terminal (7303HO1D)

34-Button Voice Terminal (7305H01D) (MD) PEC 3162-412

The 34-Button Terminal (Figure 4-9) can be desk or wall mounted and is about 10-1/4 inches, 5-1/2 inches high, and 8-1/2 inches long. *This set is available only on a reuse basis and is not orderable via the Delivery Operation Support System (DOSS) Configurator.* The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- I-Use and Status LEDs
- Test/Program Switch (two positions)
 - T—Used to test the operation of LEDs and ringer
 - P—Used to program features
- Built-in Speaker
- Speaker/Ringer Volume Control
- Seven Predefined Buttons
 - Conference
 - Drop
 - Transfer
 - Hold
 - Speaker
 - Message
 - Recall
- 34 Feature Buttons (only ten with I-Use and Status LEDs)
 - Two predefined as System Access
 - 32 programmable (default assignments for V1 are Repertory Dial [3], Flex DSS [27], Send All Calls [1], and Account Code Entry [1]; in V2, one Repertory Dial button is replaced by Last Number Dialed).

Note: The programmable buttons without LEDs should not be programmed for features that require I-Use and Status LEDs. Only the Repertory Dialing, Manual Signaling, and Account Code Entry features should be assigned to these buttons.

Adjuncts:

- 502A Headset Adapter (PEC 3164-HFA)
- S102A Speakerphone (PEC3163-HFU).

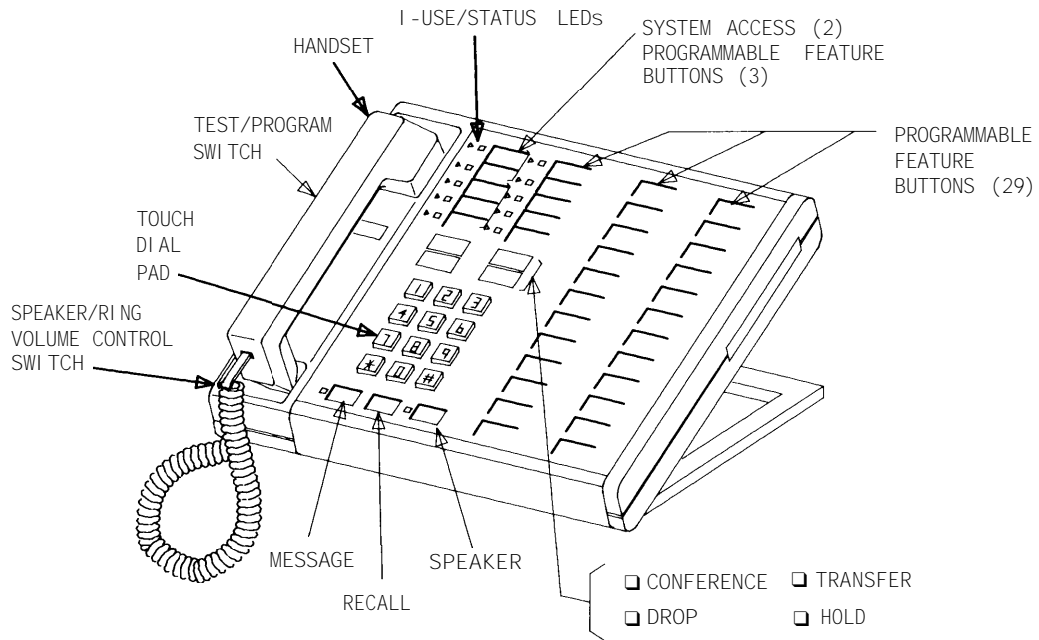


Figure 4-9. 34-Button Voice Terminal (7305HO1D) (MD)

34-Button Deluxe Voice Terminal (7305H02D) PEC 3162-417

The 34-Button Deluxe terminal (Figure 4-10) is available for general use and as a Direct Trunk Attendant Console. The voice terminal is about 10-1/4 inches wide, 5-1/2 inches high, and 8-1/2 inches long. It comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- I-Use and Status LEDs
- Test/Program Switch (two positions)
 - T—Used to test the operation of LEDs and ringer
 - P—Used to program features on the voice terminal
- Built-in Speaker
- Speaker/Ringer Volume Control
- Seven Predefined Buttons
 - Conference
 - Drop
 - Transfer
 - Hold
 - Speaker
 - Message
 - Recall
- 34 Feature Buttons (each equipped with I-Use and Status LEDs)

For General Use:

- Two predefined as System Access
- 32 Programmable (default assignments for V1 are, Repertory Dial [3], Send All Calls [1], Account Code Entry [1], and Flex DSS [27]; in V2, one Repertory Dial button is replaced by Last Number Dialed)

For Use as a Direct Trunk Attendant Console:

- Two predefined as System Access
- Six other predefined (Start, Cancel, Release, Return-On-Don't-Answer, Return-On-Busy, and Alarm)
- 26 programmable (default assignments for V1 are Repertory Dial [3], Flex DSS, Account Code Entry [1], Attendant Message Waiting [1], Night Service [1], trunk appearances [15 as Personal Lines*], Group Call Coverage [1], and Direct Facility Access† [3]; in V2, one Repertory Dial button is replaced by Last Number Dialed)

* On the primary Attendant Console, the first 15 trunks in the system are assigned button appearances on the console. If there are fewer than 15 trunks, the remaining buttons are not assigned. On the secondary Console these trunks do not receive

default assignments.

† On the primary Attendant Console, the first of the Direct Facility (Pooled) Access buttons defaults to loop-start trunks, the second to ground-start trunks, and the third to tie trunks. For any trunk type not assigned in the system, the associated button does not receive a default assignment. On the secondary Console, these buttons do not receive default assignments.

Adjuncts:

- 502A Headset Adapter (PEC3164-HFA)
- S102A Speakerphone (PEC3163-HFU).

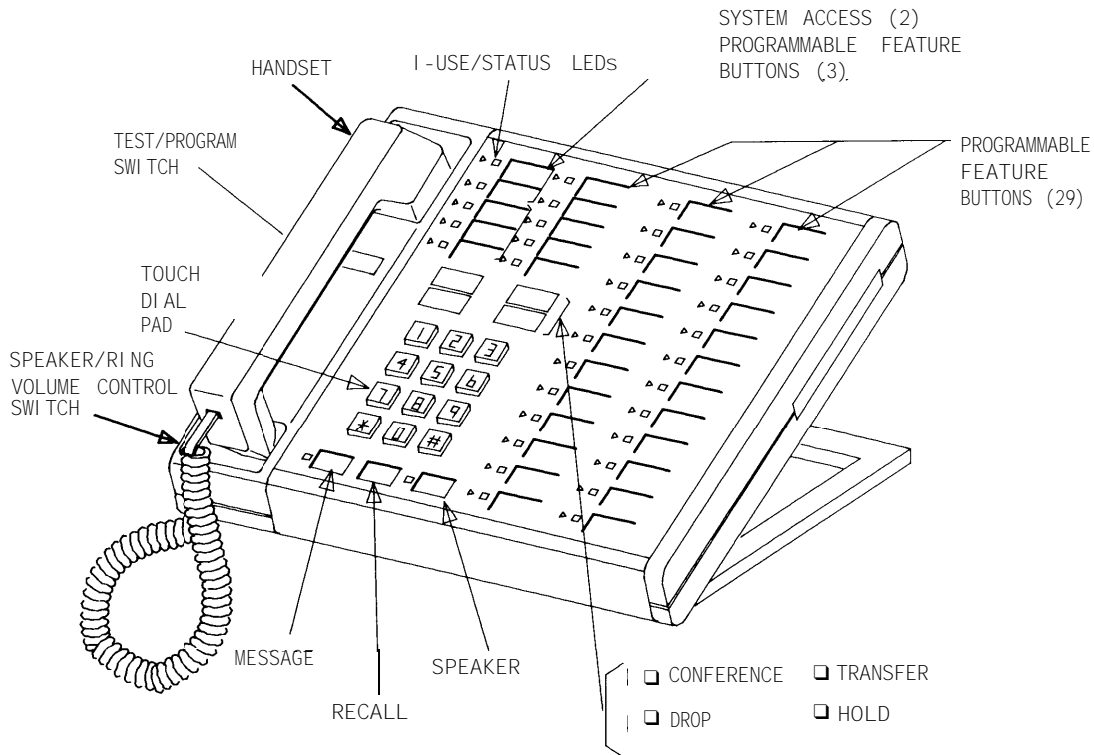


Figure 4-10. 34-Button Deluxe Voice Terminal (7305H02D)

Built-In Speakerphone (BIS) Voice Terminal (7305H03B) PEC 3162-BIS

The BIS voice terminal (Figure 4-10) can be desk or wall mounted and is about 9-1/2 inches wide, 5-1/4 inches high, and 9-1/4 inches long. The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- I-Use and Status LEDs
- Test/Program Switch (two positions)
 - T—Used to test the operation of LEDs and ringer
 - P—Used to program features
- Built-In Speakerphone
- Speakerphone Volume Control
- Built-In HFAI Microphone
- Speaker/Ringer Volume Control
- Nine Predefined Buttons
 - Conference
 - Drop
 - Transfer
 - Hold
 - Speakerphone
 - Recall
 - Message
 - Microphone
 - HFAI
- 34 Feature Buttons (only 12 have associated LEDs)
 - Two predefined as System Access
 - 32 programmable (default assignments for V1 are Repertory Dial [3], Flex DSS [27], Send All Calls [1], and Account Code Entry [1]; in V2, one Repertory Dial button is replaced by Last Number Dialed)

Note: The programmable buttons without LEDs should not be programmed for features that require I-Use and Status LEDs. Only the Repertory Dialing, Manual Signaling, and Account Code Entry features should be assigned to these buttons.

Adjunct:

- 502A Headset Adapter (PEC 3164-HFA).

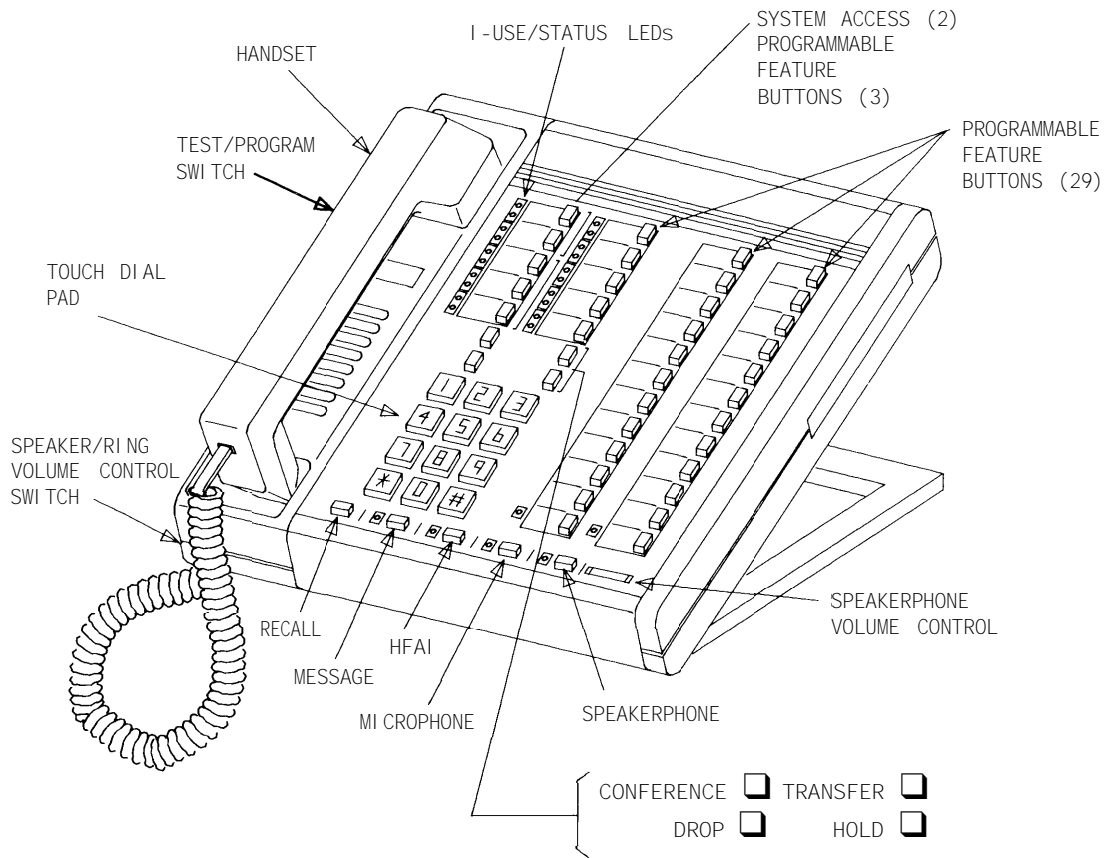


Figure 4-11. BIS Voice Terminal (7305H03B)

Built-In Speakerphone (BIS) Voice Terminal With Display (7305H04C) PEC 3162-DIS (V2 only)

The BIS Voice Terminal with Display (Figure 4-12) is available for general use and as a Switched Looped Attendant Console (SLAC). In general use, it can be desk or wall mounted; as a console it is normally desk mounted. The terminal is about 9-1/2 inches wide, 5-1/4 inches high, and 9-1/4 inches long. The set comes equipped with the following:

- Handset
- Touch Dial Pad (not DTMF)
- I-Use and Status LEDs
- Test/Program Switch (two positions)
 - T—Used to test the operation of LEDs and ringer
 - P—Inoperative on this terminal
- Built-In Speakerphone
- Built-In Display Module
 - Screen for call information and time displays
 - Clock/Timer controls
 - Contrast control for screen display
- Speakerphone Volume Control
- Built-In HFAI Microphone
- Speaker/Ringer Volume Control
- Nine Predefined Buttons
 - Conference
 - Drop
 - Transfer
 - Hold
 - Speakerphone
 - Recall
 - Message
 - Microphone
 - HFAI
- 34 Feature Buttons (only 12 have associated LEDs)

For General Use:

- Two predefined as System Access
- 32 programmable (default assignments are Repertory Dial [2], Last Number Dialed [1] Flex DSS [27], Send All Calls [1], and Account Code Entry [1])

For Use as a Switched Loop Attendant Console:

- Five predefined as loop buttons
- Twelve other predefined (Inspect, Attendant Message Waiting, Alarm, Local, Scroll, Forced Release, Start, Source, Release, Destination, Cancel, Join)
- Seventeen programmable (defaulted as Flex DSS [15], Position Busy [1], and Last Number Dialed [1])

Note: The programmable buttons without LEDs should not be programmed for features that require I-Use and Status LEDs. Only the Repertory Dialing, Manual Signaling, and Account Code Entry features should be assigned to these buttons.

Adjunct:

- 502A Headset Adapter (PEC 3164-HFA).

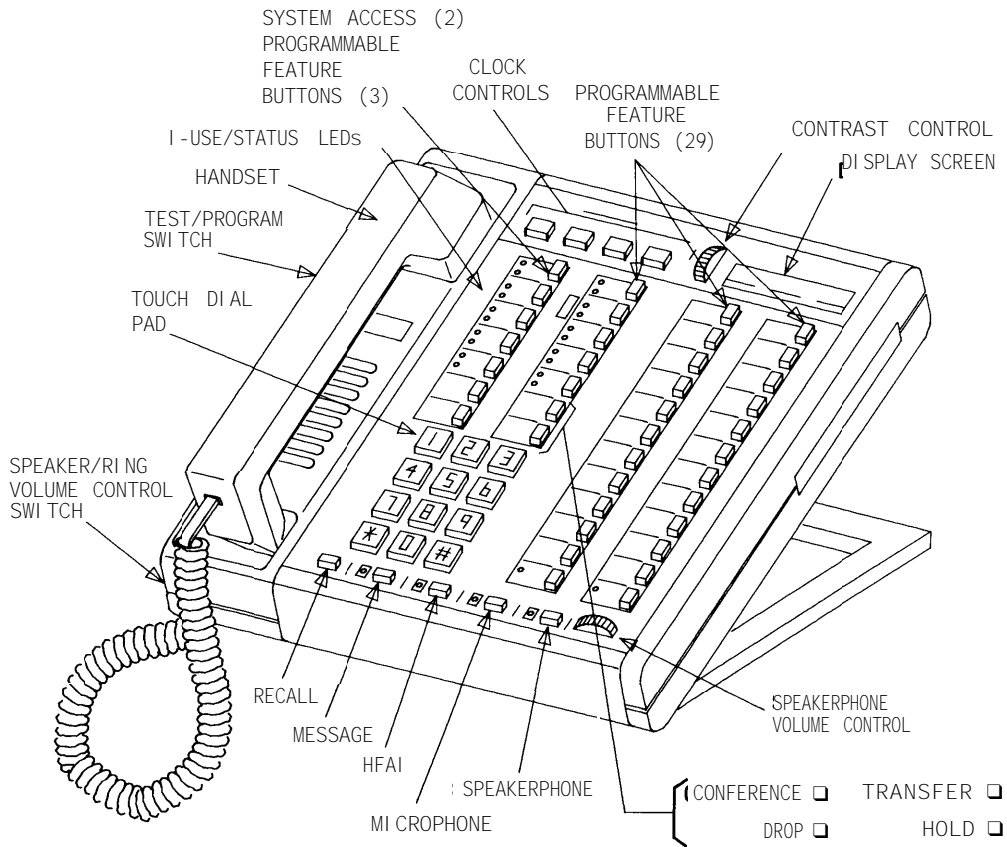


Figure 4-12. BIS Voice Terminal with Display (7305H04B)

***Hands-Free-Answer On Intercom (HFAI) Voice Terminal (7309H01B) PEC
3161-161***

The HFAI voice terminal (Figure 4-13) can be desk or wall mounted and is about 6-1/4 inches wide, 5-1/4 inches high, and 9 inches long. The set comes equipped with the following:

- Handset
- Touch dial pad (not DTMF)
- I-Use and Status LEDs
- Test/Program Switch (two positions)
 - T—Used to test the operation of LEDs and ringer
 - P—Used to program features on the voice terminal
- Built-In HFAI Microphone
- Built-In Speaker
- Speaker/Ringer Volume Control
- Eight Predefine Buttons
 - Conference
 - Drop
 - Transfer
 - Hold
 - Speaker
 - Recall
 - Message
 - HFAI Microphone
- Ten Programmable Buttons (each equipped with I-Use and Status LEDs)
 - Two predefine as System Access
 - Eight programmable (default assignment for V1 is Repertory Dial [3], Flex DSS [3], Send All Calls [1], and Account Code Entry [1]; in V2, one Repertory Dial button is replaced by Last Number Dialed).

Adjuncts: None

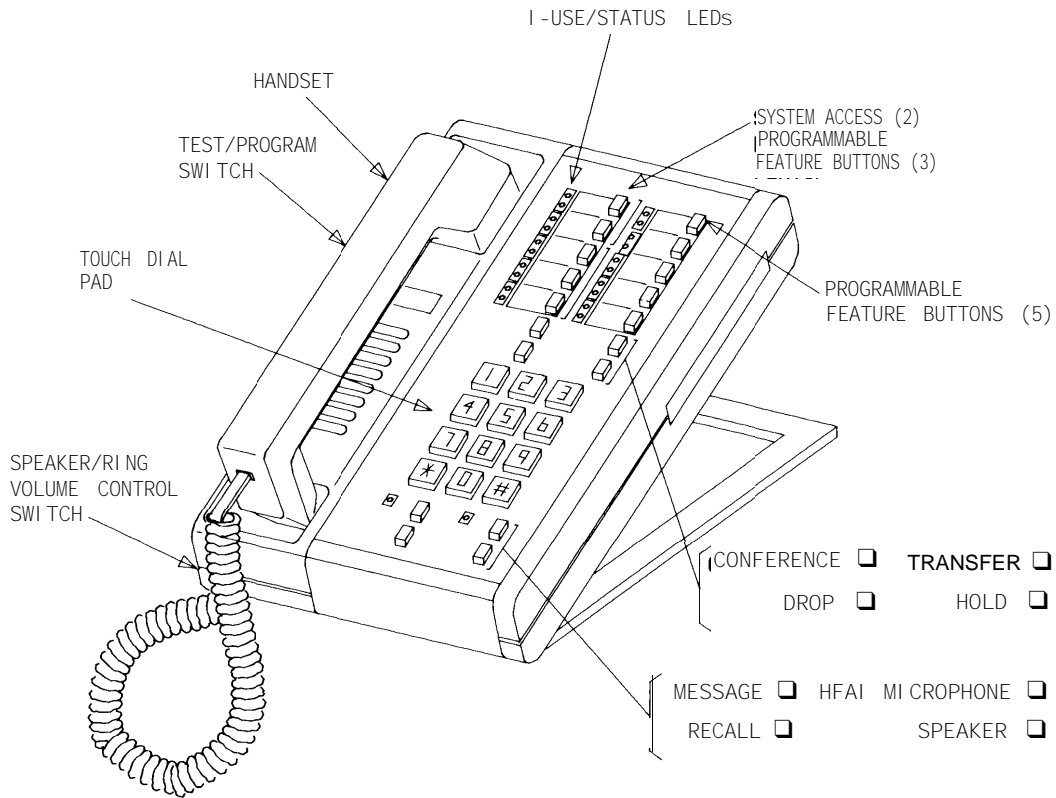


Figure 4-13. HFAI Voice Terminal (7309HO1B)

Ten Button Multibutton Electronic Telephone (MET) (2991 C/D05)

The Ten Button MET set (Figure 4-14) may be desk or wall mounted. *This set is available only on a reuse basis and is not orderable via the Delivery Operation Support System (DOSS) Configurator.* The set comes equipped with the following:

- Handset
- Touch-Tone Dial Pad (DTMF)
- I-Use and Status LEDs
- Tone Ringer Volume Control
- Six Predefined Buttons
 - Recall
 - Conference
 - Drop
 - Transfer
 - Hold
 - Message
- Five Feature Buttons (each equipped with I-Use and Status LEDs)
 - Two predefined as System Access
 - Three programmable (default assignments for V1 are Repertory Dial; in V2, one Repertory Dial button is replaced by Last Number Dialed).

Adjuncts:

- 4A Speakerphone
- MET Headset Adapter.

Note: The BLF version of this set, unless modified, will not work on System 25.

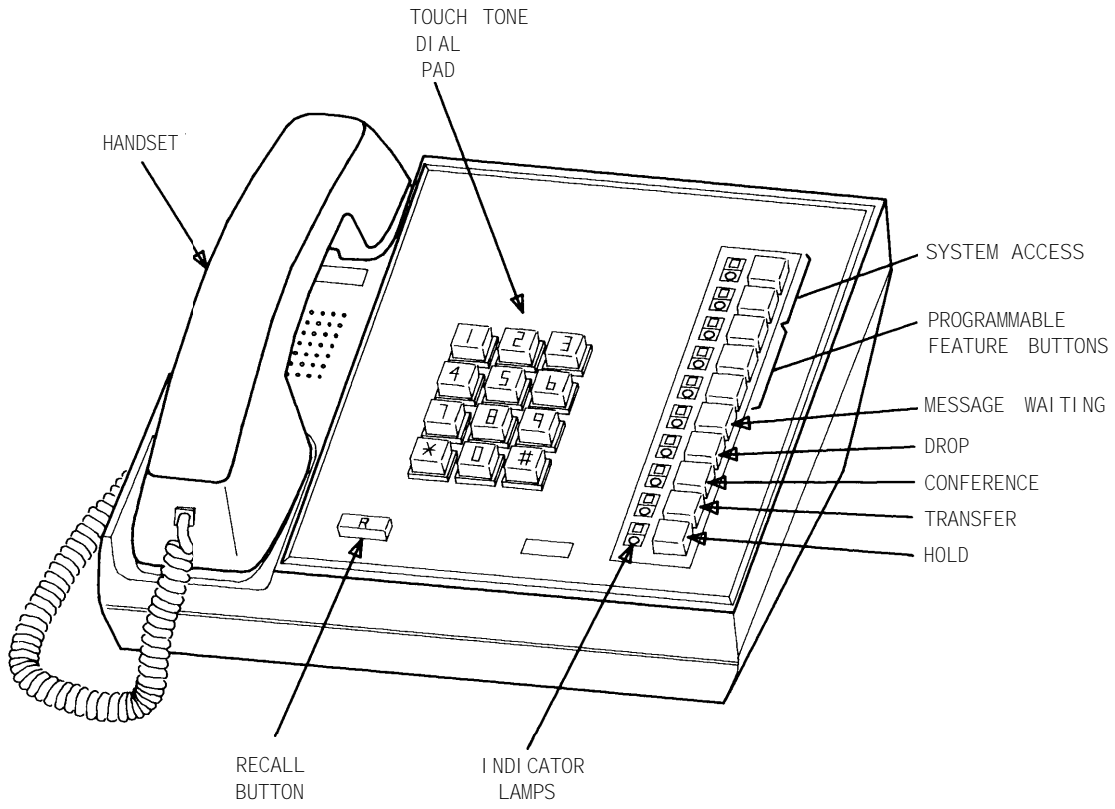


Figure 4-14. Ten Button MET Set (2991C05)

Ten Button MET With Built-In Speakerphone (2993C04)

The 10-Button MET set with BIS (Figure 4-15) can be desk or wall mounted. *This set is available only on a reuse basis and is not orderable via the Delivery Operation Support System (DOSS) Configurator.* The set comes equipped with the following:

- Handset
- Touch dial pad (DTMF)
- I-Use and Status LEDs
- Built-In Speakerphone
- Tone Ringer Volume Control
- Speakerphone Volume Control
- On/Quiet and Off Speakerphone Control Buttons
- Speakerphone Indicator Lamp
- Six Predefined Buttons
 - Recall Button
 - Conference
 - Drop
 - Transfer
 - Hold
 - Message
- Five Feature Buttons (each equipped with I-Use and Status LEDs)
 - Two predefined as System Access
 - Three programmable (default assignments for V1 are Repertory Dial; in V2, one Repertory Dial button is replaced by Last Number Dialed).

Adjuncts: None

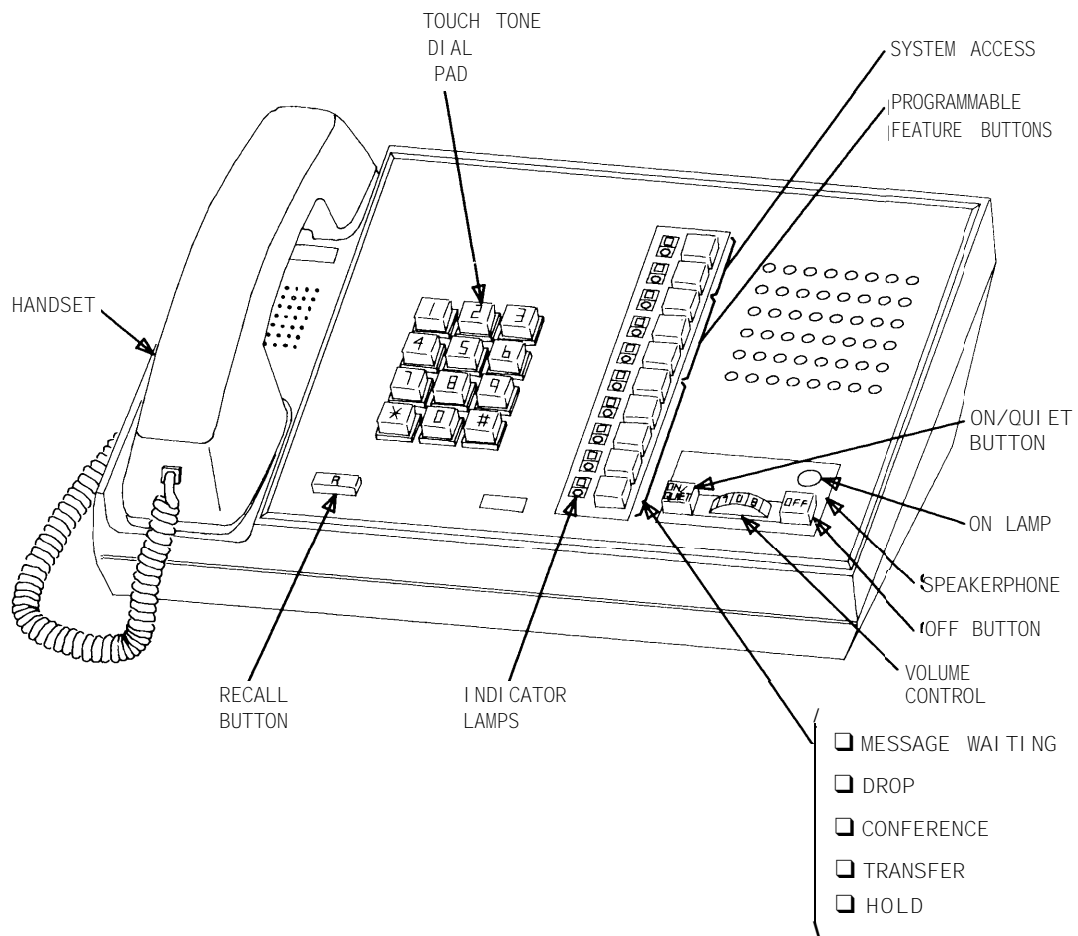


Figure 4-15. Ten Button MET With Built-In Speakerphone (2993C04)

Twelve Button MET Set (7203M)

The 12-Button MET set (Figure 4-16) is a freestanding voice terminal. *This set is available only on a reuse basis and is not orderable via the Delivery Operation Support System (DOSS) Configurator.* The set comes equipped with the following:

- Handset
- Touch-Tone Dial Pad (DTMF)
- I-Use and Status LEDs
- Tone Ringer Volume Control
- Message Waiting LED
- Seven Predefined Buttons
 - Recall
 - Conference
 - Drop
 - Transfer
 - Hold
 - Message
 - Disconnect
- Five Feature Buttons (each equipped with I-Use and Status LEDs)
 - Three programmable (default assignments for V1 are Repertory Dial; in V2, one Repertory Dial button is replaced by Last Number Dialed).

Adjuncts:

- 500A Headset Adapter
- S101A Speakerphone.

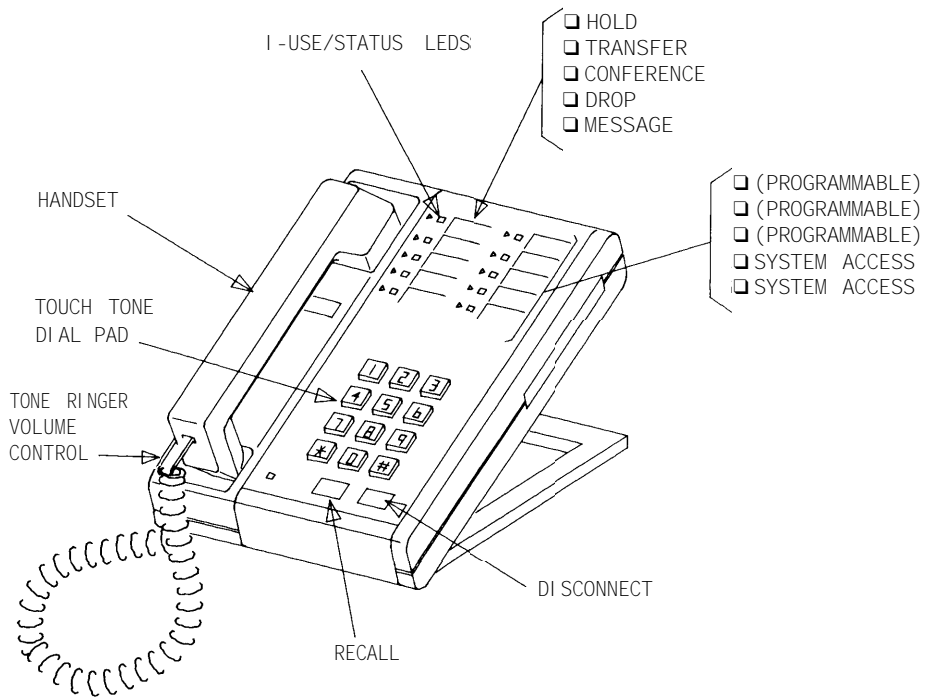


Figure 4-16. Twelve Button MET Set (7203M)

Multiline Voice Terminal Connection Information

Detailed connection information is provided in the following figures:

- Figure 4-24—On-Premises 7300H Series Multiline Voice Terminal Connections
- Figure 4-25—Off-Premises 7300H Series Multiline Voice Terminal Connections
- Figure 4-26—Ten Button MET Set Connections
- Figure 4-27—Stand-Alone Remotely Powered Voice Terminal Connections

Descriptions of the Station Interconnect Panel (SIP), Trunk Access Equipment (TAE), and associated cables and adapters, as shown on the figures, are provided under the heading “Connectivity” later in this Section.

Maximum cabling distances from the system cabinets to multiline voice terminals is provided in Section 5, “Technical Specifications.”

Feature Operations

Refer to *Multiline Terminal User Guide* (555-520-703) for information about feature operation.

Voice Terminal Adjuncts

The following adjuncts and associated power supplies are supported:

- MET Headset Adapter (for 10-Button MET Sets); refer to “Headset Adapter Adjunct” in Section 2.
- 500A/502A Headset Adapter (for 12-Button MET Sets and MERLIN voice terminals, respectively); refer to “Headset Adapter Adjunct” in Section 2.
- 4A Speakerphone System (for 10-Button MET Sets); refer to “Speakerphone Adjunct” in Section 2.
- S101A/S102A Speakerphone (for 12-Button MET Sets and MERLIN voice terminals, respectively); refer to “Speakerphone Adjunct” in Section 2.
- 55A1 Key (Ground Start Button)

A 55A1 Ground Start Button is required to obtain dial tone for each Power Failure Transfer (PFT) station that is connected to a ground start trunk.

- Acoustic Coupler

An Acoustic Coupler (349A Adapter) can be used with the 7300H series voice terminals and MET sets. The coupler provides acoustic coupling between the handset and acoustic modems.

- AT&T Answer-Record 2500

The Answer-Record 2500 (PEC 3121-050) or Code-A-Phone 2540 (PEC 3121-040) can be used as an adjunct (using a 267A “T” Adapter) to single-line voice terminals. Note, that when an answering device is bridged on to a ZTN78 Tip Ring Line CP port, the combined adjunct/terminal REN must not exceed 1.20 A/B.

- Z34A Message Waiting Indicator

The Z34A Message Waiting Indicator provides a message waiting indication at single-line sets. The indicator can be used with 2500 series voice terminals that do not have message waiting lamps.

Voice Terminal Adjunct Connection Information

Detailed adjunct connection information is provided in Section 2 with the detailed feature descriptions of the headset adapter and speakerphone adjuncts.

Descriptions of the Station Interconnect Panel (SIP), Trunk Access Equipment (TAE), and associated cables and adapters, as shown on the figures, are provided under the heading “Connectivity” later in this Section.

Voice Terminal Adjunct Power Supplies

Table 4-E provides a summary of the supplemental power supplies and their applications.

TABLE 4-E. Supplemental Voice Terminal Power Supplies

UNIT	OUTPUT	FOR USE WITH
2012D Transformer	18 V ac	ADUs (except Z3A5) if required; MET auxiliary power.
KS22911-L1 Power Unit	48 V dc	Selector Console; Z3A5 ADUs; MERLIN sets that require local power.
85B1 Power Unit	18 V ac	4A Speakerphone System

Note: The S102A Speakerphone does not require supplemental power except when connected to a 34-Button Deluxe Voice Terminal. The 502A Headset Adapter does not require supplemental power except when connected to a 34-Button Deluxe, BIS, or BIS with Display voice terminal.

Attendant Consoles

System 25 can have one of the following attendant console configurations:

- One or two Direct Trunk Attendant Consoles (DTAC)–34-Button Deluxe Voice Terminals (7305 H02B) administered for attendant service. Either or both positions can have a Direct Extension Selector Console associated with it.
- R1V2 Only: One or two Switched Loop Attendant Consoles (SLAC)–34-Button BIS/Display Voice Terminals administered for attendant service. Either or both positions can have a Selector Console.

Complete information on the Attendant Console feature can be found in Section 2.

Asynchronous Data Units (AD US)

Asynchronous Data Units, and Multiple Asynchronous Data Units (MADUs), provide an interface between ports on the TN726 Data Line CP and RS-232C Data Terminal Equipment (DTE) or Data Communications Equipment (DCE). The DTE is equipment that provides a data source, termination, or both—a host computer, printer, or a data terminal are examples of DTE. The DCE is equipment that provides the functions required to establish, maintain, and terminate data communications—modems are the most common DCE.

The Z3A series of Asynchronous Data Units (ADUs) (Figure 4-17) are DCE that allow a direct connection between DTE and port circuits on the Data Line CP (TN726). To connect an ADU to DCE equipment, a cross-over cable (“null modem”) is required.

The modular jack labeled “Wall” connects the ADU to the building wiring with a standard 4-pair modular cord. The 400B2 Adapter is required to provide supplemental AC power and can be bridged at the wall jack if required (Z3A1, 2, and 4 units only).

The modular jack labeled “Telephone” allows a voice terminal to be attached to the ADU. Separate wire pairs from the telephone to the system cabinets are provided in a single 4-pair cable run back to the SIP. The pairs separate at the SIP for connection to voice and data ports.

The Z3A series of ADUs measure about 4.5 inches in length, 2 inches wide, and 1 inch high. The ADUs available are shown in Table 4-F.

The Z3A ADUs should be installed only on lightning-protected facilities; they are not designed to be used with CO cables or with exposed outside wiring (such as aerial cables).

The Z3A series of ADUs offer the following features:

- Provide an interface to the digital switch from RS-232C device.
- Increases the distance RS-232C signals can travel over standard twisted-pair wiring. Refer to Section 5, “Technical Specification” for distance limitations.
- Data and control signals can be transmitted 2,000 feet in asynchronous full-duplex mode at speeds up to 19,200 bps. The transmission speed automatically matches that of the attached RS-232C device.
- The DC isolation via opto-couplers ensures high noise immunity, resulting in very low error rates.
- A variety of Z3AS with different connectors allows easy connection to RS-232C terminals, printers, and host computers (see Table 4-F).
- Most Z3AS can be powered from the RS-232C interface. The ADU requires 7 volts on pin 20 (DTR) to operate properly. If the RS-232C equipment cannot meet this requirement, a low-voltage power transformer and adapter(s) must be connected. Z3A5 ADUs *always* require supplemental power.
- An analog single-line voice terminal (2500 or 7100 series) or a 7300H series multiline voice terminal (Z3A5 ADU required) can be connected to the ADU, allowing the voice terminal and DTE to share a common wall jack and 4-pair cable run back to the SIP.

Note: Neither off-premises nor out-of-building service can be provided with ADUs.

For additional information on ADUs, see *Z3A Asynchronous Data Unit User Manual* (555-401-701).

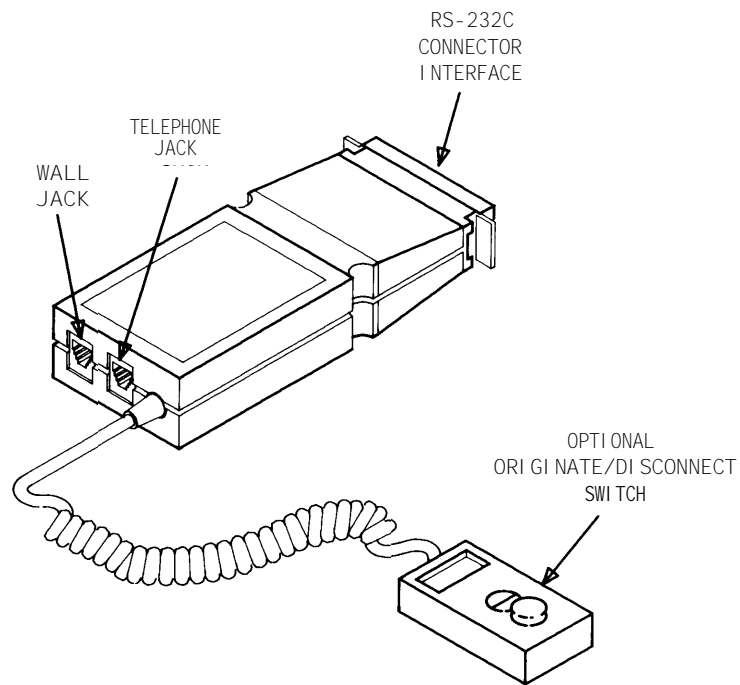


Figure 4-17. Asynchronous Data Unit (ADU)

ADU Connection Information

Detailed connection information is provided in the following figures:

- Figure 4-28—Typical ADU Connections Supporting Data Terminal And Single-Line Voice Terminal
- Figure 4-29—Typical ADU Connections Supporting Data Terminal And 7300H Series Multiline Voice Terminal
- Figure 4-30—Typical MADU Connection
- Figure 4-31—Typical Z3A1/2/4 ADU Local Power Connections.

As shown in Figure 4-31, local external power can be provided to the Z3A1, Z3A2, and Z3A4 ADUs via a 2012D power transformer and a 248B Adapter attached to a 400B2 Adapter with a D6AP-87 modular cord.

Descriptions of the Station Interconnect Panel (SIP), Trunk Access Equipment (TAE), and associated cables and adapters, as shown on the figures, are provided under the heading “Connectivity” later in this Section.

TABLE 4-F. Asynchronous Data Units

UNIT	PEC	FEATURE
Z3A1	2169-001	3-foot plug-ended EIA connector and mod jack for single-line set.
Z3A2	2169-002	EIA plug and mod jack for single-line set.
Z3A4	2169-004	3-foot receptacle-ended EIA cord and mod jack for single-line set.
Z3A5	62506	3-foot plug-ended EIA connector and mod jack for hybrid set. Requires KS-22911-L1 power unit and D6AP cord.
MADU	2169-005	Self-powered. Used for host or protocol converter connections where voice terminals are not required. No sets can be connected directly to the MADU.

Peripheral Equipment

Peripheral Equipment is equipment that connects to the Call Processor (ZTN82 or ZTN128) CP, including:

- System Administration Terminal (SAT); refer to “System Administration” in section 2.
- Digital Tape Unit (DTU); refer to “Digital Tape Unit” in Section 2.
- Station Message Detail Recording (SMDR) printer or Call Accounting System (CAS); refer to “Station Message Detail Recording” in Section 2.

Auxiliary Equipment

Auxiliary equipment supports System 25 features and services. The following equipment is supported:

- Dictation Equipment; refer to “Dictation System Access” in Section 2.
- External Alerting Equipment; refer to “External Alerts” in Section 2.
- Music Source (Music-On-Hold); refer to “Music-On-Hold” in Section 2.
- Paging Equipment; refer to “Paging System Access” in Section 2.
- Recorded Delay Announcement Equipment; refer to “Direct Group Calling Delay Announcement” and “Night Service-Delay Announcements” in Section 2.
- Optional Power Equipment.

Notes:

1. Auxiliary equipment connected to the Ztn78 Tip Ring Line CP must meet the following requirements:
 - AC impedance: 600 ohms
 - DC current less than 30 ma at 24V dc
 - Ringer Equivalent Number (REN) less than 1.20
 - Distance must not exceed 2000 feet
2. Off-premises auxiliary equipment must be connected to the TN742 Analog Line CP. If the auxiliary equipment requires a contact closure, the TN763 Auxiliary Trunk CP must be used (on-premises service only).

Optional Power Equipment

In addition to the power supplies already mentioned, the following equipment can be used with System 25.

Uninterruptible Power Supply

The AT&T 1KVA Uninterruptible Power Supply (UPS) Model 010U111 PEC 2403-004 is recommended. At maximum load the UPS will bridge a 5 minute power outage. The UPS must be connected to the common System 25 power outlet. One UPS will support a two cabinet system.

AC Power Line Surge Suppressor

The TII Model 428 Self-Restoring Powerline Surge Suppressor (PEC 8310-001, Comcode 402988950) protects against electrical surges, spikes, and transients that can cause damage to the System 25 power supply. A pilot light indicates that full protection is present. The unit plugs directly into a standard 120-volt 15-amp grounded outlet, providing a dual outlet to protected equipment.

346 Modular Bulk Power Supply

The 346 Modular Bulk Power Supply (346 MBPS) is a cost effective and flexible alternative to the KS-22911 power supply. The 346 MBPS can be used where the wall outlet mounted KS-22911 cannot be used (Canada) or where multiples units are required. The 346 MBPS consists of the 346A Power Unit (346A PU) and the 346A1 Power Panel, that is the sole method of mounting the power units. Up to three 346A PUS can be mounted per power panel. Each PU is capable of powering four terminals with adjuncts, for a total of twelve terminals per full MBPS. The 346 MBPS is intended to be installed in a closet and should be near the SIP. All connections are modular and are made with cords and adapters at the 858A Adapter of the SIP. Terminals and the Selector Console can be powered by the 346 MBPS. Refer to Figure 4-27 for a typical use of the MBPS.

Connectivity

System 25 requires four-pair building wiring that conforms to AT&T Premises Distribution System (PDS) specifications. Various cords, cables, adapters, and connecting blocks are used to facilitate the connection of equipment and associated cable and wire.

Major points of connectivity include the following:

- The system cross-connect field located on a wall adjacent to the system cabinets. The field provides mounting space for the Trunk Access Equipment (TAE), Station Interconnect Panels (SIPS), and Emergency Transfer Units (ETUs).

Refer to Section 5, “Environmental Requirements” for a typical System 25 layout including cross-connect field and associated equipment layout.

- 25-pair connectors located on the rear of each system cabinet
- Modular jacks located at each work station provide modular connections for terminals and associated adjuncts and auxiliary equipment. These jacks are connected by building wiring to the SIP. Several wiring options are described below.

Wiring Options: There are three basic PECs under which building (station) wiring is ordered:

- The 2772 (-JA1, JA2, or JC1) PECs cover new, reuse and (new) plenum wiring, respectively. These PECs apply on a per wire-run basis.
- PEC 2771-JDX covers wiring on a time and materials.
- PEC 2773-JDX covers wiring based on a PCS quote.

Consult the 2770-section of the Sales Manual for restrictions and requirements before ordering.

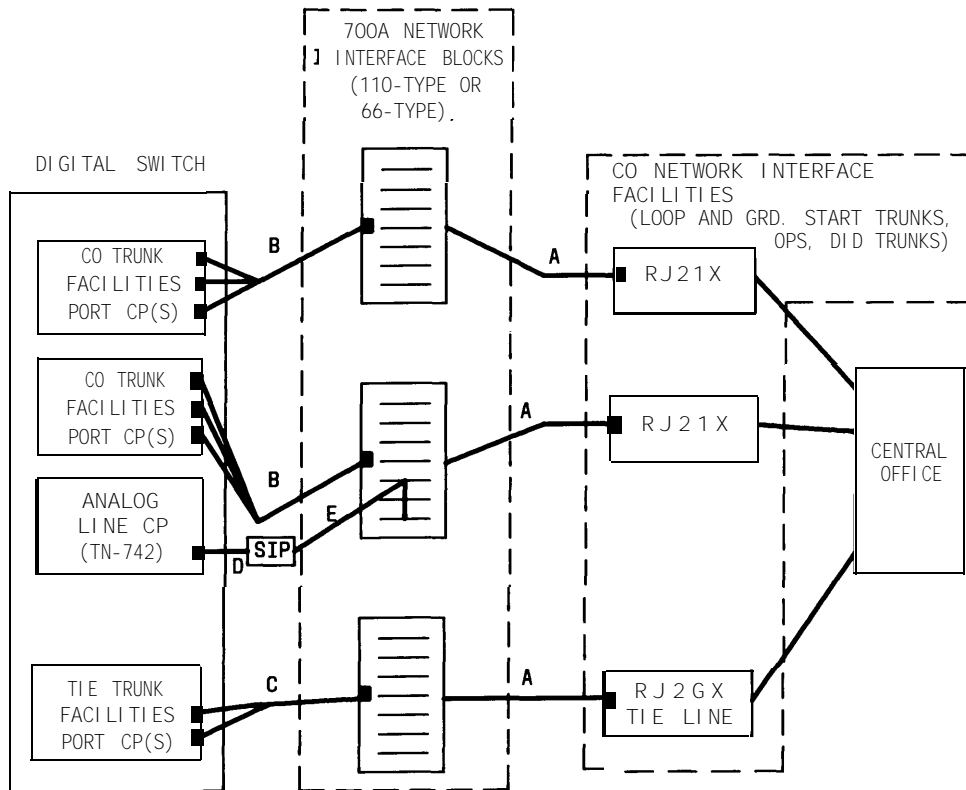
Trunk Access Equipment (TAE)

The TAE (Figure 4-18) provides for the connection of communications facilities such as Tie, Ground Start, Loop Start, and DID trunks, to the trunk ports of the system. Up to three

trunk CPs (except Tie Trunk CPs) can be connected to a three-way splitter cable (OR6016) that concentrates the CP interfaces into one 25-pair cable. Up to two Tie Trunk CPs can be connected to a two-way splitter cable (OR6015) that concentrates the CP interfaces into one 25-pair cable. Each splitter cable connects to an interface block at the TAE.

Cables are cut down or plug into the TAE blocks and plug into the telephone company provided network interface (RJ21X or RJ2GX). Trunks and tie lines are cut down by the Telco at the interface.

700A or 157B Blocks are usually used for the TAE connections (furnished by the installer).



LEGEND:

- A - SINGLE-ENDED 25-PAIR CONNECTOR CABLE (A25D)*
- B - 3 TO 1 SPLITTER CONNECTORIZED CABLE (OR6016) - PEC 2720-06X
- C - 2 TO 1 SPLITTER CONNECTORIZED CABLE (OR6015) - PEC 2720-05X
- OPS - OFF-PREMISES STATION
- SIP - STATION INTERCONNECT PANEL*
- D - OCTOPUS CABLE (WP90780) - PEC 2720-05P
- E - INSIDE WIRE*
- * - FURNISHED BY INSTALLER

Figure 4-18. Trunk Access Equipment (TAE) Connections

Station Interconnect Panel (SIP)

The Station Interconnect Panel (SIP) provides for the connection of the terminals (voice and data), peripheral equipment, and some auxiliary equipment of the system to station port CPs. This equipment includes voice terminals, attendant consoles, data terminals, System Administration Terminal, Digital Tape Unit, and Call Accounting System. The SIP is made up of 617A Panels and associated adapters.

617A Panel

The 617A Panel (Figure 4-19) is a metal plate with key slot holes on each side for mounting on a backboard. Each 617A Panel can hold eight Z210A1 or 858A Adapters, each of which can accommodate six connections to the port circuits in the cabinets. As many as five 617A Panels can be required for a maximum size system. The adapters snap into prepunched holes on the 617A Panels. (Reattached spacer buttons keep adapters from touching the metal panels.)

The cable rings located at the top of the 617A Panel route the building wiring cables to the adapters. Purse lock clips hold the building wiring cables in place. The white posts at the bottom of the 617A Panel guide the wiring from the 50A Fanning Strip to each column of adapters.

Preprinted boxes and numbers on the panel identify modular jacks for recordkeeping purposes. Letters are marked on the boxes at the top of each column by the installer. The letter (A-J) and the corresponding preprinted row number (1-24) identify the port jacks. For example, A1 identifies the modular jack located in column A row 1.

Adapters

Adapters that mount on the panel connect the following:

- Building wire runs terminated in modular jacks, 25-pair connectors, or unterminated.
- Cables from the system cabinets terminated in modular jacks or 25-pair connectors.

The following adapters can be mounted on the 617A panel:

- Z210A—Six 4-pair modular jacks to six 4-pair modular jacks. One required per six voice terminals. Connects to building wiring terminated in modular jacks. (Equivalent to six Z600A Adapters.)
- 858A—Six 4-pair modular jacks to six 110-type cut-down blocks. One required per six voice terminals. Connects to unterminated building wiring. (Equivalent to six Z601As.)

The SIP equipment is furnished by the installer.

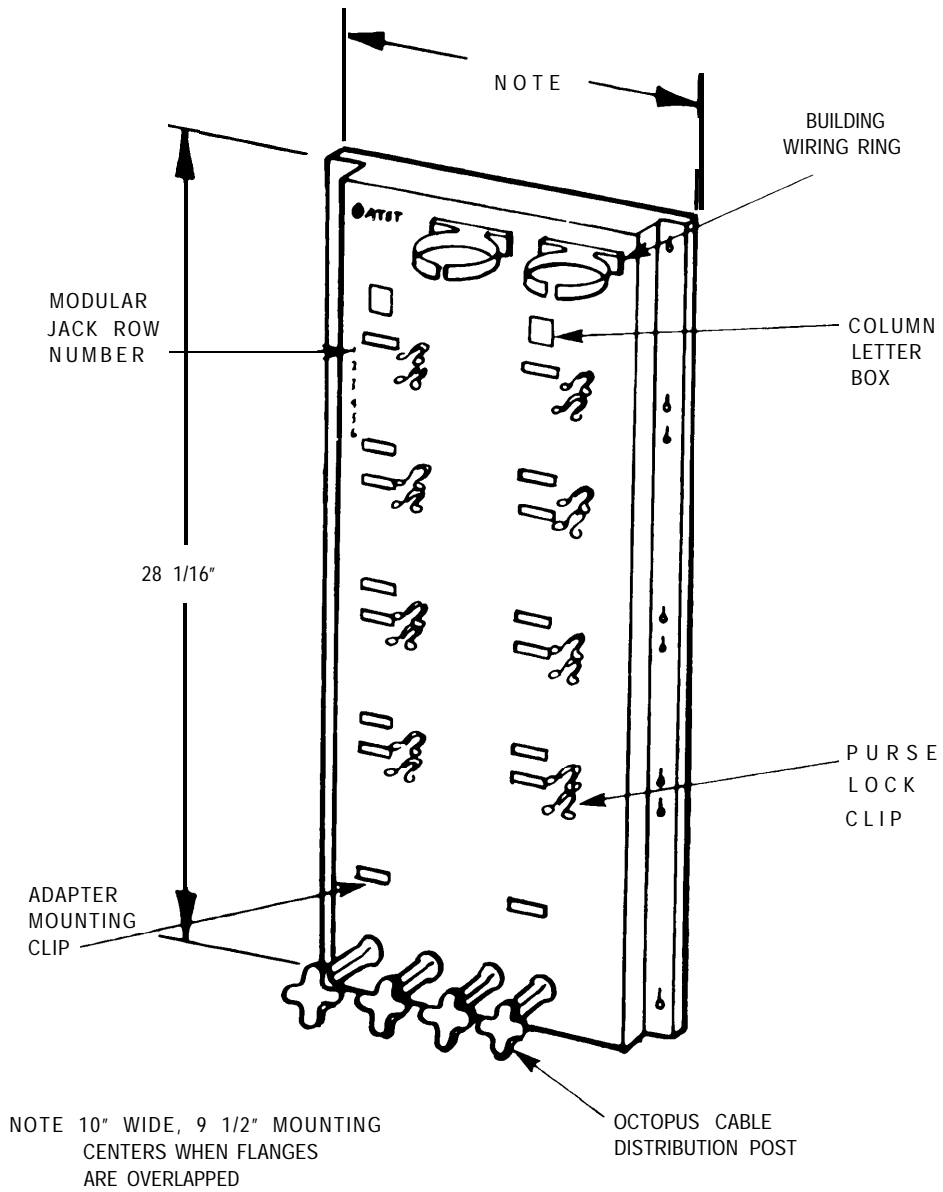


Figure 4-19. 617A Station Interconnect Panel

Figure 4-20 shows voice terminal connections to the system cabinets via the SIP. Typically, voice terminals are plugged into modular wall jacks that provide a cut-down block for building wiring. At the SIP, 858A Adapters provide a cut-down point for 4-pair wire runs. An octopus cable (WP90780) from a station CP provides 25-pair connectorized cabling to eight 4-pair modular jacks. Each jack is terminated on the SIP by a 858A Adapter. An octopus cable connects a maximum of eight voice terminals to a port CP.

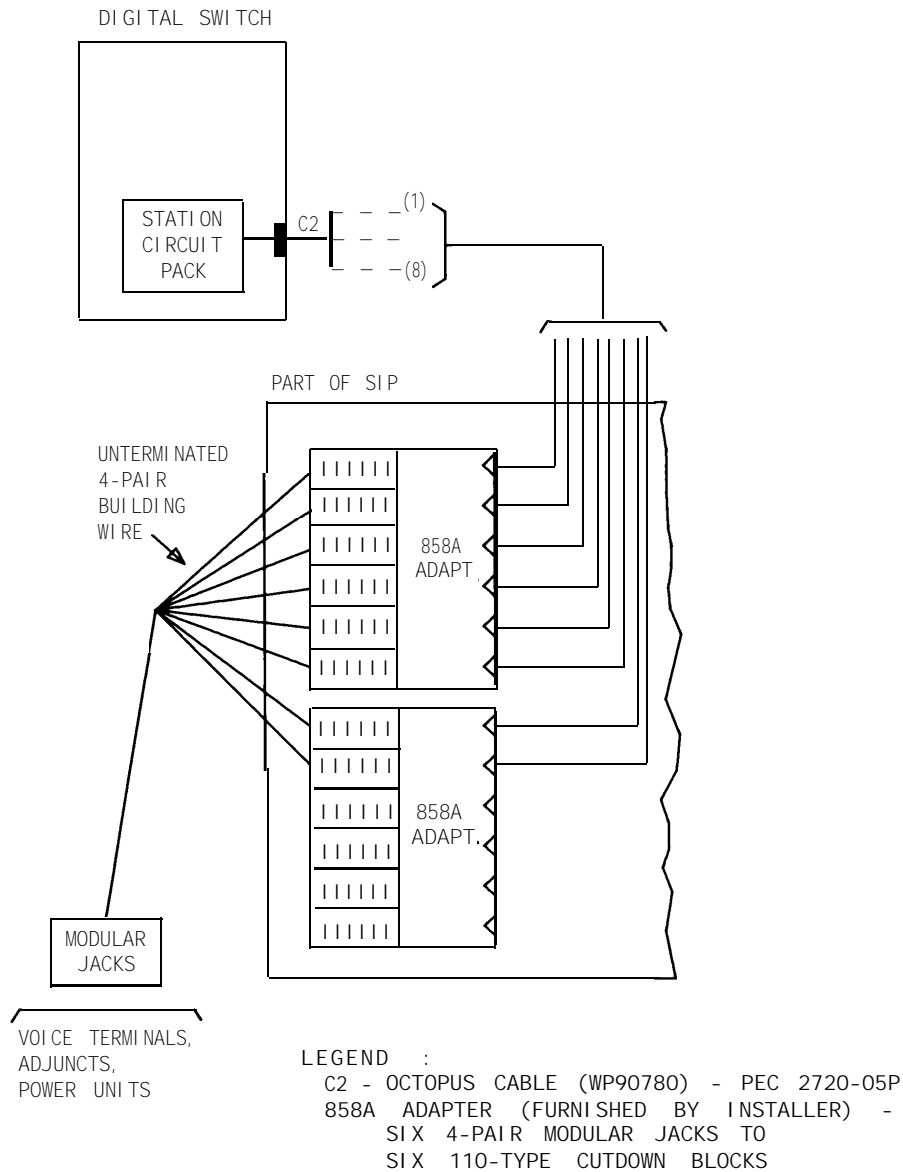


Figure 4-20. Typical SIP Connections

Connectivity Figures

Figures 4-21 through 4-31 provide connection information for various equipment. These figures have been included as an aid to understanding how equipment can be connected to System 25 and to indicate required connecting and supporting equipment. Other arrangements are possible; these figures can be useful in developing connecting arrangements for new or customer-provided equipment.

The PEC codes have been noted on the figures, as have indications of the source for obtaining non-PEC equipment (eg, from installer or furnished with other equipment). This information can be of use to Account Executives and Technical Consultants who are adding equipment to existing installations. *For new installations, the DOSS Configurator must be used to select equipment requirements.* For existing installations, you will need to determine what equipment is already installed. *You should not order equipment directly using the PECs in these figures.* The octopus cable (PEC 2720-05P), for example, supports eight terminals; you do not order one per terminal.

A list of related PECs, Apparatus, and Comcodes is provided in Section 7. Be sure to check the Sales Manual and/or DOSS before ordering since this information changes frequently.

Symbols Used in Figures: Modular jacks are shown by the triangle symbol. The 25-pair connectors are indicated by shaded blocks. Generally, only one leg of an octopus cable is shown. Unterminated wiring requiring cut down or other termination do not have symbol designations. The 103A Connecting Block is a typical modular wall jack that provides cut-down connections for building (station) wiring.

Voice Terminal and Adjunct Connections

Figures 4-21 through Figure 4-27 provide connection information for single-line and multiline voice terminals. The single-line terminals can be located on-premises, off-premises, or out-of-building. The 7300H series multiline voice terminals can be used for out-of-building service but must be within 2000 feet of the system cabinets (local power is required beyond 1000 feet). Off-premises service is not available.

Diagrams for voice terminal adjunct connections are integrated with the specific feature descriptions in Section 2.

Attendant Console Connections

Diagrams for attendant console connections are included in the “Attendant Console” descriptions in Section 2.

Peripheral Equipment Connections

Diagrams for peripheral equipment connections are integrated with the specific feature descriptions in Section 2.

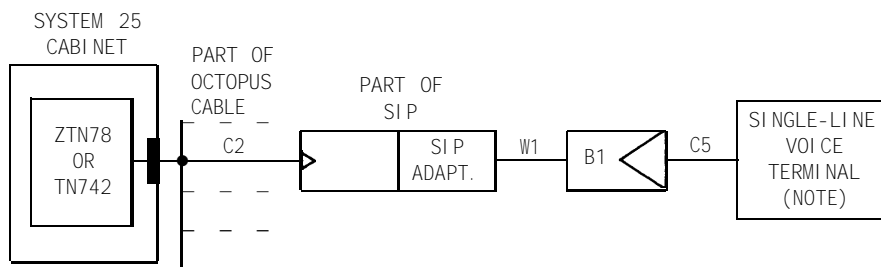
ADU Connections

Figures 4-28 and 4-29 provide connection information for data terminals and associated single-line or multiline voice terminals. The voice terminal and data terminal leads are separated at the SIP with a Y-adaptor and are connected to their respective station ports. Figure 4-30 presents a typical Multiple Asynchronous Data Unit (MADU) connection.

Figure 4-31 shows local power connections for Z3A1, Z3A2, and Z3A4 ADUs.

Auxiliary Equipment Connections

Diagrams for auxiliary equipment connections are integrated with the specific feature descriptions in Section 2.



LEGEND:

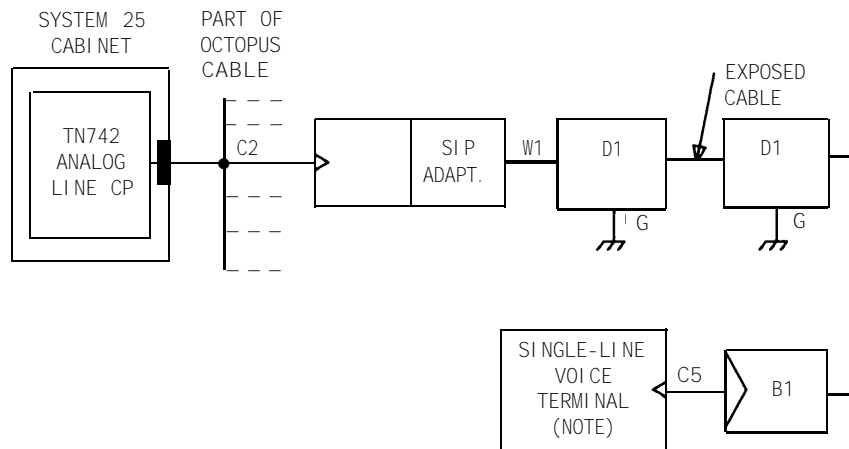
- ZTN78 - TIP RING CP
- TN742 - ANALOG LINE CP
- B1 - TYPICAL-103A CONNECTING BLOCK*
- C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
- C5 - MODULAR CORD (D4BU-87) FURNISHED WITH SET
- W1 - 4 PAIR INSIDE WIRING CABLE*

NOTE : RANGE LESS THAN 2000 FEET FROM SYSTEM CABINET, USE ZTN-78 CP.
 RANGE MORE THAN 2000 FEET BUT LESS THAN 1300 OHMS † (LOOP RESISTANCE)
 FROM SYSTEM CABINET, USE TN742 CP. FIVE SINGLE-LINE VOICE
 TERMINALS CAN BE BRIDGED WHEN USING A TN742 CP, HOWEVER, ONLY TWO
 MAY BE OFF-HOOK AT ONE TIME.

* FURNISHED BY INSTALLER

† - INCLUDES TELEPHONE/TERMINAL

Figure 4-21. On-Premises Single-Line Voice Terminal Connections



LEGEND:

- B1 - TYPICAL - 103A CONNECTING BLOCK
- C2 - OCTOPUS CABLE (WP90780)
- C5 - MODULAR CORD (D4BU-87) - FURNISHED WITH SET
- D1 - STANDARD GAS TUBE/FUSE PROTECTION PER BSP 460-100-400*
- G - APPROVED BUILDING GROUND
- W1 - INSIDE WIRING CABLE

NOTE:

MAXIMUM LOOP RESISTANCE FROM SYSTEM CABINET <1300 OHMS †
 FIVE SINGLE-LINE VOICE TERMINALS CAN BE BRIDGED; ONLY
 TWO MAY BE OFF-HOOK AT ONE TIME.

- * - FURNISHED BY INSTALLER
- † - INCLUDES VOICE TERMINAL

Figure 4-22. Out-Of-Building Single-Line Voice Terminal Connections

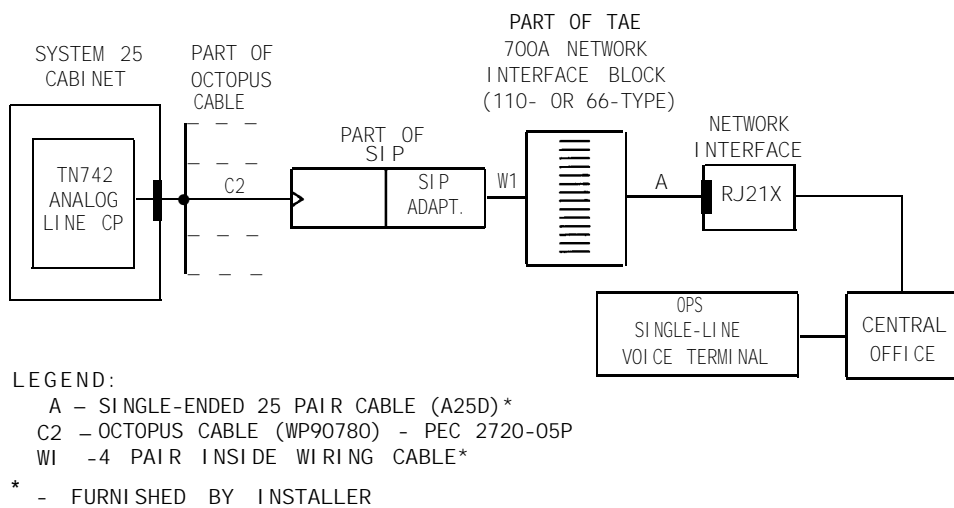
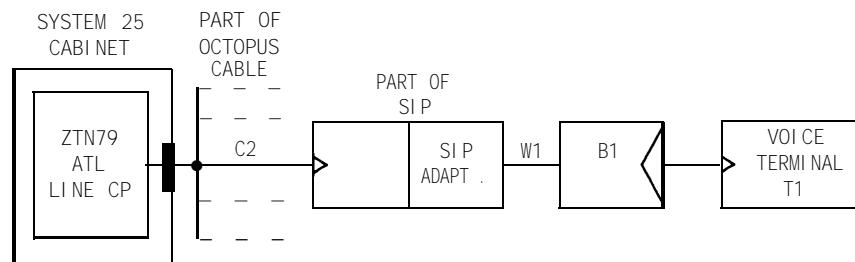


Figure 4-23. Off-Premises Station Single-Line Voice Terminal Connections



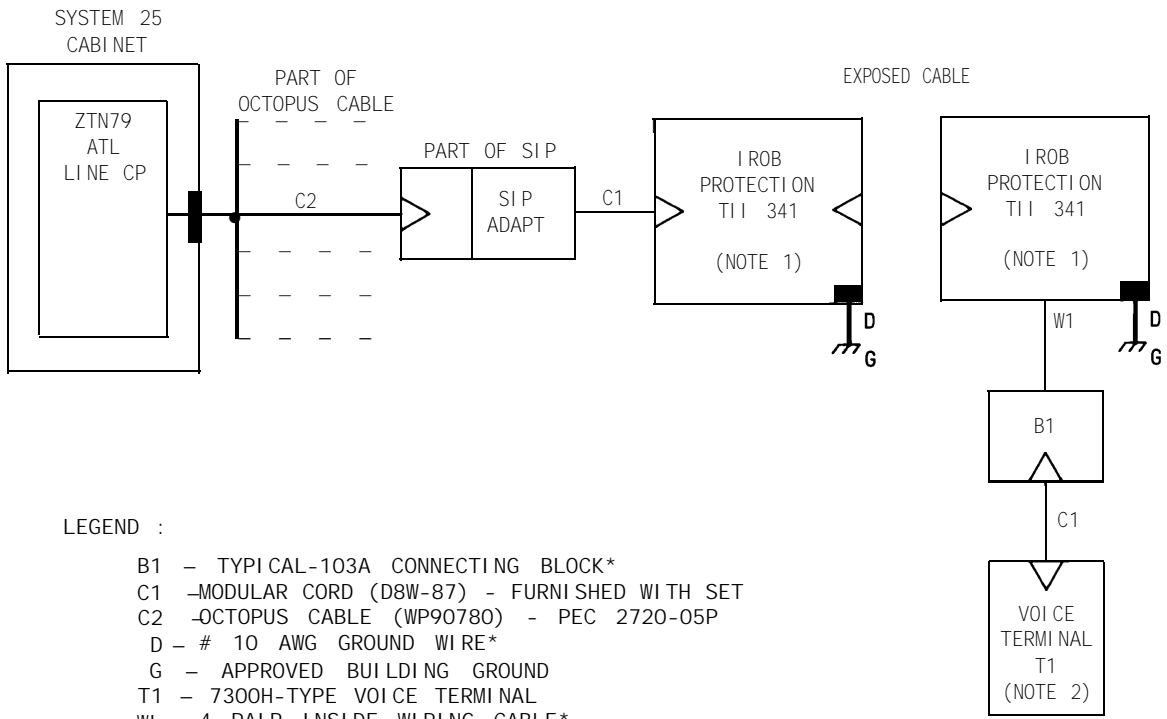
LEGEND :

- E1 - TYPICAL-103A CONNECTING BLOCK*
- C1 - MODULAR CORD (D8W-87) - FURNISHED WITH SET
- C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
- T1 - 7300H TYPE VOICE TERMINAL
- W1 - 4 PAIR INSIDE WIRING CABLE*

* - FURNISHED BY INSTALLER

NOTE: RANGE WITHIN 2000 FEET OF SYSTEM CABINET (LOCAL POWER REQUIRED >1000 FEET).

Figure 4-24. On-Premises 7300H Series Multiline Voice Terminal Connections



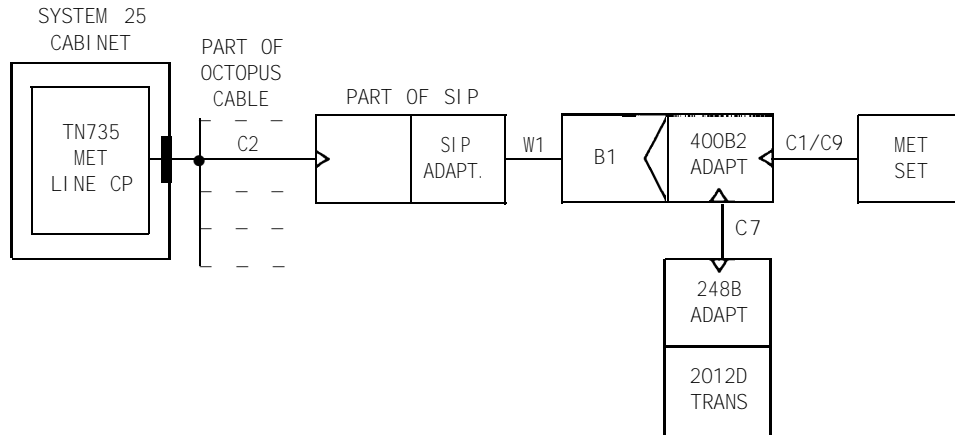
LEGEND :

- B1 - TYPICAL-103A CONNECTING BLOCK*
- C1 -MODULAR CORD (D8W-87) - FURNISHED WITH SET
- C2 -OCTOPUS CABLE (WP90780) - PEC 2720-05P
- D - # 10 AWG GROUND WIRE*
- G - APPROVED BUILDING GROUND
- T1 - 7300H-TYPE VOICE TERMINAL
- W1 - 4 PAIR INSIDE WIRING CABLE*

NOTES :

1. IN-RANGE-OUT-OF-BLDG UNIT MODEL 341 IROB'S REQUIRED - PRIMARY/SECONDARY PROTECTORS.
 2. RANGE: WITHIN 2000 FEET OF SYSTEM CABINET (LOCAL POWER REQUIRED >1000 FEET).
- * FURNISHED BY INSTALLER

Figure 4-25. Out-Of-Building 7300H Series Multiline Voice Terminal Connections



LEGEND :

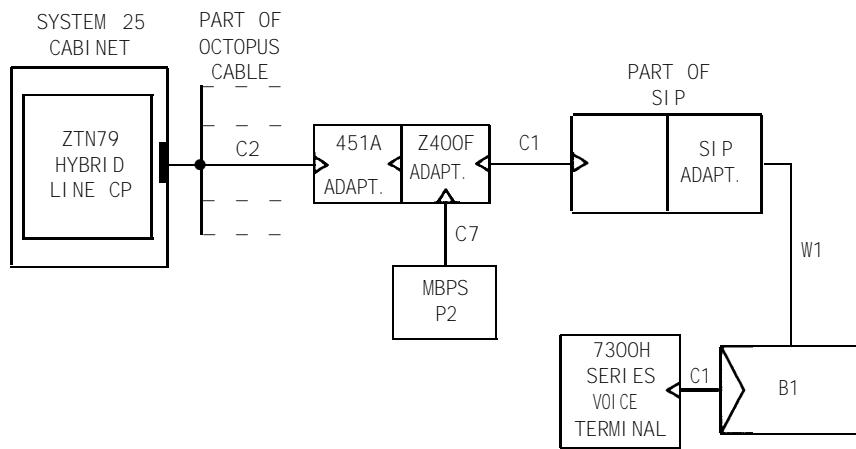
- B1 - TYPICAL-103A CONNECTING BLOCK*
- C1 - MODULAR CORD (D8W-87) - FURNISHED WITH SET
- C2 - OCTOPUS CABLE (WP90780) - PEC 2720-05P
- C9 - MODULAR CORD -(ZD8AJ-87) -COMES WITH DOSS ORDER
- MET SETS - 2991C05, 2993C04 AND 2991D05 TEN BUTTON MET SETS,
USE ZD8AJ-87 ADAPTER CORD (C9) AND C1
- MET SETS - 7203M SET - 12-BUTTON MET SET - USES C1 ONLY
- W1 - INSIDE WIRING CABLE*
- 248B ADAPTER - MODULARIZES 2012D TRANSFORMER
- 400B2 ADAPTER - POWER ADAPTER
- 2012D TRANSFORMER - 15-18V AC TRANSFORMER
- C7 -MODULAR CORD (D6AP-87)

PEC 21691

NOTE: ONLY MET SET WITH BUILT-IN SPEAKERPHONE (2993C04) REQUIRES TRANSFORMER AND ADAPTERS. OTHERWISE, PLUG C1 INTO B1 DIRECTLY.

* FURNISHED BY INSTALLER

Figure 4-26. Ten Button MET Set Connections



LEGEND:

- B 1 - TYPICAL - 103A CONNECTING BLOCK
- C 1 - MODULAR CORD (D8W-87)
- C 2 - OCTOPUS CABLE (WP90780)
- C 7 - MODULAR CORD (D6AP-87)
- P 2 - MODULAR BULK POWER SUPPLY
POWER UNIT (346A) - PEC 31760
POWER PANEL (346A-1) - PEC 31761
- W 1 - INSIDE WIRING CABLE

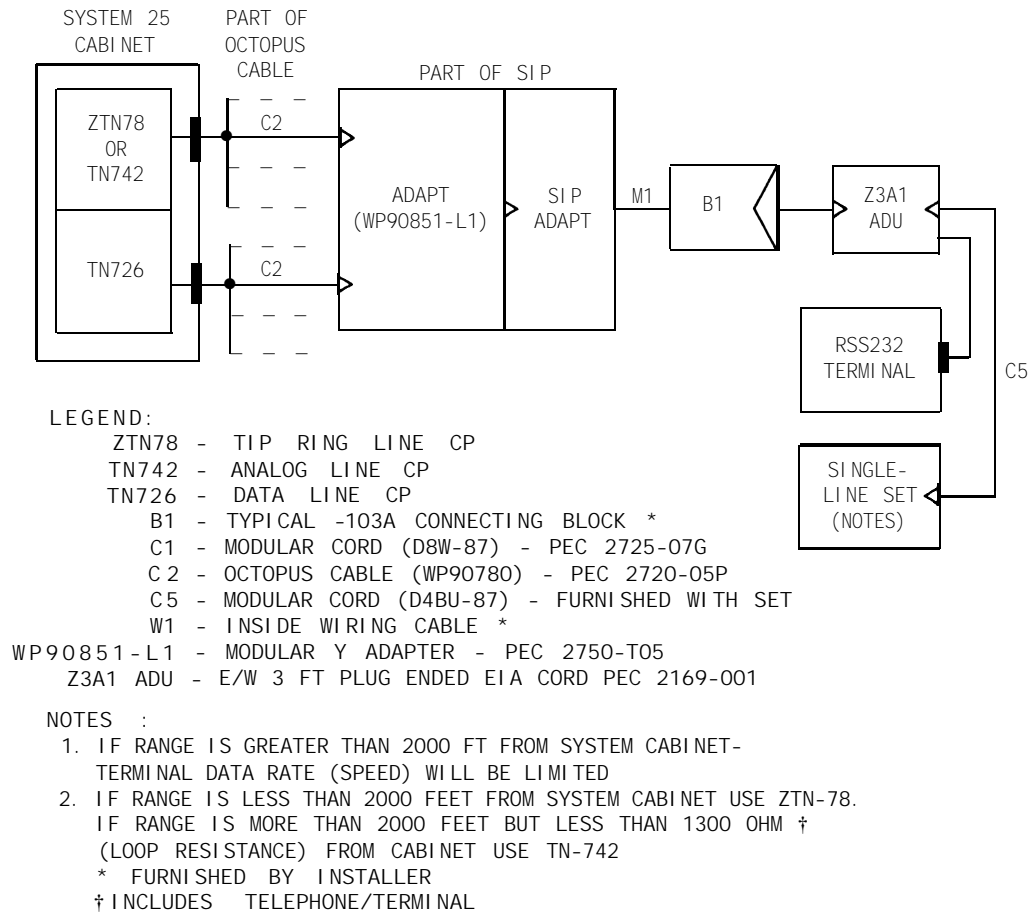


Figure 4-28. Typical ADU Connections—Supporting Data Terminal And Single-Line Voice Terminal

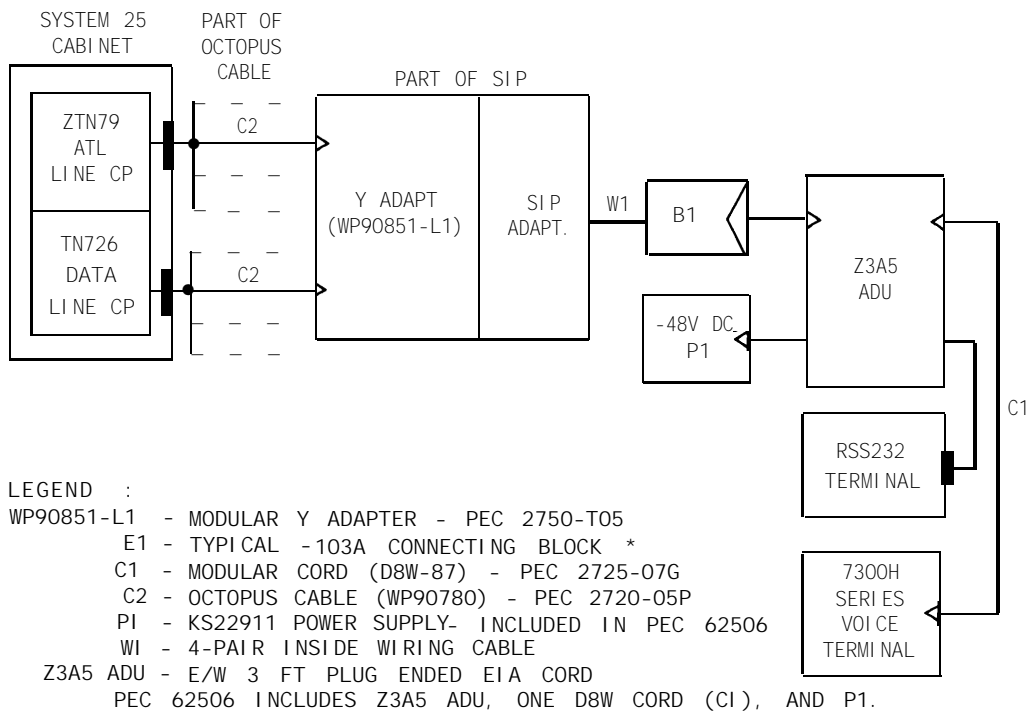
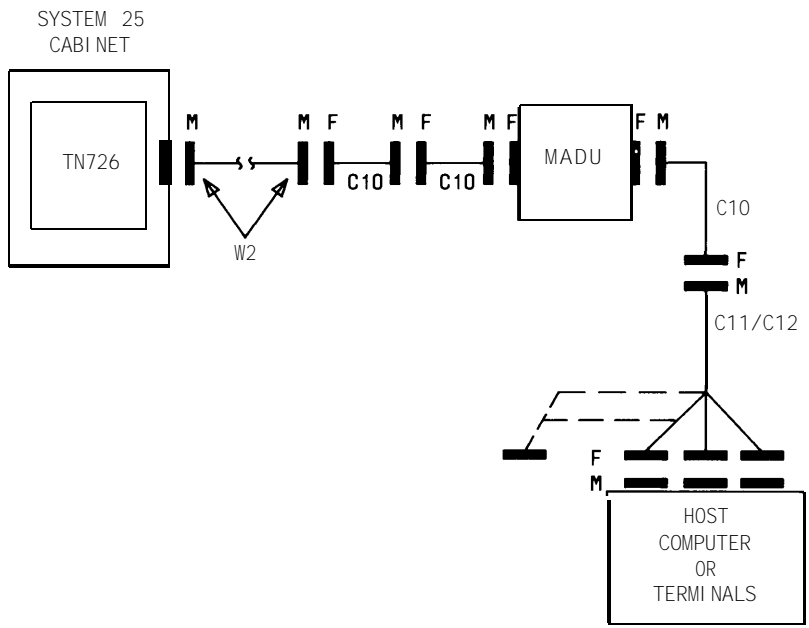
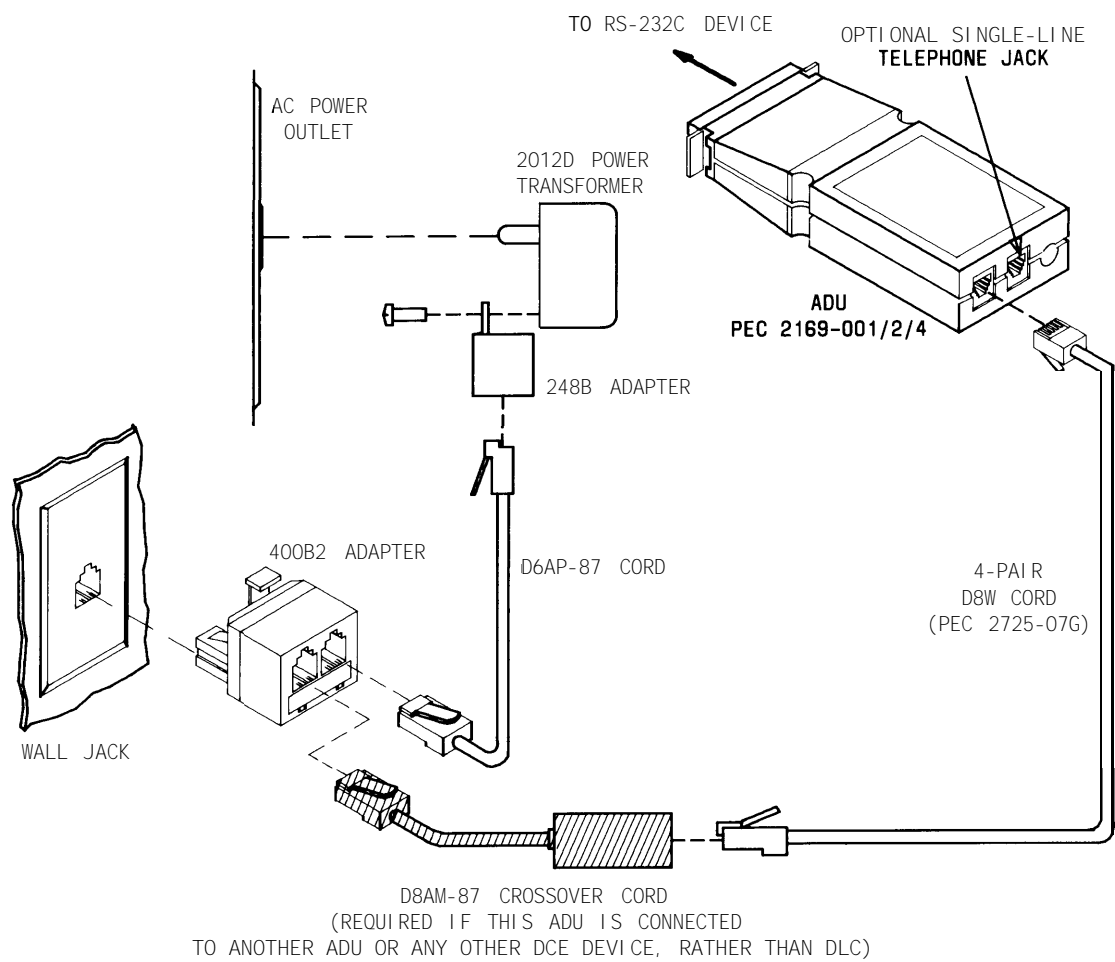


Figure 4-29. Typical ADU Connections—Supporting Data Terminal And 7300H Series Multilane Voice Terminal



- LEGEND :
- TN726 – DATA LINE CP
 - MADU – MULTIPLE ASYNCHRONOUS DATA UNIT (PEC 2169-005)
SINGLE UNIT ASSEMBLY (8 ADU PORTS)
 - W2 – BUILDING WIRING (25-PAIR CABLE)
 - C10 – 25-PAIR CENTERFEED-TO-ENDFEED CABLE (PEC 2724-78B)
(ALWAYS REQUIRED)
 - C11 – M48C OCTOPUS CABLE (PEC 2724-29G) (7-FOOT CORD WITH
EIGHT 6-INCH ARMS FOR "DTE" HOST INTERFACE
 - C12 – M48G OCTOPUS CABLE (PEC 2724-98G) (7-FOOT CORD WITH
EIGHT 6-INCH ARMS FOR "DCE" HOST INTERFACE

Figure 4-30. Typical MADU Connections



NOTE :
**PEC 21691 includes 2012D transformer, 248B and 400B2
 adapters and D6AP cord.**

Figure 4-31. Z3A1/2/4 ADU Local Power Connections

SECTION 5—TECHNICAL SPECIFICATIONS

This section provides information on the technical characteristics and capacities of the system. Some items covered here are discussed elsewhere in the manual but are repeated here for ease of reference.

Technical specifications are provided for:

- Hardware and Software Parameters
- Unit Loads
- Cable Distance Limitations
- Call Progress Tones
- Indicator Lamp Signals
- Port Specifications
- Recommended Network Facilities (Trunks)
- Analog Transmission Characteristics.

Hardware And Software Parameters

The following is a listing of maximums for hardware and software parameters.

ITEM	TOTAL
Attendant Consoles	
Direct Trunk Termination	2
Direct Extension Selector	2
Automatic Route Selection (ARS):	
Patterns	8
Subpatterns Per Pattern	2
Routes Per Subpattern	3
Facility Restriction Levels	4
Cabinets	3
Circuit Packs:	
Common Control	3
Circuit Pack Slots Per Cabinet	12
Call Coverage	
Individual Covered Stations (VI)	31
Individual Covered Stations (V2)	No limit
Call Coverage Groups	32
Receivers per covered station or group	8
Call Pickup	
Groups	16
Members per group	16
Conference Members	5
Dial Codes	600
Personal Dial Codes (PDCs)	200
Floating Personal Dial Codes (FPDCs)	300
Data Dial Codes (DDCs)	104
Virtual Facilities	10

Hardware And Software Parameters (Contd)

ITEM	TOTAL
Direct Group Calling (DGC)	
Groups	32
Members per group	10
Emergency Transfer Units (ETUs)	4
Voice Terminals per ETU	5
Modem Pool Circuit Packs per cabinet	2
Conversion Resources per circuit pack	2
Paging Zones	3
Parked Calls (System)	24
Per Voice Terminal	1
Attendant DXS Console	8
Trunk Groups	16
System Delay Announcements:	
Direct Group Calling Delay Announcement	1
Directed Night Service Delay Announcement	2
Account Code Digits SMDR or Call Accounting System (Models 100, 200, 300, or 500)	15
Speed Dialing Numbers	
System Speed Dialing Numbers (#100-#189)	90
Personal Speed Dialing Numbers (#20-#26) per station	7
Repertory Dialing plus Speed Dialing Digits	5000
System Administration Terminal	1
Toll Call Allowed Lists	4
Total Entries (all lists)	64

Hardware And Software Parameters (Contd)

ITEM	TOTAL
Traffic Data <ul style="list-style-type: none"> ● Simultaneous 2-Party Conversations ● Call Capacity <ul style="list-style-type: none"> - CCS/Hour - Busy Hour Call Capacity Reliability <ul style="list-style-type: none"> ● Mean Time Between Outages (MTBO) Power Consumption <ul style="list-style-type: none"> ● Per Cabinet, Maximum ● Thermal Dissipation 	115 4140 2500 4 Years 500 Watts 1700 BTU/Hour
Total Ports - also includes trunk and station ports [Software Limits; hardware maximum = 36 CPs] System Resources <ul style="list-style-type: none"> ● One Service Circuit CP (includes 4 TT Receivers) (8 ports allocated/CP) ● TWO Touch-Tone Receiver CPs (4 TT Receivers/CP) (8 ports allocated/CP) ● Pooled Modem (Max 6 CPs, 4 ports allocated/CP) (two modems per CP) ● STARLAN Interface CP (V2 only) (4 ports per CP) 	256 8 16 24 4
Trunks <ul style="list-style-type: none"> ● Trunk Ports <ul style="list-style-type: none"> - Tie Trunks ● Auxiliary Trunk Ports <ul style="list-style-type: none"> - Paging Access - Dictation Access 	104 32 3 8
Station Ports <ul style="list-style-type: none"> ● Data Ports ● Voice Ports <ul style="list-style-type: none"> - Single-Line Voice Terminals (200) - Multiline Voice Terminals (V2-111; V1-96) <ul style="list-style-type: none"> ● Attendant Consoles (2) ● Selector Consoles (2) ● 34-Button Sets (non-att) (V2-53; V1-38) ● Non 34-Button Sets (V2-109; V1-94) 	240 104 200

Unit Loads

A cabinet can supply no more than 80 unit loads of 48 volt power (a unit load is defined as 44 mA). Unit loading is determined by the terminal connected to the port circuits. The following table lists unit loads for various terminals.

UNIT LOADS (Note)		
EQUIPMENT	CIRCUIT PACK	UNIT LOAD PER PORT
2500 Voice Terminals 7101A Voice Terminal	ZTN78	0.5
	TN742	1.0
MET Sets	TN735	1.0
5-Btn. (7302H01D) 10-Btn. (7303 H01D) 34-Btn. (7305 H01D) HFAI (7309 H01B)	ZTN79	1.0 1.0 1.0 1.0
BIS (7305 H03B) BIS w/display (7305H04C)	ZTN79 ZTN79	1.50 2.00
34-Btn. Deluxe (7305 H02D)	ZTN79	2.0
S102A Speakerphone	ZTN79	1.0
502A Headset Adapter	ZTN79	0.75
Asynchronous Data Units	TN726	0.0
Tie Trunks	TN760B	2.0
DID Trunks	TN753	0.5
Emergency Transfer Unit	ZTN82/128	2 per ETU
Pooled Modem	TN758	Zero Unit Loads but maximum of two TN758s/cabinet allowed.
DXS Selector Console	Locally Powered	

Note: Equipment not listed above (i.e., TN763, ZTN76, ZTN77) does not affect unit loading. Any voice terminal/adjunct combination requiring more than 2 Unit Loads must be locally powered. When a voice terminal is locally powered, it places no unit load on the cabinet. Specifically, a 34-Button Deluxe voice terminal equipped for speakerphone operation requires auxiliary power. In addition, any 34-Button Deluxe, BIS, or BIS with Display voice terminal equipped for headset operation requires auxiliary power.

Cable Distance Limitations

The following specifications provide allowable cabling distances for the following devices:

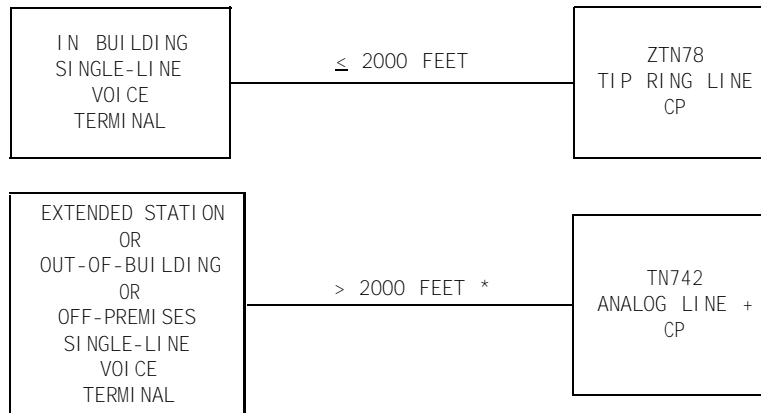
- Single-Line Voice Terminals
- Multiline Voice Terminals
- Data Terminals (RS-232) Connected to Asynchronous Data Units (ADUs).

Single-Line Voice Terminals

SUPPORTING CIRCUIT PACK	24-GAUGE WIRE (0.5106 mm)	
	FEET	METERS
ZTN78	2,000	610
TN742 *	24,000	7,320
TN742 †	15,000	4,500

* 2500-type voice terminals

† 7101-type voice terminals



* UP TO 24,000 FEET. TIP/RING LOOP RESISTANCE FROM SYSTEM CABINETS (INCLUDING VOICE TERMINAL) MUST NOT EXCEED 1300 OHMS.

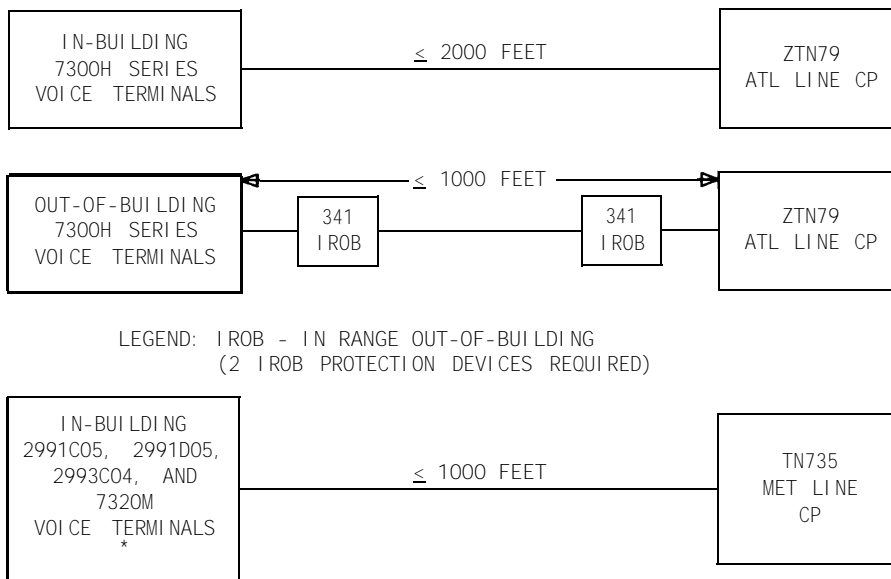
+ FIVE SINGLE-LINE VOICE TERMINALS WHEN USING THE TN-742, HOWEVER, ONLY TWO MAY BE OFF-HOOK AT ONE TIME.

Figure 5-1. Single-Line Voice Terminal Allowable Cable Distances

Multiline Voice Terminals

SUPPORTING CIRCUIT PACK	24-GAUGE WIRE (0.5106mm)	
	FEET	METERS
ZTN79 (7300H Series— in-building or out-of-building, no off-premises)	2,000 (Note)	610
TN735 (in-building MET Sets only)	1,000	305

Note: Requires local power beyond 1,000 feet.



* MODEL 2993C04 (MET EQUIPPED WITH BIS) REQUIRES
A 2012D POWER UNIT (15 - 18V AC)

Figure 5-2. Multilane Voice Terminal Allowable Cable Distances

Data Terminals (RS-232) Connected To Asynchronous Data Units

DATA RATE	24-GAUGE WIRE (0.5106 mm)	
	FEET	METERS
● 300 bps	40,000	12,200
● 1,200 bps	20,000	6,096
● 2,000 bps	12,000	3,657
● 4,800 bps	7,000	2,133
● 9600 bps	5,000	1,524
● 19,200 bps	2,000	610

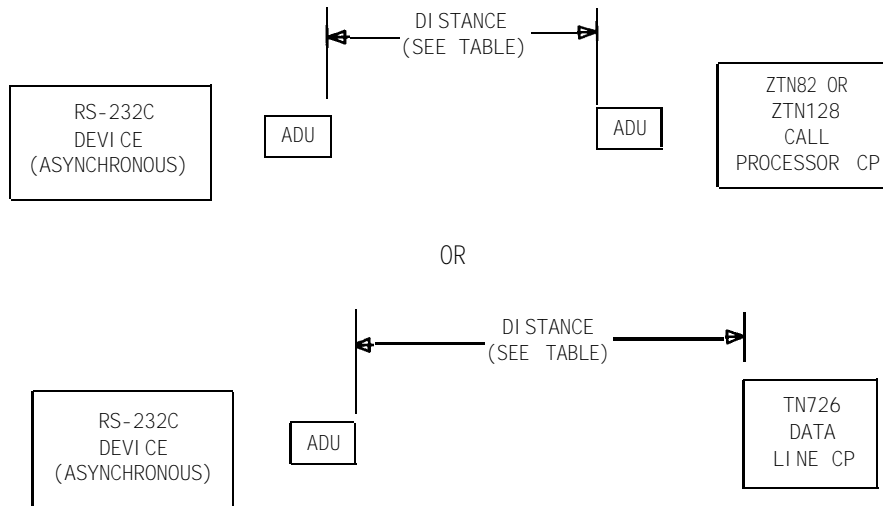


Figure 5-3. Asynchronous Data Unit Allowable Cable Distances, In-Building Only

Tones

The following call progress tones are generated by the system:

TONE	FREQUENCY	PATTERN (In Milliseconds)
Busy Tone	480 Hz + 620 Hz	500 on, 500 off; repeated
Confirmation Tone	350 HZ + 440 HZ	100 on, 100 off, 100 on, 100 off, 100 on followed by silence
Dial Tone	350 HZ + 440 HZ	Continuous
Reorder Tone	480 Hz + 620 Hz	250 on, 250 off; repeated
Ringback Tone	440 Hz + 480 Hz	1200 on, 4000 off; repeated
Call Waiting	440 Hz	200 on; not repeated
Recall Dial Tone	350 Hz + 440 Hz	100 on, 100 off, 100 on, 100 off, 100 on, 100 off, followed by continuous tone
Preferred Route (Queuing) Tone (ARS)	440 Hz	Five 50 ms tones, 50 ms apart, not repeated

Indicator Lamp Signals

The following lamp signals are provided at multiline voice terminal line appearances:

LAMP SIGNAL	PATTERN (In Milliseconds)	MEANING
Dark	Off	Inactive
Lighted	On	
Flashing	500 on, 500 off; repeated	Ringing
Broken Fluttering	50 on, 50 off; repeated, (gated on/off every 500 ms)	Transfer/Conference in progress
Wink	350 on, 50 off; repeated	Hold

Port Specifications

The following tables provides interface specifications for System 25 line and trunk port circuits: supported by System 25:

DATA TERMINAL PORTS (Note)		
STATION TYPE	CIRCUIT PACK	DATA TERMINAL SPECIFICATIONS
EIA RS-232 Device Via ADU	Data Line (TN726)	RS-232 device must furnish signals on ADU pins 2 (TD) and 20 (DTR) and ground on either pin 1 or 7. The ADU furnishes signals on pins 3 (RD) and 8 (CD). The CD signal is also tied to pins 5 (CTS) and 6 (DSR). Some data terminals may require auxiliary power when used with a Z13A1, Z3A2, or Z3A4 ADU (the Z3A5 always requires local power.) ADUs require 7 volts on pin 20 (DTR) to operate properly. The following table lists data terminals that have been tested and are known to operate properly without auxiliary power.

Note: Refer to Cable Distance Limitations for supported data rates.

Port Specifications (Contd)

DATA TERMINALS (Do Not Require Local Power)		
DATA TERMINAL	REQUIRES Z3A1 or Z3A2 ADU	REQUIRES Z3A4 ADU
AT&T 4410 4415 510A	X X X	
ADMs 3A 31	X X	
ADDS Viewpoint *	X	
ConCept HDS 108		X
Datamedia Elite 1521	X	
Hazeltine 1510	X	
Hewlett Packard 2621A 2623A 2640 2645 2645A		X X X X X
Teletype BLIT/1 (68000 based) 5620 (MAC-80 based) 5420	X X X	

* Requires Originate/Disconnect Switch.

Port Specifications (Contd)

VOICE TERMINAL PORTS		
STATION TYPE	CIRCUIT PACK	SPECIFICATIONS
<p>Tip and Ring Single-Line Sets (Analog)</p>	<p>Tip Ring Line (ZTN78)</p>	<ul style="list-style-type: none"> ● 1-Pair Interface (Tip and Ring) ● Analog signals modulated over DC loop ● Loop Voltage: 24 V dc ● Signaling: Dual Tone Multifrequency (DTMF) only ● REN (max.): 1.2 ● DC Current (max.): 35 mA ● Loop Range (24 AWG): 2,000 feet ● In-building service only
<p>Tip and Ring Single-Line Sets (Analog)</p>	<p>Analog Line (TN742)</p>	<ul style="list-style-type: none"> ● I-Pair Interface (Tip and Ring) ● Analog signals modulated over DC loop ● Loop Voltage: 48 V dc ● Signaling: Dual Tone Multifrequency (DTMF) only ● REN (max.): 5.0 ● DC Current (max.): 40 mA ● Loop Range (24 AWG): 24,000 feet ● Supports Out-of-Building, Extended (greater than 2000 feet), Off-Premises and Bridged Station services (maximum of five bridged stations, and two off-hook simultaneously).

Port Specifications (Contd)

VOICE TERMINAL PORTS (Contd)		
STATION TYPE	CIRCUIT PACK	SPECIFICATIONS
MET Sets (Hybrid)	MET Line (TN735)	<ul style="list-style-type: none"> ● 3-Pair Interface 1-Voice pair 2-Control pairs ● Analog Voice, Digital Control/Signaling ● Power: Phantom Power Over Data Pairs ● Bipolar Signaling With 0 V de Offset ● 1 MHz Nominal Signaling Rate ● Loop Range: 1000 feet (In-Building service only)
MERLIN® Sets (7300H Series) (Hybrid)	ATL Line (ZTN79)	<ul style="list-style-type: none"> ● 3-Pair Interface 1-Voice pair 1-Control pair 1-Power pair ● Analog Voice, Digital Control/Signaling ● Bipolar non-return to zero line-coding ● 40 kHz Nominal signaling rate ● Loop Range: 1000 feet ● In-Building and In-Range Out-of-Building (IROB) services only

Port Specifications (Contd)

TRUNK PORTS		
TRUNK TYPE	CIRCUIT PACK	SPECIFICATIONS
Auxiliary Trunk	TN763	<ul style="list-style-type: none"> ● Capacity: 4 Circuits ● 3-pair interface: Voice (T,R), Signaling (S, S1), Status (SZ, SZ1)
Direct Inward Dialing (DID) Trunk	TN753	<ul style="list-style-type: none"> ● Capacity: 8 Circuits ● 2-Wire (600 Ohm Fixed Impedance) Transmission ● Signaling: Wink Start, Delay Dial, or Immediate Dial. Accepts Dial Pulse Signals ● Incoming Service Only
Tie Trunk	TN760B	<ul style="list-style-type: none"> ● Capacity: 4 Circuits ● Supports Type I E&M, Type I Compatible E&M or Type V Simplex Signaling ● 4-Wire Transmission ● 3-Pair Interface Transmit Receive Signaling (E&M)

Port Specifications (Contd)

TRUNK PORTS		
TRUNK TYPE	CIRCUIT PACK	SPECIFICATIONS
Ground Start Trunk	ZTN76	<ul style="list-style-type: none">● Capacity: 8 Circuits● 2-Wire (600 Ohms or RC Balance) Transmission● Network Signaling: Ground Start● Two-way or Incoming only Service
Loop Start Trunk	ZTN77	<ul style="list-style-type: none">● Capacity: 8 Circuits● 2-Wire (600 Ohms or RC Balance) Transmission● Network Signaling: Loop Start● Two-way or Incoming only Service

Recommended Central Office Trunk Facilities

The following table provides recommendations for CO trunks based on the number of voice terminals in the system and the calling traffic.

VOICE TERMINALS	CALLING TRAFFIC		
	LIGHT TRAFFIC	MEDIUM TRAFFIC	HEAVY TRAFFIC
20	3/3	4/4	5/5
25	3/4	5/5	6/6
30	4/4	6/5	8/7
40	4/4	6/6	9/8
50	5/4	7/6	10/8
60	5/4	8/7	11/9
70	5/5	8/7	12/10
80	6/5	9/7	12/10
100	6/5	10/8	14/12
120	7/6	11/9	16/13
140	7/6	12/10	17/14
160	8/7	13/10	19/15
180	8/7	13/11	20/16
200	9/8	14/12	22/18

Notes:

1. For systems with both one-way and two-way trunks, the first number listed under "Calling Traffic" is the number of two-way trunks required, the second number is the number of one-way trunks required.

2. For systems with just two-way trunks, add the two numbers listed under Calling Traffic to determine the number of trunks required.

Analog Transmission Characteristics

Frequency Response:

(Station-To-Station or Station-To-CO-Trunk, relative to loss at 1 kHz)

FREQUENCY	LOSS
60 Hz	>20 dB
200 Hz	<5 dB
300-3000 Hz	<1 dB
3200 Hz	<1.5 dB
3400 Hz	<3 dB

Insertion Loss:

CONNECTION TYPE	LOSS
Standard Station to Standard Station	6 dB
Standard Station to Extended/Off-Premises Station	3 dB
Extended/Off-Prem Station to Extended/Off-Prem Station	0 dB
Station-to-Trunk	0 dB
Trunk-to-Trunk	0 dB

Overload Level: +3 dBm0

Crosstalk: < -70 dB

Intermodulation Distortion:

FOUR TONE METHOD	
2nd Order Tone Products	>45 dB
3rd Order Tone Products	>53 dB

Quantization Distortion:

SIGNAL LEVEL	DISTORTION LEVEL
+2 to -30 dBm0	35 dB
-40 dBm0	29 dB
-45 dBm0	25 dB

Analog Transmission Characteristics (Contd)

Sampling Rate:	8 kHz
Terminating Impedance:	600 ohms
Trunk Balance Impedance:	600 ohms or Complex Z (selectable)

Echo Return Loss:

The echo return loss of the switching equipment is infinite. The echo return loss of the station equipment can be engineered for greater than 18 db over the range of 500 Hz to 2500 HZ.

Loop Resistance:

- TN74.2—Loop resistance of up to 1300 ohms, including the station
- ZTN78—Loop resistance of up to 100 ohms **not** including the station (2000 feet with No. 24 AWG.)

Connection Bandwidth: 64 Kbits

Steady State Noise Level:

The steady state noise level presented to any busy path does not exceed 23 dBrc during the busy hour.

Impulse Noise:

The impulse noise is 0 count (hits) in five minutes at +55 dBrc during the busy hour.

Single Frequency Return Loss (Talking State):

Station to station—exceeds 12 db
Station to 4 wire trunk connection—exceeds 14 db
Station to 2 wire trunk connection—exceeds 12 db

Peak Noise Level:

Analog to analog—20 dBrc
Analog to digital—19 dBrc
Digital to analog—13 dBrc

SECTION 6—ENVIRONMENTAL REQUIREMENTS

This section provides information on floor and wall space requirements for System 25 cabinets and associated peripheral equipment. Also included are specifications for temperature, humidity, air purity, lighting, electrical noise (RFI) suppression, power, grounding and lightning protection.

Floor Plans And Layouts

Floor plan arrangements will vary depending on the available equipment area and anticipated system growth. A typical floor plan is shown in Figure 6-1.

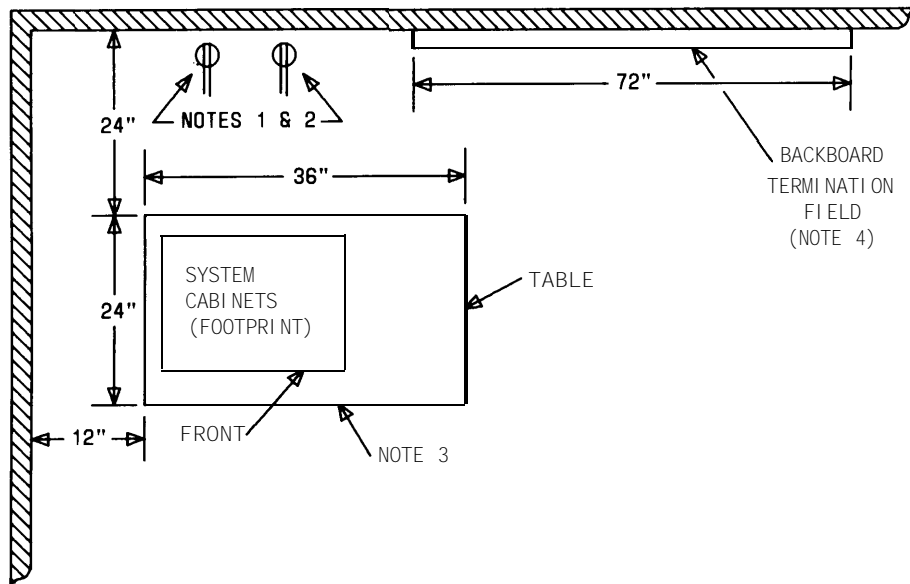
The floor must be tiled or suitably sealed, level, and free from vibration. Allow for a minimum unobstructed clearance of seven feet above the floor throughout the equipment area.

Do not locate the equipment in areas:

- Where it might be subjected to excessive vibrations or disturbed by moving equipment such as hand trucks and transporters.
- Where noise levels may exceed 90 dB.
- Susceptible to flooding.

Maintain clear access to the equipment area for both installation and maintenance purposes.

The wall behind the system cabinet must be clear of all objects (pictures, shelves, or windows) that might interfere with system installation. The entire area behind the cabinet and to the side as shown on Figure 6-2 must be reserved for the cross-connect field and cable access. Also, room for system growth should be considered.



NOTES :

1. 115V AC, 60 Hz, 15 AMP OUTLETS (HUBBELL 5262 OR EQUIVALENT) MUST BE LOCATED WITHIN SIX FEET (1.8 m) OF SYSTEM CABINETS.
2. MULTIPLE CABINET SYSTEMS REQUIRE TWO QUAD OUTLETS, SINGLE CABINET SYSTEMS REQUIRE ONE QUAD OUTLET.
3. ALLOW AT LEAST 24 INCHES OF SPACE IN FRONT OF CABINETS. TABLE MUST BE ABLE TO SUPPORT 250 POUNDS.
4. BACKBOARD IS 3/4 INCHES THICK BY 48 INCHES WIDE BY 72 INCHES LONG

Figure 6-1. Typical System 25 Equipment Area Floor Plan

Table Top Space

The following system equipment requires (customer provided) table top space in the equipment area:

- System Cabinets - Each cabinet is 13 inches high, 17 inches wide, and 21 inches deep. A three cabinet system requires a vertical space of approximately 40 inches and a 17 inch by 21 inch table top space. Each cabinet weighs approximately 75 pounds. Place the cabinets on a desk or table-top that is about 18 inches high and capable of supporting at least 250 pounds. The cabinets *must* not be placed on the floor.
- System Administration Terminal (SAT) Model 703 - The SAT should also be located near the system cabinets and plugged into the same AC outlet. It is 12 inches wide, 10 inches long, and 3 inches high.
- Digital Tape Unit - The Tape Unit (Model DC5 Digital Data Recorder) should also be located near the system cabinets. It is 5 inches wide, 2 inches high, and 10 inches long.
- SMDR or Call Accounting System (CAS) - The AT&T Model 475 printer is approximately 16 inches wide, 12 inches long, and 6 inches high. The CAS runs on the AT&T PC 6300. They should also be located near the system cabinets.

Wall Space Requirements

The customer provided backboard for the cross-connect field shall be 3/4 inch thick, 4 feet high, and 8 feet wide. Mount the board 30 inches above the floor. The board must conform to national and local fire safety codes.

If existing cross-connect hardware is reused, the space requirements and hardware requirements must be shown on the floor plan. Contact your AT&T Technical Consultant for assistance in planning for reuse of existing equipment.

Temperature and Humidity

The System 25 equipment should be installed in a well-ventilated area. The equipment *must* be located in an area with an ambient temperature between 40 degrees and 104 degrees Fahrenheit (5 and 40 degrees Celsius). The relative humidity must be less than 95%, noncondensing. These parameters shall be maintained 24 hours a day, seven days a week.

Air Purity

The cabinet should not be installed in an area where the air may be contaminated with any of the following:

- Excessive dust, lint, carbon particles, paper fiber contaminants, or metallic contaminants
- Contaminants expelled by office copying machines
- Highly corrosive atmosphere within an enclosed area or atmosphere containing vaporized chemical compounds that may condense on the equipment
- Explosive or flammable atmosphere

Lighting

Lighting should be adequate to allow administration and maintenance personnel to perform their tasks. The recommended light intensity level is 50 to 70 footcandles. This level complies with the Occupational Safety and Health Act (OSHA) standards.

Electrical Noise (RFI)

In most cases, electrical noise is introduced to the system through trunk or voice terminal cables. However, electromagnetic fields near the system cabinets may also induce noise in the system. Therefore, the system cabinets 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 volt per meter are unlikely to cause interference. Field strength can be measured by a tunable meter such as the Model R-70 meter manufactured by Electro-Metrics Division or broadband meters such as the HOLADAY* HI-3001 meter or Model EFS-1 meter manufactured by Instruments for Industry, Inc.

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 KHz).

* Trademark of Holaday Industries

AC Power Requirements

- The System 25 power service shall be a dedicated branch circuit with no other equipment served (See Figure 6-3.)
- Each cabinet requires 500 Watts at 115V ac (maximum)
- Provide a load center of appropriate current rating (ITE EQ4 typical) equipped with 120V ac, 15 ampere (AMP), single pole magnetic circuit breaker(s) (ITE QP1-B015 typical). Each breaker is to protect 2 associated wall mounted 115 V ac, 15 AMP, receptacles (HUBBELL 5262 typical). Grounding of this load center is to be provided by a “Green Wire” ground extended from the grounding electrode conductor at the AC service entrance to the load center.

The following materials are required:

A. *Single Cabinet System*

- 1 - 15 AMP 3 Wire Dedicated Branch Service
- 1 - 4“ Box (RACO 230 or Equiv.)
- 1 - 4“ Cover (RACO 807 or Equiv.)
- 1 - Ground Bar (Square D PK9GTA or Equiv.)
- 2 - Recpt. (Hubbell 5262 15 AMP or Equiv.)

B. *Multiple Cabinet System*

- 2 - 15 AMP 3 Wire Dedicated Branch Service
- 2 - 4“ Box (RACO 230 or Equiv.)
- 2 - 4“ Cover (RACO 807 or Equiv.)
- 1 - Ground Bar (Square D PK9GTA or Equiv.)
- 4 - Recpt. (Hubbell 5262 15 AMP or Equiv.)

- C. Typically, multiple cabinet systems can be powered from a single phase 120V ac, 60 Hertz service (two 15 amp circuits required.) There are no phase restrictions between cabinets. Therefore the two 15 amp circuits required may be derived from single or three-phase service.
- D. The receptacles shall be located at least 1 foot above the floor. Receptacles shall not be located further than 4 feet from the cabinets.

DANGER

Under no circumstances should this equipment be connected to 220V ac; doing so poses a serious fire hazard.

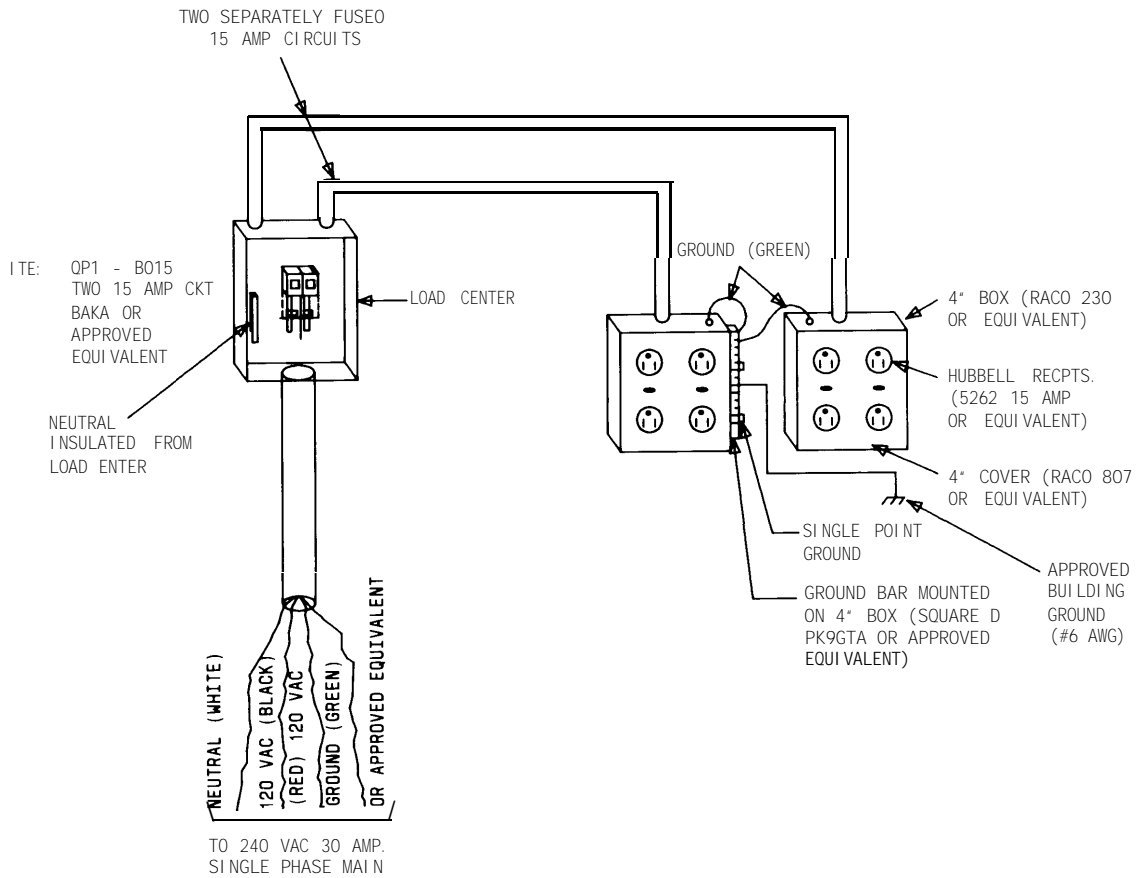


Figure 6-3. AC Power Distribution - Multiple Cabinet System

Grounding

Connection of an approved ground to the system cabinets is essential. An approved ground may consist of any of the following:

- Grounded Building Steel - The metal frame of the building.
- Water Pipe - A continuous metal water pipe, not less than 1/2 inch in diameter, that is connected to an underground metal water pipe that is in direct contact with earth for 10 feet or more.
- Concrete-Encased Ground - An electrode encased by at least 2 inches of concrete and located within and near the bottom of a concrete foundation or footing indirect contact with the earth. The foundation must consist of at least 20 feet of one or more steel reinforcing bars or rods of not less than 1/2 inch in diameter, or at least 20 feet of bare, solid copper wire not smaller than No. 4 gauge.
- Ground Ring -A ring that encircles a building or structure indirect contact with earth at a depth of at least 2-1/2 feet. The ring must consist of at least 20 feet of bare copper conductor not smaller than No. 2 gauge.

Lightning Protection

A Coupled Bonding Conductor (CBC) tie-wrapped to all trunks provides lightning protection. The CBC can be any one of the following:

- A 16 gauge ground wire
- Continuous cable sheath
- Six unused pairs of wire

The CBC should be run from the telephone company provided network interface to the system Cabinet 1's ground block. If the telephone company has not extended the CBC from the facility entrance to the network interface, the System 25 installer should run the CBC along the same route as the incoming facilities, where feasible.

SECTION 7—PARTS INFORMATION

This section contains information that may assist you in cross referencing Apparatus Codes, Comcodes and Price Element Codes (PECs). The first table is grouped by descriptive Price Element Codes (PEC). The second table is arranged by component codes (COMCODE).

Parts Listed by PEC

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
6250-011	System 25 Control Unit (V1) Carrier	includes J58901A1 L1 (e/w)	
Part of:			
6250-011	Power Supply	WP90510	403954761
6250-011	Fan (2)	WP90677 L1	845416379
6250-011	Air Filter	21985-1	403957129
6250-011	Address Plug	-----	845416635
6250-011	TDM Bus Term	J58901A1 L2,4	103810586
6250-011	Memory CP	ZTN81	103965281
6250-011	Memory CP	ZTN81B	103982740
6250-011	Memory CP	ZTN81C	105291488
6250-011	Processor CP	ZTN82	103965299
6250-011	Service Circuit CP	ZTN85	103965323
6250-012	System 25 Control Unit (V2) Carrier	includes J58901A1 L1 (e/w)	
Part of:			
6250-012	Power Supply	WP90510	403954761
6250-012	Fan (2)	WP90677 L1	845416379
6250-012	Air Filter	21985-1	403957129
6250-012	Address Plug	-----	845416635
6250-012	TDM Bus Term	J58901A1 L2,4	103810586
6250-012	Memory CP	ZTN127	105212179
6250-012	Processor CP	ZTN128	105211023
6250-012	Service Circuit CP	ZTN85	103965323
6250-012	SLAC Grap. Layer (2)		845875155
62501	System 25 Expansion Unit Carrier	includes J58901A1 L1 (e/w)	
Part of:			
62501	Power Supply	WP90510	403954761
62501	Fans (2)	WP90677 L1	845416379
62501	Air Filter	21985-1	403957129
62501	Address Plug	-----	845416635
62501	TDM Bus Cable	J58901A1 L3	403961519

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
62502	GS Trunk CP	ZTN76	103965232
62503	LS Trunk CP	ZTN77	103965240
62504	TR Line CP	ZTN78	103965257
62505	ATL Line CP	ZTN79	103965265
62506	Asynchronous Data Unit	D181521 includes	105105506
Part of: 62506		D8W-87 Cord (14 ft)	103786802
62506		Z3A5 ATL ADU	103975349
62507	Digital Tape Unit	TS-555A	404079436
62508	System Admin Terminal (SAT)	TI-703	404079428
62509	Direct Extension Selector Console	23A1-003 e/w	103969424
Part of: 62509		D6AP Cord	102937620
62509		D8W-87 Cord (14 ft)	103786802
62509		KS22911 L1 Power	403242639
62509		400B Adapter	103848859
62510	MERLIN VT local power	D181522 includes	105105514
Part of: 62510		Z400F Adapter	103942857
62510		D6AP-87 Cord	102937620
62510		KS22911 L1 Power	403242639
62511	Auxiliary Trunk Interface	D181523 includes	105105522
Part of: 62511		Block Conn 66E3-25	100009968
62511		Cable B25A 15/DE	100017334
62512	Paging/Dictation (Aux Trunk Interface)	D181524 includes	105105530
Part of: 62512		278A Adapter	103871844
62512		KS22911 L1 Pwr Unit	403242639
62512		D4BY Cord (14 ft)	102999059
62512		D181321 Kit (zener)	105031181

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
62513	MOH Interface	D181575 includes	*
Part of:			
62513		KS23395 Interface	105143186
62513		2012D Transformer	102600517
62513		36A Voice Coupler	103558916
62514	10B ETU	Emergency Trans Unit	103984118
Part of:			
62514		(2) B25A Cables DE	100017334
62514		D8W-87 Cord (7 ft)	103786786
62515	Peripherals Interface for Remote Access		
Part of:			
62515		248B Adapter	102802113
62515		355AF Adapter	105012645
62515		248B Adapter	102802113
62515		400B2 Adapter	104152558
62515		D6AP-87 Cord	102937620
62515		D8AM-87 Cord	104154430
62515		D8W-87 Cord (7 ft)	103786786
62515		M7U-87 Cord	104466616
62515		Z3A4 ADU	103964185
62515		2012D Transformer	102600517
62518	STARLAN Interface CP	ZTN84	103965315
62519	RIV2 Upgrade Kit	D181782 includes	105335657
Part of:			
62519	Memory CP	ZTN127	105212179
62519	Processor CP	ZTN128	105211023
62519	SLAC Grap. Layer (2)		845875155
62519	V2 Documentation		
62520	STARLAN/ATI. Interface	D181807 includes	105355374
Part of:			
62520	Y-Adapter	WP90851,L1	405010620
62520	Adapter	KS-23475	405462904
62520	Power Unit	KS-22911,L1	403242639
63111†	Analog Line CP	TN742	103556957
63112†	MET Line CP	TN735	103556882
63116†	DID Trunk CP	TN753	103557062
63117†	Tie Trunk CP	TN760B	103975645

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
63118†	Aux Trunk CP	TN763	103557161
63119†	Pooled Modem CP	TN758	103557112
63123†	Tone Detector CP	TN748	103976163
63130†	Data Line CP	TN726	103556791
1020-S90	ACCESS Software	*	105341218
1020-S91	CAM Software	*	105341382
1203-020	AA Software	*	105339584
2169-001	Tip Ring ADU	Z3A1	103963963
2169-004	Tip Ring ADU	Z3A4	103964185
21691	ADU Aux Power		
Part of:			
21691		248B	102802103
21691		400B	103848859
21691		D6AP-87	102937620
2301-SAA	Supplemental Alert Adapter	*	*
2610-001	STARLAN NETWORK Network Extension Unit (NEU)		527840003
2614-100	STARLAN NETWORK Network Access Unit (NAU)		527840102
2720-05P	25-pair/8-plug 15-ft CBL	WP90780L1	405010612
2720-05X	Splitter CBL, Tie-Trunk	WP90929,L3	403864150
2720-06X	Splitter CBL, CO Trunk	WP90929,L1	403836620
2724-30C	RS232 X-Over Cable	M7U-87	104246616
2724-38X	Mod(ADU) X-Over Cord	D8AM-87	104154430
2725-07G	Mod Cord	D8W-87 Cord (7 ft)	103786786
2725-075	Cord (25 ft)	D8W-87	103786828
2750-A17	MET Adapter Cord	ZD8AJ	103881421

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
2750-A24	Mod/RS232 Adapt, Male	355A	105012637
2750-A25	Mod/RS232 Adapt, Female	355AF	105012645
2750-T05	Voice/Data (Y) Adapter	WP90851L1	405010620
2781-JDX	System Wiring-Time and Materials	*	*
2782-JA1	(New) System Wiring Run, Firm Price	*	*
2782-JA2	(Reuse) System Wiring Run, Firm Price	*	*
2782-JC1	(New) System Wiring Run, Firm Price (Plenum Wiring)	*	*
2783-JDX	System Wiring-Based On a PCS Quote	*	*
3100-1TD	Basic TT Desk Tel	2500MMGB	*
3100-TWR	Basic TT Wall Tel	2554BM	103234472
3178-SYS	Basic TT Desk Tel w/Message and Recall	2500DMGC	103966255
*	Basic TT Desk Tel with Headset Jack	2514 BMW	*
3100-2TD	Basic TT Desk (4A Spkphone Compatible)	2500SM	*
3100-ORD	Basic Rtry Desk Tel	500MM	103870267
3100-ORW	Basic Rtry Wall Tel	500BMPA	103823555
3100-2RD	Basic Rtry Desk Tel (4A Spkphone Compatible)	500SM	103870416
*	Multifeature TT Desk Tel w/ Speaker, Memory & Redial	CS6402U01A	*
1121-050	AT&T Answer-Record	2500	*
1140-010	10-Button MET VT	2991C05	103871018

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
3141-BIS	MET Set-BIS	2993C04	103942146
Part of:			
3141-BIS	Transformer	2012D	102600517
3141-BIS	Kit of Parts	D181245	*
3143-12M	12-Button MET VT	Z7203M01A-003	103963310
3160-111 (MAC30 Att)	5-Button MERLIN VT	Z7302H01D-003	*
3161-172 (MAC30 Att)	10-Button MERLIN VT	Z7303H01D-003	*
3161-161	MERLIN HFAI VT	Z7309H01B-003	103982005
3162-412	34-Button MERLIN VT	Z7305H01D-003	*
3162-417 (MAC30 Att)	34-But Dlx MERLIN VT	Z7305H02D-003	*
3162-BIS	MERLIN BIS VT	Z7305H03B-003	103981965
3162-DIS	MERLIN VT With Display	Z7305H04C-003	103981981
3163-HFU	MERLIN Speaker Mod	S102A	103814356
3164-HFA	MERLIN Headset Adapter	502A	*
31710	Acoustic Coupler	349A Adapter	104010061
3170-00M	Single-Line VT w/Message Light & Recall Button	7101A01A-003	103871109
31017	Ground Start Key	55A1	100287085
31019	External Alert	*	*
:310:32	Message Adjunct	Z34A	*
	Mod. Bulk Pwr. Supply		
31760	Power Unit	346A	104174768
31761	Power Panel	346A-1	104174750
32918	IROB Unit	Mod 341	*

PEC	DESCRIPTION	APPARATUS CODE	COMCODE
8310-001	AC Power Surge Suppressor TII Model 428		402988950
Miscellaneous			
No PEC	Fanning Strip	50A	105196604
No PEC	Block, Connector	110 Female	103756334
No PEC	Block, Connector	157B Female	403613003
No PEC	Cable Conn (15' DE)	A25D	100963990
No PEC	Block, Connector	103A	103104220
No PEC	Block, Connector	104A	103116943
No PEC	Panel (Part of SIP)	617A	103982658
No PEC	110 to Mod Adpter	858A	405177791
No PEC	Mod to Mod Adpter	Z210A1	103972907
No PEC	CPU/MEM interconnect cable	*	845412956

* Not Available.

† System 75 PECs are listed. System 25 PECs may now be available.
Check before ordering.

Parts Listed by COMCODE

Throughout the following table, "part of" is abbreviated as "P/O."

<u>COMCODE</u>	<u>PEC</u>	<u>DESCRIPTION</u>	<u>APPARATUS CODE</u>
100009968	P/O 62511	Auxiliary Trunk Interface	Block Conn 66E3-25
	P/O 62514	10B ETU	(2) B25A Cables DE
100017334	P/O 62511	Auxiliary Trunk Interface	Cable B25A 15/DE
100287085	31017	Ground Start Key	55A1
100963990	No PEC	Cable Corm (15' DE)	A25D
102600517	P/O 21691	ADU Aux Power	2012D
	P/O 62515	Peripherals Interface	2012D Transformer
	P/O 3141-BIS	MET Set-BIS	2012D Transformer
	P/O 62513	MOH Interface	2012D Transformer
102802103	P/O 21691	ADU Aux Power	248B
102802113	P/O 62515	Peripherals Interface	248B Adapter
102937620	P/O 62509	Direct Extension Selector Console	D6AP Cord
	P/O 62510	MERLIN VT local power	D6AP-87 Cord
	P/o 62515	Peripherals Interface for Remote Access	D6AP-87 Cord
	P/O 21691	ADU Aux Power	D6AP-87
102999059	P/O 62512	Paging/Dictation (Aux Trunk Interface)	D4BY Cord (14ft)
103104220	No PEC	Block, Connector	103A
103116943	No PEC	Block, Connector	104A
103234472	3100-TWR	TT Wall Tel	2554BM
103556791	63130†	Data Line CP	TN726
103556882	63112†	MET Line CP	TN735
103556957	63111†	Analog Line CP	TN742

COMCODE	PEC	DESCRIPTION	APPARATUS CODE
103557062	63116†	DID Trunk CP	TN753
103557112	63119†	Pooled Modem CP	TN758
103557161	63118†	Aux Trunk CP	TN763
103558916	P/O 62513	MOH Interface	36A Voice Coupler
103756334	No PEC	Block, Connector	110 Female
103786786	P/O 62515 P/O 62514 2725-07G	Peripherals Interface 10B ETU	D8W-87 Cord (7 ft)
103786802	P/O 62509 PO 062506 2725-075	Direct Extension Selector Console Asynchronous Data Unit	D8W-87Cord (14 ft) D8W-87Cord (14 ft)
103786828	No PEC	Cord	D8W-87Cord (25 ft)
103810586	P/O 6250-011 P/O 6250-012	TDM Bus Term	J58901A1 L2,4
103814356	3163-HFU	MERLIN Speaker Mod	S102A
103823555	3100-ORW	Rtry Wall Tel	500MPA
103842050	3162-412	34-Button MERLIN VT	Z7305H01B-003
103843538	3162-417	34-Button Dlx MERLIN VT	Z7305H02B-003
103848859	P/O 21691 P/O 62509	ADU Aux Power Direct Extension Selector Console	400B 400B Adapter
103870267	3100-ORD	Rtry Desk Tel	500MM
103870416	3100-2RD	Rtry Desk Tel (4A Spkphone Compatible)	500SM
103871018	3140-010	10-Button MET VT	2991C05
103871109	3170-00M	Single-Line VT w/Message	7101A01A-003
103871844	P/O 52512	Paging/Dictation	278A Adapter

COMCODE	PEC	DESCRIPTION	APPARATUS CODE
103881421	2750-A17	MET Adapter Cord	ZD8AJ
103942146	3141-BIS	MET Set-BIS	2993C04
103942857	P/O 62510	MERLIN VT local power	Z400F Adapter
103963310	3143-12M	12-Button MET VT	Z7203M01A-003
103963963	2169-001	Tip Ring ADU	Z3A1
103964185	2169-004 P/O 62515	Tip Ring ADU Peripherals Interface	Z3A4 Z3A4 ADU
103965232	62502	GS Trunk CP	ZTN76
103965240	62503	LS Trunk CP	ZTN77
103965257	62504	TR Line CP	ZTN78
103965265	62505	ATL Line CP	ZTN79
103965281	P/O 6250-011	Memory CP	ZTN81
103965299	P/O 6250-011	Processor CP	ZTN82
103965315	62518	STARLAN Interface CP	ZTN84
103965323	P/O 6250-011 P/O 6250-012	Service Circuit CP	ZTN85
103966255	3178-SYS	Basic TT Desk Tel w/Message and Recall	2500DMGC
103969424	P/O 62509	Direct Extension Selector Console	23A1-003
103972907	No PEC	Mod to Mod Adpter	Z210A1
103975349	P/O 62506	Asynchronous Data Unit	Z3A5 ATL ADU
103975645	63117†	Tie Trunk CP	TN760B
103976163	63123†	Tone Detector CP	TN748
103981965	3162-BIS	MERLIN BIS VT	Z7305H03B-003

<u>COMCODE</u>	<u>PEC</u>	<u>DESCRIPTION</u>	<u>APPARATUS CODE</u>
103981981	3162-DIS	Z7305H04C-003	
103982005	3161-161	MERLIN HFAI VT	Z7309H01B-003
103982658	No PEC	Panel (Part of SIP)	617A
103982740	P/O 6250-011	Memory CP	ZTN81B
103984118	62514	10B ETU	Emergency Trans Unit
104010061	31710	Acoustic Coupler	349A Adapter
104152558	P/O 62515	Peripherals Interface	400B2 Adapter
104154430	2724-38X P/O 62515	Mod (ADU) X-Over Cord Peripherals Interface	D8AM-87 D8AM-87Cord
104174750	31761	Power Panel	346A-1
104174768	31760	Power Unit	346A
104246616	2724-30C P/O 62515	RS232 X-Over Cable	M7U-87
105012637	2750-A24	Mod/RS232 Adapt, Male	355A
105012645	2750-A25 P/O 62515	Mod/RS232 Adapt, Female	355AF
105031181	P/O 62512	Paging/Dictation	D181321 Kit (zener)
105105506	P/O 62506	Asynchronous	D181521
105105514	P/O 62510	MERLIN VT	D181522
105105522	P/O 62511	Auxiliary Trunk	D181523
105105530	P/O 62512	Paging/Dictation	D181524
105291488	P/O 6250-011	Memory CP	ZTN81C
105196604	No PEC	Fanning Strip	50A
105211023	P/O 6250-011 P/O 6250-012	Processor CP	ZTN128
105212179	P/O 6250-011 P/O 6250-012	Memory CP	ZTN127

COMCODE	PEC	DESCRIPTION	APPARATUS CODE
105335657	62519	RIV2 Upgrade Kit	D181782
105339584	1203-020	AA Software	*
105341218	1020-S90	ACCESS Software	*
105341382	1020-S91	CAM Software	*
105355374	62520	STARLAN/ATL Interface	D181807
402988950	8310-001	AC Power Surge Suppressor	*
403242639	P/O 62509	Direct Extension Selector Console	KS22911 L1 Power
	P/O 62510	MERLIN VT local power	KS22911 L1 Power
	P/O 62512	Paging/Dictation	KS22911 L1 Pwr Unit
	P/O 62520	STARLAN/ATL Interface	KS22911 L1 Power
403613003	No PEC	Block, Connector	157B Female
403836620	2720-06X	Splitter CBL, CO Trunk	WP9092,L1
403864150	2720-05X	Splitter CBL, Tie-Trunk	WP90929,L3
403954761	P/O 6250-011 P/O 6250-012 P/O 62501	Power Supply	WP90510
403957129	P/O 6250-011 P/O 6250-012 P/O 62501	Air Filter	21985-1
403961519	P/O 62501	TDM Bus Cable	J58901A1 L3
404079428	62508	System Admin	TS-458A
404079436	62507	Digital Tape Unit	TS-555A
405010612	2720-05P	25-pair/8-plug 15 ft CBL	WP90780L1
405010620	2750-T05	Voice/Data (Y) Adapter	WP90851L1
405177791	No PEC	ll0 to Mod Adpter	858A
405193186	P/O 62513	MOH Interface	KS23395 Interface

COMCODE	PEC	DESCRIPTION	APPARATUS CODE
405462904	P/O 62520	STARLAN/ATL Interface Adapter	KS-23475,L1
527840003	2610-001	AT&T STARLAN NETWORK Network Extension Unit (NEU)	
527840102	2614-100	AT&T STARLAN NETWORK Network Access Unit (NAU)	
845412956	No PEC	CPU/MEM Interconnect Cable	*
845416379	P/O 6250-011 P/O 6250-012 P/O 62501	Fan (2)	WP90677L1
845416635	P/O 6250-011 P/O 6250-012 P/O 62501	Address Plug	
845875155	P/O 6250-011 P/O 6250-012	SLAC Grap. Layer (2)	

* Not Available.

† System 75 PECs are listed. System 25 PECs may now be available. Check before ordering.

SECTION 8—REFERENCE DOCUMENTATION

System 25 is supported by a complete set of basic and supplementary documentation and optional software. This section provides a brief summary of the available material. Manuals not specified for Release 1 Version 2 (R1V2) cover both R1V1 and R1V2.

Basic Manuals

● Administration Manual	555-500-500
● Administration Manual for R1V2	555-520-500
● Implementation Manual	555-500-662
● Implementation Manual for R1V2	555-520-650
● Installation and Test Manual	555-520-100
● Introduction to AT&T System 25	555-520-021
● Maintenance Manual	555-520-105
● New Capabilities Manual for R1V2	555-520-205
● Reference Manual	555-520-200
● Terminal Operations Manual	555-520-710
● User Guides	
- Data Features User Guide	555-520-704
- Direct Trunk Attendant Console User Guide	555-520-701
- Multiline Terminal User Guide	555-520-703
- Single-Line Terminal User Guide	555-520-702
- Switched Loop Attendant Console User Guide	555-520-706
- Multifeature Single-Line Terminal User Guide	555-520-707

Supplementary Material

● Application Notes Binder	555-509-002
● R1V2 Upgrade Superpac	555-520-013
● AT&T Call Accounting System	
- User's Guide	555-006-201
- Implementation Guide	999-500-247
● AT&T Model 703 SAT Supplement	555-500-720
● Customer Education Leader Guide	555-520-016
● Customer Education Student Guide	555-520-014
● Customer Training Superpac	555-520-011
● System 25—Product Brochure	PM-4410
● System 25—Slim Jim Brochure	PM-4409
● System 25—Switched Loop Attendant Console Sales Literature	PM-4410-F
● System 25—AT&T STARLAN NETWORK Access Feature Sales Literature	PM-4410-H
● System 25—The Integrated Solution Sales Literature	PM-4410-E
● 420 Speakerphone Sales Literature	PM-4410-G

Optional Software

- Advanced Administration Software Package
- Communications Access Manager Software Package
- AT&T STARLAN NETWORK ACCESS Software Package

Descriptions of Basic Manuals

Administration Manual

Provides the information necessary to initialize a system and to perform on-going system administration. Explains the operation of the System Administration Terminal, the Digital Tape Unit, and the commands that allow the System Administrator to make changes and additions.

Implementation Manual

Describes how to plan the operating configuration of the system. Explains how to determine customer needs and how to convert these needs into a system configuration plan. This plan is recorded on accompanying forms that are used in conjunction with the Administration Manual to initialize the system. The Implementation Manual and associated forms are packaged together in the Administration Records Binder.

Installation And Test Manual

Provides step-by-step procedures for installing System 25 and associated equipment. Includes procedures for testing equipment and trunks and for making additions and changes to the system.

Introduction to AT&T System 25

Provides an introduction to System 25 features and services. The emphasis is on how System 25 helps solve information management, productivity, and cost control problems.

Maintenance Manual

Provides a detailed description of system operation and procedures for isolating and clearing customer affecting faults.

New Capabilities Manual

This manual describes the System 25 Release 1 Version 2 (R1V2) features that were not included in Release 1 Version 1 (R1V1). In addition, R1V1 features that have been enhanced for R1V2 and new R1V2 hardware not included in R1V1 are included.

Planning Manual

The document is a presale workbook used by the Account Team and customer to define a set of orderable equipment that will meet the customer's specific requirements. This workbook, when completed, can be used in conjunction with the Quick-Quote Price Estimator or the DOSS Configurator to obtain a price estimate or formal quote or to place an order.

Reference Manual

This document is the principal technical reference of the system. It provides reference material for sales support, system configuration and operation and for the system technician. It contains a comprehensive description of the system, emphasizing features, Components and overall capabilities and capacities.

Terminal Operations Manual

This manual is designed to help the System Administrator better understand System 25 voice terminal and data terminal operation. This manual contains the operating procedures provided in the various User Guides and provides additional explanatory material as well.

User Guides (700 Series)

These guides contain step-by-step operating procedures for System 25 attendants and voice and data terminals users.

SECTION 9—GLOSSARY

This section provides explanations for acronyms and definitions of terms used in this manual.

ADU: (Asynchronous Data Unit)

ARS: (Automatic Route Selection)

ASCII: (American Standard Code for Information Exchange)

Administer

To access and change the parameters associated with the services or features of the system.

Analog Data Endpoint

Data endpoints with customer provided built-in or stand-alone modems. They do not require the use of data modules (asynchronous data units) and are addressed similar to any voice terminal by PDC. These end-points connect to tip/ring type circuit pack ports.

Analog Voice Terminals

Voice terminals served by a single-line tip and ring circuit (2500 series and 7101A voice terminals or industry standard Dual Tone Multifrequency equivalent).

Appearance

See Call Appearance.

Asynchronous Data Transmission

A scheme for transmitting data where each character is preceded by a start bit and followed by a stop bit, thus permitting data elements to occur at irregular intervals. This type transmission is advantageous when transmission is not regular (characters typed at a keyboard).

Asynchronous Data Unit (ADU)

A data communications equipment (DCE) type device that allows direct connection between RS-232C equipment and the digital switch via ports on the Data Line Circuit Pack (TN-726).

Attendant

The operator of the attendant console.

Attendant Console

- **Direct Trunk or Switched Loop Attendant Console:** An electronic call-handling position with pushbutton control. Used by attendants to answer and place calls and to manage and monitor some system operations.
- **Direct Extension Selector Console:** Provides the attendant with a visual indication of the active or idle status of extension numbers assigned in the system. Also allows the attendant to place calls to system users by operation of appropriate Hundreds Group Select buttons and associated Direct Extension Selection (DXS) buttons.

Auxiliary Equipment

- Dictation Equipment
- Delay Announcement Devices
- External Alerting Devices (external alerts)
- Music-On-Hold Equipment
- Paging Equipment

Auxiliary Trunk

A trunk circuit used to connect auxiliary equipment to the switch, for example, music or dictation equipment.

BLF: (Busy Lamp Field)

BPS: (Bits Per Second)

Bit (Binary Digit)

One unit of information in binary notation (having two possible states or values, zero or one).

Bridge (Bridging)

The sharing of the same extension by two or more voice terminals.

Buffer

A circuit or component that isolates one electrical circuit from another. Typically, a buffer holds data from one circuit or process until another circuit or process is ready to accept the data.

Bus

A multi-conductor electrical path used to transfer information over a common connection from any of several sources to any of several destinations.

Bus, Time Division Multiplex

See Time Division Multiplex Bus.

CCS (Hundred Call Seconds)

A traffic-measuring unit that expresses the load of one or more traffic-handling devices. A device used for 1 hour without interruption generates 36 CCS which equals 1 erlang (see Erlang).

Call Appearance, Attendant Console

Two buttons, labeled System Access, used to originate, receive, and hold calls. Each button has two associated LEDs to show the status of the call appearance.

Call Appearance, Voice Terminal

A button labeled with an extension number used to place outgoing calls, receive incoming calls, or hold calls. Two LEDs next to the button show the status of the call appearance or status of the call.

Central Office

The location housing telephone switching equipment that provides local telephone service and access to toll facilities for long-distance calling.

Central Office Codes

The first three digits of a 7-digit public network telephone number. These codes are numbered from 200 through 999 and are sometimes referred to as NNXs.

Central Office Trunk

A telecommunications channel that provides access from the system to the public network through the local central office.

Channel

A communications path for transmitting voice and data.

Class of Service (COS)

Parameters used to define voice terminal, data, and trunk port capabilities and restrictions.

Common Control Switching Arrangement (CCSA)

A private telecommunications network using dedicated trunks and a shared switching center for interconnecting company locations.

Confirmation Tone

Three short bursts of tone followed by silence; indicates that the feature activated, deactivated, or canceled has been accepted.

Console

See Attendant Console.

Coverage Call

A call that is redirected from the called party's personal dial code to an alternate answering position when certain criteria are met.

Coverage Path

The order in which calls are redirected to alternate answering positions.

Coverage Point

The attendant positions (as a group), Direct Group Call (DGC) group, Coverage Answer Group, or a voice terminal extension designated as an alternate answering position in a coverage path.

Covering User

The person at an alternate answering position who answers a redirected call.

DCE: (Data Communications Equipment)

DDC: (Data Dial Code)

DDD: (Direct Distance Dialing)

DID: (Direct Inward Dialing)

DGC: (Direct Group Call)

DTE: (Data Terminal Equipment)

DTU: (Digital Tape Unit)

DTMF: (Dual Tone Multifrequency)

DXS: (Direct Extension Selector)

Data Channel

A communication path between two points used to transmit digital signals.

Data Communications Equipment (DCE)

Refers to a specific RS-232C interface connector configuration. DCE devices are designed to interface directly (pin-for-pin) to Data Terminal Equipment (DTE). The transmit and receive pins are reversed from that of a DTE interface. A modem is an example of a DCE device.

Data End Point

Two general groups; those having a DTE type interface, which encompasses almost all of the data terminal devices, and the group of DCE interface devices which are primarily modems. However, it must be noted that within each category, control interfaces may also vary. Refer to Analog Data Endpoint and Digital Data Endpoints for additional information.

Data Module

A data interface device (i.e., Asynchronous Data Unit) providing a standard interface between customer provided data equipment and a data port on the switch.

Data Terminal Equipment (DTE)

DTE refers to a specific RS-232C connector termination designed to connect directly to a DCE type connection. Typically associated with video display terminals, printers, and computers which either originate or terminate a data transmission path.

Refers to RS-232C compatible Data Terminal Equipment

Delay-Dial Tie Trunk

After a request for service (called a seizure) is detected on an incoming trunk, the system sends a momentary signal followed by a steady tone over the trunk. This informs the calling party that dialing can start. This type of trunk allows dialing directly into the system. That is, the digits are received as they are dialed.

Digital Data Endpoints

In System 25, digital data endpoints include any digital device providing a RS-232C connection interface to the switch. The connection is via Asynchronous Data Units (ADUs) to the switch.

Direct Extension Selector (DXS) Console

An option at the attendant console that allows an attendant direct access to voice terminals by pressing a Group Select button and a DXS button.

EIA: (Electronics Industries Association)

Emergency Transfer Unit

Provides direct connection of designated Power Failure Transfer (PFT) registered voice terminals to the CO during a power failure or other service interruption.

Erlang

A traffic measuring unit that expresses the load of one or more traffic-handling devices [36 CCS equals 1 erlang - see CCS (Hundred Call Seconds)].

Extension Number

One- through four-digit number assigned to each voice terminal and data end point in the system. Also see " Personal Dial Code"

External Call

A connection between a system user and a party on the public telephone network or on a tie trunk.

FRL: (Facility Restriction Level)

Facility (physical)

A transmission channel to another switching system; to a Central Office for example. By application, examples are:

- CO Trunks
- FX Trunks
- WATS Trunks
- OCC Trunks
- Tie Trunks

By technical type these include loop start, ground start, DID, automatic ringdown, etc. These facilities may be accessed by their facility access codes (FACs).

Feature

A specifically defined function or service provided by the system.

Feature Button

A labeled button on a voice terminal or attendant console designating a specific feature.

Foreign Exchange (FX)

A central office other than the one providing local access to the public telephone network.

Foreign Exchange Trunk

A telecommunications channel that directly connects the system to a central office other than its local central office.

Foreign Numbering Plan Area Code (FNPA)

An area code other than the local area code. The foreign area code must be dialed to call outside the local geographical area.

Ground-Start Trunk

On outgoing calls, System 25 transmits a request for services to the distant switching system by grounding the trunk ring lead. When the distant system is ready to receive the digits of the called number, that system grounds the trunk tip lead. When the System 25 detects this ground, the digits are sent. (Tip and ring are common nomenclature to differentiate between ground-start trunk leads.) On incoming calls, detection of ground on the tip lead is sufficient to cause the call to route to a predetermined destination, normally the system attendant group. No digits are received.

Home Numbering Plan Area Code (HNPA)

The local area code. The area code does not have to be dialed to call numbers within the local geographical area.

Immediate-Start Tie Trunk

After establishing a connection with the distant switching system for an outgoing call, the system waits a nominal 65 milliseconds before sending the digits of the called number. This allows time for the distant system to prepare to receive the digits. Similarly, on an incoming call, the system has less than 65 milliseconds to prepare to receive the digits.

In-Use Lamp

A red lamp on a multiline voice terminal that lights to show which call appearance will be selected when the handset is lifted or which call appearance is active when a user is off-hook.

Intercept Tone

An alternating high and low tone; indicates a dialing error or denial of the service requested.

Interface

A common boundary between two systems or pieces of equipment.

Internal Call

A connection between two users within the system.

LDN: (Listed Directory Number)

LED: (Light Emitting Diode)

Loop Start Trunk

After establishing a connection with the distant switching system for an outgoing call, System 25 waits for a short period of time before sending the digits of the called number. On incoming calls, the received request for service is sufficient to cause the call to route to a predetermined destination, normally the system attendant group. No digits are received.

MET: (Multibutton Electronic Telephone)

Modem

A device that modulates and demodulates signals transmitted over a communications path. Used to connect Data Terminal Equipment to the system's analog ports. The system provides a pooled modem conversion resource (12 resources maximum per system—212A compatible).

Modem Pooling

Provides shared-use conversion resources that eliminate the need for a dedicated modem when an analog data end point accesses, or is accessed by, an analog line or trunk.

Multifeature Single-Line Voice Terminal

A terminal served by one tip and ring voice circuit and having additional buttons for activating features.

Multilane Voice Terminal

A terminal equipped with several call appearance buttons for the same extension number. Allows the user to handle more than one call, on that same extension number, at the same time.

Multiplexed

The simultaneous transmission of two or more signals over a common transmission medium.

NPA: (Number Plan Area)

Network

An arrangement of inter and/or intra location circuits designed to perform specific functions.

Network Interface

Provided by the CO telephone company in two forms:

- (1) RJ21X for trunk facilities other than tie trunks.
- (2) RJ2GX for tie trunk facilities.

An arrangement provided by the local telephone company which permits remote Terminal Equipment to operate as though it was directly connected to the System 25. This tariffed service can only be provided for FCC registered single-line voice terminals.

Out-Of-Building Station

The Terminal Equipment indirectly connected to the System 25, but is not located in the same building as the common equipment. Special arrangements are made to protect the system and its users from lightning, power line crosses, etc. Only the single-line and 7300H series of voice terminal may be so connected. MET Sets can not be connected as Out-Of-Building stations.

PDC: (Personal Dial Code)

Paging Trunk

A telecommunications channel used to access an amplifier for loudspeaker paging.

Parameter

Any set of physical properties whose values determine the characteristics or behavior of something

Peripheral Equipment

System Administration Terminal (SAT), SMDR Output device such as a SMDR Printer or a Call Accounting System, Digital Tape Unit (DTU).

Personal Dial Code

Each system user is assigned a PDC and is allowed to “login” the PDC at any voice terminal in the system (optional feature) as they move about the premises. The PDC may be a 1-, 2-, 3-, or 4-digit number. There are two types of PDCs:

- PDCs assigned to voice terminals - Associated with each voice terminal in the system.
- Floating - Assigned to users and visitors who will be moving about the premises. Floating PDCs may be associated with the attendant position or may be “logged-in” by the user at a system voice terminal. Calls to the floating PDC will ring at the terminal where “logged in”

Pickup Group

A group of individuals authorized to answer any call directed to an extension number within the group.

Port

An interface circuit between System 25 and associated auxiliary and peripheral equipment. Typical references include:

- Terminal port (station port)
- Facility port (trunk port)
- Auxiliary equipment port

Private Branch Exchange (PBX)

A switching system that provides switched communications access amongst its terminals and facilities (e.g., System 25)

Private Network

A network used exclusively for handling the telecommunications needs of a particular customer.

Private Network Office Code (RNX)

The first three digits of a 7-digit private network number. These codes are numbered 220 through 999, excluding any codes that have a 0 or 1 as the second digit.

Protocol

A set of conventions or rules governing the format and timing of message exchanges to control data movement and correction of errors.

Public Network

The network that can be openly accessed by all customers for local or long-distance calling.

Queue

An ordered sequence of calls waiting to be processed.

Queuing

The process of holding calls in order of their arrival to await connection to an attendant, to an answering group, or to an idle trunk. Calls are automatically connected in first-in, first-out sequence.

Random Access Memory (RAM)

A storage arrangement whereby information can be retrieved at a speed independent of the location of the stored information.

Read Only Memory (ROM)

A storage arrangement primarily for information retrieval applications.

Recall Dial Tone

Three short bursts of tone followed by steady dial tone; indicates the system has completed some action (such as holding a call) and is ready to accept dialing.

Redirection Criteria

The information administered for each voice terminal that determines when an incoming call is redirected to coverage.

Reorder Tone

A fast-busy tone repeated 120 times a minute; indicates that at least one of the facilities, such as a trunk or a digit transmitter, required for the call was not available at the time the call was placed.

Single-Line Voice Terminal

Voice terminal served by a single-line tip and ring circuit (2500 series and 7101A voice terminals or industry standard Dual Tone Multifrequency equivalent).

SAT: (System Administration Terminal)

SIP: (Station Interconnect Panel)

SMDR: (Station Message Detail Recording)

Software

A set of computer programs that accomplish one or more tasks.

Split

A condition whereby a caller is temporarily separated from a connection with the attendant. This split condition automatically occurs when the attendant, active on a call, presses the Start button.

Status LED (lamp)

A green LED or lamp that shows the status of a call appearance or a feature button by the state of the lamp (lighted, flashing, fluttering, broken flutter, or dark).

Station

A place where terminal equipment is located or sometimes the terminal equipment itself. Each voice terminal (station) is assigned a station (extension) number. Users of the terminal are sometimes referred to as station users. Reference to the extension number is sometimes in the form PDC (Personal Dial Code) rather than station number. Though PDCs may be “logged-in” at other stations, in most discussions, though, PDCs and station numbers are interchangeable. Analogously, data stations are assigned DDCs (Data Dial Codes)

Switch

The software-controlled communications processor complex that interprets dialing pulses/tones/keyboard characters and makes the proper interconnections both within the system and external to the system. The switch itself consists of a digital computer, software, storage device (memory), and associated circuit packs and special hardware necessary to perform the actual connections.

Switchhook

The button(s) on a voice terminal located under the receiver.

System Manager

A person responsible for specifying and administering features and services for the system.

System Reload

A process that allows stored data to be written from a tape into the system memory (normally after a power outage).

TAE: (Trunk Access Equipment)

Terminal Equipment:

Equipment for changing information (sound, keystrokes) into an electrical signal compatible with the system’s port circuits (voice and data terminals are two subdivisions).

Tie Trunk

A telecommunications channel that directly connects two private switching systems.

Time Division Multiplex Bus

A special bus that is time shared by preallocating short time slots to each transmitter on a regular basis. In a PBX, all port circuits are connected to the time division multiplex bus permitting any port to send a signal to any other port.

Tone Ringer

A device with a speaker, used in electronic voice terminals to alert the user.

Translations

Specific information assigned to a terminal or to the system and customized for the user.

Trunk

A telecommunications channel between two switching systems.

Trunk Group

Telecommunications channels assigned as a group for certain functions.

Trunk Port

The hardware providing the access point to the system switching network for each circuit associated with a trunk.

Voice Terminal

A single-line or multiline voice instrument (e.g., telephone)

Voice Terminal Adjuncts

- . 500A/502A Headset Adapter
- . S101A/S102A Speakerphone
- . Acoustic Coupler

(Refer to Section 4—Hardware Description for a complete list and description of Voice Terminal Adjuncts.)

Wide Area Telecommunications Service (WATS)

A service that allows calls to a certain area or areas for a flat-rate charge based on expected usage.

Wink-Start Tie Trunk

After establishing a connection with a distant switching system for an outgoing call, the system waits for a momentary signal (wink) before sending the digits of the called number. Similarly, on an incoming call, the system sends the wink signal when ready to receive digits.

Write Operation

The processor putting information onto a storage medium such as magnetic tape.

800 Service

A service that allows incoming calls from a certain area or areas to an assigned number for a flat-rate charge based on usage.

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