AT&T SYSTEM 75
REFERENCE MANUAL

970-232
999-700-232IS
Issue 2, November 1984

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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 own expense will be required to take whatever measures may be required to correct the interference.
- The voice terminals described in Section 2 of this manual are compatible with inductively coupled hearing aids as prescribed by the FCC.

FCC REGISTRATION INFORMATION

Registration Number	AS5-93M-13283-MF-E
Ringer Equivalent	0.5A
Network Interface	RJ21X* or RJ2GX

*Ground start trunks are recommended.

INTRODUCTION

This manual provides a technical description of the system, hardware, environmental and space requirements, parameters, and features.

Purpose

This manual is intended to serve as an overall reference for the planning, operation, and administration stages of the system. It is also intended to be used with the *AT&T System 75 Implementation Manual*, 999-700-277IS, for software initialization and subsequent changes in feature assignments.

This issue replaces all previous issues of this document. Reasons for reissue include the following:

- To expand the System Description.
- To include lightning protection information and change floor loading requirements in Environmental Requirements.
- To expand System Parameters to include the following:
  - A list of Applications Processor (AP) features.
  - A list of features which can be dial or button access.
  - An expanded System Capacities list.
  - Acoustic Noise Level information.
  - Heat Dissipation information.
  - Standby Power Systems information.
- To add or expand the description of the following Voice Management features:
  - Abbreviated Dialing
  - AP Demand Print
  - Automatic Route Selection (ARS)
  - Call Coverage
  - Class of Restriction (COR)
  - Direct Department Calling (DDC)
  - Facility Test Calls
  - Station Message Detail Recording (SMDR)
  - System Measurements
  - Terminating Extension Group (TEG)
  - Uniform Call Distribution (UCD)

Organization

This manual is divided into eight sections. The remaining sections are as follows:

- SECTION 2—SYSTEM DESCRIPTION—Provides a technical description of the system including the hardware components.
- SECTION 3—SPACE AND ENVIRONMENTAL REQUIREMENTS—Provides the space and environmental requirements for installation and operation of the system.
- SECTION 4—SYSTEM PARAMETERS—Provides information relating to overall system characteristics and capacities. This section includes items that must be considered when planning for system implementation.
- SECTION 5—VOICE MANAGEMENT—Provides a detailed description of the features that relate to voice management.
- SECTION 6—DATA MANAGEMENT—Provides a detailed description of Data Management and the related features. Also includes a description of the data protocols used with the system.
- SECTION 7—GLOSSARY—Provides a glossary for the entire Reference Manual.
- SECTION 8—INDEX—Provides a permuted index for the entire Reference Manual.

An individual Table of Contents is provided for Sections 2 through 6 of this manual.

An Appendix, provided at the end of this manual, contains enhancements to be available with Release 1 Version 2 of the system. The description of Release 1 Version 2 capabilities is confined to the Appendix. All other information presented in this manual applies to Release 1 Version 1, which is currently available.

SYSTEM DESCRIPTION

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SYSTEM OVERVIEW

System 75 is a terminal-oriented advanced business communications system. The system's digital switch, using a time-space-time circuit switching technique, provides voice communications, data communications, and applications processing services.

Voice communications combine traditional telephone features, such as Hold and Transfer, with new voice features, such as Abbreviated Dialing and Leave Word Calling (see Section 5).

Data communications allow data calls through data terminal equipment connected to the digital switch. An integral part of the operation of terminals is the use of a Digital Communications Protocol (DCP). The DCP conveys both voice and data over the same link through one signaling channel and two information channels. The signaling channel conveys call control and data terminal management information between the terminal and the digital switch. The two information channels provide simultaneous voice and data connections to independent destinations.

The applications processing services are provided through an optional Applications Processor (AP). The AP provides features such as Message Center Service, Directory Service, and Electronic Documentation Communications. See Section 4 of this manual for a list of AP services.

The system is arranged exclusively for touch-tone dialing—rotary dialing is not provided except through the Power Failure Transfer feature. The system does, however, automatically convert touch-tone signals to dial pulses on trunks requiring such conversion.

Call-Handling Capabilities

System 75 can be arranged as a stand-alone system or as an access to private networks. The system will serve as a main or tributary location in an Electronic Tandem Network (ETN), Enhanced Private Switching Communications Service (EPSCS) network, Common Control Switching Arrangement (CCSA) network, or as a tandem or end location in a tandem tie trunk network.

The system can provide the following:

- Up to 400 lines that support digital, hybrid, and analog terminals and equipment
- Data switching capacity of up to 200 digital data endpoints* and 32 pooled modems with optional single-button access to the pooled facilities
- Up to 200 trunks including central office (CO) trunks, Direct Inward Dialing (DID) trunks, tie trunks, foreign exchange (FX) trunks, Wide Area Telecommunications Service (WATS) trunks, and other common carrier trunks

It should be noted, however, that the limits listed for each of these three items probably cannot be used in any one system. Allowable limits are determined according to expected call usage.

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* Digital data endpoints include Processor Data Modules, Trunk Data Modules, Digital Terminal Data Modules, 515-type Business Communications Terminals, the Applications Processor interface, and the internal data channels. A maximum of four internal data channels are available to connect equipment such as administration terminals or an output device for the Station Message Detail Recording (SMDR) feature.

System Features

The system offers the following features:

- An essentially nonblocking digital switch that supports up to 236 simultaneous port-to-port connections
- Multiple attendant console capability with up to six consoles, plus an optional night-only console
- On-line system traffic and performance measurement reports that can be optionally printed
- A dedicated System Access Terminal (SAT) that allows the client to access specific system translations and perform administration and maintenance
- An optional SAT (used on-premises with a Processor Data Module or with a digital voice terminal and a Digital Terminal Data Module, or used off-premises with a Trunk Data Module or a modem pool conversion resource) that allows the client to access specific system translations and perform administration on a dial-up basis
- Uniquely assigned passwords to prevent unauthorized use of the SAT
- Dial-up access through the Initialization and Administration System (INADS), an AT&T service that provides remote administration and maintenance
- Ability to connect equipment furnished by the client for features such as Loudspeaker Paging Access, Music-on-Hold Access, Recorded Telephone Dictation Access, and Code Calling Access
- Transfer of voice terminals to CO trunks if power fails or a major alarm condition occurs, available in multiples of 6 up to a maximum of 42 (Power Failure Transfer feature)
- Standby power for the entire system for up to 10 seconds
- Backup power for the system memory for up to 10 minutes
- Memory backup tape so that translations are saved and the client's unique system configuration is not lost if power fails

System Configuration

Figure 2-1 shows a typical system configuration.

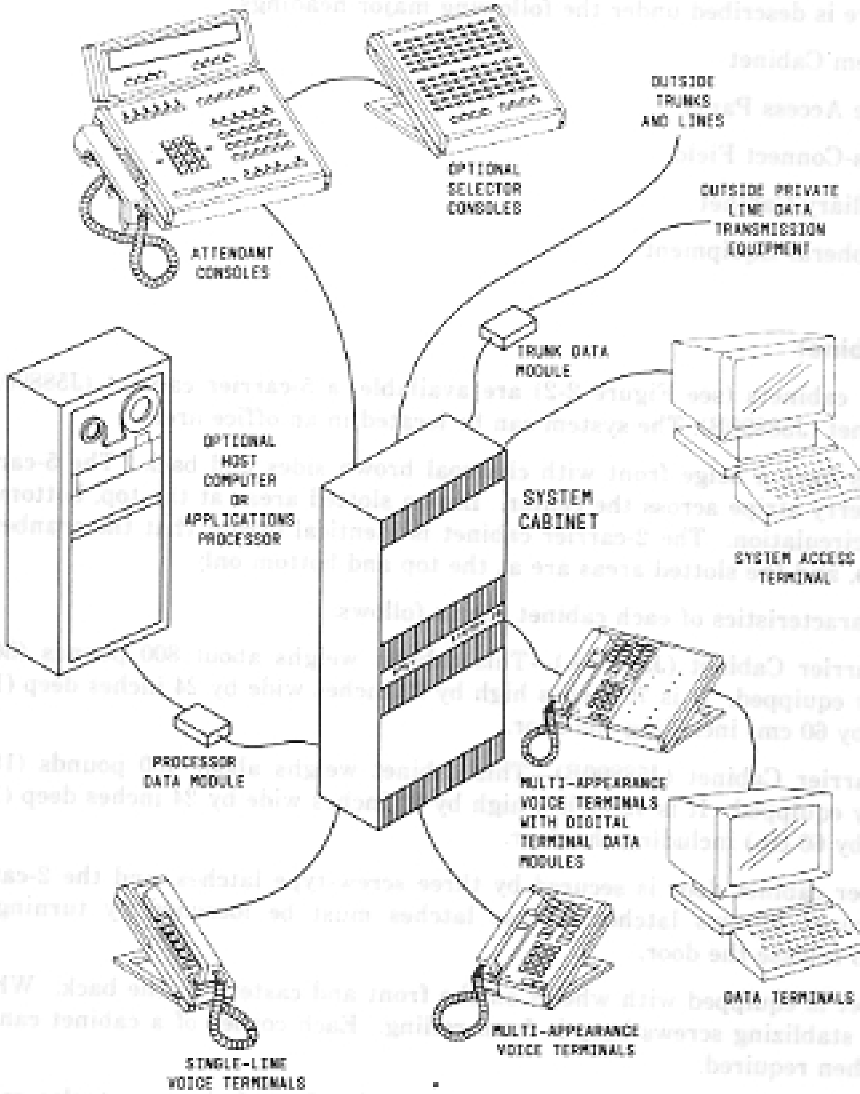


Figure 2-1. Typical System Components

SYSTEM HARDWARE

This part describes the system hardware components, their functions and interconnections. The hardware is described under the following major headings:

- System Cabinet
- Cable Access Panel
- Cross-Connect Field
- Auxiliary Cabinet
- Peripheral Equipment

System Cabinet

Two system cabinets (see Figure 2-2) are available, a 5-carrier cabinet (J58890A) and a 2-carrier cabinet (J58890B). The system can be located in an office area.

The cabinets have a beige front with charcoal brown sides and back. The 5-carrier cabinet has a cranberry stripe across the center. Brown slotted areas at the top, bottom, and center are for air circulation. The 2-carrier cabinet is identical except that the cranberry stripe is near the top, and the slotted areas are at the top and bottom only.

Physical characteristics of each cabinet are as follows:

- 5-Carrier Cabinet (J58890A)—This cabinet weighs about 800 pounds (360 kg) when fully equipped. It is 70 inches high by 32 inches wide by 24 inches deep (175 cm by 80 cm by 60 cm) including the door.
- 2-Carrier Cabinet (J58890B)—This cabinet weighs about 400 pounds (180 kg) when fully equipped. It is 42 inches high by 32 inches wide by 24 inches deep (105 cm by 80 cm by 60 cm) including the door.

The 5-carrier cabinet door is secured by three screw-type latches, and the 2-carrier cabinet door is secured by two latches. These latches must be loosened by turning the screws clockwise to release the door.

Each cabinet is equipped with wheels on the front and casters on the back. When a cabinet is in place, stablizing screws keep it from rolling. Each corner of a cabinet can be bolted to the floor when required.

Both cabinets have two ac receptacles in the back. One of the receptacles can be used to power the System Access Terminal (SAT).

The power cord on each cabinet has a heavy-duty, 3-prong plug and must be connected to a dedicated power source (115 volts ac 50 amp for the 5-carrier cabinet and 115 volts ac 20 amp for the 2-carrier cabinet). (Power requirements are described in Section 3 of this manual.)

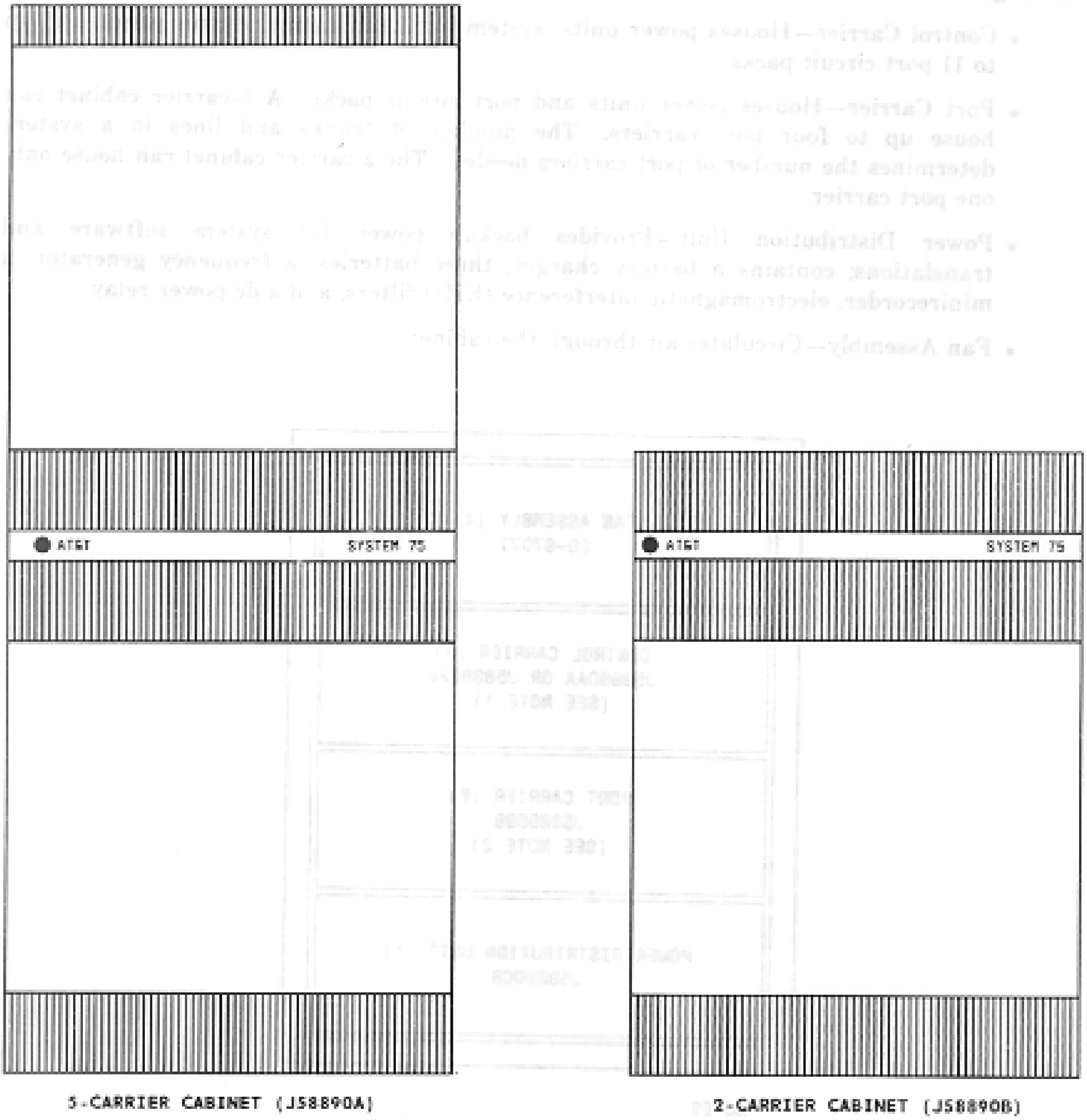
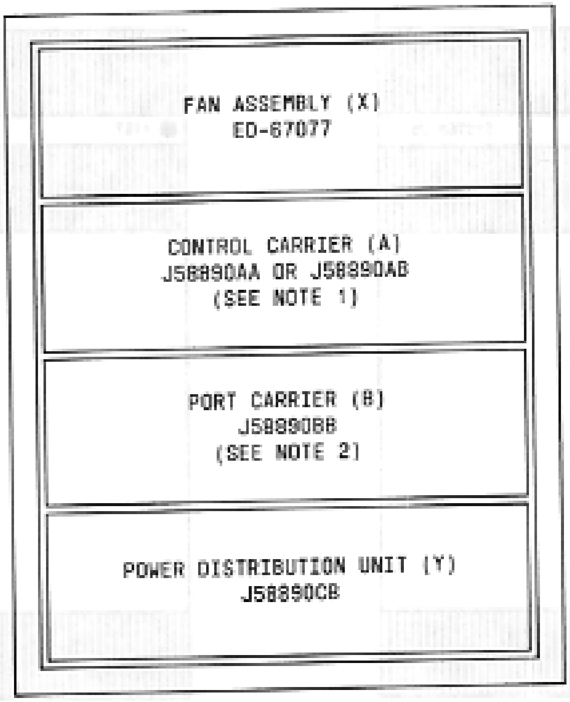


Figure 2-2. System Cabinets (J58890A and J58890B)

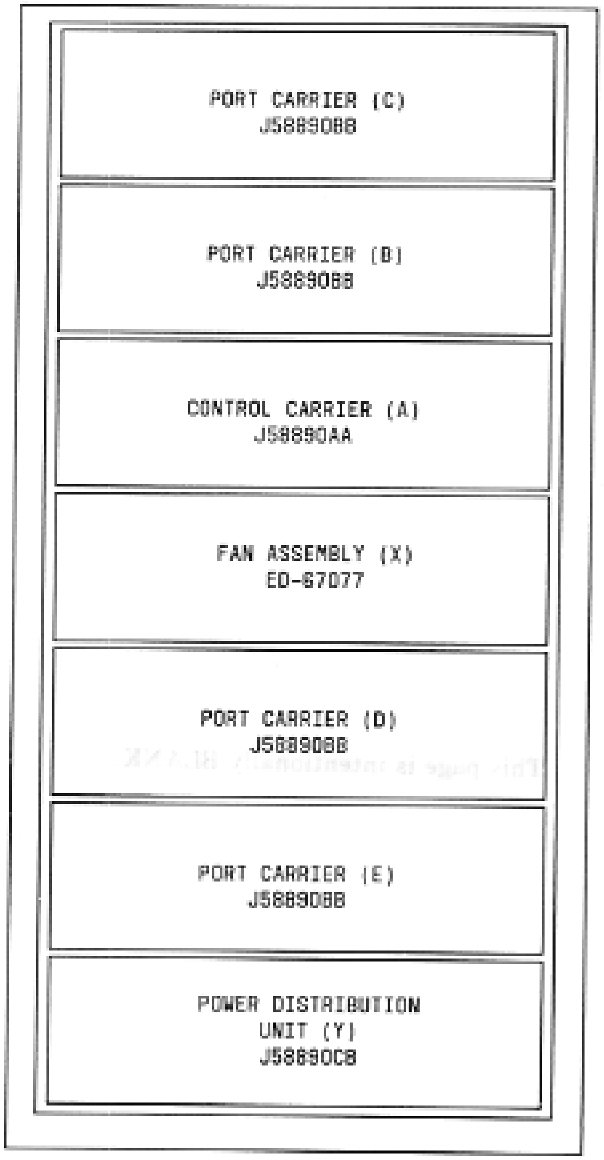
The system cabinet houses the main hardware of the system. Each fully equipped cabinet (see Figures 2-3 and 2-4) contains the following:

- Control Carrier—Houses power units, system logic and control circuit packs, and up to 11 port circuit packs.
- Port Carrier—Houses power units and port circuit packs. A 5-carrier cabinet can house up to four port carriers. The number of trunks and lines in a system determines the number of port carriers needed. The 2-carrier cabinet can house only one port carrier.
- Power Distribution Unit—Provides backup power for system software and translations; contains a battery charger, three batteries, a frequency generator, a minirecorder, electromagnetic interference (EMI) filters, and a dc power relay.
- Fan Assembly—Circulates air through the cabinet.



NOTES:
1. MODEL 1 SYSTEMS USE J58890AB.
   MODEL 2 SYSTEMS USE J58890AA.
2. MODEL 1A AND 2A SYSTEMS DO NOT USE A PORT CARRIER.

Figure 2-3. Fully Equipped 2-Carrier Cabinet (J58890B)



NOTE: MODEL 3A HAS PORT CARRIER IN POSITION B.
MODEL 3B HAS PORT CARRIERS IN POSITIONS B AND C.
MODEL 3C HAS PORT CARRIERS IN POSITIONS B, C, AND D.
MODEL 3D HAS PORT CARRIERS IN POSITIONS B, C, D, AND E.

Figure 2-4. Fully Equipped 5-Carrier Cabinet (J58890A)

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Eight system models are available. Table 2-A lists the models and corresponding cabinets. All models contain one control carrier. Models 1A and 1B do not support the Applications Processor (AP); the remaining models have AP capability.

TABLE 2-A. System Sizes

Model	Cabinet	Port Carriers	Port Circuit Packs*	AP Capability
1A	2-Carrier (J58890B)	0	11	No
1B		1	30	No
2A		0	7	Yes
2B		1	26	Yes
3A	5-Carrier (J58890A)	1	26	Yes
3B		2	45	Yes
3C		3	65	Yes
3D		4	85	Yes

* The number of ports per circuit pack are defined in this section under the heading *Circuit Packs.*

Carriers

Control Carrier

Two control carriers (J58890AA and J58890AB) are available. The J58890AA control carrier is used with all Model 2 and 3 systems. These system models support the AP.

The J58890AB control carrier is used with Models 1A and 1B. These system models do not support the AP.

The control carriers are made of structural foam. They have a slotted-steel faceplate that is grounded through a brass strip to the cabinet mounting frame. The control carriers (see Figure 2-5) contain the following:

- Circuit Pack Slots—Provide 19 slots that are always equipped with control circuit packs and can be equipped with port circuit packs. Control circuit pack slots are dedicated and cannot contain port circuit packs (see Figure 2-5).

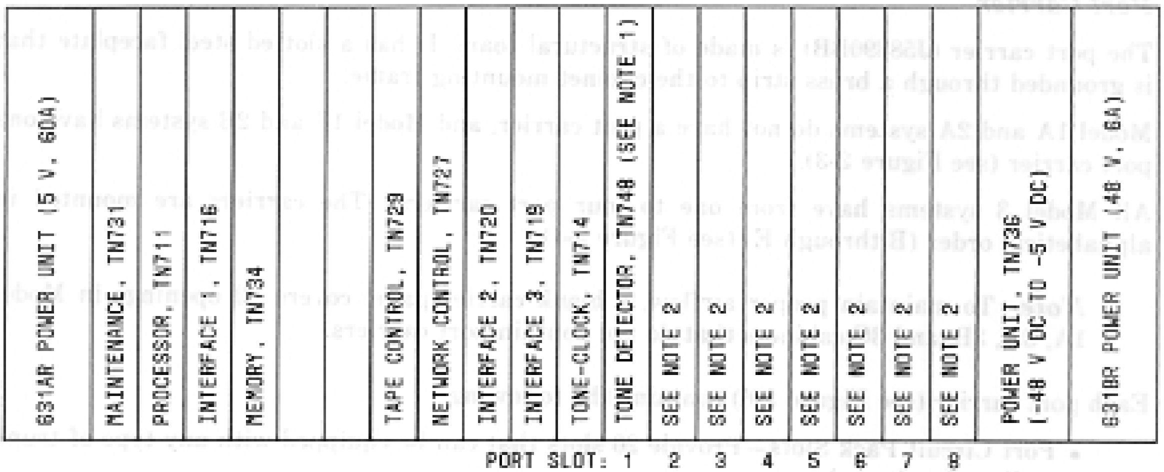
  Note: To maintain proper airflow, a Z100A apparatus blank covers all slots that do not contain circuit packs.

- 631AR Power Unit—Provides +5 volts dc power to the carrier.
- 631BR Power Unit—Provides -48 volts dc power to the carrier.
- TN736 Power Unit—Converts -48 volts dc to -5 volts dc required by the circuit packs.

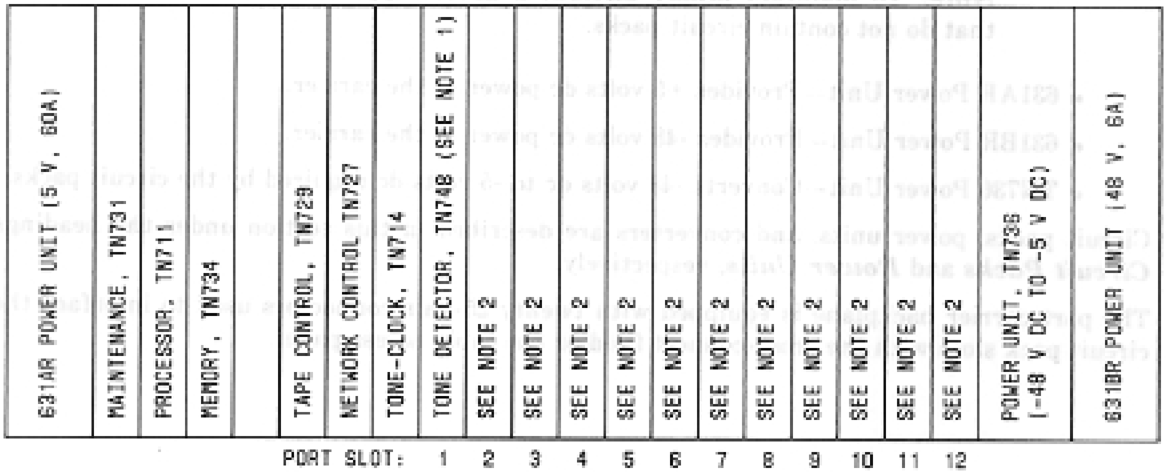
Circuit packs and power units are described in this section under the headings ***Circuit Packs*** and ***Power Units,*** respectively.

The control carrier backplanes are equipped with the following:

- Eleven 25-Pair Connectors (J58890AA) and fourteen 25-Pair Connectors (J58890AB)—Used to interface the port circuit pack slots to the cross-connect field or the cable access panel.
- Two RS-232C EIA Connectors—One connects to the System Access Terminal (SAT) and the other is reserved for future use.
- One Maintenance Connector (labeled AUXILIARY)—Used to connect control carrier input and output to the cable access panel.



CONTROL CARRIER (J58890AA)



CONTROL CARRIER (J58890AB)

NOTES:

1. TONE DETECTOR TN748 CAN BE LOCATED IN ANY PORT SLOT; HOWEVER, IT IS ALWAYS SHIPPED IN PORT SLOT 1.
2. PORT SLOTS 2 THROUGH 8 FOR J58890AA CONTROL CARRIER OR SLOTS 2 THROUGH 12 FOR J55590AB CONTROL CARRIER CAN CONTAIN THE FOLLOWING TYPES OF CIRCUIT PACKS:
   - MET LINE TN735
   - ANALOG LINE TN742
   - CO TRUNK TN747
   - DID TRUNK TN753
   - DIGITAL LINE TN754
   - POOLED MODEM TN758
   - TIE TRUNK TN760
   - HYBRID LINE TN762
   - AUXILIARY TRUNK TN763

Figure 2-5. Control Carriers (J58890AA and J58890AB)

Port Carrier

The port carrier (J58890BB) is made of structural foam. It has a slotted steel faceplate that is grounded through a brass strip to the cabinet mounting frame.

Model 1A and 2A systems do not have a port carrier, and Model 1B and 2B systems have one port carrier (see Figure 2-3).

All Model 3 systems have from one to four port carriers. The carriers are mounted in alphabetical order (B through E) (see Figure 2-4).

> ***Note:*** To maintain proper airflow, a blank carrier panel covers all openings in Model 1A, 3A, 3B, and 3C cabinets that do not contain port carriers.

Each port carrier (see Figure 2-6) contains the following:

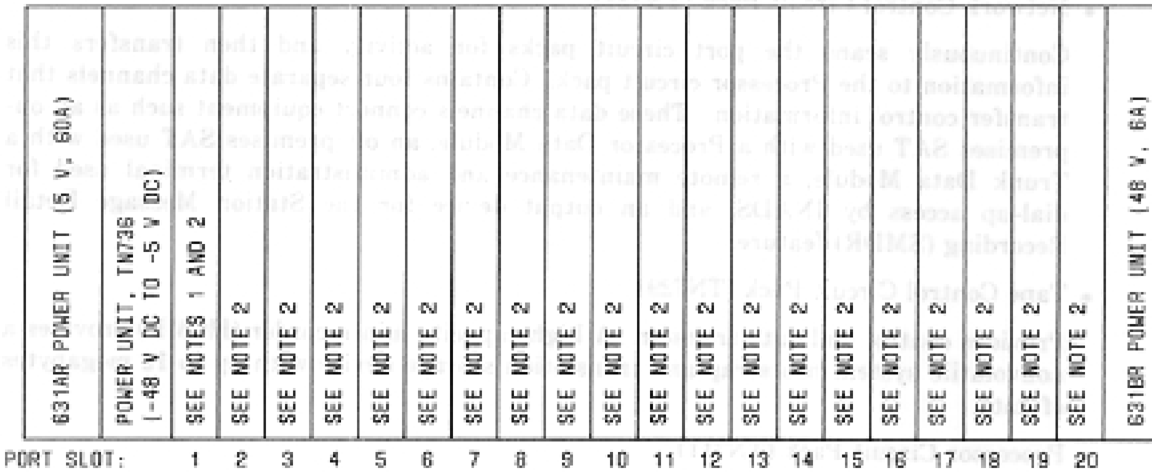
- Port Circuit Pack Slots—Provide 20 slots that can be equipped with any type of trunk or line circuit pack.

  > ***Note:*** To maintain proper airflow, a Z100A apparatus blank covers all slots that do not contain circuit packs.

- 631AR Power Unit—Provides +5 volts dc power to the carrier.
- 631BR Power Unit—Provides -48 volts dc power to the carrier.
- TN736 Power Unit—Converts -48 volts dc to -5 volts dc required by the circuit packs.

Circuit packs, power units, and converters are described in this section under the headings ***Circuit Packs*** and ***Power Units***, respectively.

The port carrier backplane is equipped with twenty 25-pair connectors used to interface the circuit pack slots with the cross-connect field or the cable access panel.



NOTES:

1. IN ADDITION TO THE TN748 TONE DETECTOR LOCATED IN THE CONTROL CARRIER, MODEL 1B, 2B, AND 3A SYSTEMS REQUIRE ONE ADDITIONAL TONE DETECTOR. MODEL 3B, 3C, AND 3D REQUIRE TWO ADDITIONAL TONE DETECTORS. THE ADDITIONAL TONE DETECTOR(S) CAN BE LOCATED IN ANY PORT SLOT; HOWEVER, THEY ARE SHIPPED IN PORT SLOT 1 OF PORT CARRIER B AND PORT CARRIER C.
2. PORT SLOTS 1 THROUGH 20 CAN CONTAIN THE FOLLOWING TYPES OF CIRCUIT PACKS:
   - MET LINE TN735
   - ANALOG LINE TN742
   - CO TRUNK TN747
   - TONE DETECTOR TN748
   - DID TRUNK TN753
   - DIGITAL LINE TN754
   - POOLED MODEM TN758
   - TIE TRUNK TN760
   - HYBRID LINE TN762
   - AUXILIARY TRUNK TN763

Figure 2-6. Port Carrier (J58890BB)

Circuit Packs

All system circuit packs have the following features:

- Solid-state logic circuitry mounted on an 8-inch by 14-inch (20.3 cm by 25.5 cm) printed wiring board.
- Color-coded faceplate labels to identify the circuit type.
- Individual circuits are fully contained on one circuit pack.
- Fault, test, and busy light-emitting diodes (LEDs).

Control Circuit Packs

The following control circuit packs, located in the control carrier, are provided with all system models:

- Maintenance Circuit Pack (TN731)

  Provides direct access to the dedicated SAT; provides alarm LEDs for system status; monitors power units and controls emergency transfer; monitors and controls Processor circuit pack conditions; monitors cabinet temperature and airflow; and originates alarms to the remote maitenance system (INADS which stands for Initialization and Administration System). This pack contains an 8088 microprocessor that implements the BX.25 protocol used for remote access. (The BX.25 protocol is described in Section 6 of this manual.) A digital signal processor provides 212-type modem emulation and a line circuit provides direct connection to the network.

- Network Control Circuit Pack (TN727)

  Continuously scans the port circuit packs for activity and then transfers this information to the Processor circuit pack. Contains four separate data channels that transfer control information. These data channels connect equipment such as an on-premises SAT used with a Processor Data Module, an off-premises SAT used with a Trunk Data Module, a remote maintenance and administration terminal used for dial-up access by INADS, and an output device for the Station Message Detail Recording (SMDR) feature.

- Tape Control Circuit Pack (TN729)

  Provides control and data transfer. A high-capacity minirecorder (HCMR) provides a nonvolatile system bootstrap and translation storage device with up to 12 megabytes of data.

- Processor Circuit Pack (TN711)

  Manages control of the entire system and executes stored programs to effect call and data processing functions. Has 32,000 bytes of Read-Only Memory (ROM)/Erasable Programmable ROM (EPROM) and 2,000 bytes of Random Access Memory (RAM).

- Memory Circuit Pack (TN734)

  Contains all system translations including addresses of all equipment connected to the switch through the port circuit packs. Handles 2 megabytes of memory and contains a memory array, on-board refresh logic, address decode logic, and bus buffers.

- Tone-Clock Circuit Pack (TN714)

  Supplies call progress tones, touch tones, answer-back tones, and trunk transmission test tones; provides 2-megahertz (MHz), 160-kilohertz (kHz), and 8-kHz clocks.

- Tone Detector Circuit Pack (TN748)

  Provides four touch-tone receivers and two general purpose tone receivers that detect call progress tones, modem answer-back tones, transmission test tones, and noise.

Note: Each control carrier has one TN748 Tone Detector circuit pack. The tone detector can be located in any port slot but is shipped in port slot 1. Model 1B, 2B, and 3A systems require one additional tone detector in port carrier B. Model 3B, 3C, and 3D systems require two additional tone detectors—one in port carrier B and one in port carrier C.

The following optional circuit packs are located in the control carrier and are required only if the system has an AP. The AP capability is available with all Model 2 and 3 systems.

- Interface 1 Circuit Pack (TN716)

  Interfaces the Processor circuit pack to the remote address of the AP; supports the System (S) bus protocol for interrupts, error reporting, and bus control.

- Interface 2 Circuit Pack (TN720)

  Contains an 8086 microprocessor, 64,000 bytes of ROM, 128,000 bytes of dynamic RAM, and timers; communicates with the Network Control circuit pack and with the Interface 3 circuit pack; forms a channel to the AP for transmission of Message Center data, Leave Word Calling data, and SMDR data (for the Call Detail Recording and Reporting feature of the AP) using the BX.25 protocol. (The BX.25 protocol is described in Section 6 of this manual.)

- Interface 3 Circuit Pack (TN719)

  Provides Data Communications Protocol (DCP) for data coming from the AP, contains special logic and several microcomputers, and includes some of the data module functions. (The DCP is described in Section 6 of this manual.)

Port Circuit Packs

Mounting spaces (slots) are provided in the port carrier for the following circuit packs:

- CO Trunk Circuit Pack (TN747)

  Provides eight ports for loop-start or ground-start central office (CO), foreign exchange (FX), or Wide Area Telecommunications Service (WATS) trunks.

- DID Trunk Circuit Pack (TN753)

  Provides eight ports for immediate-start or wink-start Direct Inward Dialing (DID) trunks.

- Tie Trunk Circuit Pack (TN760)

  Provides four ports for 4-wire E&M lead signaling tie trunks. Trunks can be automatic, immediate-start, wink-start, or delay-dial.

- Auxiliary Trunk Circuit Pack (TN763)

  Provides four ports for on-premises trunk applications such as Music-on-Hold, Loudspeaker Paging, Code Calling, and Recorded Telephone Dictation Access.

- Digital Line Circuit Pack (TN754)

  Provides eight ports for connection to multi-appearance digital voice terminals (Models 7403D and 7405D), attendant consoles, 515 Business Communications Terminals, or data modules over DCP links.

- MET Line Circuit Pack (TN735)

  Provides four ports for the Multibutton Electronic Telephone (MET) sets.

- Hybrid Line Circuit Pack (TN762)

  Provides eight ports for multi-appearance hybrid voice terminals (7303S and 7305S).

- Analog Line Circuit Pack (TN742)

  Provides eight ports for the following connections:

  — Single-line analog voice terminals (Models 2500, 2554, 7101A, and 7103A)

  — Queue warning level lamps associated with the Direct Department Calling and Uniform Call Distribution features

  — Recorded announcements associated with the Intercept Treatment feature

  — Dictation machines associated with the Recorded Telephone Dictation Access feature

  — External alerting devices associated with the Trunk Answer From Any Station feature

  — Modems

- Pooled Modem Circuit Pack (TN758)

  Provides two conversion resources per circuit pack for switched connections between digital data endpoints (data modules) and analog data endpoints (modems). A maximum of 16 Pooled Modem circuit packs (32 conversion resources) are allowed in a system.

- Tone Detector Circuit Pack (TN748)

  Provides four touch-tone receivers and two general purpose tone receivers that detect call progress tones, modem answer-back tones, transmission test tones, and noise.

  In addition to the TN748 Tone Detector located in the control carrier, Model 1B, 2B, and 3A systems require one additional tone detector. Model 3B, 3C, and 3D systems require two additional tone detectors. The additional tone detector(s) can be located in any port slot but are shipped in port slot 1 of port carrier B and port carrier C.

Power Units

The following power units are located in the control and port carriers:

- 631AR Power Unit

  This power unit provides +5 volts dc 60-amp power on the backplane on the control carrier and the port carriers.

  The power unit contains two separate converters. During normal operation, this unit converts the 115-volt ac input to +5 volts dc outputs. If ac power fails, the power unit converts the -144 volts dc supplied by the batteries to +5 volts dc outputs. The power unit contains a relay circuit that detects the highest equivalent input voltage (ac or dc) and switches to the correct converter path accordingly.

- 631BR Power Unit

  This power unit provides -48 volts dc 6-amp power on the backplanes of the carriers for the port circuit packs. The control carrier contains one 631BR Power Unit to power the circuit packs in slots 11 through 20.

  This power unit contains two separate converters. During normal operation, this unit converts the 115-volt ac input to -48 volts dc outputs. If ac power fails, the power unit converts the -144 volts dc supplied by the batteries to -48 volts dc outputs. The power unit contains a relay circuit that detects the highest equivalent input voltage (ac or dc) and switches to the correct converter path accordingly.

- TN736 Power Converter Circuit Pack

  This circuit pack provides -5 volts dc on the backplanes of all carriers for the port circuit packs.

  The Power Converter circuit pack is a dc/dc converter. It contains two identical power supplies that convert the -48 volts dc on the backplane to two -5 volts dc outputs.

Power Distribution Unit

The Power Distribution Unit (J58890CB), is located at the bottom of the system cabinet (see Figures 2-3 and 2-4). As shown in Figure 2-7, it consists of the following items:

- Three 48-Volt Batteries

  These batteries are connected in series, provide a nominal -144 volts dc, and automatically take over during an emergency transfer (Power Failure Transfer mode). These batteries power the entire system for about 10 seconds and the control carrier for about 10 minutes. The -144 volts dc power is fused at 20 amps.

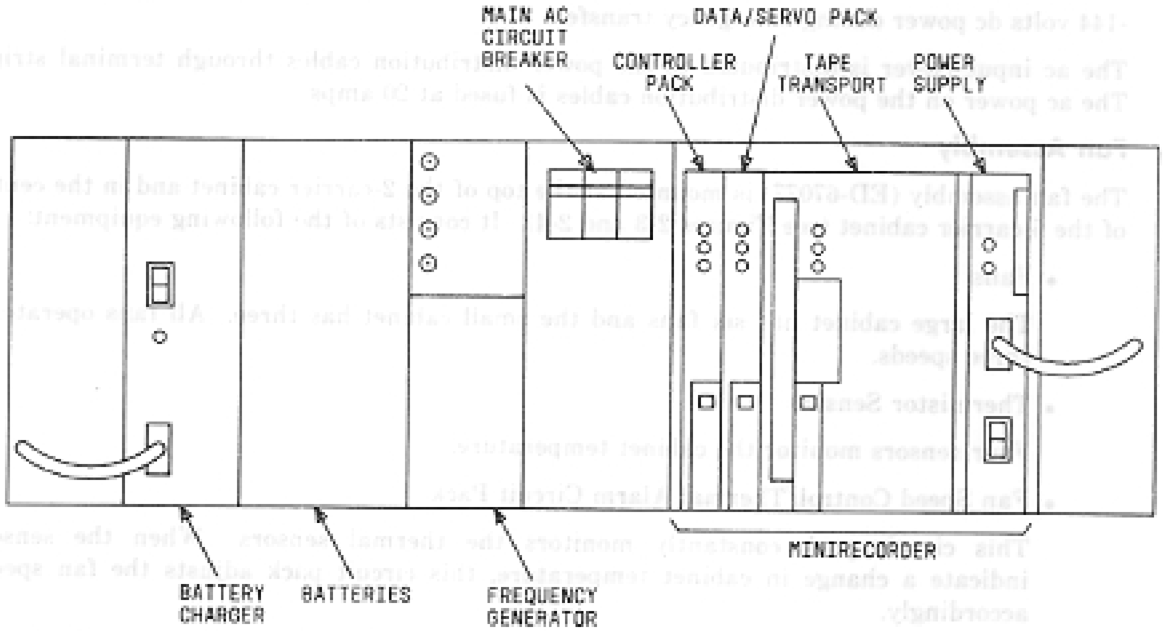


Figure 2-7. Power Distribution Unit (Cover Removed)

- 124B Frequency Generator

  This unit converts -48 volts dc input to 75- to 100-volt ac 20-hertz outputs. These outputs are distributed to all carriers through an inter-unit connecting cable. The Analog Line circuit packs use the outputs to provide ringing to their associated voice terminals.

- High Capacity Minirecorder (HCMR)

  The HCMR stores the software information for the system. If the system goes into the Power Failure Transfer mode for more than 10 minutes, information stored in memory is lost but remains on the HCMR tape. When power is restored, the recorded information is automatically transferred to the Memory circuit pack to reprogram the system.

- Four Electromagnetic Interference (EMI) Filters

  These filters suppress ac line voltage noise.

- DC Power Relay

  This relay disconnects the batteries from the system when ac power is being used. This relay also disconnects the batteries if ac power fails for more than 5 minutes. This protects the batteries from overdischarging.

- 397A Power Unit (Battery Charger)

  When power is restored following the Power Failure Transfer mode, this unit recharges the batteries. The system should fully charge the batteries within 30 hours. If the batteries are not fully charged in that time, the system continues the high rate of charge and sends a minor alarm to the Maintenance circuit pack.

Five-conductor power distribution cables connect the Power Distribution Unit to the power units in the carriers. These cables provide 115-volt ac power during normal operation and -144 volts dc power during emergency transfer.

The ac input power is distributed to the power distribution cables through terminal strips. The ac power on the power distribution cables is fused at 20 amps.

Fan Assembly

The fan assembly (ED-67077) is mounted at the top of the 2-carrier cabinet and in the center of the 5-carrier cabinet (see Figures 2-3 and 2-4). It consists of the following equipment:

- Fans

  The large cabinet has six fans and the small cabinet has three. All fans operate at three speeds.

- Thermistor Sensors

  Four sensors monitor the cabinet temperature.

- Fan Speed Control/Thermal Alarm Circuit Pack

  This circuit pack constantly monitors the thermal sensors. When the sensors indicate a change in cabinet temperature, this circuit pack adjusts the fan speeds accordingly.

  The Fan Speed Control/Thermal Alarm circuit pack sends a major or minor to the Maintenance circuit pack for the following reasons:

  — Reduced airflow in the cabinet

  — Intake air and exhaust air temperatures differ greatly

  Reduced airflow might indicate fan assembly malfunction. A minor alarm is sent when the intake versus the exhaust air temperatures differ by more than 27 degrees Fahrenheit (15 degrees Celsius). A major alarm is sent if the temperature reaches 149 degrees Fahrenheit (65 degrees Celsius). If the temperature reaches 167 degrees Fahrenheit (75 degrees Celsius), the system shuts itself down.

- Air Filters

  One air filter is located above the fan assembly and one below it. These filters can be easily removed and replaced when the cabinet door is opened and the fan assembly cover is removed.

Cable Access Panel

The cable access panel (Figure 2-8) is a single, self-contained module. Cable access panels are provided as required according to the specific system configuration. The cable access panel provides access to individual leads in certain 25-pair cables that connect to the system cabinet, network interface, cross-connect field, auxiliary cabinet, and the AP.

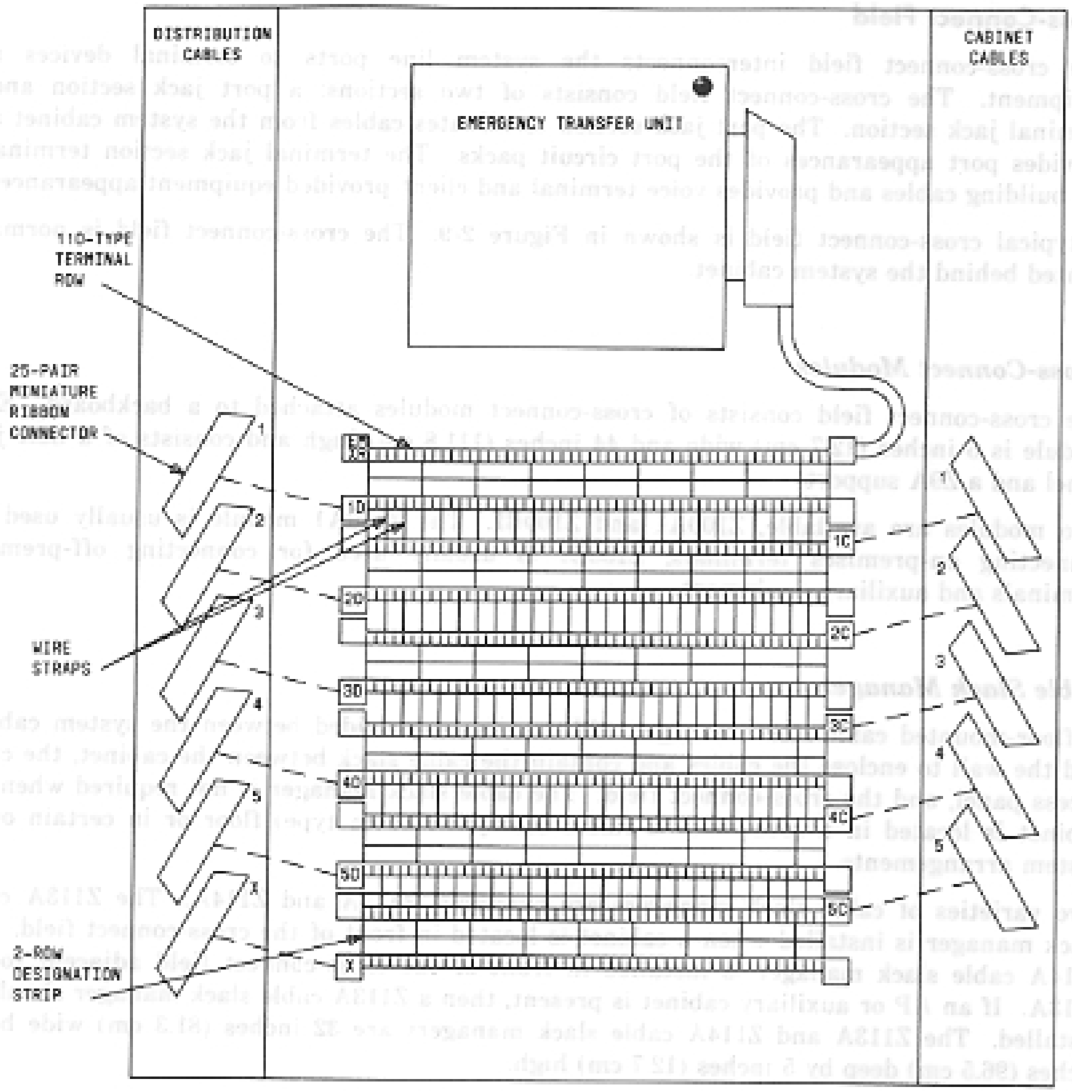


Figure 2-8. Cable Access Panel

Each cable access panel is 18 inches (45.7 cm) wide and 20 inches (50.8 cm) high. The cable access panel(s) is wall-mounted on a backboard. The panel provides the following:

- Mounting space for one Z1A emergency transfer unit. (One emergency transfer unit serves up to six Power Failure Transfer terminals.)
- Termination for up to 40 two-wire trunks (up to 32 for the first cable access panel)

  or

  Termination for up to 20 four-wire trunks (up to 16 for the first cable access panel).
- Limited cross-connection capability for trunk port rearrangement.

The *Service Manual AT&T System 75 Installation and Test*, 555-200-104IS, provides detailed information on the cable access panel.

Cross-Connect Field

The cross-connect field interconnects the system line ports to terminal devices and equipment. The cross-connect field consists of two sections: a port jack section and a terminal jack section. The port jack section terminates cables from the system cabinet and provides port appearances of the port circuit packs. The terminal jack section terminates the building cables and provides voice terminal and client-provided equipment appearances.

A typical cross-connect field is shown in Figure 2-9. The cross-connect field is normally located behind the system cabinet.

Cross-Connect Modules

The cross-connect field consists of cross-connect modules attached to a backboard. Each module is 5 inches (12.7 cm) wide and 44 inches (111.8 cm) high and consists of a Z2A jack panel and a Z9A support.

Two modules are available, Z100A1 and Z100B1. The Z100A1 module is usually used for connecting on-premises terminals; Z100B1 is usually used for connecting off-premises terminals and auxiliary equipment.

Cable Slack Manager

A floor-mounted cable slack manager will usually be provided between the system cabinet and the wall to enclose the cables and contain the cable slack between the cabinet, the cable access panel, and the cross-connect field. The cable slack manager is not required when the cabinet is located in a room with a raised (computer-room type) floor or in certain other system arrangements.

Two varieties of cable slack managers are available, Z113A and Z114A. The Z113A cable slack manager is installed when a cabinet is located in front of the cross-connect field. The Z114A cable slack manager is installed in front of the cross-connect field adjacent to the Z113A. If an AP or auxiliary cabinet is present, then a Z113A cable slack manager should be installed. The Z113A and Z114A cable slack managers are 32 inches (81.3 cm) wide by 38 inches (96.5 cm) deep by 5 inches (12.7 cm) high.

Dress Panels

Z725A dress panels are located in front of the lower part of the cross-connect field. These panels provide cable access between the cable slack manager and cross-connect modules. Each panel is 32 inches (81.3 cm) wide and is held in place by Z2A brackets.

The *Service Manual AT&T System 75 Installation and Test*, 555-200-104IS, provides detailed information on the cross-connect field.

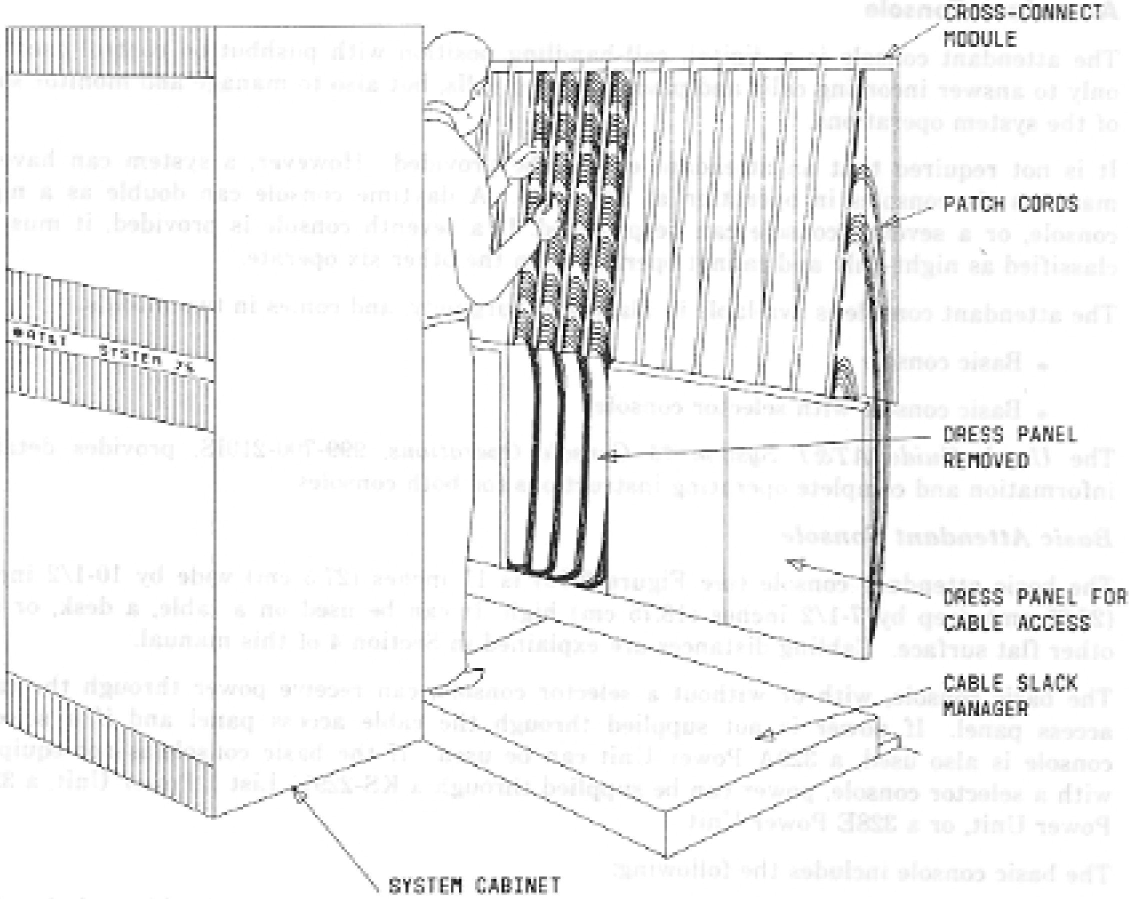


Figure 2-9. Typical Cross-Connect Field Installation—5-Carrier Cabinet System

Auxiliary Cabinet

A J58886N auxiliary cabinet can be provided to contain the equipment required for features such as Recorded Announcement, Loudspeaker Paging Access, Music-on-Hold, and Recorded Telephone Dictation Access. It is 70 inches high by 32 inches wide by 24 inches deep (175 cm by 80 cm by 60 cm) including the door.

Peripheral Equipment

Peripheral equipment is any equipment that can be connected to the system switch. The following equipment is described in this part:

- Attendant Console
- Optional Selector Console
- Voice Terminals
- Voice Terminal Adjuncts
- Data Modules

- Business Communications Terminals

Attendant Console

The attendant console is a digital, call-handling position with pushbutton control used not only to answer incoming calls and place outgoing calls, but also to manage and monitor some of the system operations.

It is not required that an attendant console be provided. However, a system can have as many as six consoles in operation at any time. A daytime console can double as a night console, or a seventh console can be provided. If a seventh console is provided, it must be classified as night-only and cannot operate when the other six operate.

The attendant console is available in black and burgundy, and comes in two models:

- Basic console
- Basic console with selector console

The *User's Guide AT&T System 75 Console Operations*, 999-700-210IS, provides detailed information and complete operating instructions for both consoles.

Basic Attendant Console

The basic attendant console (see Figure 2-10) is 11 inches (27.5 cm) wide by 10-1/2 inches (26.25 cm) deep by 7-1/2 inches (18.75 cm) high. It can be used on a table, a desk, or any other flat surface. Cabling distances are explained in Section 4 of this manual.

The basic console, with or without a selector console, can receive power through the cable access panel. If power is not supplied through the cable access panel and if a selector console is also used, a 329A Power Unit can be used. If the basic console is not equipped with a selector console, power can be supplied through a KS-22911 List 1 Power Unit, a 329A Power Unit, or a 328E Power Unit.

The basic console includes the following:

- Two Receiver or Headset Jacks—These jacks, located on the left side and the right side of the console, connect a receiver or a headset. The receiver cradle can be moved from one side to the other simply by unscrewing the knob and moving the cradle to the opposite side.
- Selector Console Jack—This jack, located on the bottom of the console, connects the optional selector console.
- Lamp Test Switch—This switch tests the lamps on the basic console and the optional selector console.
- Three Audible Tone Volume Control Switches—These switches adjust the volume of the alerting, calls waiting, and timed reminder tones.
- Alphanumeric Display Area—This display shows call-related information and optional personal-service information. Eight buttons and associated status lamps in this area are used to change the display mode. (Refer to the Attendant Display feature in Section 5 of this manual.)
- Call Processing Area—This area contains the following buttons and lamps:
  - Pushbuttons for touch-tone dialing.
  - Start, Release, and Cancel buttons—Used for call processing.

— Alarm-Acknowledge (Alm-Ack) lamps—The alarm lamp (left lamp) lights when a system alarm is detected. Both lamps light when the remote maintenance center (INADS) is notified. The Ack light flashes if the system was unable to notify the remote maintenance center. Both lamps are dark when the alarm condition is clear or when an alarm does not exist.

— Two Calls Waiting lamps—These lamps light when calls in the attendant queue are waiting to be processed. The left lamp lights when at least one call is waiting to be answered. The right lamp lights when the calls waiting exceed the limit preset by the client for the system.

— Position Available lamp—This lamp lights when the console is available for calls and goes dark when the console is not available.

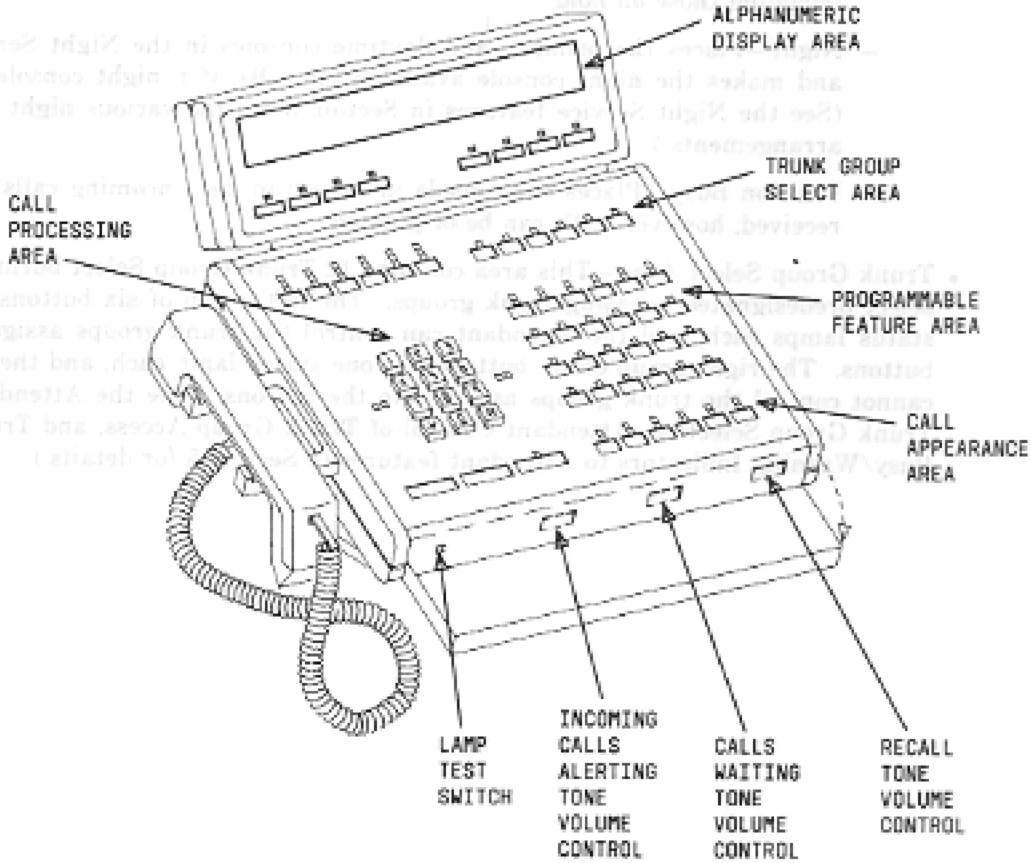


Figure 2-10. Basic Attendant Console

- Call Appearance Area—This area contains 6 buttons labeled a through f with 12 associated status lamps. The buttons are used to answer incoming calls or to originate calls. The lamps show the status of the call appearance.

- Programmable Feature Area—This area contains 24 feature buttons. The topmost 6 buttons have 2 status lamps each and the other 18 buttons have 1 status lamp each. Even though the 6 topmost buttons have 2 status lamps each, only the bottom row of lamps light. The top row of lamps is reserved for future use. The following 5 buttons are preset and the other 19 are programmable.

  — Split—Reconnects the attendant to a call that was split from the console but not from the system.

  — Hold—Places a call on hold.

  — Forced Release—Releases the attendant and disconnects all parties on a call, including those on hold.

  — Night—Places the primary and daytime consoles in the Night Service mode and makes the night console available for calls, if a night console is active. (See the Night Service features in Section 5 for the various night answering arrangements.)

  — Position Busy—Places the console in a busy mode. Incoming calls cannot be received; however, calls can be originated.

- Trunk Group Select Area—This area contains 12 Trunk Group Select buttons used to select predesignated outgoing trunk groups. The left group of six buttons has three status lamps each, and the attendant can control the trunk groups assigned to the buttons. The right group of six buttons has one status lamp each, and the attendant cannot control the trunk groups assigned to the buttons. (See the Attendant Direct Trunk Group Selection, Attendant Control of Trunk Group Access, and Trunk Group Busy/Warning Indicators to Attendant features in Section 5 for details.)

Optional Selector Console

The selector console (see Figure 2-11) is 8-1/2 inches (21.25 cm) wide by 8-3/4 inches (21.88 cm) deep by 4-3/4 inches (11.88 cm) high. The selector console is located adjacent to, and receives power from, the attendant console.

The optional selector console provides the Direct Extension Selection (DXS) With Busy Lamp Field (BLF) feature (see Section 5 of this manual). This feature provides the attendant with a visual indication of the active or idle status of the extension numbers assigned to the system. When a multi-appearance voice terminal user is active on a call, the BLF lamp will light even though other call appearances are available for incoming calls.

This feature also allows the attendant to place calls to system users by pressing a particular Group Select button and a DXS button.

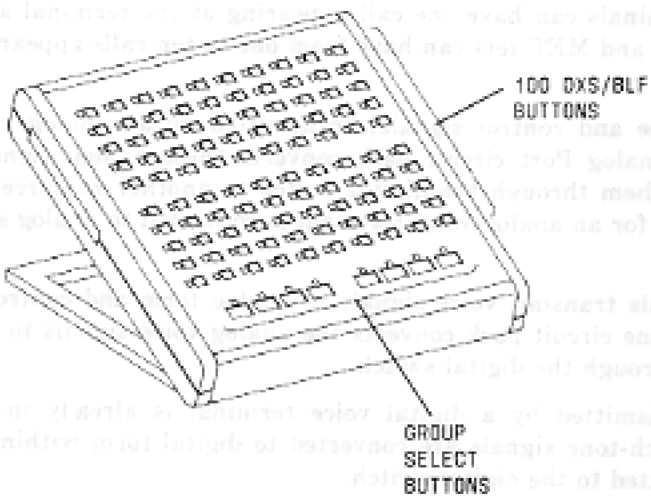


Figure 2-11. Selector Console

Voice Terminals

Voice terminals combine the capabilities of both telephone and computer and have a variety of controlling and monitoring capabilities. While providing basic telephone service (placing and answering calls), voice terminals can also be used to activate the advanced features of the system.

The following types of voice terminals can be used in the system:

- Single-line analog Models 2500, 2554, 7101A, 7103A Fixed Feature, and 7103A Programmable
- Multi-appearance hybrid Models 7303S and 7305S
- Multi-appearance digital Models 7403D and 7405D
- Multibutton Electronic Telephone (MET) sets

Single-line voice terminals can have one call appearing at the terminal at any time. Multi-appearance terminals and MET sets can have from one to ten calls appearing at the terminal at the same time.

All information (voice and control signals) transmitted by an analog voice terminal is in analog form. The Analog Port circuit pack converts these analog signals to digital form before transmitting them through the digital switch to another resource. Similarly, digital information intended for an analog voice terminal is converted to analog signals before being sent to the terminal.

Hybrid voice terminals transmit voice signals in analog form and control signals in digital form. The Hybrid Line circuit pack converts the analog voice signals to digital form before transmitting them through the digital switch.

All information transmitted by a digital voice terminal is already in digital form. The analog voice and touch-tone signals are converted to digital form within the voice terminal before being transmitted to the digital switch.

Maximum cabling distances from the switch for voice terminals are explained in Section 4 of this manual.

Single-Line Voice Terminals

- Models 2500 and 2554

  These single-line analog voice terminals (see Figure 2-12) are available in white, black, light beige, aqua blue, pastel yellow, red, moss green, and ivory. Model 2500 is freestanding and Model 2554 is wall-mounted. Each model comes equipped with the following:

  - Handset
  - Pushbuttons for touch-tone dialing
  - Volume control for the ringer

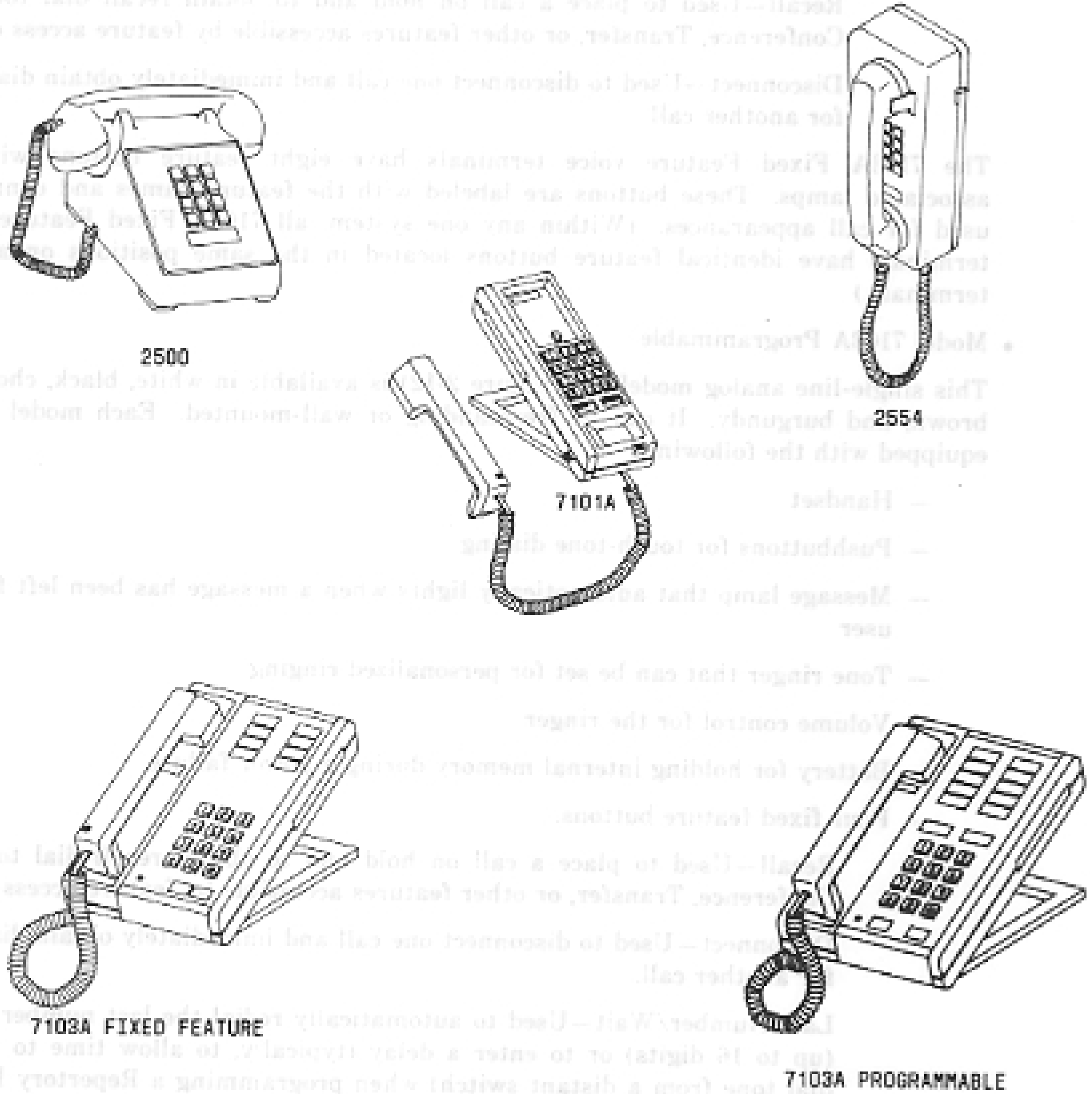


Figure 2-12. Single-Line Voice Terminal

- Models 7101A and 7103A Fixed Feature

  These single-line analog models (see Figure 2-12) are available in white, black, chocolate brown, and burgundy. They can be freestanding or wall-mounted. Each model comes equipped with the following:

  — Handset

  — Pushbuttons for touch-tone dialing

  — Message lamp that automatically lights when a message has been left for the user

  — Tone ringer

  — Volume control for the ringer

— Two fixed feature buttons:

Recall—Used to place a call on hold and to obtain recall dial tone for Conference, Transfer, or other features accessible by feature access code.

Disconnect—Used to disconnect one call and immediately obtain dial tone for another call.

The 7103A Fixed Feature voice terminals have eight feature buttons with no associated lamps. These buttons are labeled with the feature names and cannot be used for call appearances. (Within any one system, all 7103A Fixed Feature voice terminals have identical feature buttons located in the same positions on all the terminals.)

- Model 7103A Programmable

This single-line analog model (see Figure 2-12) is available in white, black, chocolate brown, and burgundy. It can be freestanding or wall-mounted. Each model comes equipped with the following:

— Handset

— Pushbuttons for touch-tone dialing

— Message lamp that automatically lights when a message has been left for the user

— Tone ringer that can be set for personalized ringing

— Volume control for the ringer

— Battery for holding internal memory during a power failure

— Four fixed feature buttons:

Recall—Used to place a call on hold and to obtain recall dial tone for Conference, Transfer, or other features accessible by feature access code.

Disconnect—Used to disconnect one call and immediately obtain dial tone for another call.

Last Number/Wait—Used to automatically redial the last number dialed (up to 16 digits) or to enter a delay (typically, to allow time to receive dial tone from a distant switch) when programming a Repertory Dialing button.

Program—Used to program a feature button for feature access or Repertory Dialing. (Repertory Dialing is a feature of the voice terminal, not of System 75. With Repertory Dialing, a user can store in-house or external numbers on one of the programmable feature buttons.)

— Ten programmable feature buttons capable of storing up to 13 digits entered by the user. These numbers can be system feature access codes or Repertory Dialing numbers.

Multi-Appearance Voice Terminals

Multi-appearance 7300 Series hybrid and 7400 Series digital voice terminals (see Figure 2-13) come in white, black, chocolate brown, burgundy, charcoal gray, midnight blue, and cinnabar. They can be freestanding or wall-mounted.

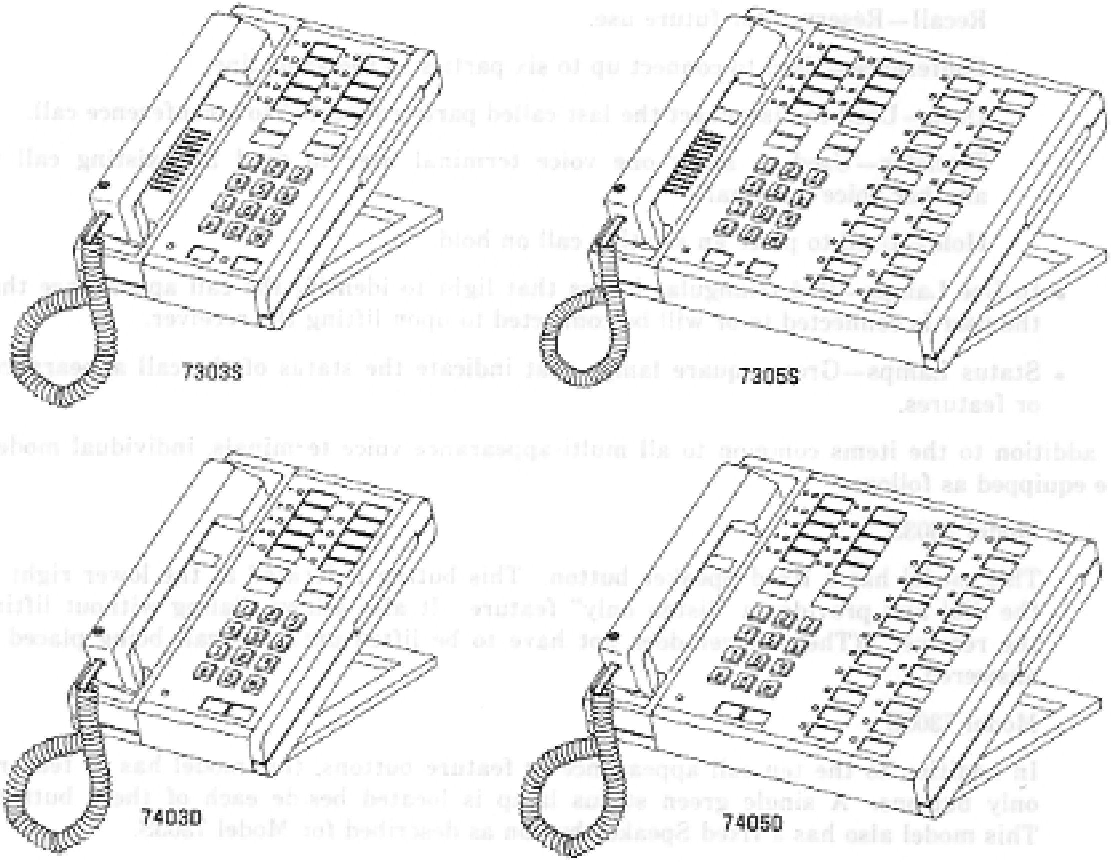


Figure 2-13. 7300 Series and 7400 Series Voice Terminals

All multi-appearance voice terminals have ten call appearance buttons. System 75 requires that two, minimum, must be assigned as such. These buttons have an associated red in-use lamp and a green status lamp located to the left of each button. If these buttons are not assigned for call appearances, they can be assigned for feature activation. (If a feature is assigned to a button, only the green status lamp functions.)

All multi-appearance digital and hybrid voice terminals are equipped with the following:

- Handset
- Pushbuttons for touch-tone dialing
- Message lamp that automatically lights when a message has been left for the user
- Tone ringer
- Volume control for the ringer
- Ten buttons used to designate features or multiple call appearances of the same extension

Note: At least two of these ten buttons are for call appearances.

- Five fixed feature buttons:

  Recall—Reserved for future use.

  Conference—Used to connect up to six parties to the same line.

  Drop—Used to disconnect the last called party connected to a conference call.

  Transfer—Used to allow one voice terminal user to send an existing call to another voice terminal

  Hold—Used to place an existing call on hold.

- In-Use Lamps—Red triangular lamps that light to identify the call appearance that the user is connected to or will be connected to upon lifting the receiver.

- Status Lamps—Green square lamps that indicate the status of the call appearances or features.

addition to the items common to all multi-appearance voice terminals, individual models e equipped as follows:

- Model 7303S

  This model has a fixed Speaker button. This button is located at the lower right of the dial and provides a "listen only" feature. It also allows dialing without lifting the receiver. (The receiver does not have to be lifted until the call being placed is answered.)

- Model 7305S

  In addition to the ten call appearance or feature buttons, this model has 24 feature-only buttons. A single green status lamp is located beside each of these buttons. This model also has a fixed Speaker button as described for Model 7303S.

- Model 7403D

  This model has a fixed Disconnect button located at the lower right of the dial. The Disconnect button disconnects one call and immediately obtains dial tone for another call. This voice terminal can be used with a Digital Terminal Data Module (see Data Communications Equipment) to provide data service.

- Model 7405D

  In addition to the ten call appearance or feature buttons, this model has 24 feature-only buttons. A single green status lamp is located beside each of these buttons. This model has a fixed Disconnect button as described for the Model 7403D. Model 7405D can be used with the following modules (see Voice Terminal Adjuncts and Data Communications Equipment):

  — Digital Terminal Data Module

  — Digital Display Module

  — Function Key Module

  — Call Coverage Module

Multibutton Electronic Telephone (MET) Sets

The MET sets (see Figure 2-14) are multi-appearance voice terminals. They have snap-in faceplates that are available in seven colors and two woodgrain finishes.

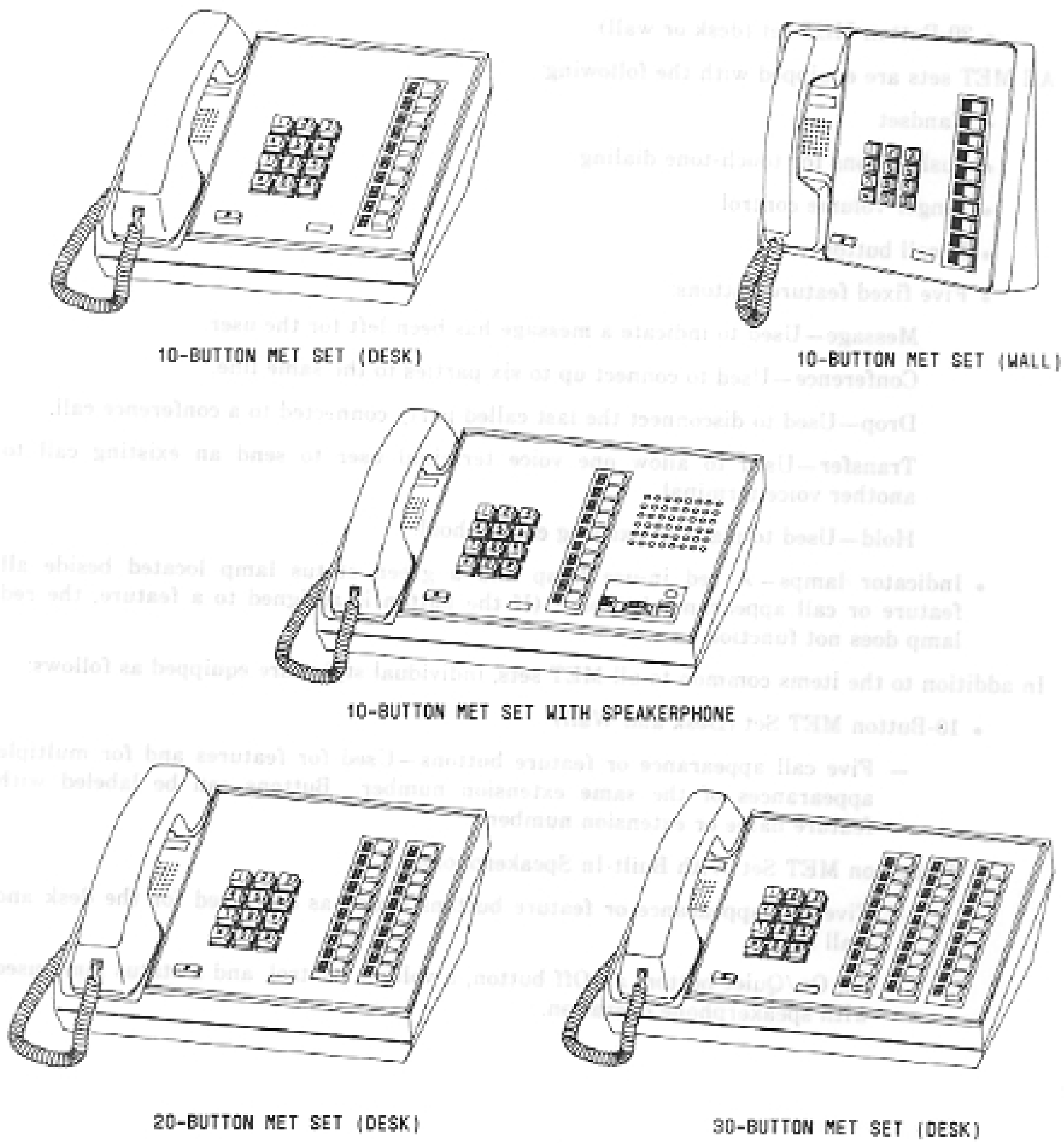


Figure 2-14. Multibutton Electronic Telephone (MET) Sets

MET sets are available in the following styles:

- 10-Button MET set (desk or wall)
- 10-Button MET set with built-in speakerphone (desk)
- 20-Button MET set (desk or wall)
- 30-Button MET set (desk or wall)

All MET sets are equipped with the following:

- Handset
- Pushbuttons for touch-tone dialing
- Ringer volume control
- Recall button
- Five fixed feature buttons:

  Message—Used to indicate a message has been left for the user.

  Conference—Used to connect up to six parties to the same line.

  Drop—Used to disconnect the last called party connected to a conference call.

  Transfer—Used to allow one voice terminal user to send an existing call to another voice terminal.

  Hold—Used to place an existing call on hold.

- Indicator lamps—A red in-use lamp and a green status lamp located beside all feature or call appearance buttons. (If the button is assigned to a feature, the red lamp does not function.)

In addition to the items common to all MET sets, individual styles are equipped as follows:

- 10-Button MET Set (Desk and Wall)
  - Five call appearance or feature buttons—Used for features and for multiple appearances of the same extension number. Buttons can be labeled with feature name or extension number.
- 10-Button MET Set With Built-In Speakerphone
  - Five call appearance or feature buttons (same as described for the desk and wall set).
  - An On/Quiet button, an Off button, a volume control, and a status lamp used with speakerphone operation.

- 20-Button MET Set

  — Ten call appearance and feature buttons—Used for features or for multiple appearances of the same extension number. Buttons can be labeled with feature name or extension number.

  — Five buttons that can be used for features only.

- 30-Button MET Set

  — Ten call appearance or feature buttons (same as described for the 20-button set).

  — Fifteen buttons that can be used for features only.

Voice Terminal Adjuncts

The adjuncts discussed here are used with various voice terminal models. These adjuncts provide additional call appearances or feature buttons and alphanumeric display, headset, and speakerphone options.

Digital Terminal Data Module

This module can be used with Model 7403D and 7405D voice terminals to provide data service. The Digital Terminal Data Module is discussed under the heading ***Data Communications Equipment*** in this section.

Digital Display Module

This module (see Figure 2-15) can be used with Model 7405D voice terminals. The 40-character alphanumeric display can be used to display call-related and personal-service information. (Refer to the Voice Terminal Display feature in Section 5 of this manual.)

The Digital Display Module is powered from a 329A Power Unit or a KS-22911 List 1 Power Unit plugged into a 120-volt ac receptacle.

A Digital Display Module and a Call Coverage Module cannot be used on the same voice terminal.

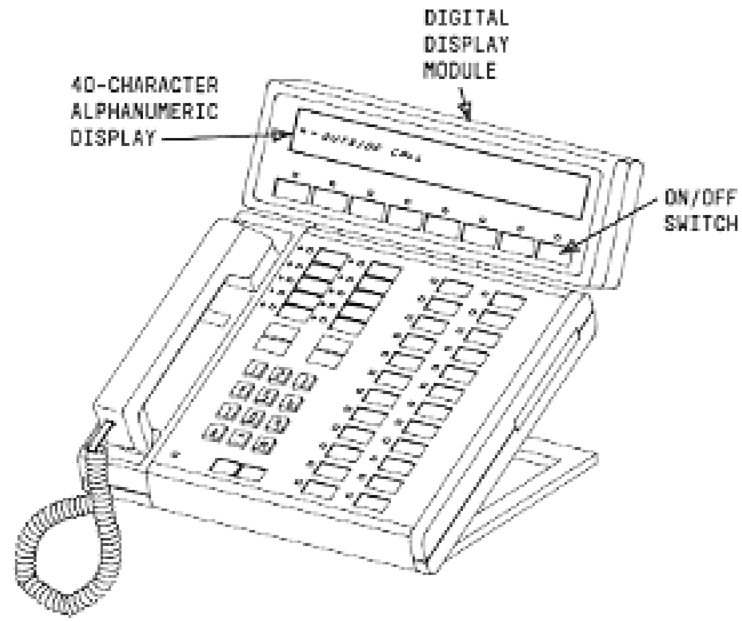


Figure 2-15. Model 7405D Voice Terminal With Optional Digital Display Module

Function Key Module

This module (see Figure 2-16) can be used with Model 7405D voice terminals. It provides 24 additional feature buttons. Each button has a status lamp.

The Function Key Module is powered from a 329A Power Unit or a KS-22911 List 1 Power Unit plugged into a 120-volt ac receptacle.

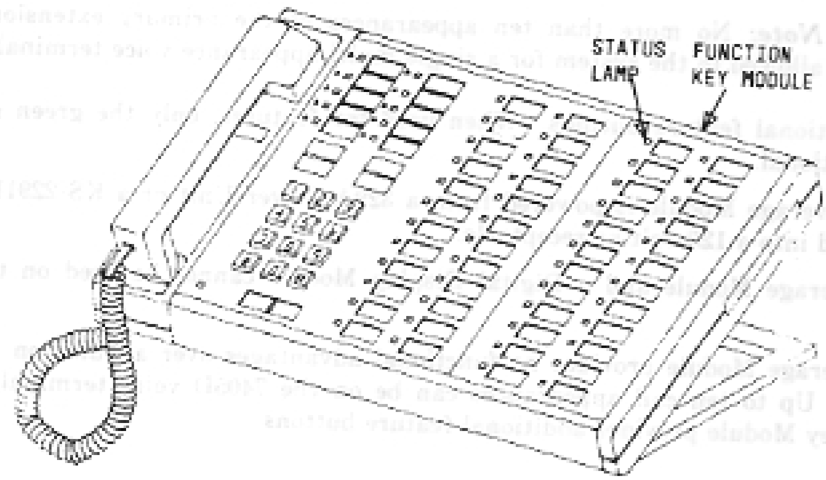


Figure 2-16. Model 7405D Voice Terminal With Optional Function Key Module

Call Coverage Module

This module (see Figure 2-17) can be used with Model 7405D voice terminals. It provides 20 call appearance or feature buttons to supplement those on the terminal. Each button has two lamps, a red triangular lamp (in-use) and a green square lamp (status). These buttons can be used as follows:

- Call appearances of the primary extension number assigned to the voice terminal.

  Note: No more than ten appearances of the primary extension number are allowed in the system for a single multi-appearance voice terminal.

- Additional feature buttons. When used for features, only the green status lamp is functional.

The Call Coverage Module is powered from a 329A Power Unit or a KS-22911 List 1 Power Unit plugged into a 120-volt ac receptacle.

A Call Coverage Module and a Digital Display Module cannot be used on the same voice terminal.

A Call Coverage Module provides no functional advantages over a Function Key Module in System 75. Up to ten call appearances can be on the 7405D voice terminal itself and the Function Key Module provides additional feature buttons.

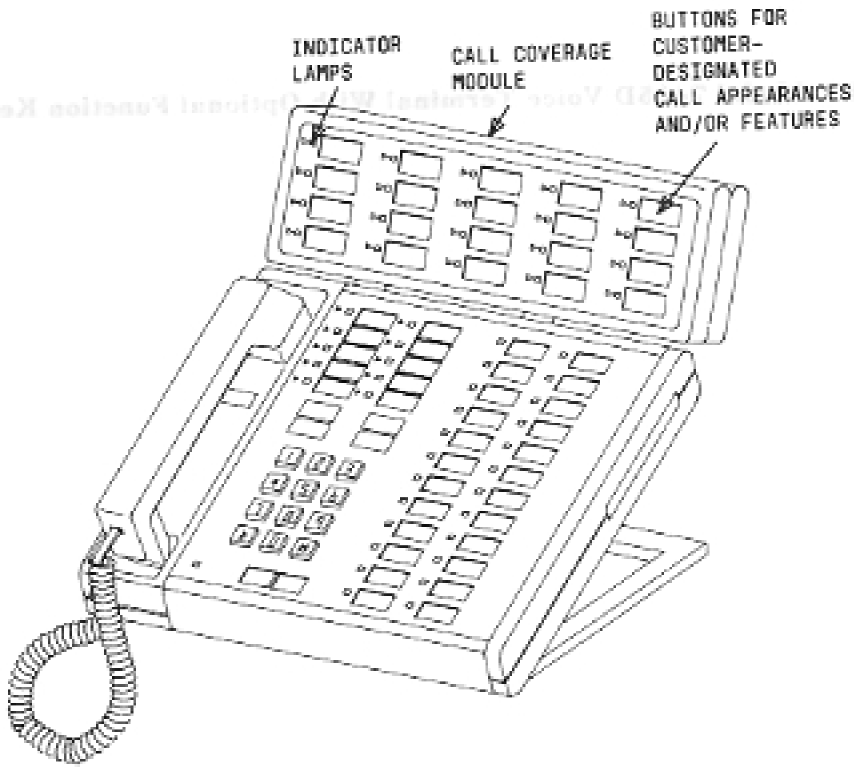


Figure 2-17. Model 7405D Voice Terminal With Optional Call Coverage Module

Headset Adapter Module

Headset adapters (see Figure 2-18) are available optionally for use with five voice terminal models and the 515 Business Communications Terminal (BCT). The 500A Headset Adapter can be used with Models 7103A, 7403D, and 7405D and the 515 BCT. The 502A Headset Adapter can be used with Models 7303S and 7305S.

Headset adapters are equipped with an 18-inch (45.7-cm) connecting cord. Connecting cords are available optionally in lengths of 4 feet and 14 feet (1.22 m and 4.3 m).

The 500A Headset Adapter can be powered remotely by a 329A Power Unit or a KS-22911 List 1 Power Unit plugged into a 120-volt ac receptacle. (These power units may also power one more adjunct.) The 500A Headset Adapter can be powered locally by a 2012D Transformer plugged into a 120-volt ac receptacle.

The 502A Headset Adapter does not require supplemental power.

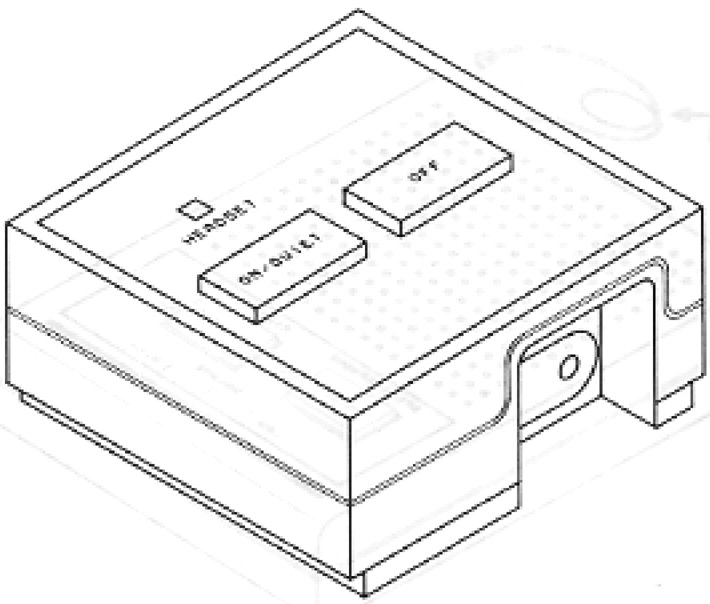


Figure 2-18. 500A or 502A Headset Adapter

Speakerphone

Speakerphones (see Figure 2-19) are available optionally for use with five voice terminal models and the 515 BCT. The S101A Speakerphone can be used with Models 7103A, 7403D, and 7405D and the 515 BCT. The S102A-185 Speakerphone can be used with Models 7303S and 7305S.

Speakerphones are equipped with a 4-foot (1.2-m) connecting cord that plugs into the voice terminal. Connecting cords are available optionally in lengths of 18 inches and 14 feet (45.7 cm and 4.3 m).

The S101A Speakerphone can be powered remotely by a 329A Power Unit or a KS-22911 List 1 Power Unit plugged into a 120-volt ac receptacle. (These power units may also power one more adjunct.) The S101A Speakerphone can be powered locally by a 2012D Transformer plugged into a 120-volt ac receptacle.

The S102A-185 Speakerphone does not require supplemental power.

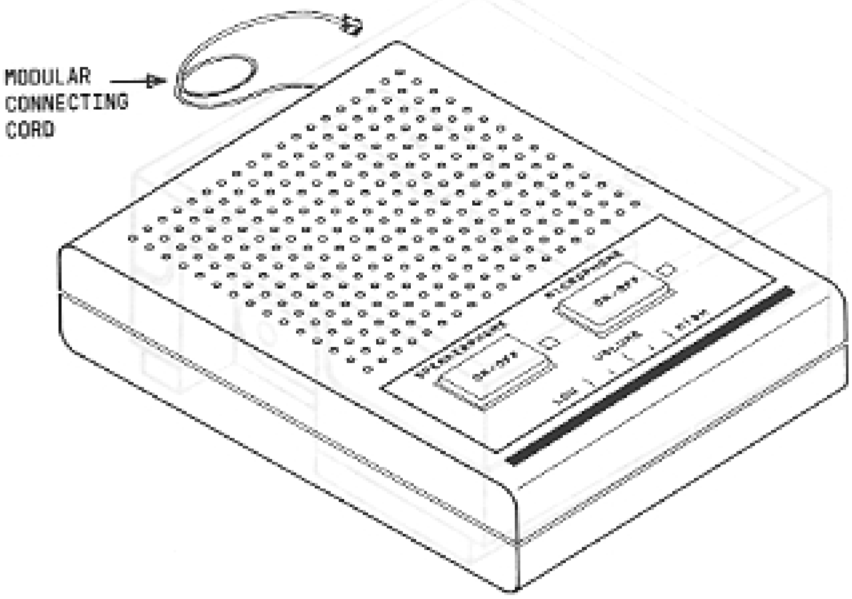


Figure 2-19. S101A or S102A-185 Speakerphone

Data Modules

Data modules provide an interface between the digital switch, Data Terminal Equip (DTE), and Data Communications Equipment (DCE). DTE is equipment that provides data source, termination, or both—a host computer or a data terminal is an example of D'. DCE is equipment that provides the functions required to establish, maintain, and termina.. a data call—a modem is an example of DCE.

Both sides of a data call require DCE and DTE. Thus, a host computer (DTE) connected to a DCE-type data module would meet the requirement on one side of a data call. If the requirement is met on the other side of the data call, the call is connected through the digital switch. The digital switch provides a Data Communications Protocol (DCP) interface to data modules on both sides of the call, and vice versa. Since the two sides of the call are compatible, DCP to DCP, the call is established.

The preceding paragraph is intended to provide a basis for understanding data modules, not to provide a synopsis of System 75 data handling and switching capabilities. See Section 6 of this manual for a detailed description of data management capabilities.

The DCE and DTE interconnect via an RS232C interface. The RS-232C interface is transparent to the code being used.

All data modules contain several option switches that are set to match the data equipment. These options are as follows:

- Synchronous or asynchronous operation
- Half- or full-duplex operation
- Standard data rates of 300 bps, 1.2 Kbps, 2.4 Kbps, 4.8 Kbps, 9.6 Kbps, and 19.2 Kbps
- Nonstandard asynchronous data rates below 1800 bps (low)
- Internal or external timing
- Parity—even, odd, or none

The data modules also provide several lamps that display operating status and test results.

The following data modules are available with the system:

- Digital Terminal Data Module (DTDM)
- Processor Data Module (PDM)
- Trunk Data Module

Digital Terminal Data Module (DTDM)

The DTDM provides a DCE interface for connection to data terminals. The DTDM is an add-on module that physically attaches to a Model 7403D or 7405D voice terminal (see Figure 2-20). The DTDM and digital voice terminal integrate data and voice into the DCP. Maximum cabling distances from the switch are explained in Section 4 of this manual.

The DTDM is contained in a molded plastic housing styled to match the voice terminals. It is 1-3/4 inches (4.4 cm) high by 5 inches (12.7 cm) wide by 8-1/2 (2.3 cm) inches deep and weighs about 1-3/4 pounds (3.86 kg).

The DTDM is powered from a 329A Power Unit or a KS-22911 List 1 Power Unit plugged into a 120-volt ac receptacle.

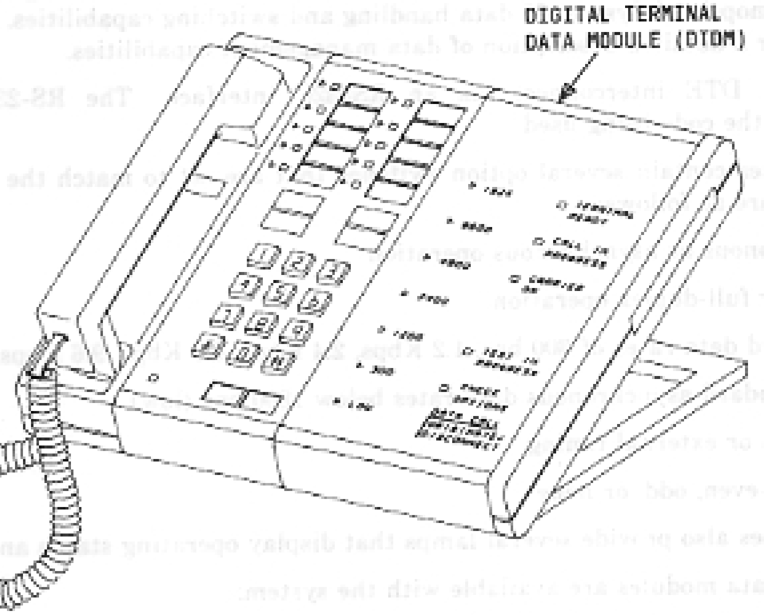


Figure 2-20. Model 7403D Voice Terminal With Optional Digital Terminal Data Module

Processor Data Module (PDM)

The PDM (see Figure 2-21) provides a DCE interface for connection to data terminals, Station Message Detail Recording (SMDR) output device, on-premises administration terminal, Applications Processor (AP), and host computers. It also provides a DCP interface for connection to the digital switch.

Maximum cabling distances from the switch are explained in Section 4 of this manual.

The PDM can be configured either in a stand-alone or multiple mount. The stand-alone version is installed in an aluminum housing that is equipped with plastic front and rear covers. Up to eight data modules (PDMs or Trunk Data Modules) may be installed in a multiple mounting.

The stand-alone version can be mounted on a surface or desk top near the associated data terminal. The multiple mount is normally contained in an auxiliary cabinet. Both mounting arrangements require power from a 120-volt ac receptacle to power the data module(s).

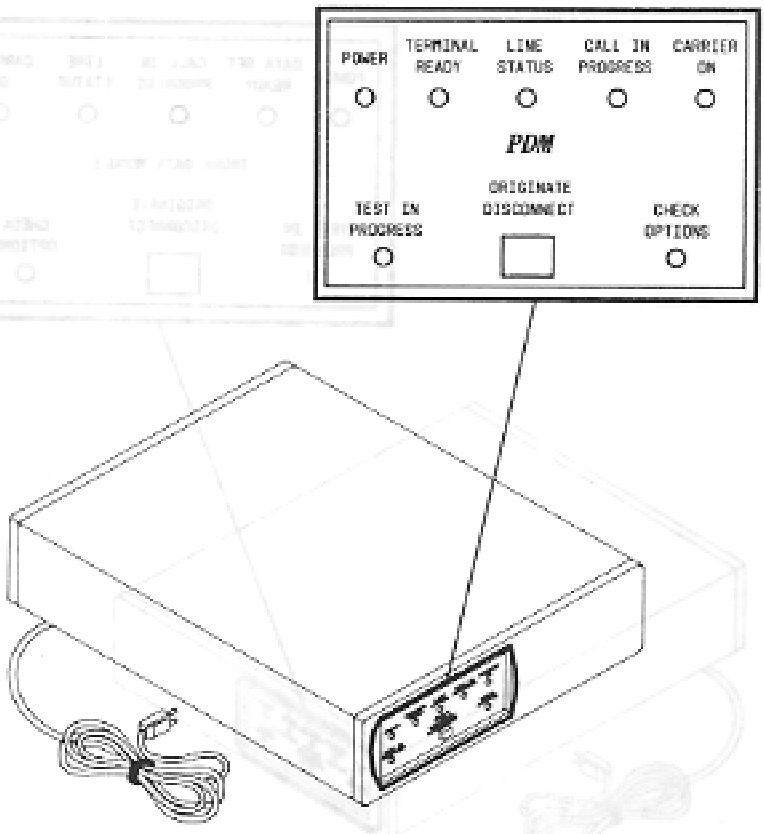


Figure 2-21. Processor Data Module (Stand-Alone)

Trunk Data Module

The Trunk Data Module (see Figure 2-22) provides an Electronic Industries Association (EIA) RS-232C DTE interface for connection to off-premises private line trunk facilities and a DCP interface for connection to the digital switch.

Maximum cabling distances from the switch are explained in Section 4 of this manual.

The Trunk Data Module can be configured either in a stand-alone or in a multiple mount. The stand-alone version is contained in an aluminum housing that is equipped with plastic front and rear covers. Up to eight data modules (Trunk Data Modules or PDMs) may be installed in a multiple mounting.

The stand-alone version can be mounted on a surface or desk top near the associated data equipment. The multiple-mounting carrier unit is normally installed in an auxiliary cabinet. Both mounting arrangements require power from a 120-volt ac receptacle to power the data module(s).

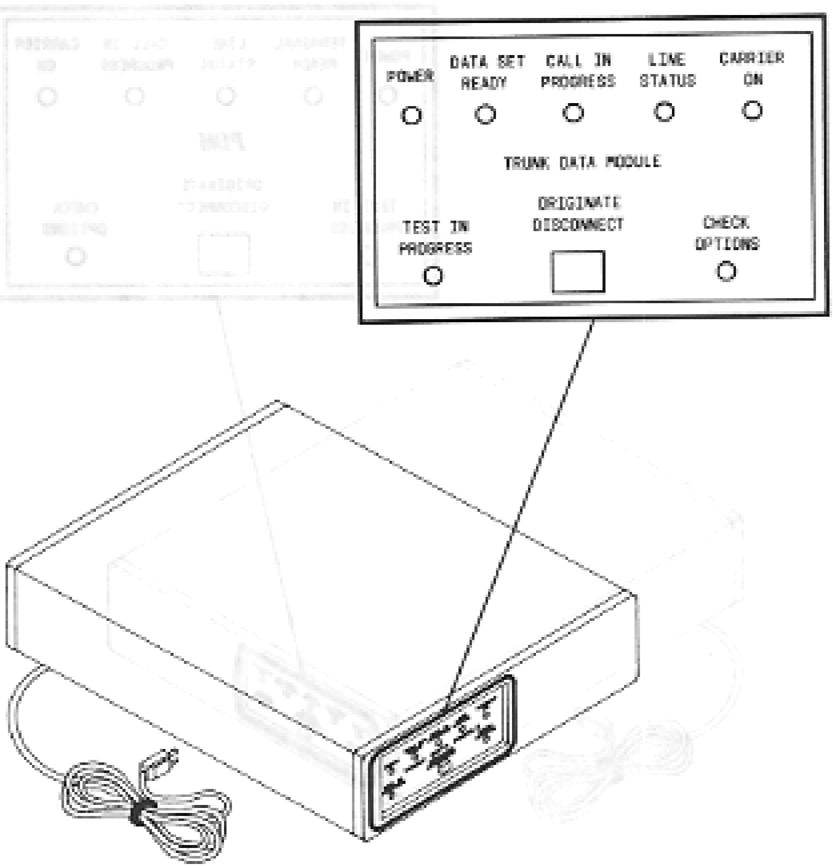


Figure 2-22. Trunk Data Module (Stand-Alone)

513 BCT

The 513 BCT (see Figure 2-23) is a general purpose asynchronous data terminal that provides a standard EIA RS-232C interface for data entry and retrieval. The 513 BCT is the dedicated System Access Terminal (SAT) for administrative and maintenance functions. When not used as a dedicated SAT, the 513 BCT can be connected through a Digital Terminal Data Module or a Processor Data Module and can serve as a remote on-premises SAT.

Maximum cabling distances from the switch are explained in Section 4 of this manual.

The 513 BCT requires 117-volt 60-hertz commercial power from a 3-wire grounded outlet. Power for the dedicated SAT can be provided by an outlet located at the bottom rear of the system cabinet or any other convenient outlet. This terminal can be used on a table or desk. It requires approximately 4 square feet (0.37 m) of space.

The 513 BCT operates in alphanumeric and graphic character set mode. It consists of the following:

- Video Display—The display is a 12-inch (30.48 cm) (diagonal measurement) screen that features inverse video, blinking, underline, and full or half intensity.

  The data portion of the screen displays 24 rows with up to 80 characters per row. Rows 26 and 27 display information associated with programmable function keys. This information is displayed in inverse video (see Figure 2-24).

- Keyboards—Two keyboards are available, a standard 89-key keyboard and an optional 103-key keyboard (see Figure 2-25). The two keyboards are alike except the 103-key keyboard has a 14-key cluster on the left side. These 14 keys are used only to transmit escape codes. Keyboards are low profile and can be stored under the display portion of the terminal.

- Memory—Up to 72 lines of memory can be displayed. Memory is read through the Scroll Mode or Page Mode. The Scroll Mode displays 24 lines at a time. If a line is scrolled off the screen, it remains in memory and can be retrieved. The Page Mode divides memory into 3 text pages with 24 lines each. Pages are displayed one at a time. A line scrolled off the screen during the Page Mode cannot be retrieved.

The *Technical Description—513 Business Communications Terminal*, 999-700-485IS, contains additional information on this terminal.

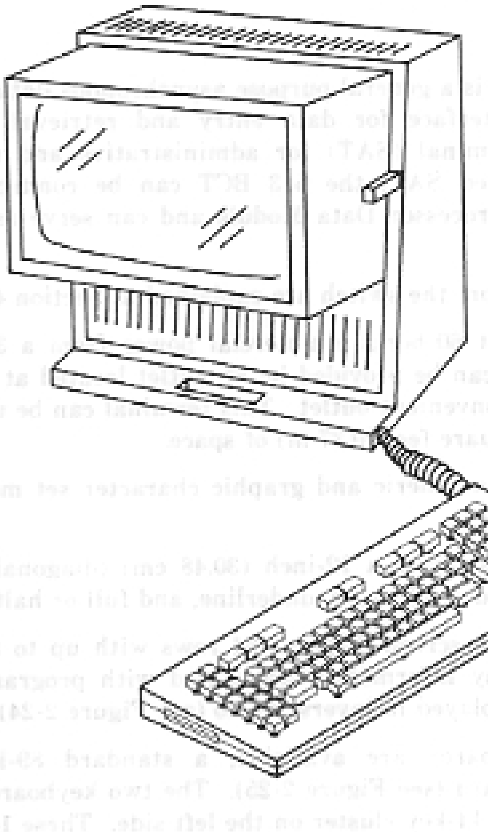


Figure 2-23. 513 BCT With Optional 103-Key Keyboard

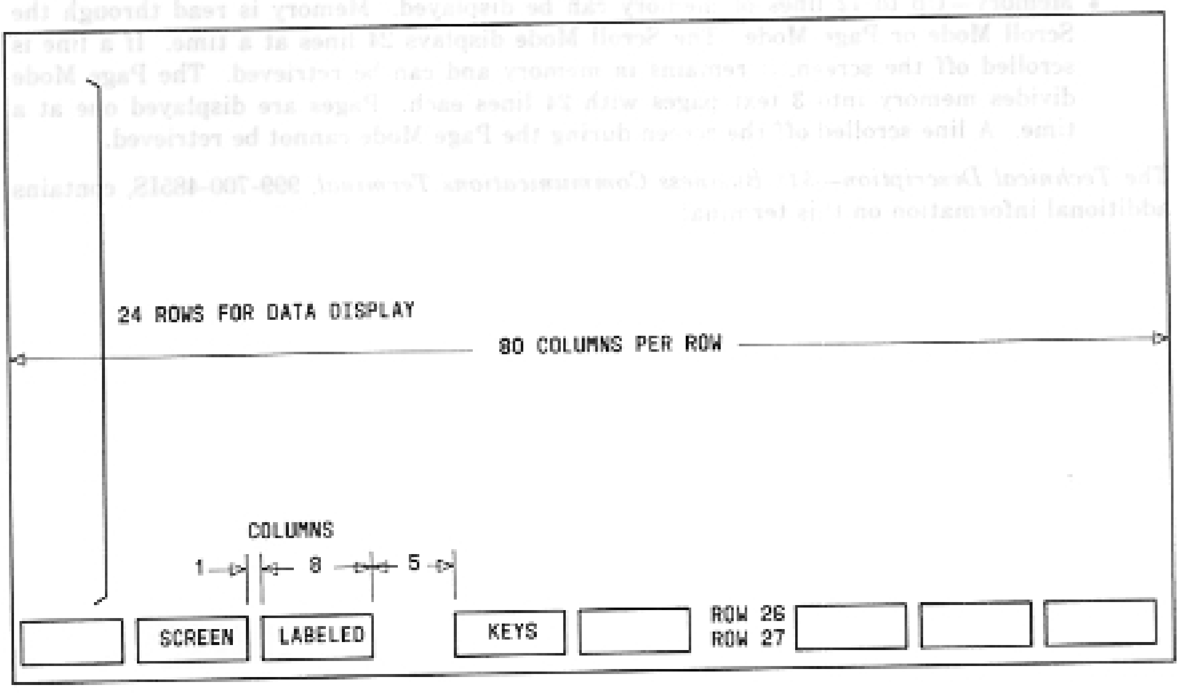
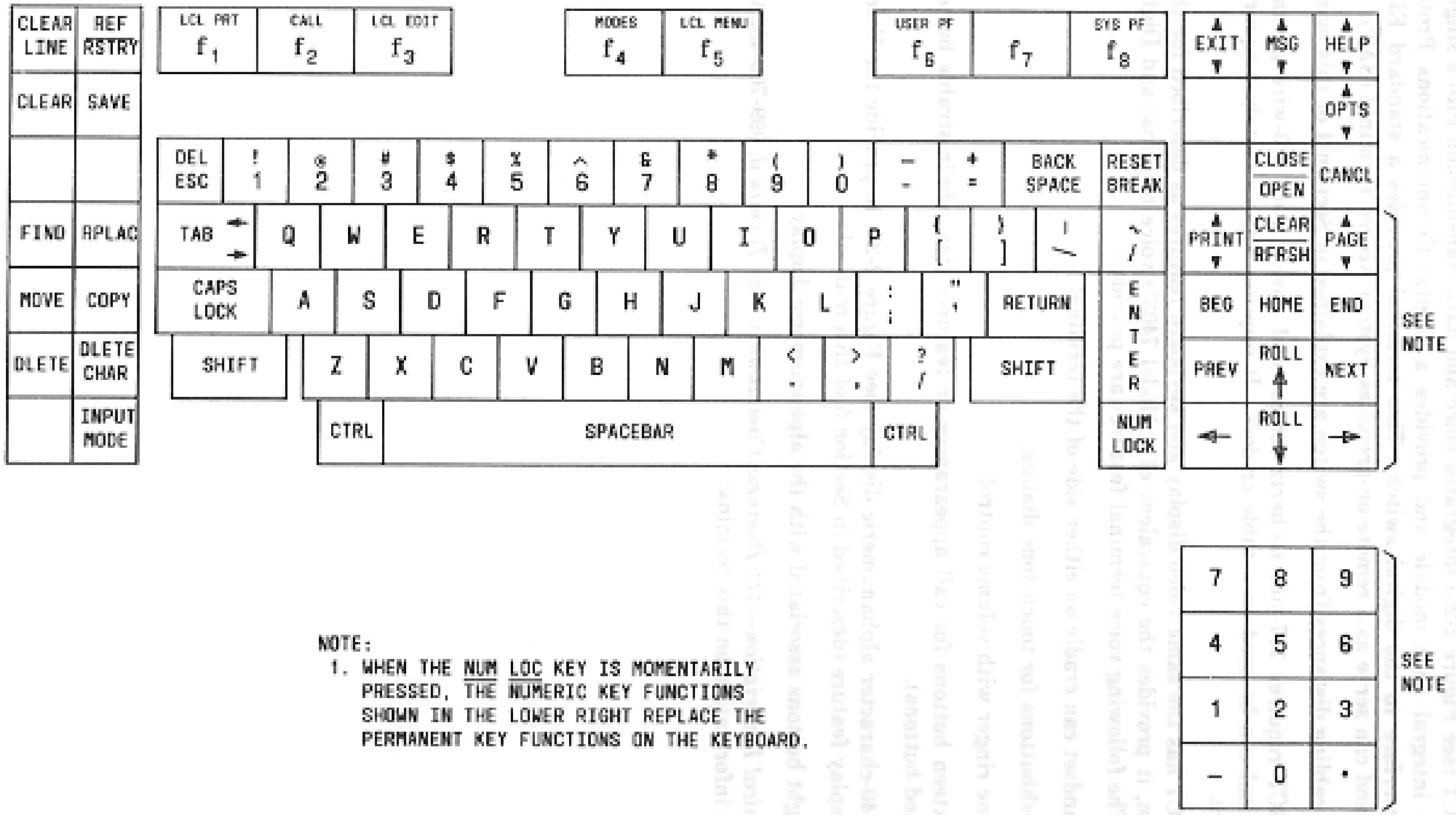


Figure 2-24. 513 BCT Screen Layout



NOTE:

1. WHEN THE NUM LOC KEY IS MOMENTARILY PRESSED, THE NUMERIC KEY FUNCTIONS SHOWN IN THE LOWER RIGHT REPLACE THE PERMANENT KEY FUNCTIONS ON THE KEYBOARD.

Figure 2-25. 515 and 513 BCT 103-Key Keyboard Arrangement

515 BCT

The 515 BCT (see Figure 2-26) integrates voice and data transmission into a single terminal. It has an integral data module and provides a Digital Communications Protocol (DCP) channel interface to the digital switch. This terminal provides a standard EIA RS-232C interface, and can serve as a remote on-premises System Access Terminal (SAT).

Maximum cabling distances from the switch are explained in Section 4 of this manual.

The 515 BCT requires 117-volt 60-hertz commercial power from a 3-wire grounded outlet. This terminal can be used on a table or desk. It requires approximately 4 square feet (0.37 m) of space.

The 515 BCT has the same video display and keyboard features as described for the 513 BCT. In addition, it provides the equivalent of a Model 7403D voice terminal and Digital Display Module. The following voice terminal features are provided:

- Handset can cradle on either side of the terminal.
- Pushbuttons for touch-tone dialing.
- Tone ringer with volume control.
- Sixteen buttons for call appearances or features (ten administrable buttons and six fixed buttons).
- A 40-character alphanumeric display (see Figure 2-27) to provide the Voice Terminal Display feature (described in Section 5 of this manual).
- Eight buttons associated with the alphanumeric display.

The *Technical Description—515 Business Communications Terminal*, 999-700-482AB, contains additional information on this terminal.

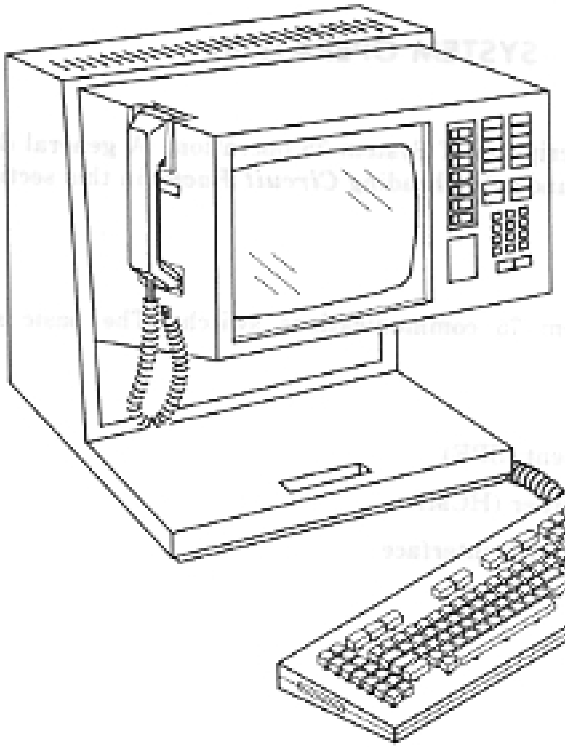


Figure 2-26. 515 BCT With 103-Key Keyboard

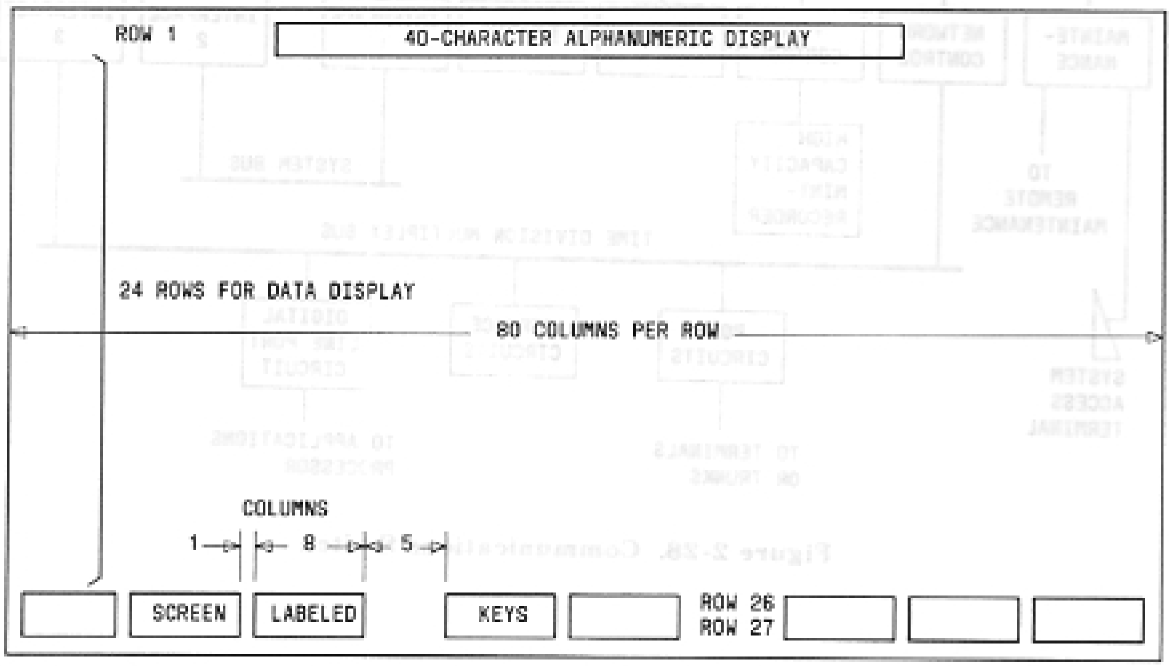


Figure 2-27. 515 BCT Screen Layout

SYSTEM OPERATION

This part provides a brief description of System 75 operation. A general description of the individual circuits is provided under the heading ***Circuit Packs*** in this section.

Switch Hardware

Figure 2-28 shows the System 75 communications switch. The basic switch hardware consists of the following:

- Switching network
- Switch processing element (SPE)
- High capacity minirecorder (HCMR)
- Applications Processor (AP) interface
- Maintenance circuit
- Power system

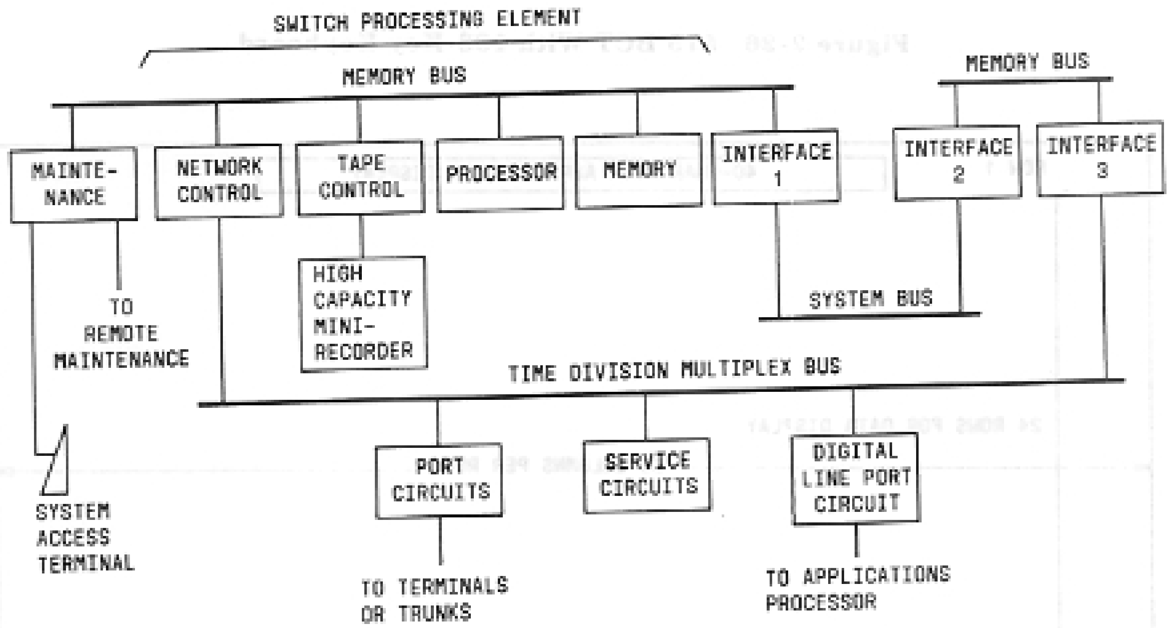


Figure 2-28. Communications Switch

Switching Network

System 75 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
- Service circuits

The port circuits connect external communications facilities to the TDM bus. The TDM bus connects the port circuits to the SPE through the network control circuit. The service circuits provide tone sources, receivers, detectors, and pooled modems.

Time Division Multiplex Bus

The TDM bus consists of two groups of eight signals (A and B buses) and five control lines with matching grounds. The port circuit packs place digitized voice [pulse code modulated (PCM)] signals on the bus. Control channel information is carried on the bus in an 8-bit Control Channel Message Set (CCMS) data signal. The CCMS data signal allows the SPE to communicate with the port circuit packs.

The TDM bus operates at 2.048 MHz. The system framing pulse is 8 kHz. This provides 256 time slots on both the A and B bus. The time slots are 488 nanoseconds wide. Time slots are generated as shown in Figure 2-29.

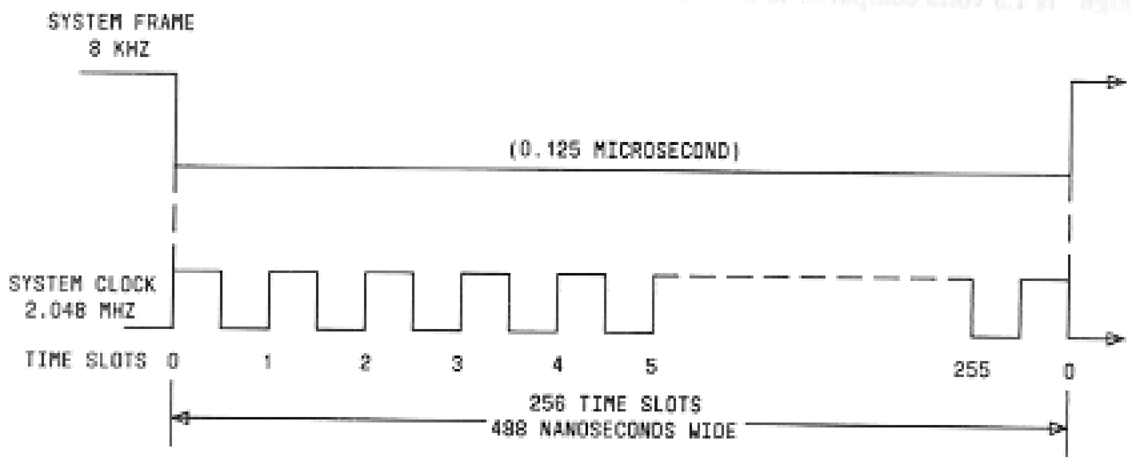


Figure 2-29. TDM Bus Time Slot Generation

Two time slots are required for a 2-party conversation. Each party transmits (talks) on one time slot and receives (listens) on another. The initial system software limits the number on a conference call to six. During a conference connection, each member of the conference transmits on an individual time slot while receiving on as many as five other time slots.

The actual switch capacity is 236 simultaneous conversations, because some slots are reserved for system use. The first five slots on Bus A (00—04) are used for the internal control channel on which the ports talk to the SPE. Seventeen slots on the B bus are used for system tones. Slots 05 through 12 on the A bus and slots 256 through 260 on the B bus are reserved for future use. The last two slots on each bus (254, 255, 510, and 511) are currently not used.

Physical Characteristics: The TDM bus contains two identical 8-bit buses (A and B). The TDM bus snakes continuously through all the carriers within the cabinet as shown in Figure 2-30. The total length is about 15 feet for a system with four port carriers. The bus is driven from any of the circuit pack slots in the carriers. Similarly, a signal on the bus can be received by any circuit pack.

Within a carrier, the bus is printed on one side of the backplane while the other side is solid ground. Between carriers, coaxial cables are used to minimize electromagnetic interference (EMI).

Electrical Characteristics: The TDM bus is an unbalanced, low characteristic impedance transmission line. Paths printed over a ground plane on the carriers and the coaxial cables between carriers maintain this impedance level over the full length of the bus.

Each end of the bus is terminated to ground with a separate resistor for each of the 16 bits. 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 volt.

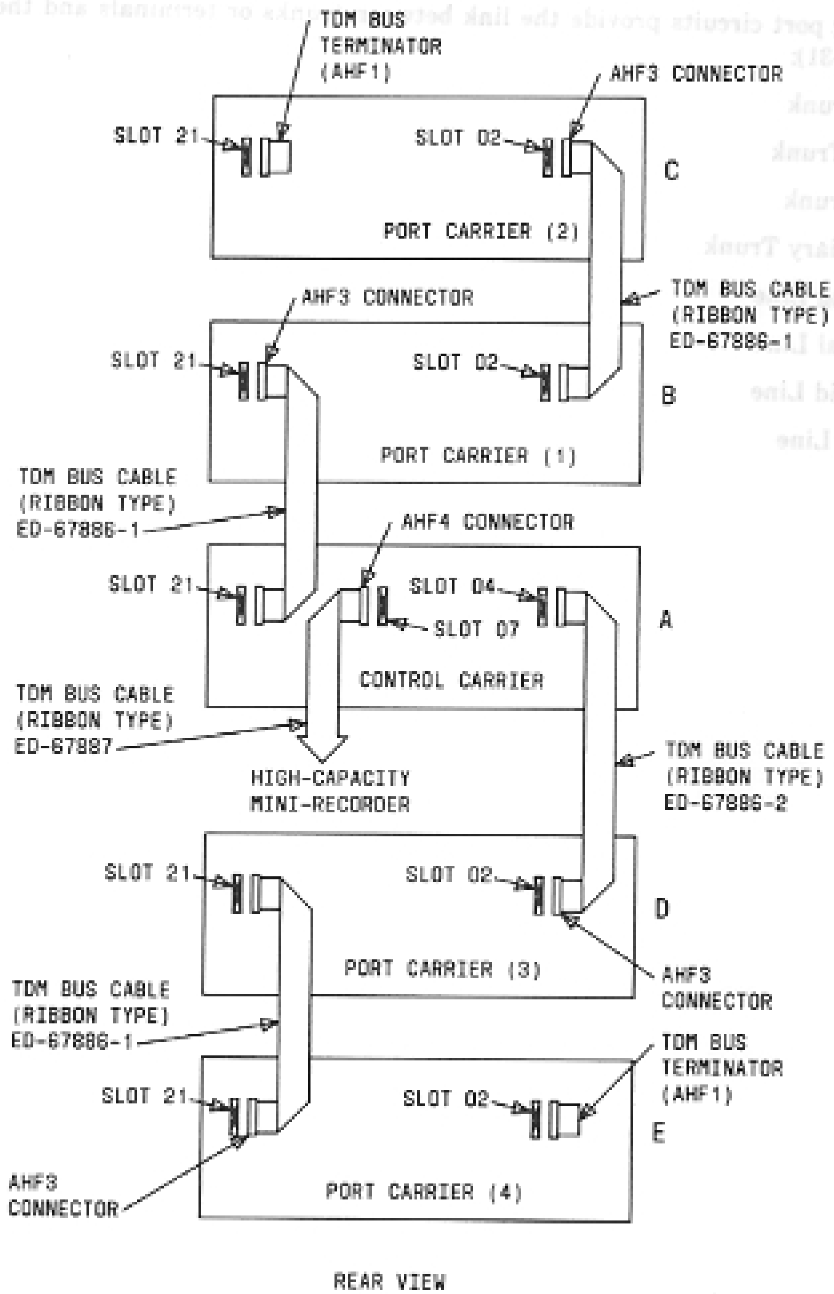


Figure 2-30. TDM Bus Wiring Diagram—Fully Loaded (4-Port Carrier and Control Carrier) Configuration

Port Circuits

The following port circuits provide the link between trunks or terminals and the TDM buses (see Figure 2-31):

- CO Trunk
- DID Trunk
- Tie Trunk
- Auxiliary Trunk
- Analog Line
- Digital Line
- Hybrid Line
- MET Line

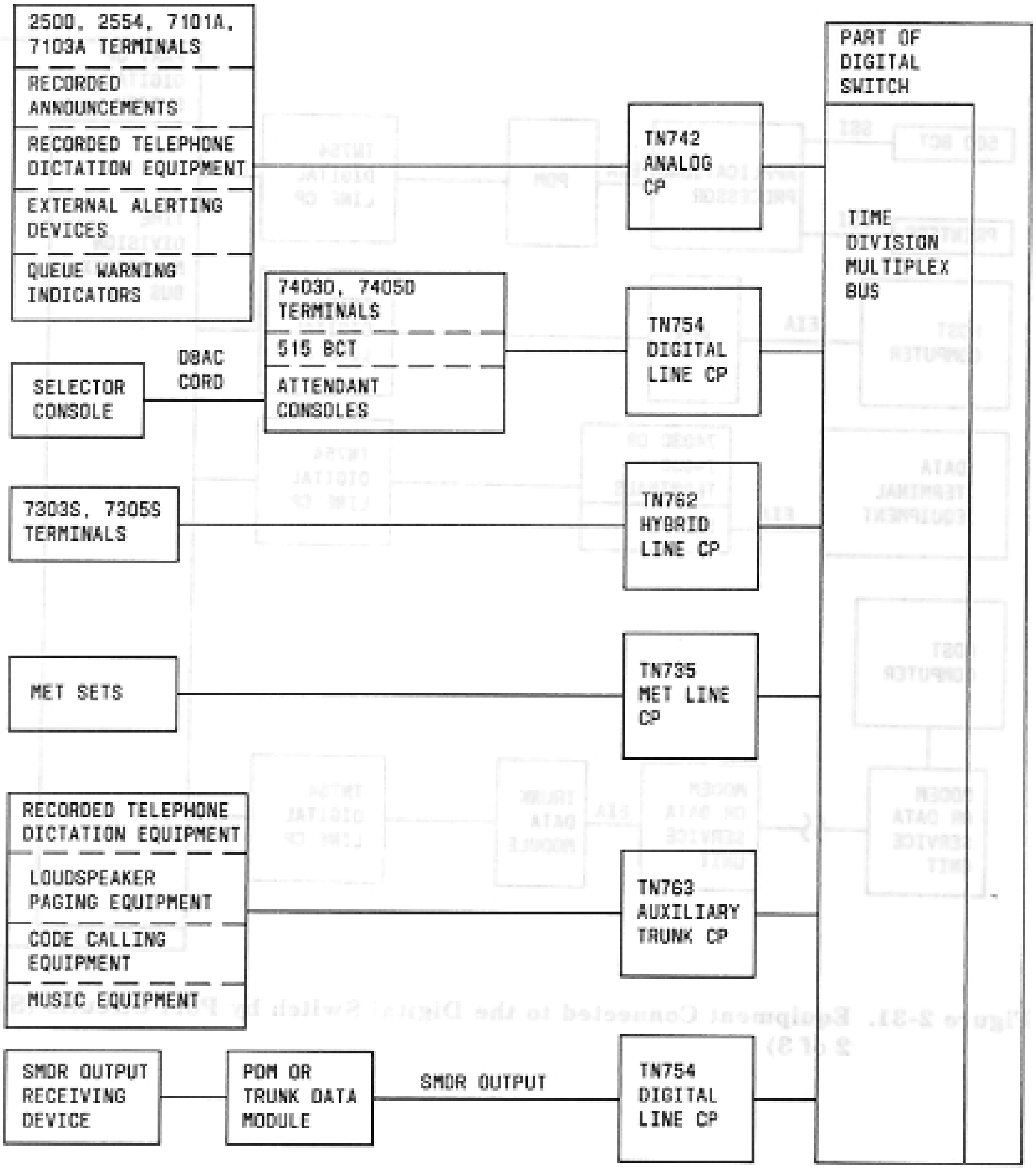


Figure 2-31. Equipment Connected to the Digital Switch by Port Circuits (Sheet 1 of 3)

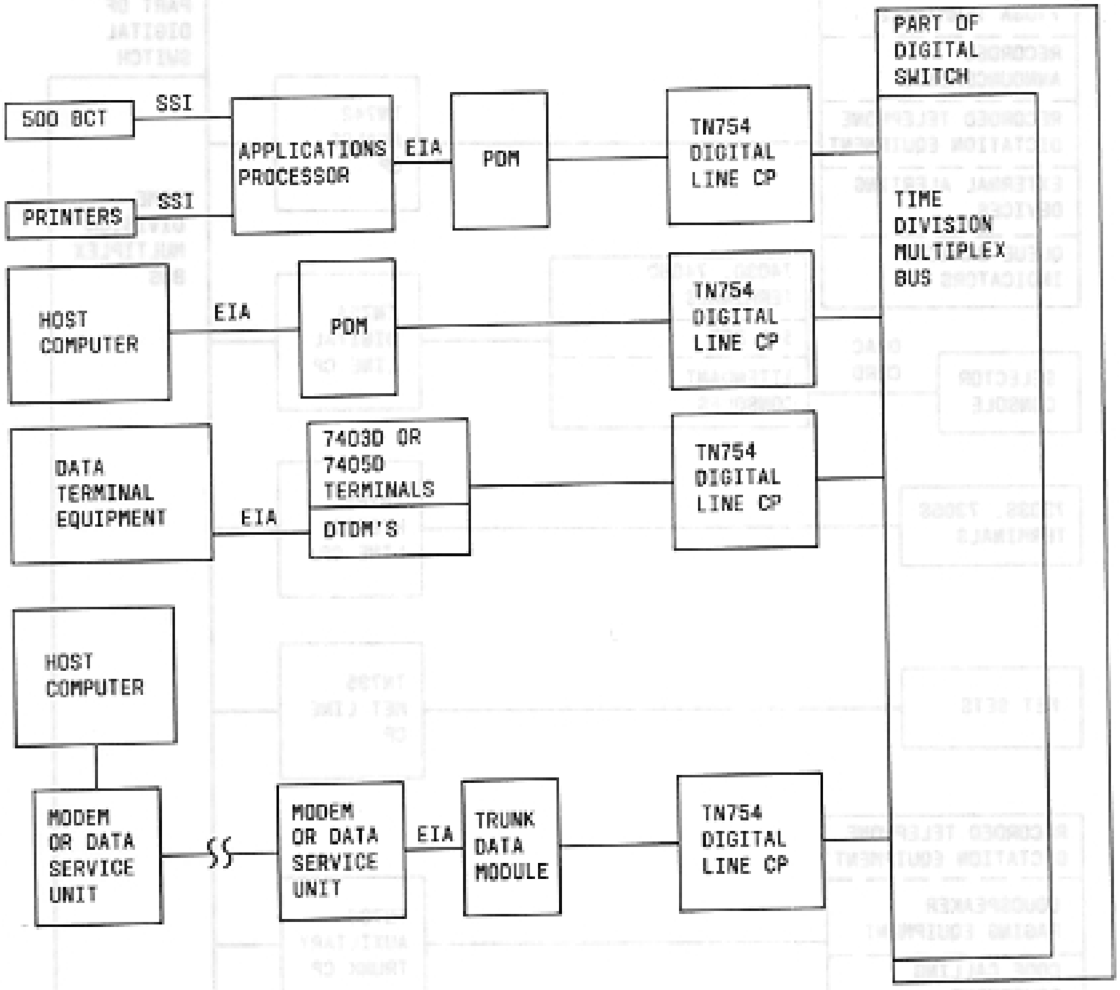
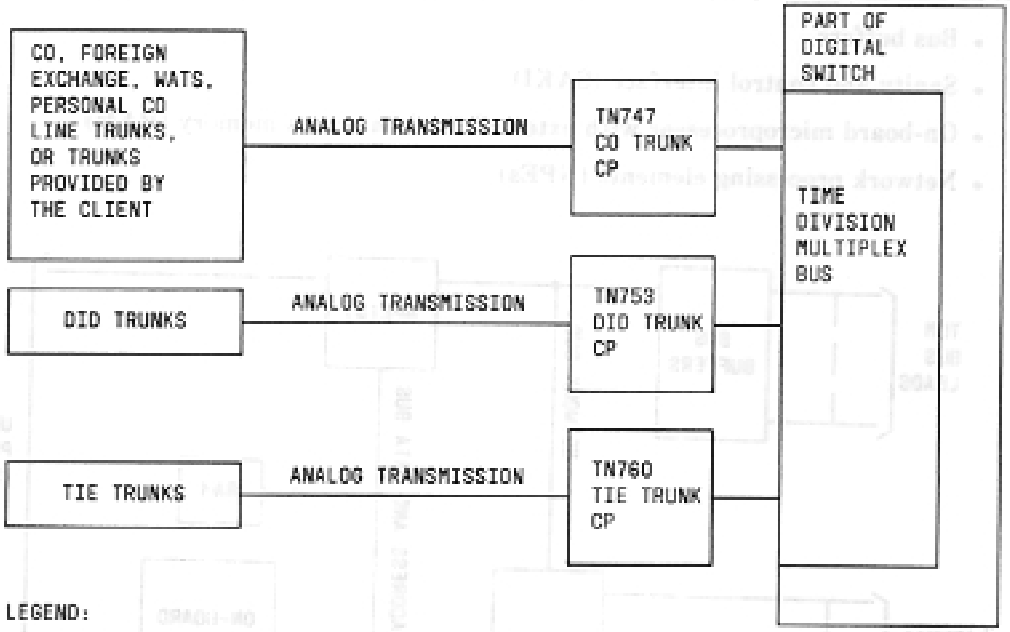


Figure 2-31. Equipment Connected to the Digital Switch by Port Circuits (Sheet 2 of 3)



LEGEND:

- BCT – BUSINESS COMMUNICATIONS TERMINAL
- CO – CENTRAL OFFICE
- CP – CIRCUIT PACK
- DID – DIRECT INWARD DIALING
- DTDM – DIGITAL TERMINAL DATA MODULE
- EIA – ELECTRONICS INDUSTRIES ASSOCIATION (CONNECTIONS SHOWN ARE 50 FT (15.3M) MAX.)
- INADS – INITIALIZATION AND ADMINISTRATION SYSTEM
- MET – MULTIBUTTON ELECTRONIC TELEPHONE
- PDM – PROCESSOR DATA MODULE
- SMDR – STATION MESSAGE DETAIL RECORDING
- SSI – STANDARD SERIAL INTERFACE
- WATS – WIDE AREA TELECOMMUNICATIONS SERVICE

Figure 2-31. Equipment Connected to the Digital Switch by Port Circuits (Sheet 3 of 3)

Each of the System 75 port circuit packs contains a number of common elements (see Figure 2-32) 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)
- Network processing elements (NPEs)

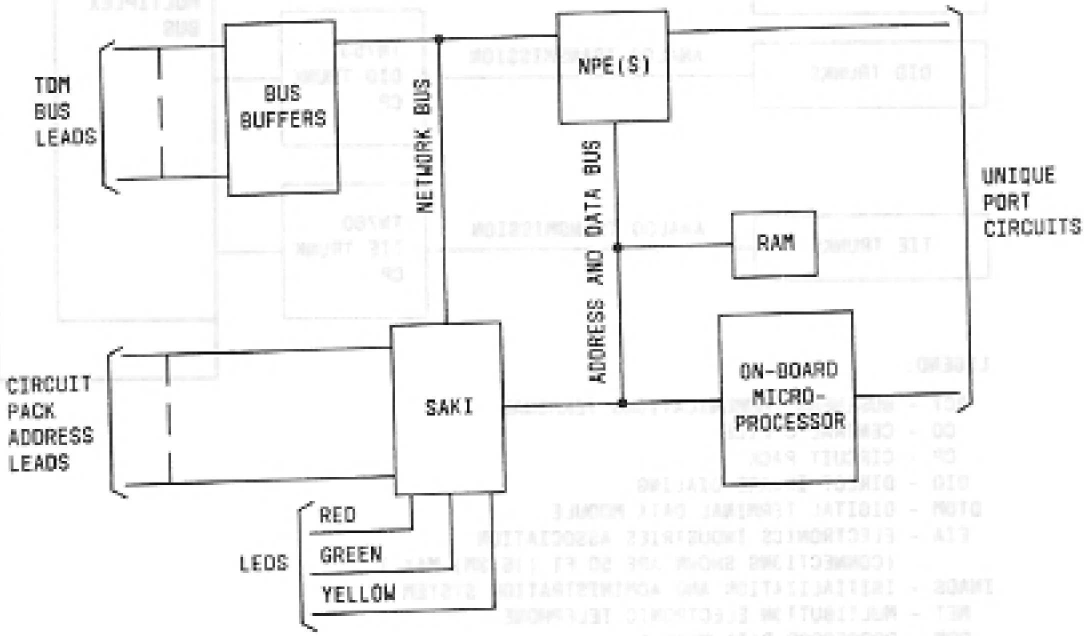


Figure 2-32. Port Circuit Pack Common Elements

The bus buffers are the digital interface between the backplane TDM bus wires (system bus) and the on-board circuitry (data bus). They receive or transmit on either of the two 8-bit TDM buses. They also receive and distribute clock and frame signals.

SAKI: The SAKI is the control interface between the SPE 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:

- Identifies the circuit pack to the SPE (location and vintage)
- Controls status indicator light-emitting diodes (LEDs)—red (failure), green (test), 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 Hybrid Line circuit pack and the formatter devices on the Digital Line circuit pack

- Takes the whole circuit pack out of service on command from the SPE 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 performs all low level functions such as scanning for changes and relay operations. In general, it carries out commands received from the SPE and reports status changes to the SPE. The external RAM stores control channel information and port-related information.

NPEs: Each port circuit pack contains one, two, or four NPEs. The Analog Line, Hybrid Line, CO Trunk, and DID Trunk circuit packs contain two NPEs. The Digital Line circuit pack contains four NPEs. The MET Line, Auxiliary Trunk, and Tie Trunk circuit packs contain one NPE.

The NPEs perform 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.

Service Circuits

The service circuits that connect to the TDM bus are as follows (see Figure 2-33):

- Tone/Clock
- Tone Detector
- Pooled Modem

As shown in Figure 2-33, the service circuits do not connect to any outside equipment. The service circuit packs contain the same common elements as the port circuit packs. These common elements perform similar functions for the service circuit packs.

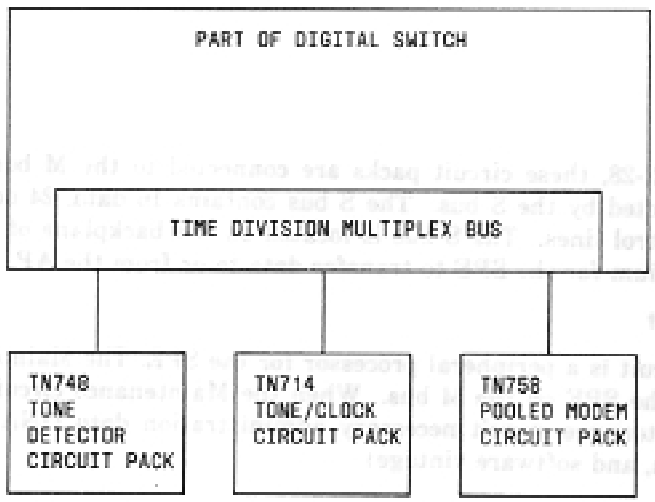


Figure 2-33. Service Circuits

Switch Processing Element

The main components of the switch processing element (SPE) are as follows (see Figure 2-28).

- Network Control
- Processor
- Memory
- Tape Control

These four components are interconnected by the 16-bit, 2-MHz M (memory) bus located on the backplane. The M bus also contains 24 address and 10 interrupt and control lines.

The Maintenance circuit pack is associated with the SPE on the same M bus. The AP interface circuit packs are also associated with the SPE via the M bus, the S (system) bus, and an identical but shorter M bus.

The Network Control circuit is the interface to the TDM bus. For this reason, it is identified separately from the remainder of the SPE in this manual.

High Capacity Minirecorder

The high capacity minirecorder (HCMR) provides a nonvolatile system bootstrap and translation storage device. The HCMR uses an incremental operating mode. In the incremental mode, data is efficiently read or written one single block of data at a time. The HCMR tape cartridge stores up to 12M bytes of data in the incremental mode.

Applications Processor Interface

The interface to an Applications Processor (AP) is provided by the following three circuit packs:

- Interface 1
- Interface 2
- Interface 3

As shown in Figure 2-28, these circuit packs are connected to the M bus. Interface 1 and Interface 2 are connected by the S bus. The S bus contains 16 data, 24 address, 5 parity, 10 interrupt, and 11 control lines. The S bus is located on the backplane of the control carrier and provides the medium for the SPE to transfer data to or from the AP.

Maintenance Circuit

The Maintenance circuit is a peripheral processor for the SPE. The Maintenance circuit pack communicates with the SPE on the M bus. When the Maintenance circuit is initialized, the SPE sends the Maintenance circuit necessary administration data (INADS phone number, product identification, and software vintage).

Power System

Figures 2-34 and 2-35 show the power distribution schemes for the 2-carrier and 5-carrier System 75 cabinets, respectively. Five-conductor power distribution cables connect the Power Distribution Unit to the power units in the carrier, to the HCMR, and to the 397A battery charger.

The power distribution cables provide 115 volts ac power on three of the leads during normal operation and -144 volts dc power on the other two leads during emergency transfer (Power Failure Transfer mode). The -144 volts dc power is not provided for the battery charger.

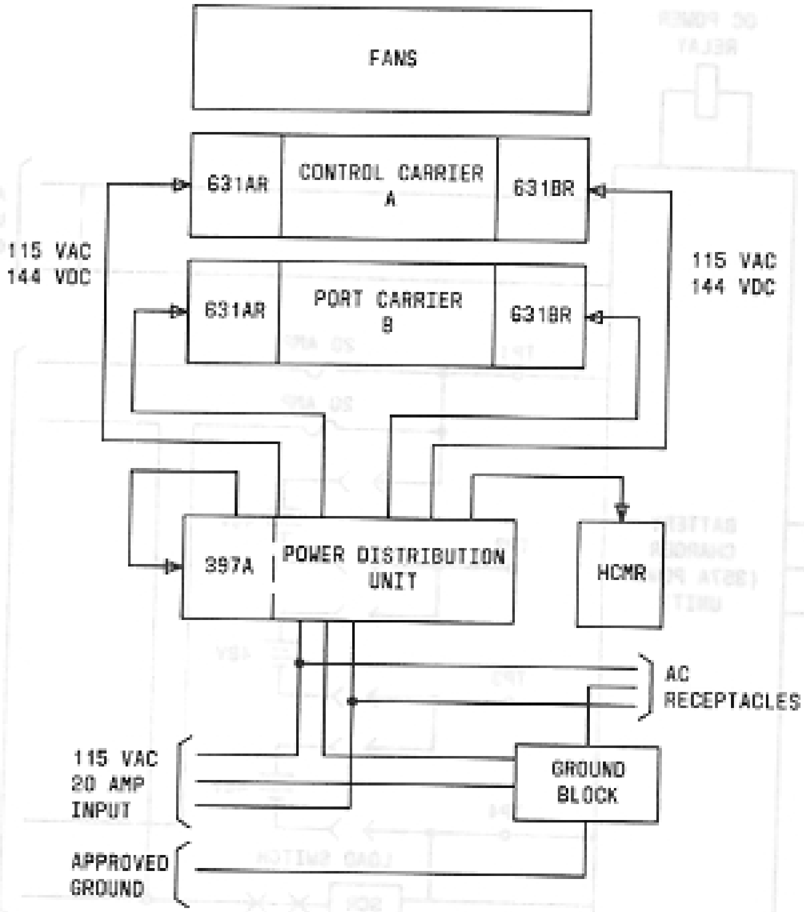


Figure 2-34. Power Distribution Scheme for 2-Carrier Cabinets

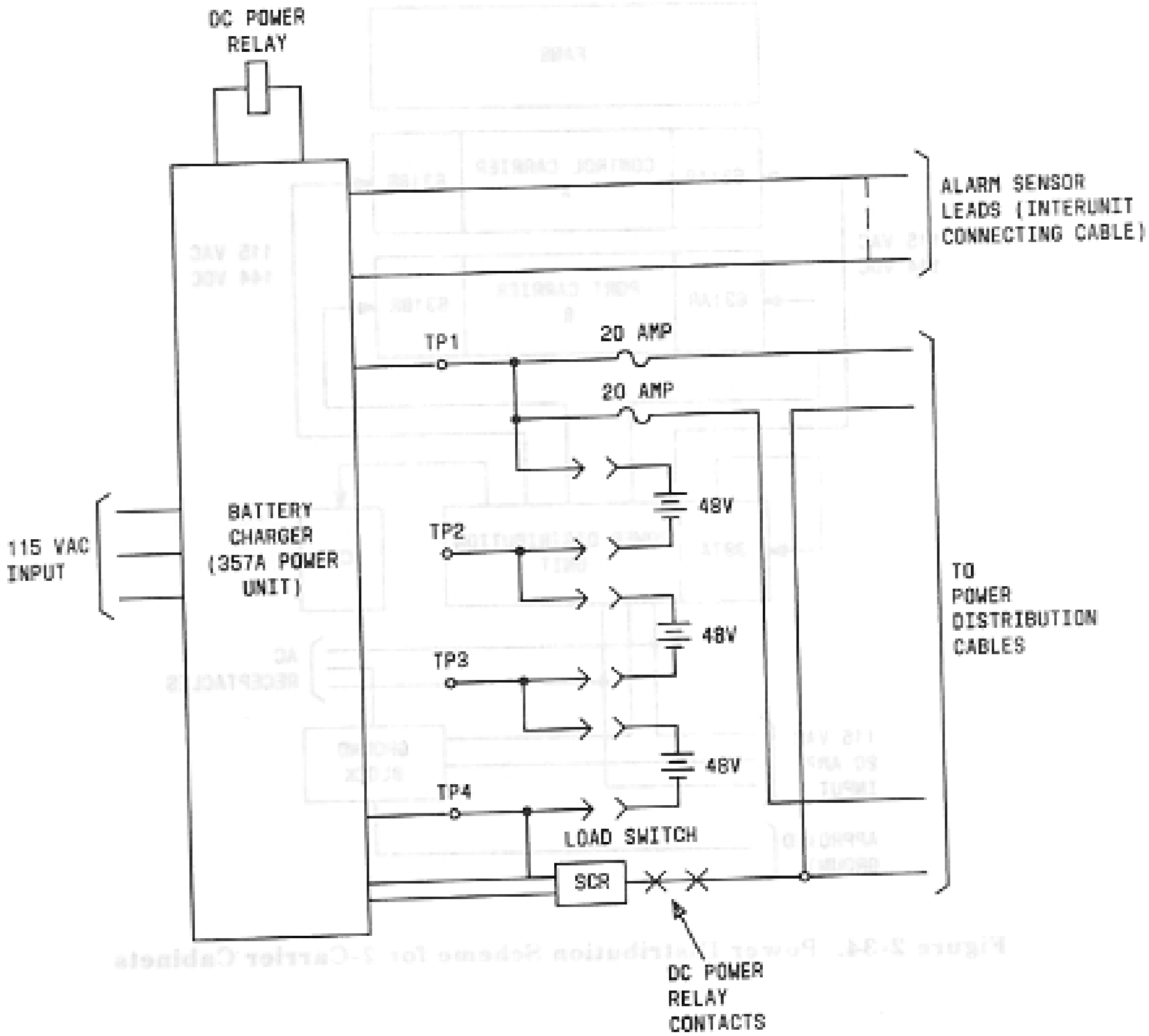


Figure 2-35. Power Distribution Scheme for 5-Carrier Cabinets

System Software

The System 75 software consists of the switched services software, administration software, and maintenance software. This software runs on top of the real-time operating system software.

Switched Services Software

The switched services software provides the call (switched voice and data) services, the message and display services, and other terminal services. The software resides in the SPE, the Network Control circuit, the Interface 2 circuit (when provided), and the 8-bit on-board microprocessors in the port and service circuits.

The message and display services of the SPE provide incoming and outgoing call identification, the Leave Word Calling feature, and the interface to AP provided Leave Word

Calling feature. The terminal services provide programming of abbreviated dialing list entries and terminal display services such as time of day.

The switched services software in the SPE uses the operating system to provide a process based, message passing, execution environment. The operating system scheduler provides SPE scheduling for the software according to process priority, activated by message dispatching.

The software in the Interface 2 circuit pack provides the protocol support for the communications link between the AP and the SPE. The Interface 2 software also translates between the System 75 commands and the format the AP accepts.

Step-By-Step Call Description

The following is a step-by-step description of a call originated by a System 75 digital voice terminal to another System 75 digital voice terminal. Calls originated by other type voice terminals or incoming calls on trunk circuits are similar. In the following description, the 8-bit microprocessors on the port and service circuits are referred to as port controllers.

1. The Network Control continually polls the port controllers on the port and service circuit packs.

2. The Digital Line port controller detects the terminal user lifting the receiver.

3. The Digital Line port controller sends an off-hook uplink message to Network Control.

4. The Network Control stores the message for the SPE.

5. The SPE software retrieves the message and interprets it as a call origination.

6. The SPE sends downlink messages via Network Control to the port controllers.

   a. The first set of messages instructs the originating port on the Digital Line circuit pack to listen for dial tone on a dedicated time slot and to light the call appearance status lamp on the terminal.

   b. The second set of messages instruct the originating port to talk on an available time slot. They also instruct a Tone Detector port controller to connect a touch-tone detector. The touch-tone detector interprets each individual touch-tone digit as it is dialed.

7. When the terminal user dials the first digit, the Digital Line port circuit converts the analog touch-tone signal to a digital signal and places the digital signal on the previously allocated time slot for that port.

8. The touch-tone detector, listening on the same time slot, interprets the tone. The Tone Detector port controller sends a digit uplink message via Network Control to the SPE.

9. The SPE sends a downlink message to the Digital Line port controller instructing it to remove dial tone from the port circuit.

10. The SPE analyzes the first digit, determines that the call is being placed to another System 75 extension, and starts collecting digits.

11. When the SPE collects enough digits to identify an extension (as specified in translations), it discontinues collecting digits.

12. The SPE recognizes that the called extension is a digital voice terminal (see Note) and sends a downlink message to the appropriate Tone Detector port controller to disconnect its touch-tone detector from the time slot.

    Note: If the extension number dialed is invalid, the SPE sends a message to the Tone/Clock port controller to place intercept tone on the time slot assigned to the originating port. Go to Step 20 for intercept tone.

13. The SPE determines if there is an available call appearance for the called digital voice terminal user and sends a message to the Tone/Clock port controller to place audible alerting or busy tone, as appropriate, on the time slot assigned to the originating port. Go to Step 20 for busy tone.

14. The SPE sends a downlink message via Network Control to the Digital Line port controller associated with the called extension. The message instructs the port controller to turn on the ringer and to flash a call appearance lamp on the called party's digital voice terminal.

15. When the called party lifts the receiver, the Digital Line port controller sends an off-hook message to the SPE as before.

16. The SPE interprets the off-hook message as an answer.

17. The SPE sends downlink messages to the Digital Line port controller to turn off the ringer and to light a call appearance lamp steady on the called party's digital voice terminal.

18. The SPE then sends downlink messages to the Digital Line port controller associated with the answering party to talk on an available time slot and to listen on the time slot assigned to the calling party.

19. The SPE instructs the Digital Line port controller associated with the called party to listen on the time slot assigned to the calling party for talking.

20. When either of the parties hangs up, the Digital Line port controller sends an on-hook message via the Network Control to the SPE.

21. The SPE interprets the on-hook message as the end of the call.

22. The SPE sends downlink messages to the Digital Line port controllers to disconnect the time slot connections and darken the lamps for the two call appearances.

Administrative Software

The administrative software provides the control for system rearrangement and change via a forms-based interface. This software resides in the SPE. Specifically, this software:

- Organizes the translation data for administrable entities in the system into forms that can be viewed and changed at the SAT or by INADS. The forms provide for administering the system, obtaining system traffic measurements, and performing maintenance operations.

- 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 indication is provided.

- Causes the translation data to be downloaded, on command, to the tape located in the HCMR. The download operation can also be administered to automatically occur daily.

Maintenance Software

The maintenance software contains two levels. A high-level subsystem exists on top of the operating system and a low-level subsystem resides independently of the operating system. The high-level maintenance software resides entirely in the SPE. The low-level maintenance software resides in the SPE and Maintenance circuit.

The high-level maintenance software operates during normal system operation. The low-level maintenance software operates when the system is in a state that it is unable to process calls, such as after a system crash (Power Failure Transfer mode) or during the initial installation.

High-Level Maintenance Software

The high-level maintenance software provides the following:

- System Initialization and Recovery—Ability of system to recover on its own from serious temporary malfunctions or failures
- Software Maintenance—Ability to recover from a process in the system software that is in an infinite loop or waiting for an event that will never occur
- Dynamic system configuration—Automatic tracking of port and service circuit pack insertion, removal, failure, and translations
- Hardware diagnostics and tests—Automatic periodic testing of System 75 hardware and an interface for the client or AT&T Information Systems technician to do the periodic tests on demand
- Maintenance Load Regulation—Ability to reduce the amount of periodic testing when there is a large amount of call processing required

Low-Level Maintenance Software

When the system is first powered up, or restarted from a system level recovery, the low-level maintenance software has control. It loads the operating system from tape, if necessary. The operating system then has control and creates the high-level maintenance software. The high-level maintenance software then starts all of the administrative and switched services software.

SPACE AND ENVIRONMENTAL REQUIREMENTS

CONTENTS

LIST OF FIGURES

SPACE REQUIREMENTS

This part provides information on the floor and wall space required for System 75 equipment and associated peripheral equipment installed in the equipment room.

Floor Plans and Layouts

Floor plan arrangements will vary depending on size and shape of the equipment room and the amount of growth planned for the system. Typical floor plans are shown in Figures 3-1 (2-carrier cabinet) and 3-2 (5-carrier cabinet).

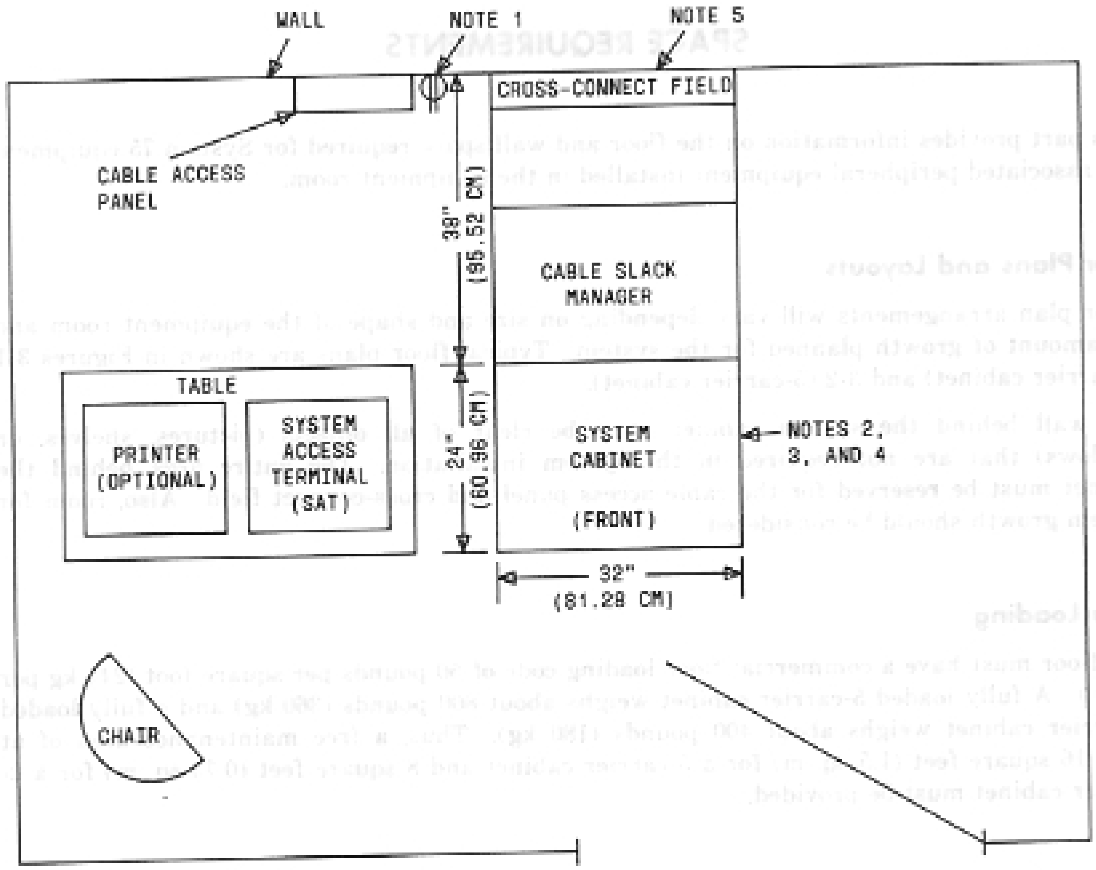
The wall behind the system cabinet must be clear of all objects (pictures, shelves, or windows) that are not required in the system installation. The entire area behind the cabinet must be reserved for the cable access panel and cross-connect field. Also, room for system growth should be considered.

Floor Loading

The floor must have a commercial floor loading code of 50 pounds per square foot (244 kg per sq. m). A fully loaded 5-carrier cabinet weighs about 800 pounds (360 kg) and a fully loaded 2-carrier cabinet weighs about 400 pounds (180 kg). Thus, a free maintenance area of at least 16 square feet (1.5 sq. m) for a 5-carrier cabinet and 8 square feet (0.75 sq. m) for a 2-carrier cabinet must be provided.

Earthquake Protection

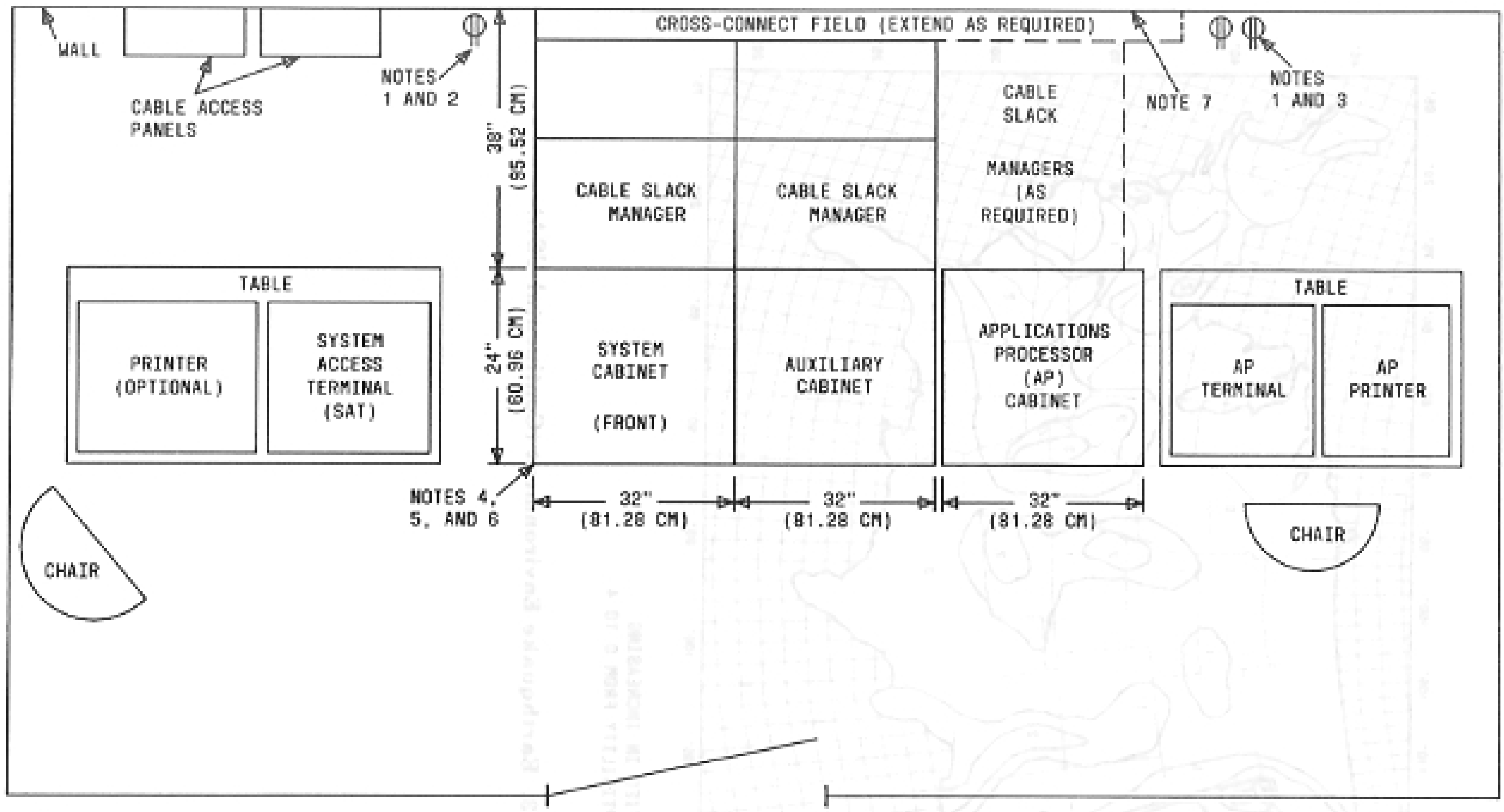
When earthquake or disaster bracing is required by law or when local engineering feels that bracing is necessary, the system cabinet can be bolted to the floor. Figure 3-3 shows the zones in the continental United States where bracing may be desirable.



NOTES:

1. 115-VOLT, 60-HZ, 20-AMP POWER OUTLET SHOULD BE LOCATED OUTSIDE THE CROSS-CONNECT FIELD AREA. POWER OUTLET MUST NOT BE UNDER SWITCH CONTROL AND MUST NOT BE SHARED WITH OTHER EQUIPMENT.
2. ALLOW AT LEAST 36 INCHES (91.4 CM) OF SPACE IN FRONT AND 6 INCHES (15.4 CM) ON THE RIGHT OF CABINET TO PERMIT DOOR TO SWING OPEN.
3. SYSTEM MUST BE GROUNDED BY ONE OF THE APPROVED METHODS LISTED IN THIS SECTION.
4. EARTHQUAKE PROTECTION MAY BE REQUIRED.
5. HORIZONTAL WALL SPACE REQUIRED FOR THE CABLE ACCESS PANEL(S) AND CROSS-CONNECT FIELD VARIES. A TYPICAL 150-LINE SYSTEM REQUIRES APPROXIMATELY 5 FEET (1.5M) OF HORIZONTAL WALL SPACE.

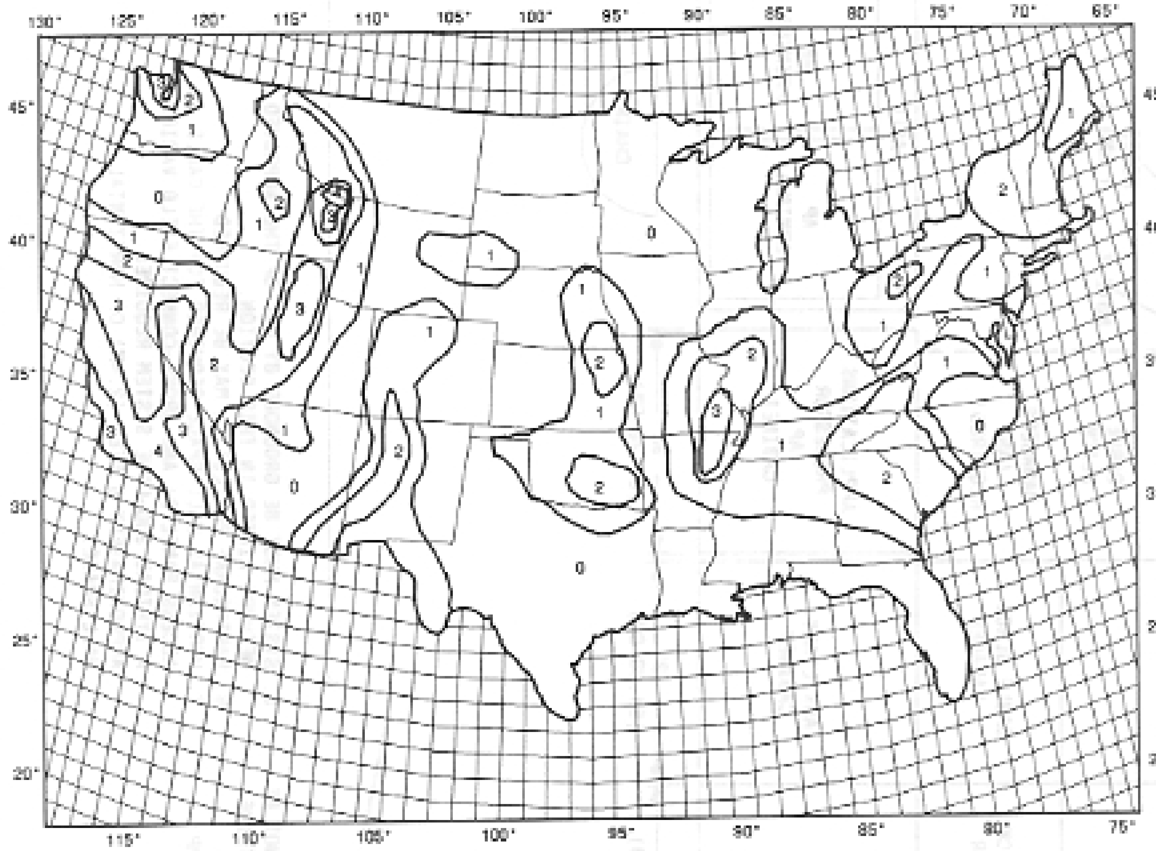
Figure 3-1. Typical Floor Plan—2-Carrier Cabinet System



NOTES:

1. POWER OUTLETS MUST NOT BE UNDER SWITCH CONTROL, MUST NOT BE SHARED WITH OTHER EQUIPMENT, AND SHOULD BE LOCATED OUTSIDE THE CROSS-CONNECT FIELD AREA.
2. SYSTEM CABINET REQUIRES SPECIAL 115-VOLT, 60-HZ, 50-AMP POWER OUTLET. SEE FIGURE 3-4.
3. AUXILIARY AND AP CABINETS REQUIRE A STANDARD 115-VOLT, 60-HZ, 30-AMP POWER OUTLET.
4. ALLOW AT LEAST 36 INCHES (91.4CM) OF SPACE IN FRONT AND 6 INCHES (15.4 CM) ON THE RIGHT OF CABINET TO PERMIT DOOR TO SWING OPEN.
5. SYSTEM MUST BE GROUNDED BY ONE OF THE APPROVED METHODS LISTED IN THIS SECTION.
6. EARTHQUAKE PROTECTION MAY BE REQUIRED.
7. HORIZONTAL WALL SPACE REQUIRED FOR THE CABLE ACCESS PANEL(S) AND CROSS-CONNECT FIELD VARIES. A TYPICAL 300-LINE SYSTEM REQUIRES APPROXIMATELY 8 FEET (2.4M) OF HORIZONTAL WALL SPACE.

Figure 3-2. Typical Floor Plan—5-Carrier Cabinet System With Applications Processor



ZONES ARE CLASSIFIED IN INCREASING
EARTHQUAKE SUSCEPTIBILITY FROM 0 TO 4

Figure 3-3. Earthquake Environment (Continental USA)

Floor Space

The following system equipment and optional peripheral equipment occupies floor space in the equipment room:

- System Cabinet and Cable Slack Manager—The system cabinet is 32 inches (80.64 cm) wide and 24 inches (59.89 cm) deep. The 5-carrier cabinet is 70 inches (175 cm) high and the 2-carrier cabinet is 42 inches (105 cm) high. The cable slack manager requires 38 inches (96.52 cm) between the cabinet and wall. The cabinet (including the door opening) and cable slack manager occupy about 22 square feet (2.04 sq. m) of floor space.
- Auxiliary Cabinet—The auxiliary cabinet is 32 inches (80.64 cm) wide, 24 inches (59.89 cm) deep, and 70 inches (175 cm) high. This cabinet (including the door opening and maintenance area behind cabinet) occupies about 22 square feet (2.04 sq. m) of floor space.
- Optional Applications Processor (AP)—The AP cabinet is 32 inches (80.64 cm) wide, 24 inches (59.89 cm) deep, and 72 inches (182.88 cm) high. This cabinet (including the door opening and maintenance area behind cabinet) occupies about 22 square feet (2.04 sq. m) of floor space.

References for Optional Equipment Requiring Floor Space

Refer to the following documents for additional information on optional equipment that can be used with the system and requires floor space:

Reference Manual—Applications Processor	999-700-407IS
User's Guide—445 Printer	999-700-023IS

Desk-Top Space

The 513 and 515 Business Communications Terminals (BCTs) can be located in the equipment room and require space on a desk or table. Sizes are as follows:

- Terminal—19 inches by 15 inches (47.5 cm by 37.5 cm)
- Keyboard—19 inches by 9 inches (47.5 cm by 22.5 cm)

These terminals occupy 4 square feet (0.37 sq. m) of space.

References for Optional Equipment Requiring Desk-Top Space

Refer to the following documents for additional information on optional equipment that can be used with the system and requires desk-top space:

User's Guide—500 Business Communications Terminal	999-700-021IS
User's Guide—443 Printer	999-700-024IS
User's Guide—450 Printer	999-700-025IS
User's Guide—460 Printer	999-700-022IS

AIR REQUIREMENTS

This part provides specifications for temperature, humidity, and air purity.

Temperature and Humidity

The System 75 equipment should be installed in a well-ventilated area. The equipment operates at an ambient temperature between 40 and 110 degrees Fahrenheit (4.4 and 43.4 degrees Celsius). The relative humidity range is 10 to 95 percent up to 78 degrees Fahrenheit (25.5 degrees Celsius). Above 78 degrees Fahrenheit (25.5 degrees Celsius), maximum relative humidity decreases from 95 percent down to 35 percent at 110 degrees Fahrenheit (43.3 degrees Celsius). Installation outside these limits may reduce system life or impede operation.

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
- Corrosive gases, such as sulfur and chlorine

ELECTRICAL REQUIREMENTS

This part provides specifications for lighting, noise suppression, power, and grounding required for the equipment room.

Lighting

Lighting should be bright enough 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.

Noise Suppression

In most cases, noise is introduced into the system through trunk or station cables, or both. However, electromagnetic fields near the system control equipment may also cause noise in the system. Therefore, the system 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 (187 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, whether synchronous or asynchronous, generally do not cause interference.

Field strengths below 1.0 volt per meter are unlikely to cause interference. These weak fields can be measured by a tunable meter such as the Model R-70 meter manufactured by Electro-Metrics Division. Field strengths greater than 1.0 volt per meter can be measured with a broadband meter such as the HOLADAY* HI-3001 meter or the 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 wave length (150 meters for a frequency of 1000 kHz).

Power Requirements

The system cabinet, auxiliary cabinet, and Applications Processor (AP) cabinet each require a separate power outlet. These outlets must not be shared with other equipment, and must not be under switch control, and should be located outside the cross-connect field area, if possible. Individual requirements are as follows:

- 5-Carrier Cabinet—This cabinet requires a special 115-volt 60-hertz 50-amp power outlet. The outlet must be located within 10 feet (3.1 m) of the cabinet.
- 2-Carrier Cabinet—This cabinet requires a special 115-volt 60-hertz 20-amp power outlet. This outlet must be located within 10 feet (3.1 m) of the cabinet.
- Auxiliary Cabinet and AP Cabinet (optional)—Each cabinet requires a standard 115-volt 60-hertz 20-amp power outlet. These outlets must be located within 12 feet (3.7 m) of the cabinets.

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* Trademark of Holaday Industries.

† Trademark of Hubbell-Harvey, Inc.

Figure 3-4 depicts a typical power and grounding layout for the system cabinet(s).

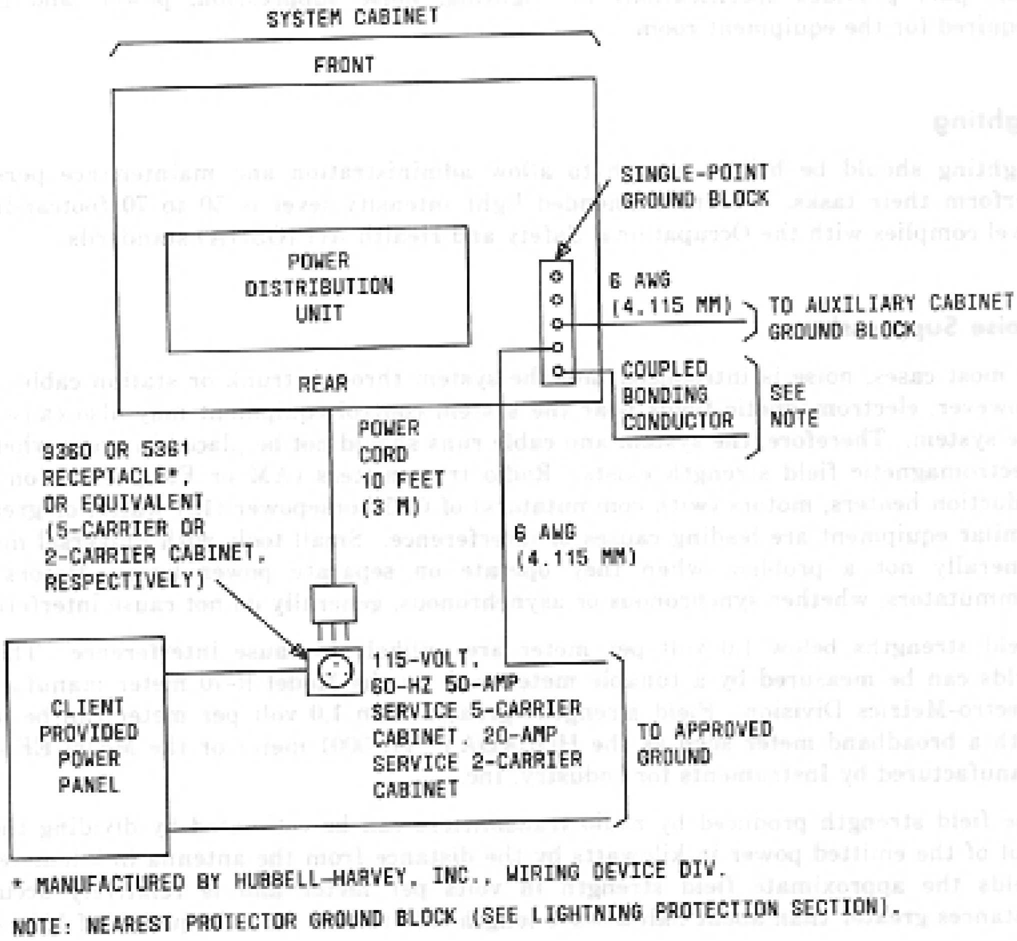

Figure 3-4. Typical Power and Grounding Layout

Grounding

An approved ground for the cabinets used in the equipment room is essential. An approved ground may consist of any of the following:

- Grounded Building Steel—The metal frame of the building where effectively grounded.
- Water Pipe—A continuous metal water pipe, not less than 1/2 inch (1.27 cm) in diameter, that is connected to an underground metal water pipe that is in direct contact with earth for 10 feet (3.1 m) or more.
- Concrete-Encased Ground—An electrode encased by at least 2 inches (5.1 cm) of concrete and located within and near the bottom of a concrete foundation or footing in direct contact with the earth. The foundation must be at least 20 feet (6.1 m) of one or more steel reinforcing bars or rods of not less than 1/2 inch (1.27 cm) in diameter, or at least 20 feet (6.1 m) of bare, solid copper wire not smaller than No. 4 gauge (5.189 mm).

- Ground Ring—A ring that encircles a building or structure in direct contact with earth at a depth of at least 2-1/2 feet (6.35 cm). The ring must consist of at least 20 feet (6.1 m) of bare copper conductor not smaller than No. 2 gauge (6.544 mm).

Lightning Protection

A coupled bonding conductor is tie-wrapped to all trunks. The coupled bonding conductor can be any one of the following:

- 10 AWG (6.86 mm) ground wire
- Continuous cable sheath
- Six unused pairs of wire

The coupled bonding conductor connects the cabinet single-point ground block and runs all the way to the protector block approved ground. The protector block ground may be from the local telephone company owned protector at the building entrance facility or from the client owned protector located in the sneak fuse panel, provided the sneak fuse panel also contains carbon block or gas tube protectors.

When an auxiliary cabinet is provided, a 6 AWG (4.115 mm) ground wire connects the system cabinet single-point ground block to the auxiliary cabinet ground block (See Figure 3-4). It is recommended that the ground wire be routed as close as possible to the cables connecting the system cabinet and the auxiliary cabinet.

If auxiliary equipment is not mounted in the auxiliary cabinet, then the power supply for this equipment must be plugged into one of the two convenience outlets located on the back of the System 75 cabinet to preserve ground integrity. The convenience outlet is fused at 15 amp. The dedicated SAT should be plugged into the other convenience outlet.

Sneak Current Protection

Sneak fuse panels, when provided, are installed between the cable access panel and the network interface. All incoming and outgoing trunks and off-premises station lines pass through the sneak fuses.

SYSTEM PARAMETERS

CONTENTS

GENERAL

This section provides information on the overall characteristics and capacities of the system.

The items presented in this section are grouped here for ease of reference. However, most items are discussed under each applicable feature. Some items, such as Cabling Distances, are not discussed in other sections of this manual and are presented here only.

ACOUSTIC NOISE LEVELS

The noise produced by the system is as follows:

- 5-Carrier Cabinet—51, 53, and 56 dB at low, medium, and high fan speeds, respectively, at a distance of 5 feet.
- 2-Carrier Cabinet—48, 50, and 53 dB at low, medium and high fan speeds, respectively, at a distance of 5 feet.
- If the system cabinet door is open, there is an additional 1 dB of noise.
- The high capacity minirecorder (HCMR) also causes additional noise. When reading data, there is an additional 2 dB of noise. When the HCMR is rewinding or fast winding, there is an additional 7 dB of noise.

CABLING DISTANCES

When the system layout is determined, maximum cabling distances to the system cabinet must be considered. The allowable intra-premises cabling distances are as follows:

Equipment	24-Gauge Wire (0.5106-mm)		26-Gauge Wire (0.4049-mm)	
	Feet	Meters	Feet	Meters
Attendant Console	3400	1037	2200	670
Business Communications Terminal—515	3400	1037	2200	670
Data Modules:				
Digital Terminal	3400	1037	2200	670
Processor	5000	1524	4000	1219
Trunk	5000	1524	4000	1219
System Access Terminal (SAT)	See Note			
Voice Terminals:				
2500-Type, 7101A, 7103A Fixed Feature and 7103A Programmable	6000	1828	3600	1100
2500-Type (Off-Premises)	14000	4267	-	-
7303S, 7305S (Without Auxiliary Power)	1000	305	750	228
7303S, 7305S (With Auxiliary Power)	2000	610	1500	457
7403D, 7405D	3400	1037	2200	670
MET Sets	1000	305	650	198

Note: The dedicated SAT distance is 50 feet (15.2 m) using standard EIA cable. When the SAT is used with a Digital Terminal Data Module or a Processor Data Module, the allowable distances are listed for the data module.

FEATURE ADMINISTRATION

Administration Not Required

Administration is not required to activate the following features.

Attendant Auto-Manual Splitting
Attendant Call Waiting
Attendant Conference
Attendant Recall
Attendant Release Loop Operation
Data Call Answering
Hold
Line Lockout
Recall Signaling
Straightforward Outward Completion
Temporary Bridged Appearance
Terminal Conference
Through Dialing
Touch-Tone Calling Senderized Operation
Touch-Tone Dialing (for Terminals)
Transfer
Uniform Numbering

Administration Required

Administration is required to activate the following features.

Abbreviated Dialing
Alerting
Attendant Control of Trunk Group Access
Attendant Direct Extension Selection With Busy Lamp Field
Attendant Direct Trunk Group Selection
Attendant Display (Buttons only)
Automatic Callback
Automatic Route Selection
Call Coverage
Call Forwarding All Calls
Call Park
Call Pickup
Call Waiting Termination
Code Calling Access
Data Privacy
Data Restriction
Dial Access to Attendant
Direct Department Calling
Direct Inward Dialing
Direct Outward Dialing
Facility Busy Indication
Hot Line Service
Hunting
Integrated Directory
Intercept Treatment
Intercom
Last Number Dialed

- Leave Word Calling
- Loudspeaker Paging Access
- Manual Message Waiting
- Manual Originating Line Service
- Manual Signaling
- Modem Pooling
- Multi-Appearance Preselection and Preference
- Multiple Listed Directory Numbers
- Music-on-Hold Access
- Night Service
- Off-Premises Station
- Personal Central Office Line
- Power Failure Transfer
- Priority Calling
- Privacy—Attendant Lockout
- Privacy—Manual Exclusion
- Recorded Telephone Dictation Access
- Remote Access
- Restrictions
- Ringback Queuing
- Station Message Detail Recording
- Terminating Extension Group
- Timed Reminder
- Touch-Tone Dialing (for Trunks)
- Trunk Group Busy/Warning Indicators to Attendant
- Trunk-to-Trunk Transfer
- Uniform Call Distribution
- Voice Terminal Display

FEATURE ACCESS

Dial Access

The following features or feature options can be activated and/or deactivated by dialing the assigned Feature Access Code or Trunk Access Code.

- Abbreviated Dialing:

  List 1
  List 2
  List 3
  Program

- Controlled Restriction:

  Single Voice Terminal (activate and deactivate)
  Group of Voice Terminals (activate and deactivate)

- Automatic Route Selection
- Automatic Callback (activate and deactivate) (applies to single-line voice terminals only)
- Call Forwarding All Calls (activate and deactivate)
- Call Park and Call Park Answer Back
- Call Pickup
- Code Calling Access
- Data Origination (associated with Data Call Setup and Pooled Modem)
- Data Privacy (associated with Data Call Setup and Pooled Modem)
- Facility Test Calls
- Hunt Group Make Busy (activate and deactivate) (associated with Direct Department Calling and Uniform Call Distribution)
- Last Number Dialed
- Leave Word Calling:

  Cancel a Message
  Display Module Lock
  Display Module Unlock
  Store a Message

- Loudspeaker Paging Access
- Private Network Access
- Priority Calling
- Public Network Access
- Recorded Telephone Dictation Access
- Send All Calls (associated with Call Coverage)
- Station Message Detail Recording (Account Code)
- Trunk Answer From Any Station (associated with Night Service)

Button Access

The following features or feature options must be assigned to a button. Feature Access Codes cannot be provided.

- Automatic Callback (applies to multi-appearance voice terminals only)
- Call Coverage:

  Consult
  Coverage Callback
  Coverage Message Retrieval
  Go to Cover
- Data Extension (associated with Data Call Setup)
- Display—Attendant or Voice Terminal:

  Date and Time
  Timer (Elapsed Time)
  Inspect
  Integrated Directory
  Normal
  Stored Number (associated with Abbreviated Dialing)
- Facility Busy Indication
- Intercom:

  Automatic
  Dial
- Leave Word Calling:

  Delete Message
  Message Retrieval
  Next Message (also used with Integrated Directory)
  Return Call (also used with Integrated Directory)
- Manual Message Waiting
- Manual Signaling
- Personal Central Office Line
- Privacy—Manual Exclusion
- Special Characters (associated with Abbreviated Dialing)

  Pause, Wait, Mark, and Suppress can each be assigned to a button or a Function Entry button can be assigned. Pressing Function Entry and then dialing 1, 2, 3, or 4 depicts Pause, Wait, Mark, or Suppress, respectively.
- Terminating Extension Group

Dial and Button Access

The following features or feature options can be activated and/or deactivated by dialing the assigned Feature Access Code or Trunk Access Code; they can also be assigned to a button for button access.

- Abbreviated Dialing:

  List 1
  List 2
  List 3
  Program

- Call Park and Call Park Answer Back
- Call Pickup
- Hunt Group Make Busy (activate and deactivate) (associated with Direct Department Calling and Uniform Call Distribution)
- Last Number Dialed
- Leave Word Calling:

  Cancel a Message
  Display Module Lock
  Store a Message

- Priority Calling

  The Priority Calling access code and extension number to be called, or the Priority Calling access code only, can be assigned to an Abbreviated Dialing (AD) button.

- Send All Calls (associated with Call Coverage)

Feature Status Button Indicators

The following buttons are not operational, but can be assigned to indicate the status of a feature or feature option. The lamp associated with the button lights when the assigned feature or option is active or is in use.

- Group Call (Lights to indicate that an incoming call is associated with a Call Coverage Answer group, a Direct Department Calling group, or a Uniform Call Distribution group.)
- Lock (Associated with the Voice Terminal Display; lights when activated and means that Leave Word Calling message retrieval will be denied from that terminal. Other display modes still work, including Coverage Message Retrieval.)
- Terminating Extension Group (Lights to indicate that the incoming call is for the associated group.)

FEATURE HARDWARE REQUIREMENTS

The following features require additional hardware:

- Attendant Direct Extension Selection With Busy Lamp Field—Requires a selector console.
- Code Calling Access—Requires loudspeaker paging equipment and one port on a TN763 Auxiliary Trunk circuit pack for each assigned zone. (This equipment can be shared with the Loudspeaker Paging Access feature.)
- Data Call Setup—Requires data modules to interconnect the desired data communications and data terminal equipment. Each module requires one port on a TN754 Digital Line circuit pack. The TN754 port is shared (voice and data) with a 7405D voice terminal with a Digital Terminal Data Module or with a 515 Business Communications Terminal. Analog modems, when used, require one port on a TN742 Analog Line circuit pack or conversion resource on a TN758 Pooled Modem circuit pack.
- Direct Department Calling—Requires one port on a TN742 Analog Line circuit pack for each queue warning level lamp.
- Direct Inward Dialing (DID)—Requires one port on a TN753 DID Trunk circuit pack for each DID trunk.
- Direct Outward Dialing—Requires one port on a TN747 Central Office (CO) Trunk circuit pack for each assigned trunk.
- Intercept Treatment—If recorded announcements are provided, requires announcement equipment and one port on a TN742 Analog Line circuit pack for each announcement.
- Loudspeaker Paging Access—See Code Calling Access.
- Modem Pooling—Requires a TN758 Pooled Modem circuit pack for each two analog to Digital Communications Protocol conversion resources provided.
- Music-on-Hold—Requires the music source(s) and one port on a TN763 Auxiliary Trunk circuit pack for each source.
- Night Console Service—Requires an attendant console.
- Off-Premises Station—Requires cross-connection through a Z100B1 cross-connect module and one port on a TN742 Analog Line circuit pack.
- Personal Central Office Line (PCOL)—Requires one port on a TN747 CO Trunk circuit pack for each CO, foreign exchange (FX), or Wide Area Telecommunications Service (WATS) trunk or one port on a TN760 Tie Trunk circuit pack for each tie trunk assigned as a PCOL.
- Power Failure Transfer—Requires rotary dial or touch-tone (2500-series) telephone set; requires a ground start key for each assigned telephone if CO trunks are ground start; and requires one power failure transfer unit for every six trunks assigned to this feature.
- Private Network Access—Requires one port on a TN760 Tie Trunk circuit pack for each trunk assigned.
- Public Network Access—Requires one port on a TN747 CO Trunk circuit pack for each trunk assigned.

- Recorded Announcement—See Intercept Treatment.
- Recorded Telephone Dictation Access—Requires dictation equipment (machine) and one port on a TN742 Analog Line circuit pack for each dictation machine assigned.
- Station Message Detail Recording—Requires output device (such as a printer), a Processor Data Module or a Trunk Data Module to interconnect the device, and one port on a TN754 Digital Line circuit pack.
- Trunk Answer From Any Station (associated with Night Service)—Requires external alerting device(s) (gong, bell, or chime) and one port on a TN742 Analog Line circuit pack.
- Uniform Call Distribution—Requires one port on a TN742 Analog Line circuit pack for each queue warning level lamp.
- Voice Terminal Display—Requires a Model 7405D voice terminal, a Digital Display Module, and one port on a TN754 Digital Line circuit pack.

HEAT DISSIPATION

The System 75 equipment should be installed in a well-ventilated area. A fully-loaded 5-carrier system dissipates approximately 4250 BTUs per hour. However, the typical average for a 5-carrier cabinet is a dissipation of 3000 BTUs per hour. The heat dissipation of a 2-carrier system is about half that of the 5-carrier system.

APPLICATIONS PROCESSOR (AP) FEATURES

The following features are available with the AP:

- Automated Building Management
- Directory
- Electronic Document Communications
- Message Center
- Terminal Emulation

SYSTEM CAPACITIES

The following is a synopsis of significant hardware, feature, and function capacities for System 75.

Hardware, Feature, or Function	Maximum
Abbreviated Dial Lists:	502
Personal Lists	400
Group Lists	100
System List	1
7103A List	1
Applications Processors	
With Information Exchange Between AP and System 75 switch	1
Without Information Exchange Between AP and System 75 switch	8
Attendant Consoles:	
Daytime Consoles	6
Night-Only Console	1
Automatic Route Selection (ARS):	
ARS Patterns	16
Trunk Groups per Pattern	6
Toll Lists	4
3-Digit Translation Tables	2
6-Digit Translation Tables	4
Cabinets (Basic)	1
Call Coverage:	
Coverage Paths per System	200
Coverage Points per Path	3
Coverage Answer Groups	100
Members per Coverage Answer Group	8
Call Pickup Groups	200
Members per Group	25
Total Members	400
Calls Per Hour	1800
Carriers:	
Control	1
Port (5-Carrier Cabinet)	4
Port (2-Carrier Cabinet)	1
Classes of Restriction	64
Classes of Service	16
Digital Data Endpoints (Endpoints include Processor Data Modules, Trunk Data Modules, Digital Terminal Data Modules, 515 terminals, the AP interface, and internal data channels)	200
Extension Numbers	600

Hardware, Feature, or Function	Maximum
Facility Busy Indicators	1000
Feature Access Codes	50
Hunt Groups (DDC and UCD Combined)	32
Members per Group	32
Members per System	448
Queue Slots per Group	35
Intercom Groups (Automatic and Dial Combined)	32
Members per Group	32
Members per System	128
Leave Word Calling (Switch-Based, No AP):	
Messages Stored	1000
Messages per User	127
Systemwide Message Retrievers	10
Remote Message Waiting Indicator:	
Per Extension Number	1
Per System	50
Outgoing Trunk Queue Slots	100
Personal Central Office Lines	25
Pooled Modems (16 Circuit Packs)	32
Port Circuit Packs	87
Recorded Announcements	10
Calls Connected per Announcement	1
Terminating Extension Groups	32
Members per Group	4
Time Slots:	
Total	512
Call Switching	472
Simultaneous Conversations	236
Traffic Handling Capability [in Hundred Call Seconds (CCS)]	8500
Trunk Access Codes	100
Trunks	200
Trunk Groups	50
Trunks per Group	60
Voice Terminals (Includes all voice terminals, 515 terminals, and MET sets. See Note.)	400

Note: The total number of Digital Display Modules, Call Coverage Modules, Function Key Modules, Attendant Consoles, 20/30 Button MET sets, and 7405D, 7305S, and 515 terminals in a system cannot exceed 125. A maximum of 62 Digital Display Modules can be provided.

STANDBY POWER SYSTEM

Standby power provides an alternate, independent source of on-premises power to maintain the System 75 for a limited time during a commercial power failure. A battery supply and an inverter are used to provide standby power for up to 8 hours after a commercial power failure.

When standby power is provided, the following items must be taken into consideration:

- Size and weight of the batteries
- Size and weight of the inverter(s)
- Heat dissipation
- Air flow and circulation
- Items of equipment to receive power

A standby power system contains the following:

- Inverter
- Batteries
- Battery stand

System 75 requires a 115 V ac input which is provided by the inverter. The size of the inverter is determined by the System 75 carrier configuration and the additional equipment to be provided with power in the event of a commercial power failure. The size of the battery supply required depends upon the length of time power is to be provided and the particular power demands of the system.

Refer to the *Standby Power Configuration Guide* for information on the requirements and worksheets for sizing the standby power system.

VOICE MANAGEMENT

CONTENTS

LIST OF FIGURES

LIST OF TABLES

GENERAL

This section defines the system features associated with Voice Management. The information for each feature is presented under four headings: Description, Considerations, Hardware and Software Requirements, and Interactions.

- Description

  Defines the feature, tells what it does for the user or how it serves the system, and briefly describes how it is used.

- Considerations

  Discusses other factors to be considered when the feature is used. These factors include administrable items such as feature button, Class of Service (COS), and Class of Restriction (COR) assignments.

  A COS and a COR are administered to all voice and data terminals. A COR is administered to all trunk groups. A COS allows or denies access to certain features. A COR assigns the various restrictions to voice terminals and/or trunk groups.

- Hardware and Software Requirements

  Lists any additional hardware and/or software requirements needed for the feature.

- Interactions

  Lists and briefly discusses other features that may significantly affect the feature being described. Interacting features are as follows:

  - 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 normal operation of one feature modifies, or is modified by, the normal operation of the other feature.

ABBREVIATED DIALING

Description

Provides lists of stored numbers that can be accessed to place local, long-distance, and international calls; to activate features; or to perform end-to-end signaling. (End-to-end signaling allows access to remote computer equipment.) Stored numbers can be accessed by voice terminal users, data terminal users, and incoming tie trunk groups.

Desired called numbers are stored in any of three types of lists, and each stored number is one list entry. To use abbreviated dialing, a user merely accesses the appropriate list, either through a dial access code or button, then dials the 1- or 2-digit list entry number, where the desired called number is stored. The number is then dialed automatically by the system. For a frequently called number, the list and list entry number can be stored on an abbreviated dialing button. In this case, simply pressing the button places the call.

The three types of lists where desired called numbers are stored are as follows.

- Personal Number Lists

  Allow users to have a personal set of stored numbers. A user can have one Personal Number List with 5 or 10 list entries. As many as 400 Personal Number Lists are allowed in the system. The user, or the System Manager, programs the Personal Number Lists. The System Manager sets which users will have a personal list.

- Group Number Lists

  Allow access by a group of users, such as purchasing or personnel departments, who frequently dial the same numbers. As many as 100 Group Number Lists are allowed in the system. Each Group Number List can have from 5 to 15 list entries in increments of 5. An individual user can access 0 through 3 specific Group Number Lists, as set by the System Manager.

- System Number List

  The System Number List can have up to 50 entries (in multiples of 5). The System Number List can contain any number or dial access code. The System Manager programs the System Number List and sets which users can access the list.

List entries for the Personal Number Lists are numbered 1 through 9, and 0. List entries for the Group Number Lists are numbered 11 through 25. List entries for the System Number List are numbered 11 through 60. This numbering scheme is used because the system expects either one or two digits to identify entries on a given list, not a mixture.

Each extension number, when implemented, can be assigned up to three Abbreviated Dialing Lists—List 1, List 2, and List 3. Each of these three lists is designated as being either Personal, Group, or System. The three lists may be any combination of the above, as long as there is no more than one Personal List or System List in the combination. When a list is designated as being a Group List, the particular number of the Group List is specified (for example, group list 42).

To access Abbreviated Dialing, the user accesses List 1, List 2, or List 3 either by dialing the access code or by using a button programmed with the access code. The access codes for the List 1, List 2, and List 3 are the same systemwide. Therefore, it is possible for a system list or a particular group list to have a different access code at different voice terminals. For example, suppose the feature access codes for List 1, List 2, and List 3 are 101, 102, and 103, respectively. One voice terminal may have List 1 administered as "personal," List 2 administered as "group 42," and List 3 administered as "system." Another voice terminal may have List 1 administered as "group 42," List 2 administered as "system," and List 3 administered as "group 21." In this case, the access code for "group 42" is 102 for the first voice terminal and 101 for the second voice terminal. Likewise, the access code for the "system" list is 103 for the first voice terminal and 102 for the second voice terminal.

All Group Number Lists and the System Number List can be designated as Privileged by the System Manager. Calls automatically dialed from a Privileged List are completed without Class of Restriction or Facilities Restriction Level (FRL) checking. (FRLs are associated with the Automatic Route Selection feature.) This allows access to selected numbers that certain voice terminal users might otherwise be restricted from manually dialing. For example, a voice terminal user may be restricted from making long-distance calls. However, the number of another office location may be long distance. This number could be entered in a list designated as privileged. The user could then call the office location using Abbreviated Dialing, while still being restricted from making other long-distance calls.

A Privileged List is also very practical from a tie trunk user's perspective. A tie trunk user cannot access a Personal Number List. The user can, however, access Group Number Lists and the System Number List. For example, if a tie trunk user is normally toll restricted, toll calls cannot be made. However, one or more numbers that are toll calls may be used for business purposes. These numbers, if included in a Privileged List, can be called by the tie trunk user without restriction.

A number stored in a Personal, Group, or System Number List can be a combination of numerical digits and special characters. A special character instructs the system to take a different action when dialing reaches the point where the character is stored. The following special characters can be stored.

- Pause

  When a Pause precedes, or is included in, a string of stored digits to be outpulsed over a trunk, outpulsing of the digit(s) following the Pause will be delayed 1.5 seconds. Outpulsing will automatically resume after expiration of the delay timing.

  The Pause is useful when the probability of dial tone being returned within 1.5 seconds is high. Typical applications include tandem switching through private networks and end-to-end signaling over the public or a private network.

- Wait

  When a Wait precedes, or is included in, a string of stored digits to be outpulsed over a trunk, outpulsing of the digit(s) following the Wait will be delayed 4 seconds or until the user initiates an End-Wait signal, whichever occurs first. Outpulsing will resume after the End-Wait signal is received or when delay timing expires.

  The user can initiate End-Wait (after hearing dial tone) by pressing the Wait button, if provided. When a Wait is encountered in a stored number, the status lamp associated with the Wait button lights and goes dark when the Wait button is pressed or when delay timing expires.

  The Wait is useful in cases where dial tone delays of variable length and/or network blocking outside the system are frequently experienced. Typical applications include tandem switching through private networks and end-to-end signaling over the public or a private network.

- Mark

  When a Mark precedes, or is included in, a string of stored digits, all digits following the Mark are treated as end-to-end signaling digits to be outpulsed over an outgoing trunk in touch-tone signal form even if the trunk involved requires dial pulses.

- Suppress

  When a Suppress precedes, or is included in, a string of stored digits, the system treats all digits following the Suppress the same as any other digits for call setup and digit outpulsing. The Suppress character only affects the display of the stored number. Stored numbers are normally shown when an alphanumeric display is provided through the Voice Terminal Display feature; however, the digits following the Suppress character are not displayed. The display shows the lowercase s instead of the stored digits.

Stored numbers can be accessed by any of the following options:

- Abbreviated Dialing-Code (AD Code)

  This option allows users to access a stored number by dialing the AD feature access code and a list entry number. Each AD code automatically dials the number stored in the list the user accessed.

- Abbreviated Dialing-Button (AD Button)

  This option allows multi-appearance voice terminal users to access stored numbers by pressing one or more buttons. Each AD button automatically dials the number stored in the list and the list entry number administered to the button. Access to any list and associated list entry number can be programmed in an AD button.

  The System Manager administers the AD button. If the button is administered to access a number in the user's Personal Number List, the user can change the number that is assigned to the button. However, if the number assigned to the button accesses a Group or the System Number List entry, only the System Manager can make the change.

A separate list, called the 7103A Group Number List, is used only by 7103A Fixed Feature voice terminal users as a group. This list allows button access to stored numbers and can have eight list entries. Any number can be stored in the 7103A Group Number List; however, it is intended primarily for feature access codes. The System Manager programs the 7103A Group Number List.

All users can program their Personal Number List, and users with an assigned AD button can program the button. Programming is done by dial access or by pressing the Program button, if assigned.

To program an entry in a Personal Number List, the user dials the AD Program access code or presses the AD Program button, then dials the list entry number, and the number to be stored (up to 16 digits), and then presses the # button. Confirmation tone is heard when the number is stored. While in the program mode, users can program all Personal Number List entries, if desired. To exit the program mode, the user simply hangs up.

To program an AD button administered to access a particular entry in the Personal Number List, the user dials the AD Program access code or presses the AD Program button, if assigned. The user then presses the AD button, dials the desired number (up to 16 digits), and then presses the # button. Confirmation tone is heard when the number is stored. While in the program mode, the user can program as many assigned AD buttons as desired. To exit the program mode, the user simply hangs up.

Only the System Manager and multi-appearance voice terminal users can program special characters. Voice terminal users need Pause, Wait, Mark, and Suppress buttons or a Function Entry button to program special characters. Pressing a Pause, Wait, Mark, or Suppress button programs the special character administered to the button. Pressing the AD Function Entry button and then dialing 1, 2, 3, or 4 programs Pause, Wait, Mark, or Suppress, respectively.

Considerations

A maximum of 502 lists and a maximum of 2500 entries are allowed for the system.

A number stored in any list can contain up to 16 digits. A special character used for Pause, Wait, Mark, or Suppress counts as two digits.

The Pause and Wait special characters are not needed to delay outpulsing of the initial digits following access of an outgoing trunk; the system always knows when to start outpulsing over a trunk. These characters are used when outpulsing should be delayed until dial tone is returned from a distant point reached through a switched connection outside the system.

Users can be assigned access to three AD lists: one Personal Number List and two Group Number Lists; one Personal Number List, one Group Number List, and the System Number List; the System Number List and two Group Number Lists; or three Group Number Lists.

A Personal Number List must be assigned to a voice terminal before the System Manager can establish that list. For example, during implementation, a voice terminal must first be assigned a Personal Number List on the individual voice terminal form. The actual list can then be established on the Abbreviated Dialing personal list form.

Incoming tie trunks cannot access a Personal Number List.

The Abbreviated Dialing feature cannot be administered to an attendant console.

The following items can be administered:

- Feature access codes (three codes, one each for List 1, List 2, and List 3)
- AD buttons (per multi-appearance voice terminal)
- Wait, Mark, Pause, Suppress, and Function Entry buttons (per multi-appearance voice terminal)
- Access to as many as three lists (per voice terminal)
- The lists themselves (maximum of 502), including 400 Personal Number Lists, 100 Group Number Lists, a 7103A Group Number List, and a System Number List)

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Last Number Dialed

  This feature will place a call to the same number as called previously, even if Abbreviated Dialing was used on the previous call. If any special characters (Mark, Wait, Pause, and/or Suppress) are included in the previous call, they are also included on the Last Number Dialed call.

ALERTING

Description

Helps voice terminal users and attendants distinguish between various types of incoming calls.

The associated call types, types of users, and alerting cycles are as follows:

Associated Call Type	User	Alerting Cycle (In Seconds)
Internal voice terminal, internal tie trunk, and Remote Access	All voice terminals	1-burst alerting (1.2 on, 4.0 off repetitive)
Intercom	Single-line voice terminals	
Attendant-extended, attendant-originated, and incoming trunk, including external tie trunk	All voice terminals	2-burst alerting (0.2 on, 0.4 off; 0.6 on, 4.0 off repetitive)
Automatic Callback, Priority Calling, and Ringback queuing callback	Single-line voice terminals	3-burst alerting (0.2 on, 0.1 off; 0.2 on, 0.1 off; 0.6 on, 4.0 off repetitive)
	Multi-appearance voice terminals	3-burst alerting (0.1 on, 0.1 off; 0.1 on, 0.3 off; 0.6 on, 4.0 off repetitive)
Intercom	Multi-appearance voice terminals	Single tone (0.6 on, 4.6 off repetitive)
Manual Signaling	Multi-appearance voice terminals	Single tone (2.0 on)
Redirection notification	All voice terminals	Single tone (0.2 on)

The following call types and their alerting cycles are received at attendant consoles:

Call Type	Alerting Cycle (In Seconds)
Incoming call	Low-pitched tone (0.4 on, 1.2 off repetitive)
Attendant Recall call and when any call associated with a timed reminder interval returns to the console	High-pitched tone (0.4 on, 1.2 off repetitive)
Calls waiting in queue	Low-pitched tone (0.25 on, 0.8 off repetitive)

Considerations

The 2- and 3-burst alerting is optional only on 2500-series voice terminals. If Alerting is disabled, the user will hear a 1-burst repetitive tone for all incoming calls. This is useful for equipment interfaced by analog lines, especially if the Off-Premises Station feature is used.

The following items can be administered:

- Redirection notification (used with the Call Forwarding All Calls and Call Coverage features) is optional on a per-terminal basis.
- The Alerting feature is optional for 2500-series voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

AP DEMAND PRINT

Description

Allows the voice terminal user to print his or her own undelivered messages without calling the Message Center.

The Message lamp at each voice terminal indicates whether or not any undelivered messages are waiting for the voice terminal user. The lamp is lighted when there are undelivered messages and goes dark when there are no undelivered messages. When the Message lamp is lighted at a voice terminal, the voice terminal user can have the message(s) printed.

The voice terminal user whose messages are to be printed is called the requesting extension. (Messages are requested by entering the extension number of the person for whom the messages were left.) The requesting extension can activate the AP Demand Print feature or another extension can activate the AP Demand Print feature for the requesting extension. The extension that activates the AP Demand Print feature is called the originating extension. Thus, if a user activates the feature from his or her own voice terminal to print his or her own messages, then the assigned extension number is both the requesting and originating extension.

Each requesting extension has an assigned printer which is used to print the AP Demand Print messages. However, some extensions may be associated with an overriding printer. If the originating extension is associated with an overriding printer, the messages will be printed on that printer instead of the printer assigned to the requesting extension.

A requesting extension can be an individual voice terminal, a Personal CO Line Group, a Uniform Call Distribution Group, a Direct Department Calling Group, or a Terminating Extension Group. Each requesting extension is assigned an authorization password. The password consists of four digits. Each digit can be 0 through 9. This password allows the originating extension to access the requesting extension's messages.

AP Demand Print is activated either by dialing the feature access code or by pressing the Print Msgs button. After the originating extension does this, the requesting extension's number and the requesting extension's authorization password must be entered. The messages are then printed. If an overriding printer is used by an originating extension, the system attempts to print the messages on that printer. Otherwise, the system attempts to use the printer assigned to the requesting user. In either case, if the printer is inoperable, the messages are routed to the AP-system default printer. After the messages are printed, the message waiting lamp at the requesting extension goes dark.

Considerations

The following items can be administered:

- Authorization passwords (per requesting extension)
- Feature access code
- Print Msgs buttons (per voice terminal)
- Printer assignment (per requesting extension)
- Overriding printers (maximum of 10)

Feature access codes and Print Msgs buttons are assigned in System 75 translations. The remaining items are set in AP translations.

Hardware and Software Requirements

This feature requires an AP. No additional software is required.

Interactions

In addition to Message Center messages, all undelivered Leave Word Calling messages can be printed with AP Demand Print.

When an AP is provided, Leave Word Calling is provided by the AP. Operation of this feature, when AP-based, differs from the operation described elsewhere in this section.

ATTENDANT AUTO-MANUAL SPLITTING

Description

Allows the attendant to announce a call or consult privately with one party without being heard by the other party on the call.

This feature is activated automatically when the attendant, active on a call, presses the Start button, a Group Select button and a Direct Extension Selection button (if provided), or a Trunk Group Select button. Any of these actions temporarily separates the party on the call from the connection and allows the attendant to call and talk privately with another party.

The connection is reestablished when the attendant presses one of the following buttons:

- Cancel—Cancels the call attempt and reconnects the attendant and the separated party.
- Split—Establishes a 3-way conversation with the attendant, the separated party, and the called party.
- Release—Connects the separated party and the called party and disconnects the attendant.

Considerations

None.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

ATTENDANT CALL WAITING

Description

Allows the attendant to extend a call to a busy single-line voice terminal and be free to handle other calls.

When Attendant Call Waiting is activated, the attendant hears a special audible ringback tone and the busy voice terminal user hears a 2-burst tone. The 2-burst tone is heard only by the called voice terminal user.

The call waits until the voice terminal is idle or until the administered interval expires. If the interval expires, the call returns to the console. The call in progress at the voice terminal cannot be placed on hold. It must be terminated.

Considerations

Attendant Call Waiting applies only for calls to single-line voice terminals within the system. Only one call per voice terminal can wait at a time.

The call waiting interval is administered through the Timed Reminder feature.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Automatic Callback

  If Automatic Callback is activated at the called voice terminal, Attendant Call Waiting is denied.

- Call Coverage

  If Call Coverage is assigned to a voice terminal, and if Send All Calls is activated or coverage criteria are met, the call may redirect to the coverage path instead of waiting. In some cases the call can wait and then redirect to coverage. In other cases the call returns to the console, rather than redirecting to coverage. Operation is as follows:

  — The Coverage Don't Answer interval (2 to 9 alerting cycles, or the equivalent time) specifies how long a call remains directed to the called voice terminal before redirecting to coverage. This interval applies to both the Busy and Don't Answer criteria. (The Active criteria should not be assigned to single-line voice terminals.) If Attendant Call Waiting is applicable on the call, this feature is active for the duration of the Don't Answer interval only. At the expiration of this interval, the call redirects to coverage.

  — If the Timed Reminder interval (10 to 1020 seconds) expires before the Don't Answer Interval expires, the call does not go to coverage but returns to an attendant console. If the Don't Answer interval expires first, the call redirects to coverage instead of returning to the attendant console.

  — If Send All Calls is active, or if the redirection criterion is Cover All Calls, the call immediately redirects to coverage instead of waiting.

— An attendant can release from an extended call at any point during the call, without affecting the preceding operations.

- Data Privacy

  If Data Privacy is activated at the called voice terminal, Attendant Call Waiting is denied.

- Direct Department Calling (DDC) and Uniform Call Distribution (UCD)

  Calls to a DDC or UCD group do not wait; however, such calls can enter the group queue, if provided.

- Loudspeaker Paging Access

  If Loudspeaker Paging Access is activated at the called voice terminal, Attendant Call Waiting is denied.

- Recorded Telephone Dictation Access

  If Recorded Telephone Dictation Access is activated at the called voice terminal, Attendant Call Waiting is denied.

- Timed Reminder

  The Timed Reminder interval (10 to 1020 seconds) determines how long a call will wait before returning to an attendant console. If the call is not answered or does not redirect to coverage before this interval expires, the call returns to the attendant console.

ATTENDANT CONTROL OF TRUNK GROUP ACCESS

Description

Allows the attendant to control trunk groups, and prevents voice terminal users from directly accessing a controlled trunk group. If a user dials a controlled trunk group, the call is redirected to the attendant who then decides whether or not to allow the call to go through.

Considerations

This feature can be activated for any trunk group assigned to a Trunk Group Select button with an associated control lamp. One attendant can control access to as many as six trunk groups.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Attendant Direct Trunk Group Selection

  This feature must be assigned with Attendant Control of Trunk Group Access.

- Attendant Display

  When a call redirects to the console because Attendant Control of Trunk Group Access is activated, the alphanumeric display identifies the calling party and call purpose.

- Automatic Route Selection

  Activating Attendant Control of Trunk Group Access removes the controlled trunk group(s) from the Automatic Route Selection patterns. Deactivating the feature reinserts the group(s) into the patterns. Automatic Route Selection calls are not routed to the attendant.

- Trunk Group Busy/Warning Indicators to Attendant

  This feature keeps the attendant informed of trunk group status. This status can be used to determine when to activate control.

ATTENDANT DIRECT EXTENSION SELECTION WITH BUSY LAMP FIELD

Description

Allows the attendant to place or extend calls to all extension numbers assigned to the system by pressing a Group Select button and a Direct Extension Selection (DXS) button instead of dialing the extension number.

When the Group Select button is pressed, the busy lamp field shows the idle or active status of the extension number associated with a voice terminal or a group of voice terminals.

Considerations

After the Group Select button is pressed, if the lamp adjacent to the desired DXS button is lighted to indicate busy status, the call can still be placed or extended. Attendant Call Waiting can be activated for a single-line voice terminal. A multi-appearance voice terminal user receives the call on an idle appearance.

Hardware and Software Requirements

Requires a selector console. No additional software is required.

Interactions

- Attendant Display

  When the attendant uses the Direct Extension Selection With Busy Lamp Field, the call is identified on the alphanumeric display through the Attendant Display feature.

- Call Coverage

  If Send All Calls is activated, or if the Call Coverage redirection criteria are met, then an extended call will redirect to the coverage path.

ATTENDANT DIRECT TRUNK GROUP SELECTION

Description

Allows the attendant direct access to an idle outgoing trunk by pressing the button assigned to the desired trunk group.

Considerations

The attendant console has 12 Trunk Group Select buttons. Loudspeaker Paging zones can be assigned to Trunk Group Select buttons.

The Trunk Group Busy/Warning Indicators to Attendant feature provides a visual indication of the busy or idle status of the trunk group or Loudspeaker Paging Access zone assigned to the Trunk Group Select button.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

If the Attendant Control of Trunk Group Access feature is provided, this feature must also be provided.

ATTENDANT DISPLAY

Description

Shows call-related information that the attendant needs for efficient operation of the console. Also shows personal-service and message information. Information is shown on the 40-character alphanumeric display on the attendant console.

The following display modes can be assigned to buttons in the display area or to programmable buttons on the console. The Normal and Test modes are always provided; the others are optional.

- Normal Mode

  Displays call-related information for the active call appearance. The alphanumeric display is in the Normal mode unless the attendant selects one of the other modes.

- Inspect Mode

  Displays call-related information for a call on hold.

- Stored Number Mode

  Displays the number assigned to a button administered through the Facility Busy Indication feature.

- Date/Time Mode

  Displays the current date and time of day.

- Test Mode

  Displays a test pattern representing each of the 40 characters that can be displayed. The Lamp Test switch is provided on the console; an additional button assignment is not needed.

- Elapsed Time

  Displays elapsed time in hours, minutes, and seconds. The timing starts or stops when the button is pressed.

- Integrated Directory

  Turns off the touch-tone signals and allows the touch-tone buttons to be used to key in the name of a system user. After a name is keyed in, the display shows that name and associated extension number. (Refer to the Integrated Directory feature.)

- Coverage Message Retrieval Mode

  Retrieves and displays Leave Word Calling and Call Coverage messages for system users. Messages can be retrieved at any time. The attendant does not need to lift the receiver to retrieve messages. Also, the attendant can be active on a call and still retrieve messages.

Three additional buttons can be assigned to the console when the Coverage Message Retrieval mode or the Integrated Directory mode is assigned. These buttons and their functions are as follows:

- Next Message

  Retrieves and displays the next message or displays END OF FILE, PUSH Next TO REPEAT when in the Coverage Message Retrieval Mode. Displays the next name in the alphabetical listing when in the Integrated Directory mode. This button must be assigned when the Retrieval mode button is assigned.

- Delete

  Deletes the currently displayed message. This button must be assigned when the Retrieval mode button is assigned.

- Return Call

  Automatically returns the call requested by the currently displayed message or the currently displayed name and extension number. This button is optional.

The system provides the following call-related information:

- Call Appearance Identification

  The six attendant call appearance buttons are labeled a through f. The display shows, for example, a= for a call incoming on the first call appearance button, b= for a call incoming on the second call appearance button, and so on.

- Calling Party Identification

  When the incoming call is internal, the identification consists of the caller's extension number or name. When the incoming call is external, the display shows OUTSIDE CALL or, as an option, a unique trunk identification such as DENVER.

- Called Party Identification

  When the attendant places an internal call, the display shows the called party's extension number or name. When the attendant places an external call, the display shows the trunk group name that was set by the System Manager.

- Internal Caller's Class of Restriction (COR)

  The COR display shows a 2-digit number followed by a hyphen and a 4-character restriction identifier. The restriction identifiers are as follows:

  ORIG—Origination restriction

  OTWD—Outward restriction

  TOLL—Toll restriction

  CODE—Code restriction

  NONE—No restriction

- Call Purpose

  This refers to calls that are directed, redirected, or returning to the console. The call purpose identifiers are as follows:

  hc—Held Call—Indicates that the administered interval for a held call expired and the call has returned to the console.

  rt—Return Call—Indicates that an attendant-extended call was not answered within the administered interval and the call has returned to the console.

  rc—Recall Call—Indicates that an internal user, active on a call held on the console, is requesting attendant assistance.

  tc—Trunk Control—Indicates that an internal user attempted to access an attendant-controlled trunk and the call was redirected to the console.

  priority—Indicates that a call has priority status.

  f—Call Forwarding—Indicates that an internal user has calls forwarded automatically to the attendant.

When the Call Coverage feature is active and the attendant is a covering user, the following call purpose identifiers are displayed:

s—Send All Calls—Indicates that the called voice terminal user is temporarily sending all calls to coverage.

c—Go to Cover—Indicates that the calling voice terminal user has sent the call to coverage.

b—Busy—Indicates that the called voice terminal user is active on a call.

d—Don't Answer—Indicates that the called voice terminal user is not available.

Some typical displays are as follows.

- Internal call originated by the attendant:

```
a=3602                               04-TOLL
```

or

```
a=      TOM BROWN                    04-TOLL
```

- Outgoing trunk call originated by the attendant:

```
b=87843541
```

Where 8 is the trunk access code and 784-3541 is the number dialed.

then

```
b=    OUTSIDE CALL
```

or

```
b=              WATS
```

- Incoming trunk call to the attendant:

```
a=    OUTSIDE CALL
```

- Conference call originated by the attendant:

```
b=   CONFERENCE 4
```

Where 4 is the number of conferees. The number does not include the attendant.

- Internal call redirected to coverage:

```
b=          3174  to  3077              c
```

or

```
b=      BOB SMITH  to  JOYCE THOMAS     c
```

Where c indicates that Go to Cover was activated by the calling voice terminal user.

- Incoming trunk call redirected to coverage:

```
a=  OUTSIDE CALL  to  DON SMITH        s
```

Where s indicates that Send All Calls was activated by the called voice terminal user.

- Coverage message retrieval:

```
IN PROGRESS
```

then

```
MESSAGES FOR BETTY R. SIMS
```

then

```
  JOE JONES 10/16 11:40a 2 CALL 3124
```

This message means that Joe Jones called Betty Sims the morning of October 16. The second message was stored at 11:40. Joe wants Betty to call his extension number, 3124.

- Integrated Directory mode:

```
CARTER, ANN          3408          3
```

This display shows the name and extension number as administered in the system. The 3 indicates that three buttons were pressed to reach this particular display.

Considerations

If the attendant group is administered for systemwide message retrieval, attendants can retrieve messages for voice terminal users. Permission to have coverage message retrieval must also be administered for the voice terminal user. It is not possible for selected attendants to retrieve messages for selected voice terminal users.

The following buttons can be administered:

- Coverage Message Retrieval
- Date and Time (one button)
- Delete Message (must be assigned if the Retrieval button is assigned)
- Elapsed Time
- Inspect Mode
- Integrated Directory
- Next Message (must be assigned if the Retrieval button is assigned)
- Normal Mode
- Return Call (optional, used with the Retrieval mode or the Integrated Directory mode)
- Stored Number

The display must be in the Normal mode for the attendant to answer incoming calls.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

ATTENDANT RECALL

Description

Allows voice terminal users on a 2-party call, or on an Attendant Conference call held on the console, to recall the attendant for assistance.

Single-line users press the Recall button or flash the switchhook to recall the attendant.

Multi-appearance users press the Conference or Transfer button to recall the attendant, and will remain on the connection when either button is used.

Considerations

The call must be held on the console.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

ATTENDANT RELEASE LOOP OPERATION

Description

Allows the attendant to hold the connection of any call off the console if completion of the call is delayed (such as a call extended to a busy single-line voice terminal or to a voice terminal that does not answer). This feature frees the attendant to handle other calls.

Timed Reminder starts once the call is off the console. If the called terminal user does not answer before the administered interval expires, the call returns to the attendant queue for further processing. A timed reminder tone is heard at the console chosen by the queue, and the alphanumeric display shows the call identification.

Considerations

None.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Alerting

  Timed reminder tone is provided by this feature.
- Attendant Display

  Call identification is provided by this feature.
- Timed Reminder

  The time-out interval is provided by this feature.

AUTOMATIC CALLBACK

Description

Allows internal users who placed a call to a busy or unanswered internal voice terminal to be called back automatically when the called voice terminal becomes available.

Before the system originates an Automatic Callback call, it checks to see that both the calling and called voice terminals are available. When the Automatic Callback call is originated, the calling voice terminal receives 3-burst alerting and the called party receives the same alerting provided on the original call. The alerting at the called voice terminal occurs immediately after the calling voice terminal user lifts the receiver.

When Automatic Callback is activated, the system assumes that the called voice terminal is not available for a call. In other words, the system must detect that the user hung up before the callback call is originated.

A single-line voice terminal user activates Automatic Callback by pressing the Recall button and dialing the Automatic Callback access code. Only one Automatic Callback call can be activated at any given time by a single-line user.

A multi-appearance voice terminal user can activate Automatic Callback for the number of Automatic Callback buttons assigned to the terminal. After placing a call to a voice terminal that is busy or that is not answered, the caller simply presses an idle Automatic Callback button and hangs up.

Considerations

An Automatic Callback button will also activate the Ringback Queuing feature.

An Automatic Callback request will be canceled for any of the following reasons:

- The called party is not available within 30 minutes.
- The calling party does not answer the callback call within the administered interval (2 to 9 alerting cycles).
- The calling party decides not to wait and presses the same Automatic Callback button a second time (multi-appearance voice terminal) or dials the Automatic Callback cancellation code (single-line voice terminal).

The system can process a maximum of 40 callback calls at one time.

Automatic Callback is administered to individual voice terminals by their Class of Service, and cannot be assigned to the attendant(s).

Multi-appearance voice terminals must have an Automatic Callback button to activate the feature.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Call Coverage

  Automatic Callback calls do not redirect to coverage.

- Call Pickup

  A group member cannot answer a callback call for another group member.

- Call Forwarding All Calls

  Automatic Callback cannot be activated toward a voice terminal that has Call Forwarding activated. However, if Automatic Callback was activated before the called voice terminal user activated Call Forwarding, the callback call attempt is made toward the called party and is not redirected.

- Attendant Call Waiting and Call Waiting Termination

  If Automatic Callback is activated to or from a single-line voice terminal, the Call Waiting features are denied.

Voice terminals with the following features cannot activate Automatic Callback:

- Hot Line Service
- Manual Originating Line Service
- Origination Restriction

Automatic Callback cannot be activated to the following:

- The attendant console group
- A voice terminal assigned Termination Restriction
- A data terminal (or data module)
- A Direct Department Calling group
- A Uniform Call Distribution group
- A Terminating Extension Group

AUTOMATIC ROUTE SELECTION

Description

Automatic Route Selection (ARS) is an optional feature that routes long-distance calls over the public network based on the preferred (normally the least expensive) route.

ARS selects patterns for the following types of calls:

- Calls made to central offices (COs) within the Home Numbering Plan Area (HNPA) Code. The HNPA is commonly called the Home Area Code.
- Calls made to Foreign Numbering Plan Area (FNPA) Codes (CO codes located outside the local Area Code). These are calls to Area Codes other than the Home Area Code.
- Calls made to COs located within the FNPA (calls to these COs are referred to as Remote Home Numbering Plan Area [RHNPA]) calls.
- Special service calls such as 911.
- International calls.

An ARS pattern may contain any or all of the following types of trunk groups for routing a call: Wide Area Telecommunications Service (WATS), Foreign Exchange (FX), Central Office (CO), and tie trunks.

Routing is controlled by as many as 16 routing patterns. A routing pattern is an ordered list of the routes (trunk groups plus a Facility Restriction Level [FRL], discussed later) the system can use to complete a particular call. Each pattern can contain up to six trunk groups arranged in the order of preference. Usually the first-choice trunk group is the least expensive trunk group.

The system selects a routing pattern from one of the six tables stored in memory. The six tables are: one HNPA, one FNPA, and four RHNPAs.

The HNPA table contains the ARS pattern number corresponding to the individual office codes within the Home Area Code that provides the trunk groups that can be used to place a call within the HNPA. Each ARS pattern number must be administered.

The FNPA table contains the 160 possible 3-digit Area Codes and Service Codes that can be accessed via ARS. The table also contains the ARS pattern number which must be used to make the call or the RHNPA table number. If the RHNPA table is entered, the system selects the ARS pattern number from the RHNPA table. The ARS pattern or the RHNPA table number is administrable for the FNPA.

The RHNPA table contains the CO codes that can be dialed for a particular FNPA. This table is used when an ARS pattern is not specified in the FNPA tables. This table gives the pattern number that must be used to call the CO codes. The ARS pattern must be administered on the RHNPA table.

Each trunk group assigned to an ARS pattern has an associated FRL number from 0 to 7. The same trunk group can be used in more than one pattern and can have a different FRL in each pattern. Each voice terminal user has an FRL assigned to it via the Class of Restriction (COR) form. A voice terminal assigned an FRL of 0 has the least privileges, an FRL of 7 has the greatest privileges. A trunk group with an FRL of 0 is least restricted, an FRL of 7 is most restricted.

The FRL denies or allows access to a trunk group in a pattern. When a user places a call the system selects a routing pattern based on the first three digits dialed (office or area code), or the first six digits (area and office code). The route pattern is selected from one of the six translation tables stored in memory.

Once a pattern is selected for the type of call being placed, the system attempts to select a trunk group to handle the call. The trunk groups in the ARS pattern are listed in the order of ascending FRLs. The trunk group with the lowest FRL number is checked first. The system compares the FRL associated with the caller to the trunk group FRL. The system will allow the caller to access the trunk group if the caller FRL is equal to or greater than the FRL of the trunk group.

Once the call satisfies the FRL requirements, the system automatically checks for an available trunk within the trunk group. If an idle trunk is selected the call is placed. If all trunk groups are busy, the call can queue on the first choice trunk group if queuing is provided for this trunk group. If the FRL requirements are met on a call, but no trunk group or queuing is available, reorder tone is returned to the caller. If the originating FRL does not authorize access to any trunk groups in the pattern, intercept tone is sent back to the caller. In this case the caller should not try the call again, because the call will always be denied regardless of how many trunks in the group are idle.

ARS Prefix Codes

ARS has four prefix codes which specify the ARS pattern to be used for 0, 0+, 01, and 011 calls. All 0-type calls must use the same pattern. The 0-type codes are listed below:

- 0 indicates operator access
- 0+ indicates operator assisted calls
- 01 indicates international operator
- 011 indicates international direct

The ARS prefix form is used to assign a pattern for these codes.

Considerations

The following items must be established, defined, and administered before Automatic Route Selection (ARS) can be activated:

- Facilities Restriction Levels (FRLs)

  Eight FRLs, numbered 0 through 7, allow and deny access to facilities. An FRL is assigned to each trunk group route within each routing pattern. A route assigned an FRL of 0 is the least restricted, a route assigned an FRL of 7 is the most restricted.

  Originating FRLs are assigned by Class of Restriction to a voice terminal, an incoming tie trunk group, a remote access trunk group, and the attendant group. An originating FRL of 0 has the least calling privileges, an originating FRL of 7 has the most calling privileges.

- Routing Patterns

  As many as 16 routing patterns, containing up to 6 routes each, can be used to control routing. Routing patterns are numbered from 1 to 16.

  Patterns are selected from the HNPA, FNPA, or RHNPA tables. Patterns are administered on the FNPA, HNPA, or RHNPA forms.

  Patterns 2, 3, 4, and 5 are default patterns for the RHNPA Tables 1, 2, 3, and 4, respectively. Pattern 1 is default for the HNPA Table Entry.

  The same trunk group can be assigned to several routing patterns. The FRL assigned to the individual route controls whether a call is allowed or denied.

The FRLs associated with each trunk group must be assigned in order of preference on the ARS pattern form. This keeps the system from searching the entire pattern to find the least expensive route.

- Home Numbering Plan Area (HNPA) and Foreign Numbering Plan Area (FNPA) tables

  The HNPA table defines up to 800 local CO codes, numbered from 200 through 999. A corresponding routing pattern number is also assigned.

  The FNPA table defines up to 160 foreign area codes and service codes. A corresponding routing pattern number or a reference to a Remote Home Number Plan Area (RHNPA) table also is assigned. To administer an RHNPA for a particular FNPA, the System Manager must enter r1, r2, r3, or r4 instead of the ARS pattern number.

- RHNPA Tables

  Up to four RHNPA tables can be established. These tables are normally reserved for FX trunks or for 6-digit translations. Each table can include the 800 possible CO codes, numbered from 200 through 999, and a corresponding routing pattern number. These tables will be associated with an area code referenced from the FNPA table. A pattern number must be entered in the RHNPA if an RHNPA table is used. The system defaults Tables 1, 2, 3, and 4 to ARS patterns 2, 3, 4 and 5, respectively.

- ARS Access Code

  The ARS access code is used to gain access to the ARS features. The access code is assigned on the Feature Access Code form. Only one code can be assigned. This code must be dialed before the 7- or 10-digit number can be dialed.

- Tie Trunk

  Tie trunks that are assigned as Advanced Private Line Termination (APLT) cannot be assigned as a trunk group in an ARS pattern. ARS inserts the HNPA if not dialed since private networks require ten digits on all calls routing to the public network.

- Toll Tables

  Up to four toll tables can be established. Toll lists are needed when an FX trunk group terminates at a step-by-step CO and requires the digit 1 on all toll calls.

  Toll tables are numbered from 1 through 4. The number of the toll table must be administered to the associated trunk group. They must be assigned to CO, FX, and WATS trunk groups.

- Prefix Table

  This table specifies the routing pattern number used for calls beginning with zero. This includes operator-assisted calls and directly dialed international calls. The ARS pattern used for these calls is assigned on the ARS Prefix Codes form.

The following tones are associated with ARS:

- Confirmation—Indicates that the call has queued.
- Busy—Indicates that the called number is busy.
- Reorder (fast busy)—Indicates that all trunks are busy and the queue is full or queuing is not provided on the most preferred route.
- Intercept—Indicates that the originating FRL is not sufficient to allow the call.

The ARS default patterns are as follows.

- Pattern 0 is the fixed intercept pattern. It is included in the ARS data existing in the system at all times. Intercepted ARS calls are routed to intercept tone. The intercept pattern is the automatic default pattern for all 160 entries of the FNPA table until changed for a specific system through the System Access Terminal (SAT). Intercept, Pattern 0, can also be the destination for any entry of the other NPA tables.
- Pattern 1 is the default pattern for the 800 office codes that make up the HNPA table. Therefore, when defining patterns, Pattern 1 should contain only local CO trunks to provide for completion of local calling area office codes. If less expensive facilities (that is, FX or WATS) have been provided for HNPA long-distance calls, another pattern(s) should be created and that (those) pattern number(s) should replace the Pattern 1 route associated with those office codes in the HNPA table. This is done using the series of ARS change commands at the SAT.

  Any nonworking home area office codes should remain routed to Pattern 1 (local CO trunks) so that they may be intercepted in the CO. This application eliminates the need to continually monitor and update working and nonworking office codes. When the nonworking code is activated in the CO, calls will automatically complete. Again, this arrangement can be changed using the SAT.
- Patterns 2, 3, 4, and 5 exist in ARS software as default patterns for RHNPA tables r1, r2, r3, and r4, respectively. If "r" tables are not used, these default patterns are also unused, and the pattern numbers (2, 3, 4, and 5) can be used to define any route. It is recommended, however, that these pattern numbers be held in reserve for reasons explained in the following paragraphs.

Primarily, the RHNPA tables exist to support the four possible FX trunk groups that can be included in an ARS configuration. Like the HNPA table, an RHNPA table contains 800 possible office codes including service codes (such as 411 and 911) associated with the CO where the FX trunks terminate. The RHNPA tables are administered and translated in the same manner as the HNPA table.

It should be noted, however, that FX trunk groups may have the same NPA as the local CO trunk group that serves the system. In this case, the HNPA table will serve the FX trunk group and an additional pattern(s) should be created to designate the FX trunk group as the first choice. This additional pattern(s) should then be assigned to the affected HNPA office codes to provide the least expensive route for long-distance calling.

When the FX trunk group terminates in an area code (NPA) other than the HNPA, calls to office codes that are long distance to the terminating office should be routed through the least expensive route. Service codes should be routed locally through the FX trunk group. Like the HNPA table, nonworking CO codes should be arranged so that they are intercepted in the CO.

When FX trunks are not provided, all "r" tables are available for 6-digit translation applications. Only one area code may be associated with an "r" table and, once initialized, routes for each of the possible 800 office codes must be considered. That is, the default pattern entry must be changed to reflect the appropriate pattern.

Hardware and Software Requirements

ARS is an optional feature that permits efficient use of trunk groups provided with the system. No additional hardware is required. ARS software is required.

Interactions

The following features interact with ARS:

- Abbreviated Dialing

  The ARS access code may be stored in an Abbreviated Dialing Group or System List. If the group or system list is privileged, the caller's Class Of Restriction (which contains the FRL) is never checked and any number in the list will be processed.

- Attendant Console

  If the attendant dials an ARS code for an outgoing call for voice terminal user, the system checks the attendant FRL to determine if the call can be made.

- Attendant Control of Trunk Group Access

  Activation of this feature removes the trunk group from ARS patterns, and deactivation reinserts the trunk group into the routing pattern. ARS calls do not redirect to the attendant.

- Controlled Restriction

  All ARS calls are denied when either Controlled Outward Restriction or Controlled Total Restriction has been activated for the calling extension.

- Miscellaneous Trunk Restriction

  This feature, if provided, does not apply to ARS calls. The route FRL is the controlling factor.

- Origination and Outward Restrictions

  These restrictions prohibit access to ARS.

- Personal Central Office Line Group (PCOLG)

  The PCOLG cannot be assigned to an ARS pattern.

- Remote Access

  The FRL of the incoming trunk group or the Remote Access barrier code, if used on the call, serves as the originating FRL. This FRL is contained in the Class of Restriction assigned to the trunk group or barrier code.

- Ringback Queuing

  When all accessible trunk groups in a routing pattern are busy, the call will queue on the most preferred trunk group (if queuing is provided and the queue is not full). Queuing is automatic for single-line voice terminals; however, multi-appearance voice terminal users must press an Automatic Callback button to activate the Ringback Queuing feature.

  Calls will queue on the trunk group only if the trunk group is administered for queuing. The number of calls that can be queued can range from 1 to 100. A zero indicates no calls will be held in queue. The system defaults the trunk group queue to zero. This applies to all trunk groups except Direct Inward Dialing.

- Station Message Detail Recording (SMDR)

  If SMDR generation is administered for a trunk group assigned to a routing pattern, data will be collected for all calls routed through the trunk group. If an SMDR account code is to be dialed an ARS call, it must be dialed before the ARS access code.

CALL COVERAGE

Description

Provides automatic redirection of certain calls to alternate answering positions in a Call Coverage path.

A Call Coverage path is a list of one, two, or three alternate answering positions (covering users) that will be accessed, in sequence, when the called individual or group (principal) is not available to answer the call. Any of the following can be assigned a Call Coverage path, and are thus eligible to have their calls redirected to coverage.

- Voice terminal
- Uniform Call Distribution (UCD) group
- Direct Department Calling (DDC) group
- Terminating Extension Group (TEG)
- Personal Central Office Line (PCOL) group

The System Manager establishes the coverage paths and sets the redirection criteria at the time the system is implemented. These paths and criteria can be changed at later dates. A coverage path can include any of the following:

- Voice terminal.
- Attendant group.
- UCD group.
- DDC group.
- Coverage Answer group, which is a group of up to eight voice terminals specifically established to answer redirected calls. All group members are alerted simultaneously. Any group member can answer the call.
- Message Center (if an Applications Processor is provided with the system).

The redirection criteria determine the conditions under which a call redirects from the principal (called) extension number to the first position in the coverage path. The criteria and conditions that apply are as follows:

- Active

  Redirects calls to call coverage immediately when the principal is active on at least one call appearance. For a voice terminal with only one appearance or a single line extension, the Busy criterion (discussed below) should be assigned instead of the Active criterion.

- Busy

  Redirects calls to coverage when all available call appearances at the principal extension are in use. For multi-appearance voice terminals, one call appearance is reserved for outgoing calls or incoming priority calls (discussed later). The remaining assigned call appearances are available for other incoming calls. An incoming call (other than a priority call) will redirect to coverage only when all of these unreserved call appearances are in use. If at least one unreserved call appearance is idle at the principal extension, the call will remain at that idle appearance.

  A TEG is considered busy if any voice terminal in the group is active on a call.

For a UCD or DDC group, each voice terminal in the group must be active on at least one call appearance in order for the call to be redirected to coverage. If any voice terminal in the group is idle (not active on any call appearance) the call directs to that voice terminal. If no voice terminal is available, the call can queue if queuing is provided. If queuing is not provided or if the queue is full, the call routes to coverage. Queued calls will remain in queue for a time interval equal to the Don't Answer Interval (discussed next).

- Don't Answer

  Redirects calls to coverage if unanswered during the assigned Don't Answer Interval (2 to 9 alerting cycles). A call will ring for the assigned Don't Answer Interval and then redirect to coverage.

- Cover All Calls

  Redirects all incoming calls to coverage. This criterion has precedence over any other criterion previously assigned.

- No Coverage

  Occurs when none of the above criteria have been assigned. Calls are only redirected to coverage when the principal has activated Send All Calls or the caller has activated Go to Cover. Both of these overriding criteria are discussed later.

Redirection criteria can be assigned in combinations; Active/Don't Answer and Busy/Don't Answer can be useful. Other combinations are not possible or do not provide any useful function. For example, Active/Busy does not accomplish anything. A busy voice terminal is always active.

Redirection criteria is assigned separately for internal and external calls. Thus, Busy/Don't Answer can be assigned for internal calls and Active can be assigned for external calls. Similarly, Busy/Don't Answer could apply for external calls and No Coverage could apply for internal calls. In the latter case, internal calls remain directed to the called terminal or group.

Certain overriding criteria are possible. These criteria, of course, are checked before the redirection criteria are checked. The overriding criteria are:

- Go to Cover

  This is optionally assigned to a button on a voice terminal and is activated by the internal calling party. If activated prior to alerting, the call does not attempt to direct to the called voice terminal or group, but goes directly to coverage. Go to Cover can be used later in a call. This is discussed later.

- Send All Calls

  Allows principals to temporarily direct all incoming calls to coverage regardless of the assigned redirection criteria. For example, if the redirection criteria are administered so that no calls redirect, all incoming calls will terminate at the principal's voice terminal unless Send All Calls is activated. Also, activating Send All Calls allows covering users to temporarily remove their voice terminals from the coverage path.

  Send All Calls is activated by pressing the Send All Calls button or by dialing the Send All Calls access code. The option is deactivated by pressing the button a second time or by dialing the deactivate code.

  Send All Calls is similar to Cover All Calls, discussed previously. However, Cover All Calls is set by the System Manager and would be used for screening the principal's call. The principal may or may not be alerted of an incoming call, depending on how this function is assigned. Send All Calls is controlled by the principal and is normally used when the principal will be away temporarily.

- Send Term

  This is the same function as Send All Calls, except Send Term is for a TEG. Since a TEG cannot be in a coverage path, this function only applies to a directly called TEG.

- Call Forwarding All Calls

  Call Forwarding provides a temporary override of the redirection criteria. The call attempts to complete to the forwarded-to extension number before redirecting to coverage. If the principal's redirection criteria are met at the forwarded-to extension, then the call is redirected to the principal's coverage path.

Call Coverage provides redirection of calls from the called principal or group to alternate answering positions when certain criteria are met. Yet the call is intended for the called principal or group. Certain provisions allow calls to direct to and/or be answered by the principal even though the redirection or overriding criteria are met. These provisions are:

- If no answering positions are available in the Coverage Path, the call alerts the called voice terminal, if possible; otherwise, the calling party receives busy tone. This applies even if the Cover All Calls redirection criterion or the Send All Calls overriding criterion is active.

- Similarly, calls directed to a UCD or DDC group will queue, if queuing is available, when no group members are available to answer the call. The call remains in queue for a time interval equivalent to the Don't Answer Interval before routing according to the Coverage Path. If no points on the path are available, the call remains in queue. The worst case is that group queuing and the coverage points are both unavailable. In this case, the caller receives busy tone.

- If the redirection criterion is Active or Cover All Calls, a called principal can receive a redirection notification signal (a short burst of alerting) when the call routes to coverage. (Redirection Notification is optional on a per-terminal basis.) Note that in the Active, Cover All Calls, and Don't Answer cases, the principal could answer the call. The Busy case means there are no call appearances available to answer the call. Redirected calls maintain an appearance on the called voice terminal, if possible. The call appearance status lamp flashes to indicate an incoming call before the call redirects. When the call does redirect, the status lamp lights steadily. The user can answer the call by pressing the call appearance button. If the call has already been answered, the principal is bridged onto the call. This provision is called Temporary Bridged Appearance.

- Priority Calling, Dial Intercom, and Automatic Intercom Calls always route directly to the principal's voice terminal unless the calling party has activated Go to Cover. These calls take precedence over the redirection criteria.

An internal calling party is informed that a call is redirecting to coverage by a single, short burst of alerting, called a Call Coverage tone. This tone is followed by an optional period of silence, called a Caller Response Interval. This interval allows the calling party time to decide what to do: hang up or activate Leave Word Calling, Automatic Callback, or Go to Cover. Activating Go to Cover cancels the remaining interval.

For specific Call Coverage needs, the following options are available to voice terminal users:

- Consult

  Allows the covering user, by first pressing the Conference or Transfer button and then the Consult button, to call the principal (called party) for private consultation. These two actions place the calling party on hold and establish a connection between the principal and the covering user. If the principal wishes, the covering user can complete the conference and add the calling party to the conversation. Similarly, the call can be transferred to the principal. (Consult calls use the Temporary Bridged appearance maintained on the call, if there is one. If not, the Consult call is denied.)

- Coverage Callback (Implied Principal Addressing)

  Allows a covering user, by pressing the Cover Callback button, to leave a message for the principal to call the calling party. The calling party must be an internal caller. The principal receives no indication that the covering user handled the call.

  Alternatively, if the covering user presses the Leave Word Calling button, a "call me" message is left for the principal. The principal calls the covering user to get the message. This method is used when an external call is received or when an internal caller wants to leave a message but will not be available for a return call.

- Coverage Answer Group

  A Coverage Answer Group can have up to eight members. When a call is redirected to a Coverage Answer Group, all voice terminals in the group alert (ring) simultaneously. Anyone in the group can answer the call. A Coverage Answer Group member already handling a group call will not be alerted when another call is redirected to that Coverage Answer Group. However, if a Coverage Answer Group member is also a member of another Coverage Answer Group, he or she can still receive calls for the other group.

- Coverage Incoming Call Identification (ICI)

  A Coverage ICI button can be assigned to multi-appearance voice terminal users without a display in a Coverage Answer Group.

  The Coverage ICI status lamp simply identifies a call incoming to that Coverage Answer Group. If a Coverage Answer Group is assigned to more than one Call Coverage path, the path number cannot be identified. Likewise, if a given path is assigned to more than one principal, the individual principals cannot be identified. To provide unique path and principal identification, the System Manager must establish a unique path for each principal and a unique Coverage Answer Group to be included in the path.

When a call meets the redirection criteria of the principal, the call attempts to route to one of up to three points in the coverage path, beginning with point 1. If no coverage points are available, the call may revert to the called principal or group. If any point in the path is available, the call either alerts the individual voice terminal or member of a group specified for that point or queues on the group. Once a call is alerting or queued at any point in a coverage path, the call never reverts to the called principal or group, or to the previous point. A call remains at a coverage point for a time equal to the Don't Answer Interval for Subsequent Redirection (2 to 9 alerting cycles or equivalent time interval). At the end of this time, the call attempts to route to any remaining points in the coverage path. If no other point is available to accept the call, the call will remain queued or continue alerting the current coverage point.

Call Coverage is an extremely flexible feature and allows various combinations of coverage points. To illustrate the usefulness of Call Coverage, three typical coverage arrangements are given here as an example.

- Executive Coverage

  Provides a principal with call redirection to covering users having a close working relationship with the principal. Because of the status of the principal, personalized answering should be provided. Also, the principal may or may not choose to answer his or her own calls.

A typical example of this form of coverage is when a principal's calls are redirected to a secretary. The secretary would be informed of the principal's daily schedule and other pertinent facts such as the importance of certain calls. The secretary could provide personalized answering by answering calls with the principal's name.

If the secretary is unavailable to answer the coverage call for the principal, the call redirects to a backup answering position. Personalized answering should also be provided at the backup position.

- Middle Manager Coverage

  Provides a group of principals with call redirection to one or more covering users (such as a secretary). The secretary should have some knowledge of the principal's daily schedule. A backup answering position should be provided in case the secretary is unavailable.

- General User Coverage

  Provides less-personal coverage for a broader spectrum of users. Covering users typically consist of a group or pooled answering arrangement. With this type of arrangement, coverage calls may be distributed among the members of the answering group.

As an example of how to provide a particular cover arrangement, the following provisions for the Executive Coverage arrangement are given.

- Determine if the secretary and backup position have a call display capability.
  - If so, Coverage Answer Groups are not required.
  - If not, establish a unique Coverage Answer Group for each one without a display. Specify only the applicable extension number. The Coverage Answer Group will contain only one member. Establish two groups, if required. Note that if the secretary and/or the backup answering position are in a Coverage Answer Group, each will receive only one redirected call for the executive at any given time. Calls do not alert a Coverage Answer Group member already busy on a call to the group. For frequently called executives, it is desirable that the secretary and possibly the backup answering position have a digital display capability.
- Establish a unique Call Coverage Path for the executive.
  - If the secretary will screen the calls, specify Cover All Calls as the redirection criteria.
  - If the executive will answer calls, specify Active, Busy, Don't Answer, Active/Don't Answer, or Busy/Don't Answer as desired.
  - Specify the secretary and the backup position (or the Coverage Answer Group(s) containing the secretary's and backup position's extension numbers) as the coverage points in the path.
- Optionally, specify a Send All Calls button on the executive's voice terminal. If someone else answers the executive's calls, the button is not needed.
- Specify a Send All Calls button and a Consult button on the secretary's voice terminal. Specify a Coverage ICI button if the secretary doesn't have a call display capability. Send All Calls is needed if the secretary will be unavailable for a period of time. Consult is needed to enable private consultation with the executive during an established call. Coverage ICI is needed to identify the call as a call to the executive rather than a personal call to the secretary.

- Specify a Consult button and a Coverage ICI button on the backup position's voice terminal for the same reasons these buttons were specified for the secretary.

Considerations

Call Coverage provides the means 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 Send All Calls and Consult, accommodate the day-to-day variations that occur in an employee's work schedule. Call Coverage was designed on the premise that incoming calls are intended for the called party, but suitable alternatives must be available if the called party cannot, or does not wish to, answer his or her own calls.

The following items should be considered when planning the Call Coverage feature:

- Caller Response Interval

  This interval can be from 0 to 10 seconds. If 0 is administered, the Caller Response Interval does not apply.

- Coverage Answer Groups

  The system allows for as many as 100 Coverage Answer groups with up to 8 voice terminals in each group.

  A Coverage Incoming Call Identification (ICI) button can be assigned to each multi-appearance voice terminal user without a display in a Coverage Answer group.

- Coverage Paths

  Up to 200 coverage paths can be established. Each coverage path can have one, two, or three coverage points. The same coverage path can be used for as many voice terminal users as desired.

- Don't Answer Interval and Don't Answer Interval for Subsequent Redirection

  The Don't Answer Interval specifies the number of alerting cycles heard at the principal's terminal before the call is redirected to the first coverage point. This interval is recommended to be two alerts, but can be administered from two to nine alerts. All principals with the same coverage path are assigned the same Don't Answer Interval.

  The Don't Answer Interval for Subsequent Redirection specifies the number of alerts at a covering terminal before the call attempts to redirect to the next coverage point. This interval is recommended to be two alerts, but can be administered from two to nine alerts. This interval is administered as a system parameter.

- Redirection Notification Signal

  This signal is administered on a per-terminal basis. If administered, the signal also applies to forwarded calls. With Call Coverage, the signal indicates to the caller that the call is being redirected to coverage because of the Active or Cover All Calls redirection criteria.

The following buttons can be assigned to multi-appearance voice terminals:

- Consult
- Coverage Callback
- Go to Cover

- Coverage ICI
- Send All Calls (can also be dialed using a feature access code)

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Attendant Display and Voice Terminal Display

  These features provide call identification for the covering user.

- Automatic Callback and Ringback Queuing

  Callback calls do not redirect to coverage. The caller can activate Automatic Callback when alerting, redirection notification signal, or busy signal is heard.

- Automatic Intercom, Dial Intercom, and Priority Calling

  Calls using these features will not redirect to coverage unless the caller presses the Go to Cover button.

- Call Forwarding All Calls

  Call Forwarding provides a temporary override of the redirection criteria. Normally, calls forward instead of redirecting to coverage. However, if a forwarding extension number's redirection criteria are met at the designated (forwarded-to) extension number, the forwarded call is handled as if Call Forwarding had not been activated. When the forwarded call goes to coverage, however, a temporary bridged appearance remains at the forwarded-to voice terminal until the call is answered.

  If Cover All Calls is part of the coverage redirection criteria and if Call Forwarding is active at a voice terminal, incoming Priority Calling calls forward to the designated extension number.

  The Redirection Notification Signal applies to both Call Coverage and Call Forwarding.

- Call Pickup

  Any call redirected to a covering user who is a member of a Call Pickup group can be answered by other members of the Call Pickup Group.

- Leave Word Calling

  Call Coverage can be used with or without Leave Word Calling. However, the two features complement each other. When a covering user activates Leave Word Calling during a coverage call, a message is left for the principal to call the covering user. When a covering user activates Coverage Callback during a coverage call, a message is left for the principal to call the internal caller.

- Temporary Bridged Appearance

  Calls redirected to coverage maintain an appearance on the called voice terminal if a call appearance is available to handle the call. The called party can bridge onto the call at any time.

Consult calls use the Temporary Bridged Appearance maintained on the call. At the conclusion of a consult call, the bridged appearance is no longer maintained. If the principal chooses not to talk with the calling party, the principal cannot bridge onto the call later.

- Direct Department Calling (DDC) and Uniform Call Distribution (UCD)

  When Send All Calls is activated or deactivated, the Make Busy function associated with DDC or UCD is also activated or deactivated simultaneously. However, activating or deactivating the Make Busy function does not activate or deactivate Send All Calls.

CALL FORWARDING ALL CALLS

Description

Allows all calls to an extension number to be forwarded to a selected extension number or to the attendant. This feature is activated or deactivated by dial access code.

Considerations

Attendants cannot forward calls. However, calls can be forwarded to the attendant group.

Calls can be forwarded only once. Calls forwarded to a designated (forwarded-to) extension number do not forward again. These calls alert the designated extension number, if possible; redirect if the forwarding party's Call Coverage criteria are met; or return busy tone to the calling party.

When Call Forwarding All Calls is activated at a voice terminal and a call for that terminal is forwarded, the terminal can receive a redirection notification signal that a call is being forwarded.

Leave Word Calling and Coverage Callback can be activated as usual when a call is forwarded.

The individual voice terminal Class of Service allows or denies access to this feature.

The following items can be administered:

- Class of Service (per voice terminal)
- Dial access codes (activate and deactivate)
- Redirection notification signal (per voice terminal)

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Automatic Callback and Ringback Queuing

  Callback calls do not forward.

- Call Coverage

  If a forwarding extension number's Call Coverage criteria are met at the designated (forwarded-to) extension number, the forwarded call is handled as if Call Forwarding had not been activated. When the forwarded call goes to coverage, however, a temporary bridged appearance remains at the forwarded-to voice terminal until the call is answered.

  If Cover All Calls is part of the coverage redirection criteria and if Call Forwarding is active at a voice terminal, incoming Priority Calls forward to the designated extension number and all other calls redirect according to the Call Coverage path.

  When a covering user has activated Call Forwarding, a coverage redirected call does not forward to the designated extension number. Instead, the call is redirected to the next point in the principal's coverage path, if available. If no other coverage point is available, the call will remain at the principal's voice terminal.

- Code Calling Access and Call Park

  Calls using these features override Call Forwarding. Code Calling Access and Call Park calls complete to the called extension number even if Call Forwarding is active.

- Hot Line Service and Manual Originating Line Service

  Voice terminals assigned these features cannot activate Call Forwarding. However, calls can be forwarded to these terminals.

CALL PARK

Description

Allows users to put a call on hold and then retrieve the call from any other voice terminal within the system.

When a voice terminal user, active on a call, needs to go to another location for information, the call can be placed in Call Park and retrieved at the other location.

Conference calls can also be placed in Call Park.

Call Park can be activated by any of the following:

- A single-line voice terminal user—Flash the switchhook (2500-series) or press the Recall button, dial the Call Park access code, and hang up or press the Recall button again.
- A multi-appearance voice terminal user with an assigned Call Park button—Press the Call Park button. The call is parked on the user's extension number.
- A multi-appearance voice terminal user without an assigned Call Park button—Press the Transfer or Conference button, dial the Call Park access code, and press the Transfer or Conference button again. The call is parked on the user's extension number.
- An attendant—Press Start, dial the Call Park access code followed by any extension number, and press Release. The call will be parked on the number dialed. An attendant can use the Direct Extension Selection With Busy Lamp Field feature instead of dialing the extension number.
- The system—When Code Calling Access is used, the call is automatically parked on the paged party's extension number.

Calls are retrieved by dialing the Call Park Answer Back access code and the extension number where the call is parked, or by pressing the same Call Park button used to park the call.

Considerations

Only one call per extension number can be parked at a time, even if the extension number has multiple appearances. However, a conference call with five parties (multi-appearance voice terminal) or two parties (single-line voice terminal) can be parked. The sixth and third conferees, respectively, will be the retrieving parties.

Calls cannot be parked on a group extension number. If a group member places a call in Call Park, the call will be parked on the member extension number. Group members include the following:

- A Coverage Answer group member
- A Direct Department Calling group member
- A Terminating Extension Group member
- A Trunk Answer From Any Station answering user
- A Uniform Call Distribution group member

An expiration interval for a parked call can be administered for the system. When the interval expires, the parked call will redirect to an attendant console and will no longer be parked on the extension number.

Up to 10 common shared extension numbers can be assigned to the attendant console group. These extension numbers are not assigned to a voice terminal, but are used by the attendant to park a call. These extensions should be assigned to the optional selector console in the 00 through 09 block (bottom row) in any hundreds group that the attendant can easily identify. The lamp associated with the extension number will identify call parked or no call parked (instead of active or idle status).

This feature can be used by any voice terminal user or attendant.

The following items can be administered:

- Call Park access code
- Answer Back access code
- Call Park expiration interval (from 1 to 10 minutes in intervals of 5 seconds)
- Call Park button (multi-appearance voice terminals only)
- Common shared extension numbers for the attendant group (from 1 to 10)

The common shared extension numbers are stored in system translations.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Data Privacy and Data Restriction

  These features are automatically deactivated when a call is parked.

- Loudspeaker Paging Access

  Calls to paging zones cannot be parked.

- Music-on-Hold

  Music can be provided to one party on a parked call. However, music cannot be provided to a multiple-party (conference) parked call.

- Remote Access

  A Remote Access caller cannot park a call. However, the Code Calling Access feature, an answering attendant, or a voice terminal user can park an incoming Remote Access call.

CALL PICKUP

Description

Allows voice terminal users to answer calls to other extension numbers within the user's specified Call Pickup group.

Considerations

The Leave Word Calling and Coverage Callback features can be activated as usual when Call Pickup group members answer calls.

The following items can be administered:

- Up to 200 Call Pickup groups with up to 25 members per group. (A voice terminal can be a member of only one Call Pickup group.)
- Call Pickup group identifying number (per group).
- Extension numbers within each group.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Automatic Callback and Ringback Queuing

  Callback calls cannot be answered by Call Pickup group members.

- Hot Line Service and Manual Originating Line Service

  Voice terminals assigned these features can be Call Pickup group members so their incoming calls can be answered. However, voice terminal users with these features assigned cannot answer calls for other group members.

- Call Forwarding All Calls

  A forwarded call cannot be picked up at the forwarded-to voice terminal unless the forwarding and forwarded-to voice terminals are in the same pickup group.

CALL WAITING TERMINATION

Description

Provides for calls to busy single-line voice terminals to wait until the voice terminal is idle. The called party hears one quick burst of tone when a call from another voice terminal user is waiting, two quick bursts of tone when an attendant-handled or an outside call is waiting, or three quick bursts of tone when a Priority Call is waiting. The called party hangs up on the current call and immediately receives ringing from the waiting call.

The calling party hears special audible ringback tone if the call is allowed to wait. If Call Waiting is denied, the calling party hears busy tone. Only one call can wait at a time.

Considerations

The burst(s) of tone heard by the called voice terminal user is not heard by other parties on the call.

Calls to multi-appearance voice terminals are routed to an idle call appearance and do not wait.

An internal caller can activate Leave Word Calling or Automatic Callback after Call Waiting has been activated.

A Priority Call and an attendant-handled call can wait for the voice terminal to become idle even if the Call Waiting Termination feature is not assigned.

Calls to a Direct Department Calling or Uniform Call Distribution group voice terminal cannot wait. However, such calls can enter the group queue, if provided, and if the queue is not full.

The Call Waiting Termination feature can be administered to any single-line voice terminal.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

Call Waiting is denied when the following features are activated at the single-line voice terminal:

- Automatic Callback (to or from the voice terminal)
- Data Privacy
- Data Restriction

CLASS OF RESTRICTION

Description

The Class of Restriction (COR) parameters define up to 64 different classes of call origination and termination privileges. Systems may have only a single COR, one with no restrictions. Other systems may have as many CORs (up to 64) as necessary to effect the desired restrictions.

A COR is assigned to each of the following:

- Trunk group
- Voice terminal
- Data module
- Loudspeaker Paging Access zone
- Code Calling Access zone
- Remote Access barrier code
- Attendant consoles (as a group)
- Terminating Extension Group
- Uniform Call Distribution group
- Direct Department Calling group

Use of CORs can be categorized as follows:

- Calling party restrictions
- Called party restrictions
- Miscellaneous restriction groups
- Selective denial of public network calling through a Common Control Switching Arrangement (CCSA) or Enhanced Private Switched Communications Service (EPSCS) network
- An Automatic Route Selection (ARS) Facilities Restriction Level (FRL) for control of call routing

Features assignable as calling party restrictions are as follows:

- Code Restriction
- Origination Restriction
- Outward Restriction
- Toll Restriction

Features assignable as called party restrictions are as follows:

- Inward Restriction
- Manual Terminating Line Restriction
- Termination Restriction

Use of CORs

CORs can be used to assign a variety of restrictions to a variety of facilities. The types of restrictions which can be assigned are discussed in the following paragraphs. As an aid to understanding CORs, the screen form used to administer CORs is shown in Figure 5-1. The values shown on the form are the default values. However, these values can be changed to implement the desired restrictions.

```
                                                                    Page 1 of 1
                              CLASS OF RESTRICTION
COR Number: _

     APLT?: y                                                        FRL: 7
                                           Calling Party Restriction: none
                                            Called Party Restriction: none
CALLING PERMISSION (Enter "y" to grant permission to call specified COR)
     0? y      8? y     16? y     24? y     32? y     40? y     48? y     56? y
     1? y      9? y     17? y     25? y     33? y     41? y     49? y     57? y
     2? y     10? y     18? y     26? y     34? y     42? y     50? y     58? y
     3? y     11? y     19? y     27? y     35? y     43? y     51? y     59? y
     4? y     12? y     20? y     28? y     36? y     44? y     52? y     60? y
     5? y     13? y     21? y     29? y     37? y     45? y     53? y     61? y
     6? y     14? y     22? y     30? y     38? y     46? y     54? y     62? y
     7? y     15? y     23? y     31? y     39? y     47? y     55? y     63? y
```

Figure 5-1. Example of Screen Form Used for Implementing CORs

Calling Party and Called Party Restrictions

Calling party restrictions prevent specified users from placing certain calls or accessing certain features. Features assignable as calling party restrictions are Code Restriction, Origination Restriction, Outward Restriction, and Toll Restriction. These individual features are fully described elsewhere in this section. A brief description is given here:

- Code Restriction—Denies the specified voice terminal completion of outgoing calls to selected office and area codes.
- Outward Restriction—Prevents callers at specified voice terminals from activating the Public Network Access feature. Calls can be placed to other voice terminal users, to an attendant, and to tie trunks.

- Origination Restriction—Prevents callers at specified voice terminals from originating calls. Voice terminal users can, however, receive calls.

- Toll Restriction—Prevents callers at specified voice terminals from placing toll calls unless the called office code, area code, or service code is on an allowed calls list. This list can contain up to ten entries.

Called party restrictions prevent specified users from receiving certain calls. Features assignable as called party restrictions are Inward Restriction, Manual Terminating Line Restriction, and Termination Restriction. These individual features are fully described elsewhere in this section. A brief description is given here:

- Inward Restriction—Restricts users at specified voice terminals from receiving public network, attendant-originated, and attendant-extended calls.

- Manual Terminating Line Restriction—Restricts users at specified voice terminals from receiving calls other than those from an attendant.

- Termination Restriction—Restricts users of specified voice terminals from receiving any calls.

Looking at the screen form used to administer CORs (see Figure 5-1), the field labeled "Calling Party Restriction" and the field labeled "Called Party Restriction" are both administered as "none." However, the field for "Calling Party Restriction" could be administered as Code, Origination, Outward, or Toll. Likewise, the field for "Called Party Restriction" could be administered as Inward, Manual (Manual Terminating Line), or Termination. Including "none" as a choice of restrictions, as many as 20 combinations of calling and called party restrictions are possible. However, it is unlikely that all 20 combinations will be needed in any one situation. Therefore, only the required ones should be established.

Calling and called party restrictions are the basis for all CORs. In cases where no restrictions are needed, a single COR could be assigned with calling and called party restrictions of "none." This same COR could be used for unrestricted voice terminals, trunk groups, terminating extension groups, UCD groups, DDC groups, and the attendant group.

The following are typical examples of calling and called party restrictions which may be assigned to a COR:

- Long distance calling is to be limited by the Code Restriction feature, but there will be no restrictions on incoming calls.

  - Calling party restriction=Code

  - Called party restriction=None

- A voice terminal in a storeroom should not be used for outside calling. Also, all incoming calls should be from internal callers.

  - Calling party restriction=Outward

  - Called party restriction=Inward

- A voice terminal in a certain department cannot be used for outside calling. Incoming calls must be from the attendant (assuming that department cannot be dialed directly from the outside).

  - Calling party restriction=Outward

  - Called party restriction=Manual Terminating Line

- Certain voice terminals are to be included in a Uniform Call Distribution group for answering business calls only. These terminals are not to be used individually.
  - — Calling party restriction=Origination
  - — Called party restriction=Termination

The called party restriction is checked only at the called terminal, module, attendant console, zone, or group. For example, if a call redirects from one voice terminal to another, as through the Call Coverage feature, the called party restriction of the called (redirected from) voice terminal is the only one checked.

Each COR is established as needed and is arbitrarily identified by a number, 0 through 63. For example, if the class of restriction for the storeroom example is 12, the storeroom voice terminal(s) is assigned COR 12.

Selective Denial of Public Network Calling Through a Common Control Switching Arrangement (CCSA) or Enhanced Private Switched Communications Services (EPSCS) Network (APLT Field)

Public network calling via the private CCSA or EPSCS network (commonly referred to as off-network calling) is optional on a per-private network basis. If off-network calling is not provided, then the APLT field can be ignored. If off-network calling is provided, then permission or denial to access the off-network capability is set via the APLT field. Users assigned a COR that has APLT set to "y" (yes) can use off-network calling. Users assigned a COR that has APLT set to "n" (no) cannot. If there is a need for both yes and no choices in a system, separate CORs must be assigned to reflect this.

Looking at the screen form used to administer CORs (Figure 5-1), the field labeled APLT is preset as "y." This means that, yes, a facility with this COR is allowed to access CCSA or EPSCS off-network capabilities for public network calling. An "n" in this field would indicate that the facility cannot access CCSA or EPSCS off-network capabilities.

Automatic Route Selection (ARS) Facilities Restriction Level (FRL) for Control of Call Routing

If the system does not use ARS to determine the most preferred routing of public network calls, then the FRL field can be ignored. If ARS is used, then an FRL is used to either allow or deny access to certain routes. The FRL for the outgoing (trunk) side of the call is provided in the ARS Routing Pattern. Although each outgoing trunk group has a COR and each COR has an FRL, this FRL is not used. Call routing is determined by a comparison of the FRLs in the ARS Routing Pattern and the FRL in the COR of the call originator (typically, a voice terminal user).

The FRL field (see Figure 5-1) is preset to 7. However, this field can have a value of 0 through 7. An originating FRL of 0 has the least calling privileges, whereas an originating FRL of 7 has the most calling privileges. Each of the up to 6 routes in each of the up to 16 ARS Routing Patterns also has an FRL. These route FRLs can also have a value of 0 through 7. A route FRL of 0 is the least restrictive, whereas a route FRL of 7 is the most restrictive. An FRL of 0 will be checked before the other routes in a given ARS routing pattern. To access a route, the originating FRL must be greater than or equal to the route FRL. Determination of appropriate FRL values must be made with respect to the outgoing routes from a specific system and the desired levels of calling privileges. This is part of ARS customization. The FRL of the call originator is contained in the COR assigned. The FRL field in a COR assigned to an outgoing trunk group is never checked and should be ignored.

Assuming ARS has been customized for a system, the System Manager must establish unique CORs for each of the up to eight levels of ARS calling privileges that will be used in the system. However, these CORs must maintain the desired restrictions dictated by the other fields on the screen form. The simplest case is a COR specifying no restriction. Ordinarily, this COR can be assigned to all unrestricted users. However, if some subset(s) of these users requires different FRLs, separate CORs must be established for each different FRL required.

Both ARS and FRLs are fully described under Automatic Route Selection elsewhere in this section.

Miscellaneous Restriction Groups

Miscellaneous Trunk and Miscellaneous Terminal Restriction groups restrict access to a terminal, module, zone, attendant console, or group. This is accomplished via the COR assigned to the calling and the called facilities. When a COR is administered, access by that COR to each of the 64 CORs is either allowed or denied. Since a given COR can be assigned to both calling and called facilities, calling to one's own COR can be restricted. This is fully explained below.

The simplest way to understand miscellaneous restrictions is to look at the screen form used during implementation (see Figure 5-1). When a COR is established, the assigned number is entered in the COR Number field. If this COR is assigned to a facility that originates a call, such as a voice terminal, the calls to CORs associated with terminating facilities can be prohibited. The Miscellaneous Restriction group information is found in the Calling Permission field. A "y" entry in this field indicates that the COR specified at the top of the form can call the COR numbers that contain a "y." If an "n" is entered, the specified COR cannot be called by the COR number at the top of the form. On the screen form in Figure 5-1, no restrictions apply because all 64 CORs are specified as "y."

The screen form in Figure 5-2 gives an example of Miscellaneous Restriction groups. This form is for COR 6 as is indicated in the COR Number field. The 64 COR numbers in the Calling Permission field relate which CORs can or cannot receive a call from a facility with a COR of 6. In the example shown:

- A facility with a COR of 3, 7, or 10 cannot be called by a facility with a COR of 6.
- A facility with any COR other than 3, 7, or 10 can be called by a facility with a COR of 6.

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                                                              Page 1 of 1

                          CLASS OF RESTRICTION

  COR Number: 6                                                  FRL: 7

       APLT?: y                          Calling Party Restriction: none

                                          Called Party Restriction: none

  CALLING PERMISSION (Enter "y" to grant permission to call specified COR)

      0? y    8? y   16? y   24? y   32? y   40? y   48? y   56? y

      1? y    9? y   17? y   25? y   33? y   41? y   49? y   57? y

      2? y   10? n   18? y   26? y   34? y   42? y   50? y   58? y

      3? n   11? y   19? y   27? y   35? y   43? y   51? y   59? y

      4? y   12? y   20? y   28? y   36? y   44? y   52? y   60? y

      5? y   13? y   21? y   29? y   37? y   45? y   53? y   61? y

      6? y   14? y   22? y   30? y   38? y   46? y   54? y   62? y

      7? n   15? y   23? y   31? y   39? y   47? y   55? y   63? y
```

Figure 5-2. Screen Form Used to Explain Miscellaneous Restriction Groups

Miscellaneous Restriction groups apply on a per-COR basis. However, the same COR can be assigned to more than one facility. Facilities with the same COR may be like facilities (such as two voice terminals) or different facilities (such as a voice terminal and a trunk group). In either case, the same restrictions apply to both facilities.

Certain facilities, such as voice terminals, can originate and receive calls. Call origination and termination restrictions are specified via a single COR. Miscellaneous Restrictions can prevent calling any COR, including one's own COR. If, in Figure 5-2, COR 6 in the Calling Permission field is set to "n," then an originating facility with a COR of 6 cannot call any facility with a COR of 6. This means that two voice terminals, each with a COR of 6, cannot call each other.

When a COR is administered, the allowance or denial of access from that COR to each of the 64 CORs applies only to the COR being administered. For example, if COR 6 is administered with access denied to COR 3, this only specifies that COR 3 cannot receive a call from COR 6. Whether or not COR 3 can be accessed by any other COR (for example, COR 7) is determined when that COR (COR 7) is administered. From this, it follows that a single COR cannot be used to provide both unrestricted service and miscellaneous restrictions.

Assignment of Restrictions

A COR is assigned to each of the following:

Voice Terminals

All voice terminals must be assigned a COR. The same COR may be assigned to all voice terminals or a unique COR may be assigned to a particular voice terminal or group of voice terminals. This COR applies individually to each voice terminal and is independent of all other COR applications, such as Miscellaneous Restriction groups or Uniform Call Distribution (UCD) groups.

The main items of concern for individual voice terminals are calling party restrictions and called party restrictions (discussed previously under "Use of CORs"). If no restrictions are needed for a certain group of voice terminals, "none" can be specified for both calling party and called party restrictions. If it is desired to restrict a group of voice terminals from making outside calls, a COR specifying a calling party restriction of "outward" should be established.

Additionally, miscellaneous restrictions, restrictions to CCSA and EPSCS off-network calling capabilities, and FRLs also apply. A separate COR must be established for each unique set of restrictions.

Trunk Groups

Each trunk group is assigned a COR. Trunk groups are assigned CORs mainly for the use of miscellaneous restrictions. For example, in Figure 5-2, access to trunk groups with a COR of 3, 7, or 10 is denied to facilities with a COR of 6.

Calling party and called party restrictions should be "none." Whether or not a trunk group is toll or code restricted is specified on the trunk group form used during implementation.

For Toll Restriction to apply on a call, either the trunk group or the originating facility must specify "toll." For Code Restriction to apply on a call, both the trunk group and the originating facility must specify "code." The originating facility can be specified as "code" or "toll" via the Calling Party Restriction field of the COR assigned to the facility.

Attendant Consoles (as a group)

Attendants are normally allowed full access to the system's capabilities. Therefore, calling and called party restrictions will usually be "none." Also access to the attendant is normally allowed to all CORs. This is accomplished via a "y" (yes) for the attendant's COR in the Calling Permission field on the screen form for each assigned COR.

Data Module, Loudspeaker Paging Access Zone, Code Calling Access Zone, and Remote Access Barrier Code

Each data module, Loudspeaker Paging Access zone, Code Calling Access zone, and Remote Access barrier code is assigned a COR. Normally, the calling and called party restrictions of CORs assigned to these facilities will be "none." Through Miscellaneous Restriction groups certain users are allowed access to certain facilities, while other users are denied access. For example (looking at Figure 5-2), if a loudspeaker paging access zone has a COR of 3, 7, or 10, then a voice terminal with a COR of 6 cannot access that Loudspeaker Paging Access zone.

Terminating Extension Group, Uniform Call Distribution Group, and Direct Department Calling Group

These groups are set up to receive calls. A COR is assigned to each group. This COR is distinct and separate from CORs assigned to the individual group members. The group COR allows or denies calls to the group. Since Miscellaneous Restriction groups are normally used to restrict calling, called party restrictions should be specified as "none." Since a group cannot originate a call, calling party restrictions do not apply. However, for simplicity, "none" is normally specified. For calls by group members or calls to individual group members, the COR assigned to the voice terminal applies. The group COR has no effect on calls directly to or from a group member.

The important aspect of these CORs is that they allow the called party restrictions of the group (normally none) to be different from the called party restrictions of the individual group members (Inward, Manual Terminating Line, or Termination).

Class of Restriction Examples

The examples given below are designed to help in the understanding of CORs and to illustrate some of the practical aspects of CORs. These are, however, only examples. In reality, each system must be administered to meet its individual needs.

Example Using Miscellaneous Restrictions

As an illustration of miscellaneous restrictions, assume a System 75 installation provides the following:

- Central office trunks
- WATS
- FX trunks
- Data modules
- Attendant service
- Voice terminals
- Direct Inward Dialing (DID) trunks
- Remote Access

In an unrestricted environment, each of the above facilities could have the same COR. However, suppose the following requirements exist:

- Attendants cannot make data calls.
- Remote Access can be used for data calls only.
- DID cannot be used for data calls except through Remote Access. (A dedicated Remote Access trunk group is not required, although one or more could be provided. This example assumes all Remote Access is via DID.)
- There are three classes of voice terminals:
  - — Those that can call anywhere, any time.
  - — Those that can place local central office and in-house calls only.
  - — Those that can place local central office, FX, and in-house calls only.

To implement the above requirements, a COR must be assigned to each facility or group of facilities. For simplicity, each can have a unique COR. The CORs are arbitrarily assigned as follows:

- COR 30—Local central office trunks
- COR 31—WATS trunks
- COR 32—FX trunks
- COR 33—Data modules
- COR 34—Attendant group

- COR 35—Unrestricted voice terminals
- COR 36—Voice terminals that can place in-house and local calls only (no FX or WATS calls)
- COR 37—Voice terminals that can place in-house, local, and FX calls only (no WATS calls)
- COR 38—DID trunk group
- COR 39—One of the remote access barrier codes (can be up to ten)

With the CORs defined, it should be individually determined which CORs cannot call other CORs. This is done as follows:

- COR 30 (local central office trunks)—No restrictions were specified for these trunks. The default values on the screen form (see Figure 5-1) are sufficient. No action is required, except to specify a COR number of 30.
- COR 31 (WATS)—CORs that cannot use WATS are specified as they are encountered. WATS itself is an outgoing service without any calling capabilities. Thus, Miscellaneous Restrictions are not specified on this form. The Calling Party Restriction should be "none" (although this restriction does not really have any meaning for an outgoing facility). Similarly, the Called Party Restriction applies to facilities capable of answering a call. Since this is not the case with WATS, "none" should be specified. Again, the default values are sufficient, so only the COR number needs to be specified.
- COR 32 (FX)—According to the requirements for this example, no restrictions apply. Reasons are the same as for WATS. Only the COR number needs to be specified.
- COR 33 (data modules)—No restrictions apply for reasons similar to the reasons why no restrictions were assigned for WATS. Only the COR number needs to be specified.
- COR 34 (attendant group)—The attendant group cannot call COR 33 (data modules). Specify an "n" beside COR 33 in the Calling Permission field. Specify 34 in the COR Number field.
- COR 35 (unrestricted voice terminals)—Since no restrictions were specified, only the COR number needs to be entered.
- COR 36 (no FX or WATS calls)—This COR cannot call COR 32 (FX) or COR 31 (WATS). Specify an "n" beside CORs 32 and 31 in the Calling Permission field. Specify 36 in the COR Number field.
- COR 37 (no WATS calls)—This COR cannot call COR 31 (WATS). Specify an "n" beside COR 31 in the Calling Permission field. Specify 37 in the COR Number field.
- COR 38 (DID)—This COR cannot call COR 33 (data modules). Specify "n" beside COR 33 in the Calling Permission field. Enter 38 in the COR Number field.
- COR 39 (Remote Access barrier code)—This COR can be used for data calls only. Thus, this COR can call COR 33, but not CORs 30 (local central office), 31 (WATS), 32 (FX), 34 (attendant group), 35, 36, or 37 (voice terminals). Specify an "n" beside CORs 30, 31, 32, 34, 35, 36, and 37 in the Calling Permission field. Enter 39 in the COR Number field. (The CORs listed in the Calling Permission field can be viewed as terminating or screening CORs that can or cannot be called by the originating COR. Since COR 38 [DID] is neither a terminating or screening COR, it did not have to be considered when assigning the barrier code COR.)

Example Using Calling Party Restrictions, Called Party Restrictions, and Miscellaneous Restrictions

To illustrate the use of both Calling and Called Party restrictions, and Miscellaneous restrictions, assume a System 75 installation provides the following:

- Central office trunks (outgoing)
- WATS
- FX trunks (outgoing)
- Voice terminals
- Data modules
- Terminating Extension Groups
- Loudspeaker Paging

Suppose that the following requirements exist:

- Only the attendant can access loudspeaker paging.
- Terminating Extension Groups can only accept calls from internal voice terminals.
- There are six classes of voice terminals:
  - Those that are toll restricted.
  - Those that cannot call outside to a public network (outward restricted).
  - Those that can receive calls only from an attendant.
  - Those that can call anywhere, any time.
  - Those that cannot place FX or WATS calls.
  - Those that cannot place WATS calls.

To implement the above requirements, a COR must be assigned to each facility or group of facilities. For simplicity, each can have a unique COR. The CORs are arbitrarily assigned as follows:

- COR 40—Local central office trunks
- COR 41—WATS trunks
- COR 42—FX trunks
- COR 43—Attendant group
- COR 44—Data modules
- COR 45—Terminating Extension Groups
- COR 46—Loudspeaker Paging Access Zones
- COR 47—Unrestricted voice terminals
- COR 48—Voice terminals that are toll restricted
- COR 49—Voice terminals that are outward restricted
- COR 50—Voice terminals that can only receive calls from an attendant
- COR 51—Voice terminals that cannot place FX or WATS calls
- COR 52—Voice terminals that cannot place WATS calls

With the CORs defined, it should be determined individually which CORs cannot call other CORs. This is done as follows:

- COR 40 (local central office trunks)—Restrictions that prohibit access to this COR are assigned when the originating CORs are considered. Only the COR number has to be specified on this form.
- COR 41 (WATS)—This is the same case as described in the previous configuration example. Only the COR number needs to be specified.
- COR 42 (FX)—Again, only the COR number needs to be specified.
- COR 43 (attendant group)—No restrictions were stated, so only the COR number needs to be specified.
- COR 44 (data modules)—No restrictions were stated, so only the COR number needs to be specified.
- COR 45 (Terminating Extension Group)—This COR can receive internal voice terminal-originated calls only. Since no tie trunks are specified for this example, the Inward Restriction feature can provide the desired restriction. Specify "inward" as the Called Party Restriction. If dial repeating tie trunks were provided, Miscellaneous Restrictions could be used to deny trunk access to the group. Also, specify 45 as the COR number.
- COR 46 (Loudspeaker Paging Access zones)—Since this COR can be accessed by an attendant only, the Manual Terminating Line feature can provide the restriction. Specify "manual" as the Called Party Restriction. Specify 46 as the COR number.
- COR 47 (unrestricted voice terminals)—No restrictions were stated, so only the COR number needs to be specified.
- COR 48 (toll restricted voice terminals)—Specify "toll" as the Calling Party Restriction. Specify 48 as the COR number.
- COR 49 (outward restricted voice terminals)—Specify "outward" as the Calling Party Restriction. Specify 49 as the COR number.
- COR 50 (voice terminals that can only receive calls from an attendant)—Specify "manual" as the Called Party Restriction. Specify 50 as the COR number.
- COR 51 (voice terminals that cannot place WATS or FX calls)—None of the Calling Party Restrictions uniquely prohibit WATS and FX calls, so Miscellaneous Restrictions are used. Enter an "n" beside COR 41 (WATS) and COR 42 (FX) in the Calling Permission field. Leave the Calling Party Restriction as "none" and specify 51 as the COR number.
- COR 52 (voice terminals that cannot place WATS calls)—Enter an "n" beside COR 41 (WATS) in the Calling Permission field. Leave the Calling Party Restriction as "none" and specify 52 as the COR number.

Another method to determine COR assignment is to consider the restrictions to be assigned. The requirements given for the second example were as follows:

- Only the attendant can access loudspeaker paging.
- Terminating Extension Groups can only accept calls from internal voice terminals.
- The six classes of voice terminals are:
  - Those that are toll restricted.
  - Those that cannot call outside to a public network (outward restricted).

— Those that can receive calls only from an attendant.

— Those that can call anywhere, any time.

— Those that cannot place FX or WATS calls.

— Those that cannot place WATS calls.

Assignments for these requirements could be made as follows:

- COR 60—Manual Terminating Line Restriction
- COR 61—Inward Restriction
- COR 62—Toll Restriction
- COR 63—Outward Restriction

  Note: A new Manual Terminating Line Restriction for voice terminals was not established. COR 60, above, can be assigned.

- COR 64—Unrestricted
- COR 65—Provides Miscellaneous Restrictions for WATS and FX (not yet complete since a COR has not been assigned to WATS or FX)
- COR 66—Provides Miscellaneous Restriction for WATS (not yet complete since a COR has not been assigned to WATS)
- COR 67—COR for WATS. Enter an "n" beside COR 67 in the Calling Permission field on the form for COR 65 and COR 66.
- COR 68—COR for FX. Enter an "n" beside COR 68 in the Calling Permission field on the form for COR 65.

Now assign the appropriate COR to each physical or screening facility:

- Central office trunks—COR 64 (unrestricted)
- WATS—COR 67 (WATS COR)
- FX—COR 68 (FX COR)
- Attendant group—COR 64 (unrestricted)
- Voice terminals—COR 62 (toll), COR 63 (outward), COR 60 (manual), COR 64 (unrestricted), COR 65 (WATS and FX miscellaneous), or COR 66 (WATS miscellaneous), as required
- Data Modules—COR 64 (unrestricted)
- Terminating Extension Group—COR 61 (inward)
- Loudspeaker Paging trunks—COR 60 (manual)

This latter method is probably more difficult to use, but minimizes the number of CORs established. This method required 9 CORs to effect the same restrictions as 13 CORs with the previous method.

Considerations

COR provides the means to consolidate assignment and administration of the various restriction features available with the system.

All items associated with a COR are distinct and separate. A unique COR must exist for each needed combination of FRLs, CCSA/EPSCS off-network restrictions, calling party restrictions, called party restrictions, and miscellaneous restrictions. Up to 64 CORs can be established, as required, to provide the needed combinations.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- ARS

  Originating FRLs are assigned via a COR.

- Code Restriction

  This feature is assigned to an originating facility via a COR. (Code Restriction is assigned to an outgoing trunk group on the trunk group form.)

- Controlled Restriction

  Restrictions assigned via the Controlled Restriction feature override the calling and called party restrictions via a COR.

- Inward Restriction

  This feature is assigned via a COR.

- Manual Terminating Line Restriction

  This feature is assigned via a COR.

- Origination Restriction

  This feature is assigned via a COR.

- Outward Restriction

  This feature is assigned via a COR.

- Private Network Access and Public Network Access

  Access to the public network via the private network is allowed or denied via a COR (assuming the private network provides the capability to access the public network).

- Termination Restriction

  This feature is assigned via a COR.

- Toll Restriction

  This feature is assigned to an originating facility via a COR. (Toll Restriction is assigned to an outgoing trunk group on the trunk group form.)

CLASS OF SERVICE

Description

A Class of Service (COS) parameter defines whether or not voice terminal users may access four features:

- Automatic Callback
- Call Forwarding All Calls
- Data Privacy
- Priority Calling

There are only two choices for each feature, a voice terminal **can** or **cannot** activate the feature. Four features with two choices each yields 16 possible combinations. A COS parameter is preassigned for each of these 16 combinations. Although the system does allow changing these parameters, there is no need to do so. Any change will result in unneeded duplication. Which COS numbers represent which combination of allowed/denied features are given in the *AT&T System 75 Implementation Manual,* 999-700-277IS. To assign a COS, simply choose the COS number, 0 through 15, that represents the desired allowed/denied combination of features and indicate that number when implementing voice terminals.

COS has no other use in System 75. Restriction groups and call origination/reception privileges are defined and assigned by a Class of Restriction (COR), not a COS.

Considerations

COS is used to assign four features; Automatic Callback, Call Forwarding All Calls, Data Privacy, and Priority Calling. Each voice terminal is assigned one of 16 COSs to determine whether or not it will have any or all of these four features. COS serves no other purpose than to assign these four features.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Automatic Callback

  This feature is assigned via COS.
- Call Forwarding All Calls

  This feature is assigned via COS.
- Data Privacy

  This feature is assigned via COS.
- Priority Calling

  This feature is assigned via COS.

CODE CALLING ACCESS

Description

Allows attendants, voice terminal users, and tie trunk users to page with coded chime signals.

A paging party dials the Code Calling Access code and the extension number assigned to the person to be paged. The system translates the number to a chime code and then plays the code over loudspeakers. The paging party is automatically parked (through the Call Park feature) on the paged party's extension number. To answer the call from any voice terminal within the system, the paged party dials the Call Park Answer Back access code and their own extension number.

Considerations

The system provides access to any type of loudspeaker equipment.

The following items can be administered:

- Up to nine individual areas (zones) plus one zone to activate all zones simultaneously.
- As many as 125 three-digit called party codes can be provided. The codes are combinations of the digits 1 through 5.
- Trunk access code.

Hardware and Software Requirements

Requires loudspeaker paging equipment and one port on a TN763 Auxiliary Trunk circuit pack for each individual zone. (These hardware requirements can be shared with the Loudspeaker Paging Access feature. Activation of each feature is by the assigned trunk access code.)

No additional software is required.

Interactions

- Call Park

  This feature is automatically provided with Code Calling Access.
- Controlled Restriction

  The Total restriction prohibits use of Code Calling Access.
- Miscellaneous Trunk Restriction

  Voice terminals and tie trunks with this restriction cannot use Code Calling Access.
- Origination Restriction

  This restriction prohibits use of Code Calling Access.

CONFERENCE—ATTENDANT

Description

Allows the attendant to set up a conference call for as many as six conferees including the attendant. Conferees from inside and outside the system can be added to a conference call.

Considerations

The attendant can set up only one conference call at a time. The attendant can hold a conference call on the console or release from it. If the attendant releases from a conference call involving only trunk conferees, the trunks are also disconnected.

The attendant cannot handle any other calls while setting up a conference call.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

CONFERENCE—TERMINAL

Description

Allows multi-appearance voice terminal users to set up 6-party conference calls without attendant assistance. Single-line voice terminal users can set up 3-party conference calls without attendant assistance.

Considerations

If a voice terminal user releases from a conference call involving only trunk conferees, the trunks are also disconnected if trunk-to-trunk connections are disallowed through administration.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

DIAL ACCESS TO ATTENDANT

Description

Allows voice terminal users to access an attendant by dialing 0. Attendants can then extend the call to a trunk or to another voice terminal.

Considerations

A voice terminal user calling the attendant by dial access cannot be added to an existing conference by the attendant.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

Origination Restriction (administered to a voice terminal by the Class of Restriction) prohibits placing any calls, including Dial Access to Attendant calls.

DIAL PLAN

Description

The dial plan is the system's guide to digit translation. When a digit is dialed, the system must know what to expect, based on that digit. For example, if a voice terminal user dials a 4, the system must know how many more digits to expect before the code will be processed.

The dial plan, or first-digit table, established during administration for each system provides this information. The table defines the intended use of a code beginning with a specific first digit and relates to the system how many digits to collect before processing the code. The choices of a first digit are 0 through 9, *, and #. Permissible code uses and the allowable number of digits are as follows:

- Extension Numbers

  Flexible numbering allows 2-, 3-, or 4-digit extension numbers. The first digit in the extension number tells the system how many digits to expect the extension number to have. Therefore, all extension numbers beginning with the same digit must be the same length.

  Extension numbers can have a first digit of 1 through 9. For example, if a 3-digit extension number is administered and the first digit is a 4, the extension numbers might be 432, 434, 438, and so on. Also, if a 4-digit number with a 6 as the first digit is administered, the extension numbers might be 6321, 6422, 6428, and so on.

- Attendant

  Dial access to the attendant is always by the single digit "0."

- Trunk Access Codes

  A minimum of one digit and a maximum of three digits can be used. Trunk access codes can have a first digit of 1 through 9. For example, 9 could be used for local trunks, 8 for WATS trunks, and 7 for tie trunks.

- Feature Access Codes

  A minimum of one digit and a maximum of three digits can be used. The * and # buttons can be used as part of a feature access code and, when used, must be the first digit. The * or # counts as one digit. For example, *2 could be used to activate Call Forwarding All Calls and #2 used to deactivate Call Forwarding All Calls.

  Feature access codes can also have a first digit of 1 through 9. For example, 32 could be used to activate Call Forwarding All Calls and 33 used to deactivate Call Forwarding All Calls.

The first digit administered for one type of entry in the first-digit table cannot also be administered as the first digit of another entry. For example, when a 9 is used as a trunk access code, 9XX cannot be used as an extension number or as a feature access code.

Considerations

The entire Dial Plan is dependent on the first digit dialed. The 12 possible choices of a first digit are 0 through 9, * and #.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

All dial access features and services provided by the system require the Dial Plan.

DIRECT DEPARTMENT CALLING

Description

Allows direct inward access to an answering group other than the attendant even if the system does not have the Direct Inward Dialing (DID) feature.

One extension number is assigned to all voice terminals in a group or department, that is, to a set of voice terminals that serve the same function and require call distribution among the members of the group. Incoming calls to a Direct Department Calling (DDC) group can be internal or external. An incoming call alerts the first available voice terminal in the administered sequence. If the first terminal in the sequence is active (busy), the call routes to the next terminal, and so on. In other words, incoming calls always try to complete at the first terminal in the administered sequence. Therefore, the calls are not evenly distributed among the DDC group members.

When DDC, or the similar Uniform Call Distribution (UCD) feature, is not provided, incoming Listed Directory Number (LDN) calls, foreign exchange calls, 800 service calls, and automatic tie trunk calls are normally directed to an attendant, who must extend the call. When DDC is provided on a trunk group, incoming calls are automatically directed to the desired DDC group by the System 75 switch. Attendant intervention is not required.

A queue can be established for a DDC group. When all voice terminals within the group are active, the queue allows incoming calls to await an idle terminal.

When a call enters the queue, a delay announcement interval is started. This interval (1 to 999 seconds) indicates how long a call will remain in queue before the call is connected to a recorded announcement. If Call Coverage is provided, the Don't Answer Interval (2 to 9 alerting cycles) also begins when the call enters the DDC group queue. After these intervals have begun, one of the following occurs:

- If the Coverage Don't Answer Interval expires before the delay announcement interval expires, the call is redirected to coverage. If no coverage point is available to handle the call, the call remains in queue and may then be connected to delay announcement.
- If the delay announcement interval expires before the Coverage Don't Answer Interval, the call is connected to a delay recorded announcement, if available. If the announcement is already in use, the delay announcement interval is reset. This process (as described above) continues until the call is answered, goes to coverage, is connected to a delay announcement, or the calling party hangs up.

Calls connected to a delay recorded announcement remain in queue while the announcement is heard by the caller. If the call has not been answered by the time the announcement is over, the call is connected to music (if provided) or there will be silence, as long as the call remains in queue.

The queue length can be set from 0 (no queue) to 35 calls. If queuing is not provided, if the queue is full, or if all group members have activated the Make Busy option (discussed later), calls to a busy group receive busy tone or redirect via the Call Coverage feature. Lamp indicators may be used to give a warning when the number of calls waiting in the queue reaches a predetermined limit (queue warning limit). The queue warning level can be 0 to 35; however, it cannot exceed the queue length. Although it is possible, the queue warning level should not be set to 0, as this would result in the indicator lamp lighting at all times.

When the queue warning level is reached, the indicator lamp lights and remains lighted until the calls waiting in queue are fewer than the queue warning level. A queue warning level lamp may be provided for each DDC group queue. The lamp can be installed at any location convenient for the group.

As an example of queue warning level and delay announcement operation, assume that there is an incoming call to a DDC group with the following parameters.

- Queue length is 10 calls.
- Queue warning level is 5 calls.
- Recorded announcement delay is 20 seconds.

Also assume the following:

- All DDC group voice terminals are busy.
- The call is the fifth call in the queue.

Since all voice terminals in the DDC group are busy, the incoming call enters the queue. The incoming call, being the fifth call in the queue, causes the queue warning level to be reached. This causes the queue warning level lamp to light.

From the indicator lamp, the DDC group members know the queue warning level has been reached and try to complete their present calls. Meanwhile, the incoming call has been in the queue for 20 seconds and hears the delay recorded announcement. The caller may decide to hang up or may decide to remain in the queue. Assume the caller remains in the queue. When a DDC group member becomes idle, the longest queued call is directed to that voice terminal. The queue warning level lamp may or may not be lighted at that time depending on the number of other calls that have been queued. Also, the first four calls in the queue will have heard the delay announcement after being queued for 20 seconds. The queue warning level and delay announcement capabilities are independent of each other.

Considerations

DDC is particularly useful when the answering group assigned receives a high volume of incoming calls. Call completion time is minimized and attendant assistance is not required. This feature can also minimize the use of DID trunks.

Calls incoming on a non-DID trunk group can route to a DDC group instead of to an attendant. Calls incoming on any non-DID trunk group can have only one primary destination; therefore, the trunk group must be dedicated to the DDC group.

If DID is provided and the DDC group extension number is within the range of extension numbers that can be dialed directly, then the group can be called the same as any voice terminal.

Any voice terminal can be administered to one or more DDC groups. Each voice terminal in a group also has its own unique extension number and can be alerted individually. Multi-appearance voice terminals can have an assigned status lamp that identifies an incoming DDC call. However, the voice terminal must be idle (not active on any call appearance) before a group call will be directed to the terminal.

If DDC and UCD groups are both used in the system, the number of combined groups and the number of voice terminals per group is determined by the size of the system and call traffic requirements. A maximum of 32 groups with up to 32 members per group can be provided. The system maximum, however, is 448 group members.

The DDC group queue is optional on an individual group basis, and the maximum queue size is 35. If a queue is not assigned or if an assigned queue is full, incoming calls can be redirected through the Call Coverage feature.

An optional queue warning level lamp can be provided for each DDC group. The warning level can be administered for 0 to 35 calls waiting in queue. The lamp lights when the warning level is reached. Each queue warning level lamp requires a port on an Analog Line circuit pack. The lamp can be placed at a location convenient to the group.

The system provides access to music sources and delay announcement equipment, if used.

Each System 75 can contain up to ten different recorded announcements. Each group queue can be assigned one of these announcements as a delay announcement. A delay announcement can be shared among the DDC groups, UCD groups, or a combination of these groups. However, only one caller can be connected to an announcement at any one time. Callers are always connected at the beginning of the announcement. More efficient use of the announcements is realized if the announcements are brief.

If a delay announcement is used, answer supervision is sent to the distant office when the caller is connected to the announcement. Charging for the call, if applicable, begins when answer supervision is returned.

Multi-appearance voice terminals can receive only one DDC call at a time. A voice terminal is idle for a DDC call only if all call appearances are idle.

A Make Busy option can be administered for the system. When a voice terminal user dials the Make Busy activation code followed by the DDC group extension number, or presses the Make Busy button, the terminal appears busy to the DDC group. This effectively removes the terminal from the group until the user dials the Make Busy cancellation code or presses the button again. The Make Busy button can be assigned to multi-appearance voice terminals only.

The last available member of a DDC group cannot activate the Make Busy option if any calls are remaining in the queue. An attempt by the last available group member to activate the Make Busy option results in the following:

- New calls to the DDC group either receive busy tone or redirect to coverage.
- Calls already in the queue continue to route to the last available voice terminal until the queue is empty.
- At the last available voice terminal, the status lamp associated with the Make Busy button, if provided, flashes until the queue is empty. When no more calls remain in the queue, Make Busy is activated and the status lamp, if provided, lights steadily. (The same sequence applies when Make Busy is dial activated instead of button activated, except there is no status lamp.)

Leave Word Calling messages can be stored for a DDC group and can be retrieved by a member of the DDC group, a covering user of the group, or a systemwide message retriever. The Voice Terminal Display feature and proper authorization must be assigned to the message retriever. Also, a Remote Message Waiting Indicator can be assigned to a group member to provide a visual indication that a message has been stored for the group. One Remote Message Waiting Indicator is allowed per DDC group. The status lamp associated with this button informs the user that at least one message has been left for the group.

Hardware and Software Requirements

Each queue warning level lamp requires one port on a TN742 Analog Line circuit pack. A 21C-49 indicator lamp may be used as a queue warning level lamp. This lamp is approximately 2 inches in diameter and has a clear beehive lens. The lamp operates on ringing voltage and can be mounted at a location convenient to the group.

Each delay announcement requires announcement equipment and one port on a TN742 Analog Line circuit pack. If music is to be heard after the delay announcement, a music source and a port on a TN763 Auxiliary Trunk circuit pack is required. Announcement equipment and music sources are not provided by the system.

No additional software is required.

Interactions

- Attendant Call Waiting

  An attendant can originate or extend a call to a DDC group. Attendant Call Waiting cannot be used on such calls. However, such calls can enter the group queue, if provided.

- Call Coverage

  Calls can redirect to or from a DDC group.

  When the Send All Calls button is pressed, the associated voice terminal appears busy to the DDC group until the button is pressed again. Activating Send All Calls also lights the lamp associated with the Make Busy button, if provided. Likewise, deactivating the Send All Calls function deactivates the Make Busy function. However, activating or deactivating the Make Busy function does not activate or deactivate the Send All Calls function.

  For a call to a DDC group to be directed to Call Coverage, each voice terminal in the group must be active on at least one call appearance and the queue, if there is one, must be full. If the queue is not full, a call will enter the queue when no voice terminal is available. Queued calls remain in queue for a time interval equal to the Coverage Don't Answer Interval before redirecting to coverage. If any voice terminal in the group is idle, the call directs to that voice terminal.

- Call Forwarding All Calls

  When activated, the activating voice terminal appears busy to the DDC group.

- Multi-Appearance Preselection and Preference

  All assigned call appearances must be idle before a DDC group call is directed to a voice terminal.

- Music-on-Hold Access

  A call placed in a DDC group queue can receive a delay announcement followed by music.

- Priority Calling

  A priority call directed to a DDC group is treated the same as a non-priority call, except that the distinctive 3-burst alerting is heard.

- Terminating Extension Group

  A Terminating Extension Group cannot be a member of a DDC group.

- Voice Terminal Display

  On calls dialed directly to a DDC group extension number, the DDC group's identity is displayed at the calling extension.

DIRECT INWARD DIALING

Description

Connects calls from the public network directly to the dialed extension number without attendant assistance.

Reduces the attendant's workload and provides the calling party immediate contact with the called party.

Considerations

Requires Direct Inward Dialing (DID) trunk group(s) from the local telephone company central office.

Hardware and Software Requirements

Requires one port on a TN753 DID Trunk circuit pack for each DID trunk. No additional software is required.

Interactions

The Inward Restriction, Manual Terminating Line Restriction, and Termination Restriction features (administered by the Class of Restriction) prevent receiving DID calls at the restricted voice terminal.

DIRECT OUTWARD DIALING

Description

Allows voice terminal users to access the public network without attendant assistance.

Considerations

Requires trunks to the local telephone company central office (CO), a Wide Area Telecommunications Service (WATS) serving office, or a foreign exchange (FX).

Only one CO trunk group is accessible by a single dial access code. An all-busy trunk group cannot redirect calls to another trunk group. Therefore, if more than one trunk group is provided, a separate access code must be established for each.

Hardware and Software Requirements

Requires one port on a TN747 CO Trunk circuit pack for each assigned trunk. No additional software is required.

Interactions

Calling party restrictions (assigned by the Class of Restriction) prevent placing Direct Outward Dialing calls from the restricted voice terminal.

FACILITY BUSY INDICATION

Description

Provides multi-appearance voice terminal users with a visual indication of the busy or idle status of an extension number, a trunk group, or any loudspeaker paging zone, including all zones. The Facility Busy Indication button provides the voice terminal user direct access to the extension number, trunk group, or paging zone.

When the lamp associated with the Facility Busy Indication button is lighted, the tracked resource is busy. If the lamp is dark, the resource is idle. If the lamp is flashing, the tracked resource is placing a call to the voice terminal with the button.

Pressing the Facility Busy Indication button when the lamp is dark automatically selects an idle call appearance and places a call to the resource.

Considerations

Extension numbers, trunk group access codes, and Loudspeaker Paging Access codes can be stored in a Facility Busy Indication button. An access code followed by other numbers is not allowed.

A maximum of 1000 Facility Busy Indication buttons are allowed in the system, and as many as 100 of these buttons can be administered to track the same resource. A new state of the tracked resource (that is, a change from idle to busy) is updated within 5 seconds after the system detects the change.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

FACILITY TEST CALLS

Description

Provides a voice terminal user with the capability of making test calls to access specific trunks, touch-tone receivers, time slots, and system tones. A local voice terminal user can make a test call by dialing an access code. An Initialization and Administration System (INADS) terminal user can make a test call over a trunk.

Four types of Facility Test Calls can be made:

- Trunk test call

  Accesses specific Tie or CO trunks. Direct Inward Dialing (DID) trunks cannot be accessed.

- Touch-tone receiver test call

  Accesses and tests the four touch-tone receivers located on a Tone Detector circuit pack.

- Time slot test call

  Connects the voice terminal user to a specific time slot located on the Time Division Multiplex (TDM) buses or out-of-service time slots.

- System tone test call

  Connects the voice terminal user to a specific system tone.

Considerations

Since test calls provide access to trunks, security should be provided to insure that only authorized users make test calls. Security for test calls is provided by making the feature access code and test procedures known only to authorized users.

A touch-tone voice terminal must be used to make test calls.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

HOLD

Description

Allows multi-appearance voice terminal users to disconnect from a call temporarily, use the voice terminal for other call purposes, and then return to the original call.

Considerations

Multi-appearance voice terminal users can hold a call on each call appearance.

One party on hold can hear music if the Music-on-Hold feature is provided. The music is removed when the voice terminal user reenters the call.

A single-line voice terminal user can place a call on hold by pressing the Recall button or by flashing the switchhook. The user can then place another call or activate a feature, and return to the original call by pressing Recall or flashing the switchhook again.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

A held multi-appearance voice terminal user can activate Leave Word Calling toward the holding user.

HOT LINE SERVICE

Description

Allows single-line voice terminal users, by simply lifting the receiver, to automatically place a call to a preassigned extension number, public or private network telephone number, or feature access code.

Considerations

The Hot Line Service destination number is stored in an Abbreviated Dialing Group Number List. When the Hot Line Service user lifts the receiver, the system automatically routes the call to the stored number and the call is completed as though manually dialed. If the appropriate feature access code is prefixed to the stored number, Automatic Route Selection (ARS), Data Privacy, or Priority Calling can be used on the call. Also, if the Public or Private Network Access code is the stored number, the voice terminal user will be connected to an outgoing trunk and can dial the outside number.

A Hot Line Service voice terminal receives calls allowed by its Class of Restriction. Call reception is not affected by Hot Line Service. Likewise, the Hot Line Service destination is not affected by Hot Line Service.

A Direct Department Calling (DDC), a Uniform Call Distribution (UCD), or a Terminating Extension Group (TEG) extension number or any individual extension number within any of the groups can be a Hot Line Service destination. Also, any extension number within a DDC group, UDC group, or TEG can have the Hot Line Service feature assigned.

If a Hot Line Service call accesses a trunk group with Ringback Queuing assigned, the call can queue unless the voice terminal is termination restricted by its Class of Restriction. Queuing, when applicable, is automatic on single-line voice terminals; dialing is not required.

Loudspeaker Paging Access can be used with Hot Line Service to provide automatic access to paging equipment.

The number of voice terminals that can be assigned Hot Line Service is not limited, and the number of voice terminals that can be assigned the same destination is not limited. The limit, if any, would be on the number of entries that can be stored in the Abbreviated Dialing lists.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

A Hot Line Service user cannot activate any feature unless the access code is, or is part of, the destination number.

HUNTING

Description

Checks for the active or idle status of extension numbers in one or more ordered groups. If all terminals in a group are active, the call can route to another group via Call Coverage.

Considerations

Hunting is administered through the Call Coverage, Direct Department Calling, and Uniform Call Distribution features. The order of hunting is defined under each individual feature.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

INTEGRATED DIRECTORY

Description

Allows internal system users to access the system data base, use the touch-tone buttons to key in a name, and retrieve an extension number from the system directory. The directory contains an alphanumeric listing of the names and extension numbers assigned to all voice terminals administered in the system.

The Integrated Directory feature can be accessed by users who have the Voice Terminal Display or Attendant Display feature and an assigned Integrated Directory button.

The names in the directory will be those administered by the System Manager on the individual voice terminal forms. Names cannot exceed 15 characters (including spaces and commas) and can be entered in one of the following three formats.

- Last name, comma, first name, space, then middle name or initial, if desired. For example, the following entries are acceptable:

  Jones,Betty Ann
  Smith,A E I
  Thomas,John J
  Abbott,Lynn

- First name, space, second name or initial, and then last name. For example, the following entries are acceptable:

  Betty Ann Jones
  A E I Smith
  John J Thomas
  Lynn Abbott

- A single entry is also acceptable:

  Cafeteria
  1J409
  2F816
  Purchasing

The use of a hyphen, an apostrophe, or a period is not acceptable. For example, the name Tom O'Connell should be entered as Tom OConnell, and the name Yih-Nan Yang should be entered as Yih Nan Yang.

The following is an example of a typical Integrated Directory data base:

1J409
Abbott,Lynn A
Brown,Kent J
Cafeteria
Carr,Danny
Carter,Ann
2F816

Purchasing
Barbara Quincey
Roberson,Don T
William Ruoff
Smith,A E I
Streck,R T

The touch-tone buttons are used to key in the numbers and letters labeled on them. The following exceptions apply:

- Button 7 (PRS) is also used for a Q.
- Button 9 (WXY) is also used for a Z.
- Button * is used for a space or comma.
- Button # is not used.

To activate the Integrated Directory feature, the user presses the Integrated Directory button. This puts the voice terminal in the Integrated Directory mode and turns off the tones normally generated when a touch-tone button is pressed. The touch-tone buttons are now used exclusively for keying in names and not for dialing.

After the Integrated Directory button is pressed, the alphanumeric display will show DIRECTORY — PLEASE ENTER NAME. Names are always keyed in the following order: Last name, comma, and then first name or initial. When searching for a single entry, the letters or numbers would be keyed in order. Several letters might be needed to get the correct entry.

When a button is pressed, the display will show the first name that matches the first letter on the button. For example, if a user is searching for the name Ann Carter and presses button 2 to key in the letter C, the display might show Abbott,Lynn A and an extension number. (Button 2 matches A before it matches C.) If the user presses button 2 again to key in the letter A, the display will stay the same. (Again, AB is matched before CA.) If the user now presses button 7 to key in an R, the display might show Carr,Danny and an extension number.

At this point, the user can press button 8 to key in the letter T or can press the Next Message button on the alphanumeric display. Pressing Next Message displays the next name in the directory and, in this case, might be Ann Carter.

When the desired name and extension number are displayed, the user can automatically place a call to that person by pressing the Return Call button.

If a name is entered but not found in the directory, the display will show NO MATCH — TRY AGAIN. To search for another name, the user presses the Integrated Directory button again, and the feature is reactivated.

To exit the Integrated Directory mode, the user presses one of the other mode buttons assigned to the alphanumeric display module; for example, the Normal mode button.

Considerations

The entire directory cannot be searched by pressing button 2. Pressing button 2 and then continually pressing Next Message will display, one by one, all entries beginning with A, B, C, and 2. If all entries have been displayed and Next Message is pressed again, the display will repeat from the first entry in the listing associated with button 2.

When the voice terminal is in the Integrated Directory mode, it can still be used to activate other features or to place calls if dialing is not required. Also, a user can enter the Integrated Directory mode while active on a call, and calls can be received when the Integrated Directory mode is active.

The maximum size of the directory is 400 entries. The maximum length of the name is 15 characters (including spaces and commas). The extension number cannot exceed four digits.

The Integrated Directory mode button is administered through the Attendant Display and Voice Terminal Display features.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Touch-Tone Dialing

  Call origination and feature access by dial code is not allowed when the Integrated Directory feature is active.

INTERCEPT TREATMENT

Description

Provides an intercept tone or a recorded announcement or routes the call to an attendant for assistance when calls cannot be completed or when use of a feature is denied.

- Intercept Treatment—Tone

  Provides a siren-type tone to internal calls that cannot be completed as dialed.

  Intercept Tone is provided to voice terminals when users lift the receiver and do not dial within 10 seconds, pause longer than 10 seconds between digits during the dialing process, or remain connected to Loudspeaker Paging for longer than an administered interval.

  When a single-line voice terminal user receives Intercept Tone for 30 seconds and does not hang up or does not hang up within 10 seconds after other parties have disconnected, the user receives dial tone for a new call origination.

  When multi-appearance voice terminal users receive Intercept Tone for 30 seconds and do not hang up, the call appearance returns to idle. If the multi-appearance user is the last party left on a call, the call appearance immediately returns to idle.

- Intercept Treatment—Recorded Announcement

  Provides a recorded announcement to Direct Inward Dialing and incoming Private Network Access calls that cannot be completed as dialed. The System Manager selects and records the message.

  Toll charges do not apply to Direct Inward Dialing and Private Network Access calls routed to Recorded Announcement.

- Intercept Treatment—Attendant

  Allows attendants to provide information and assistance to callers on all Direct Inward Dialing or incoming Private Network Access calls that cannot be completed as dialed.

  Normal toll charges apply to these calls.

Considerations

Ten recorded announcements can be used with the system. None, some, or all of these announcements can be used for Intercept Treatment. The announcement equipment and the appropriate messages must be furnished by the client.

Only one person can be connected to an announcement at any given time. The caller is always connected to the beginning of the announcement.

Hardware and Software Requirements

Requires announcement equipment and one port on a TN742 Analog Line circuit pack for each announcement. No additional software is required.

Interactions

Attendant Intercept and Recorded Announcement Intercept (both optional) cannot be used together. Direct Inward Dialing calls cannot be assigned Intercept Treatment—Tone.

INTERCOM—AUTOMATIC

Description

Provides a talking path between two voice terminal users. Calling users press the Automatic Intercom button and lift the receiver, or vice versa. The called user receives a unique intercom alerting signal, and the status lamp associated with the Intercom button, if provided, flashes.

Considerations

Single-line voice terminal users can receive Automatic Intercom calls, but cannot originate them.

A combination of Automatic and Dial Intercom can be used between terminals so that Automatic Intercom applies in one direction and Dial Intercom applies in the other.

The following items can be administered:

- Automatic Intercom button (per multi-appearance voice terminal)
- Intercom group number
- Length of dial code
- Extension numbers within the group
- Dial code to access Intercom group member

The following must be administered for the Automatic Intercom button:

- Intercom group number to be accessed
- Dial code assigned to group member to be accessed

Two terminals with Automatic Intercom to and from each other or terminals with combined Automatic and Dial Intercom to each other must be in the same Intercom group.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Call Coverage

  Intercom calls are redirected only if the caller activates Go to Cover.

- Dial Intercom

  This feature must be provided. Users assigned an Automatic Intercom button must be a member of the same Dial Intercom group as the destination extension number.

INTERCOM—DIAL

Description

Allows multi-appearance voice terminal users to gain rapid access to as many as 32 other voice terminal users within an administered group. Calling voice terminal users lift the receiver, press the Dial Intercom button, and dial the 1- or 2-digit code assigned to the desired party. The called user receives alerting tone, and the status lamp associated with the Intercom button, if provided, flashes.

Considerations

Single-line voice terminals can receive Dial Intercom calls, but cannot originate them.

A combination of Dial and Automatic Intercom can be used between terminals so that Dial Intercom applies in one direction and Automatic Intercom applies in the other.

A Dial Intercom user can place an intercom call to all members in the group, including Automatic Intercom members.

The following can be established, defined, and administered:

- Dial Intercom button (per multi-appearance voice terminal)
- Intercom group numbers (up to 32 groups including up to 32 members each, with a system maximum of 128 members)
- Dial code for each group member (1-digit code if group is less than 10, 2-digit number if group is 10 or more)

Two terminals with Dial Intercom to and from each other or two terminals with combined Dial and Automatic Intercom to and from each other must be in the same Intercom group.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Call Coverage

  Intercom calls are redirected to Call Coverage only if the caller activates Go to Cover.

- Automatic Intercom

  Users assigned this feature must be a member of a Dial Intercom group.

LAST NUMBER DIALED

Description

Automatically redials the last number dialed when users press the Last Number Dialed button or dial the Last Number Dialed feature access code.

The system saves the first 16 digits of the last number dialed whether the call attempt was manually dialed or an Abbreviated Dialing button was pressed.

Considerations

Special characters (Pause, Wait, Mark, or Suppress) stored in an Abbreviated Dialing button are recognized by the system and will be outpulsed when such a number is automatically redialed by the Last Number Dialed feature.

When a manually dialed number is redialed automatically, a delay in dialing is not recorded. The system will outpulse the numbers as one continuous digit string. Thus, to accomplish automatic redialing, the distant end must accept the outpulsed digits without delay.

Last Number Dialed information is not saved on tape and can be used only for the next call origination. End-to-end signaling digits manually dialed are never saved.

Automatic Callback can be used after the Last Number Dialed feature is used on a call to an internal voice terminal.

The following items can be administered:

- Last Number Dialed button (multi-appearance voice terminals only)
- Last Number Dialed feature access code

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

LEAVE WORD CALLING

Description

Allows internal system users to leave short messages for other internal users. Users can activate Leave Word Calling (LWC) at any time during a call attempt.

The LWC feature electronically stores a standard message; for example, CARTER, ANN 2/7 10:45a 2 CALL 3124. This message means that Ann Carter called two times, the last time on the morning of February 7, and wants a return call to extension 3124.

When a message is stored, the Message lamp on the called voice terminal automatically lights. This lamp is referred to as an Automatic Message Waiting lamp since the status of the lamp is controlled automatically by the system.

Another voice terminal may also receive an indication that an LWC message has been left for the called party. This is accomplished via a Remote Message Waiting Indicator at another voice terminal. The Remote Message Waiting Indicator is a status lamp associated with a button assigned for this purpose. The Remote Message Waiting Indicator lights at the same time that the Message lamp lights at the called voice terminal. A common use of a Remote Message Waiting Indicator is to provide an indication of an executive's message on a secretary's voice terminal. If the executive calls from outside to receive any messages, the secretary knows at a glance if any messages have been left. Remote Message Waiting indicators also allow an indication of LWC messages left for a Direct Department Calling (DDC) group, Uniform Call Distribution (UCD) group, Terminating Extension Group (TEG), and Personal Central Office Line (PCOL) group

When identical messages are entered in the system, the date, time, and number of messages are updated. When nine or more identical messages accumulate, the count remains at nine but the date and time are updated.

Messages can be stored by calling, called, and covering users. A covering user can be through the Call Coverage, Call Pickup, or Call Forwarding All Calls features. Messages are stored as follows:

- Storage by Calling User
  - Before dialing the desired extension number, the user presses the LWC button or dials the LWC access code and then dials the desired number.
  - After dialing the desired number but before the call is answered, a multi-appearance voice terminal user presses the LWC button or a single-line voice terminal user presses the Recall button and dials the access code.
  - After the call has been answered by any user, the calling user presses the LWC button or the Recall button and dials the access code.
- Storage by Called User
  - After answering the call, the called user presses the LWC button. This leaves a message for the calling user to call back. (This cannot be done by dial access code.)
- Storage by Covering User
  - After answering the call, the covering user presses the Coverage Callback button. This stores a message for the called user to call the calling user.
  - After answering the call, the covering user presses the LWC button. This leaves a "call me" message for the originally called user.

In addition, a user placed on hold can activate LWC and leave a message for the holding user to place a return call.

Messages are retrieved by users who have the Voice Terminal Display or Attendant Display feature. Users without the Voice Terminal Display feature have their messages retrieved by systemwide message retrievers or by covering users in their Call Coverage path.

A calling user who left an LWC message can cancel that message if it has not already been accessed. The calling user lifts the receiver, presses the LWC Cancel button or dials the access code, and dials the extension number of the called party. This deletes the message (even if the count was more than one) and causes all Message lamps associated with the called voice terminal to go dark (if the called user has no other messages).

Messages are protected by restricting unauthorized users from displaying, canceling, or deleting messages. A Lock function restricts a voice terminal, and an Unlock function releases the restriction. The Lock function is activated by dialing a systemwide access code. The Lock function is canceled by dialing a systemwide access code and then an Unlock security code unique to the voice terminal. These functions apply only to the voice terminal where the operation is performed. A status lamp can be assigned to show the locked or unlocked status of the voice terminal.

Considerations

If an Applications Processor (AP) is provided with the system, LWC messages can be retrieved by a Message Center agent or by authorized users through the AP Demand Print feature.

If the following conditions are met, messages for users can be retrieved by selected voice terminal users or any attendant:

- The retriever must be in the called user's Call Coverage path or must be administered as a systemwide retriever.
- Permission to retrieve messages must be administered to the called voice terminal.

Ten terminals, or nine terminals and the attendant console group, can be administered as systemwide message retrievers.

A system maximum of 1000 messages can be stored, and a systemwide maximum number of messages not to exceed 127 per user can be administered.

If the system does not have an AP and if the stored message level reaches 95 percent of capacity, the status lamp associated with all Coverage Message Retrieval buttons in the system will flash. These lamps will continue to flash until the stored message level drops below 85 percent of capacity. Authorized retrievers can selectively delete messages to gain storage space. Old messages are not automatically purged by the system.

All buttons associated with the display modes are administered through the Attendant Display and Voice Terminal Display features.

The following items are administrable for the LWC feature:

- AP Demand Print button (per voice terminal)
- Identities of authorized systemwide LWC retrievers
- Locking and unlocking message retrieval and cancellation (per voice terminal)
- Lock dial access code (systemwide)
- Lock status lamp (per voice terminal)

- LWC activation (per voice terminal and the attendant group)
- LWC activation dial access code (systemwide)
- LWC button (per voice terminal)
- LWC Cancel button
- LWC cancellation dial access code (systemwide)
- LWC reception (per voice terminal and per Hunt group; that is, DDC group, UCD group, TEG, and PCOL group)
- Maximum number of messages not to exceed 127 per user (systemwide)
- Remote Message Waiting Indicator on another voice terminal (one allowed per extension number, including an extension number for a DDC group, UCD group, TEG, and PCOL group; 50 allowed per system)
- Retrieval permission for covering users (per voice terminal)
- Unlock dial access code (systemwide)
- Unlock security code (per voice terminal)

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Call Coverage

  The LWC feature can be used with or without Call Coverage. However, the two features complement each other. The Coverage Callback option of the Call Coverage feature is provided by the LWC feature. Also, a caller can activate LWC for the called party even if the call has been answered by a covering user.

- Conference

  A member of a conference call cannot activate LWC because that user cannot be uniquely identified.

LINE LOCKOUT

Description

Removes single-line voice terminal extension numbers from service when users fail to hang up after receiving intercept tone for 30 seconds and then dial tone for 10 seconds.

Line Lockout occurs as follows:

- A user does not hang up after the other party on a call is disconnected.
- A user pauses for 10 seconds between digits while dialing.

In either of the above two cases, users will receive intercept tone for 30 seconds and then will receive dial tone for 10 seconds. After this time if the receiver is still lifted, the voice terminal is taken out of service.

The out-of-service condition does not tie up switching facilities or call processing time, and it remains in effect until the voice terminal user hangs up.

Considerations

None.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

Call intercept is provided by the Intercept Treatment—Tone feature.

LOUDSPEAKER PAGING ACCESS

Description

Provides attendants and voice terminal users dial access to voice paging equipment.

As many as nine individual paging zones can be provided. (A zone is the location of the loudspeakers; for example, conference rooms, warehouses, or storerooms.) In addition, one zone can be provided to activate all zones simultaneously.

Considerations

The system provides access to any type of loudspeaker paging equipment.

One Direct Trunk Group Select button on the attendant console can be assigned to access each individual zone or all zones for Loudspeaker Paging. For example, a Direct Trunk Group Select button could be assigned for all-zones paging and the other zones could be paged by dialing the trunk access code.

Loudspeaker Paging Access codes can be stored in Abbreviated Dialing lists. Abbreviated Dialing buttons can be administered to multi-appearance voice terminals.

A Listed Directory Number or Direct Inward Dialing call cannot be connected to the paging facility. However, the attendant can make the page and park the incoming call using the Call Park feature.

The following items can be administered:

- Up to 10 (1 per zone) Loudspeaker Paging Access buttons (per multi-appearance voice terminal and attendant console). Buttons are assigned through the Attendant Direct Trunk Group Selection, Abbreviated Dialing, and Facility Busy Indication feature.
- Trunk access codes and Class of Restriction (per zone provided).
- Paging expiration interval (from 10 seconds to 5 minutes).
- Station Message Detail Recording activation.

Hardware and Software Requirements

Requires loudspeaker paging equipment and one port on a TN763 Auxiliary Trunk circuit pack for each individual zone. Paging interface equipment, consisting of a 278A adapter and a 36A voice coupler, is also required for each individual zone. (This hardware can be shared with the Code Calling Access feature. Each feature is activated by the assigned trunk access code.)

No additional software is required.

Interactions

The following features cannot be used with Loudspeaker Paging:

- Attendant Conference
- Terminal Conference
- Data Call Setup
- Hold
- Ringback Queuing
- Transfer

MANUAL MESSAGE WAITING

Description

Enables multi-appearance voice terminal users, by pressing a designated button on their own terminals, to light the status lamp associated with the Manual Message Waiting button at another multi-appearance voice terminal. Activating the feature causes the lamp to light on both the originating and receiving voice terminals. Either terminal user can cause the lamp to go dark by pressing the button.

Considerations

This feature can be administered only to pairs of voice terminals, such as a secretary and an executive. The secretary might press the designated button to signal the executive that a call needs answering. The executive might press the button to indicate "Do Not Disturb" or "Not Available" to the secretary. (The button can be marked to reflect the intended use.)

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

MANUAL ORIGINATING LINE SERVICE

Description

Provides single-line voice terminals that automatically place a call to the attendant when the user lifts the receiver.

Considerations

A zero (0) is stored in an Abbreviated Dialing list. When the Manual Originating Line Service voice terminal user lifts the receiver, the system automatically routes the call to the attendant.

A Manual Originating Line Service user can receive calls allowed by the assigned Class of Restriction. Call reception is not affected by Manual Originating Line Service.

When a Night Service feature is activated, the Manual Originating Line Service call redirects.

The number of voice terminals that can be assigned Manual Originating Line Service is not limited.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

A Manual Originating Line Service call is a Hot Line Service call to the attendant.

A Manual Originating Line Service voice terminal user cannot activate features that require dialing.

MANUAL SIGNALING

Description

Allows a voice terminal user to signal another voice terminal user. The receiving voice terminal user hears a 2-second burst of tone.

Considerations

When a voice terminal user presses the Manual Signaling button, the associated status lamp lights for 2 seconds.

The signal is sent each time the button is pressed. If the receiving voice terminal is already being alerted of an incoming call, Manual Signaling is denied. The status lamp associated with the Manual Signaling button at the originating voice terminal will flutter briefly to indicate the denial.

A Manual Signaling button must be assigned to the originating voice terminal.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

MULTI-APPEARANCE PRESELECTION AND PREFERENCE

Description

Provides multi-appearance voice terminal users with options for placing or answering calls on selected appearances.

- Alerting Appearance Preference

  When a user lifts the receiver to answer an incoming call, the system automatically connects the user to the alerting call appearance. If more than one call is incoming, the user is automatically connected to the eldest (first-in) alerting call appearance. The in-use (red) lamp tracks the alerting appearance and the answered appearance.

- Idle Appearance Preference

  When a user lifts the receiver to place a call, the system automatically connects the user to an idle appearance even if an incoming call is alerting at another appearance. The in-use (red) lamp tracks an idle appearance when the receiver is lifted.

- Preselection

  Before lifting the receiver to place or answer a call, the user can manually select an appearance (press a call appearance button or a feature button) where the in-use lamp is dark. Preselection is used, for example, when the user wants to reenter a held call or activate a feature. Preselection also activates the speakerphone if the voice terminal is so equipped.

Considerations

The Idle Appearance Preference option is administered on a per-terminal basis. If Idle Appearance Preference is not administered, the voice terminal will have Alerting Appearance Preference. Both Preference options cannot be used on the same voice terminal, and no preference is not an option. Administratively, Idle Appearance Preference (yes or no) is the only choice. No, which is the system default, selects Alerting Appearance Preference.

Multi-appearance voice terminals can have from 2 to 10 call appearances. One of these call appearances is reserved for placing calls or for receiving a Priority Calling call. If a voice terminal has two call appearances and one of them is active, a non-Priority call cannot access the other call appearance, even if the call appearance is idle. Also, the reserved call appearance is not a fixed-position button. It is simply the last idle call appearance. For example, assume a voice terminal has 10 call appearances. Any nine can be in use, but the tenth (last) one is reserved. This aspect of system operation should be considered when determining the number of call appearances for a voice terminal. The default value and recommended number of call appearances is 3.

All incoming and outgoing calls require a call appearance. There are no hidden or free call appearances. For example, consider a member of a Call Pickup group with a Call Pickup button. When a call alerts some other group member, it can normally be answered by pressing the Call Pickup button. However, pressing the button selects a call appearance for the call, if available. If a call appearance is not available, the call cannot be picked up. Similarly, calls originated using the Facility Busy Indication feature calls also require a call appearance. In this case, the call cannot be completed unless an idle call appearance is available. A Facility Busy Indication button on a called voice terminal provides a visual indication of the busy or idle status of another facility. It does not provide a talking path. These facts should be considered when determining the number of call appearances for a voice terminal.

The Preselection option overrides both Preference options. If the user does not lift the receiver within 5 seconds after using Preselection, the selected appearance returns to idle.

Preselection can be used with a feature button. For example, if an Abbreviated Dialing button is pressed, a call appearance is automatically selected and, if the user lifts the receiver within 5 seconds, the call is automatically placed.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- If Cover All Calls is the redirection criteria to be used for a voice terminal, Idle Appearance Preference should also be assigned to the voice terminal. This allows the principal (called party) to lift the receiver without being accidentally connected to a call which should be screened.

MULTIPLE LISTED DIRECTORY NUMBERS

Description

Allows up to 50 publicly published numbers for any system.

Considerations

Some or all of the incoming and/or 2-way (incoming side) foreign exchange (FX) and/or local central office (CO) trunk groups can route to one of the following:

- The attendant group
- A Direct Department Calling group
- A Uniform Call Distribution group
- Remote Access

All of the 50 trunk groups available with the system can be administered as incoming or 2-way FX and/or local CO trunk groups. Since separate Listed Directory Numbers (LDNs) can be assigned to access each trunk group from the FX or local CO, up to 50 LDNs can be assigned (one for each trunk group).

In systems with Direct Inward Dialing (DID) trunk groups, up to eight DID LDNs can be assigned to route to the attendant group.

A unique display for incoming call identification can be provided for each LDN, including the DID numbers.

The following can be administered for each trunk group:

- The incoming destination
- A unique name for each LDN (optional, for display purposes)

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

MUSIC-ON-HOLD ACCESS

Description

Provides music to one party on hold, waiting in a queue, or parked. The music lets the waiting party know that the connection is still in effect.

Considerations

If a multiple-party connection is on hold, waiting in queue, or parked, music is not provided.

The system provides access to the music source.

The number of parties that can be connected to Music-on-Hold simultaneously is not limited.

Hardware and Software Requirements

Requires the music source(s) and one port on a TN763 Auxiliary Trunk circuit pack for each source. No additional software is required.

Interactions

When any one of the following features is activated, music is provided when one party is waiting or held:

- Hold
- Terminal Conference
- Transfer

In addition to these three features, a single party in Call Park can receive music. Also, a call placed in queue for a Direct Department Calling or Uniform Call Distribution group can receive a delay announcement followed by music.

NETWORK ACCESS—PRIVATE

Description

Allows calls to be connected to the following types of networks:

- Common Control Switching Arrangement (CCSA)
- Electronic Tandem Network (ETN) Access
- Enhanced Private Switched Communications Service (EPSCS) Access
- Tandem Tie Trunk Network

A private network provides call routing over facilities dedicated to the client.

Considerations

A total of 50 trunk groups can be assigned to the system, including private network trunk groups.

Unless prohibited by the Class of Restriction (COR), all incoming Private Network trunks except CCSA can access outgoing trunks without attendant or terminal user assistance. All incoming CCSA calls must route to an attendant or a terminal user.

When off-network capability is specified as part of the CCSA and EPSCS service, long-distance calls route as far as possible over these networks before terminating on the public network. Thus, charges for toll calls are reduced. The COR administered to individual system users determines whether access to this capability is allowed or denied.

Hardware and Software Requirements

Requires one port on a TN760 Tie Trunk circuit pack for each trunk assigned. No additional software is required.

Interactions

None.

NETWORK ACCESS—PUBLIC

Description

Provides voice terminal users and attendants with access to and from the public network.

Outgoing access is provided to the following:

- Local central offices (COs)
- Foreign exchange (FX) offices
- Wide Area Telecommunications Service (WATS) offices
- Other long-distance carriers

Incoming access is provided from the following:

- Local COs
- FX offices
- 800 Service offices

Considerations

The Automatic Route Selection feature can be used to select the most preferred route, where possible, for outgoing calls to the public network. Alternatively, trunk access codes can be assigned to outgoing routes for manual route selection.

Hardware and Software Requirements

Requires one port on a TN747 CO Trunk circuit pack for each trunk assigned. No additional software is required.

Interactions

None.

NIGHT SERVICE—NIGHT CONSOLE SERVICE

Description

Directs all calls for the primary and daytime attendant consoles to a night console. All attendant-seeking calls and calls waiting in queue are directed to the night console.

Considerations

Only one night console is allowed in the system. The night console can be activated only when the primary and daytime consoles have been deactivated. The attendant activates the night console and deactivates all other consoles by pressing the Night button on the primary console.

The night console must be identical to, and have the same features as, the primary console. A daytime console can double as the night console.

Hardware and Software Requirements

Requires an attendant console. No additional software is required.

Interactions

None.

NIGHT SERVICE—NIGHT STATION SERVICE

Description

Redirects incoming attendant-seeking trunk calls to designated extension numbers whenever the console is unattended.

This feature is activated under the following two conditions:

- The attendant has pressed the Night button on the primary console.
- A night console is not assigned or not operational.

Considerations

Any existing extension number can be assigned to receive night calls on a given incoming central office, foreign exchange, or 800 Service trunk group. A different extension number can be assigned for each trunk group. The assigned extension number can be one voice terminal or an answering group; that is, Direct Department Calling group, Uniform Call Distribution group, or Terminating Extension Group.

When the Night Station Service feature is active but extension numbers have not been established, the Trunk Answer From Any Station feature can be activated.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Inward Restriction

  Inward-restricted voice terminals can be administered for Night Station Service. Night Service features override Inward Restriction.

- Remote Access

  The Remote Access extension number can be specified as the Night Station extension number on an incoming, non-DID, trunk group.

NIGHT SERVICE—TRUNK ANSWER FROM ANY STATION

Description

Allows voice terminal users to answer all incoming attendant-seeking trunk calls when the attendant(s) is not on duty and when other voice terminals have not been designated to answer the calls.

The incoming call activates a gong, bell, or chime. A voice terminal user dials an access code and answers the call from any unrestricted voice terminal.

Considerations

Trunk Answer From Any Station (TAAS) is activated only under the following three conditions:

- The attendant has pressed the Night button on the primary console.
- A night console is not assigned or not operational.
- The Night Station Service feature is not active.

Hardware and Software Requirements

Requires an alerting device and one port on a TN742 Analog Line circuit pack. No additional software is required.

Interactions

Inward-restricted voice terminals can activate TAAS for incoming trunk calls. Night Service features override Inward Restriction.

OFF-PREMISES STATION

Description

Allows a remotely located inter-premises 2500-series voice terminal to be connected to the system.

Considerations

Off-Premises Stations are administered the same as on-premises 2500-series voice terminals.

The maximum loop distance for Off-Premises Stations is 26,000 feet, without repeaters.

Hardware and Software Requirements

Requires cross-connection through a Z100B1 cross-connect module and one port on a TN742 Analog Line circuit pack. Separate cross-connect field labels are used to specify Off-Premises Stations. No additional software is required.

Interactions

The Alerting feature might function improperly at an Off-Premises Station due to the distance. However, the Alerting feature can be disabled when the Off-Premises Station is administered. If the Alerting feature is not used with an Off-Premises Station, the terminal will receive 1-burst alerting for all calls.

PERSONAL CENTRAL OFFICE LINE

Description

Provides a dedicated trunk for direct access to or from the public network for multi-appearance voice terminal users. Typical users are executives, dispatchers, or buyers with a high volume of calls going outside the system, and businesses with specialized incoming calls (such as a service department).

Each Personal Central Office Line (PCOL) can have an appearance at up to four multi-appearance voice terminals. Users assigned this feature press the PCOL feature button to answer and place calls—dial access is not provided. The status lamp associated with the PCOL button indicates the busy or idle status of the trunk.

An incoming PCOL call alerts all voice terminals assigned the feature. The PCOL button status lamp flashes even if all call appearances at the voice terminal are active. If a call appearance is idle, the status lamp associated with that appearance also flashes.

Considerations

Central office (CO), foreign exchange (FX), and Wide Area Telecommunications Service (WATS) trunks can be assigned to this feature.

The system will support 25 PCOLs. These lines (trunks) are not included in the 50 trunk groups supported by the system. They are, however, included in the 200-trunk system limit.

PCOLs are not assigned a Class of Restriction.

When a PCOL is shared (assigned to a group), any group member can bridge onto a PCOL call through the Temporary Bridged Appearance feature. The Privacy—Manual Exclusion feature can be activated on such a call if the voice terminal is assigned an Exclusion button.

The Station Message Detail Recording (SMDR) feature can be activated for PCOL calls; however, the SMDR record will not identify the call as PCOL. The call will be recorded to the extension number assigned to the voice terminal where the call was originated or answered.

Leave Word Calling messages can be stored for a PCOL group. The messages are retrieved by an authorized systemwide message retriever. When a message is stored, the Remote Message Waiting Indicator assigned for the PCOL group lights. One Remote Message Waiting Indicator is allowed per group.

The following items can be administered for PCOLs:

- Group number (from 1 to 25)
- Group type (CO, FX, or WATS)
- Group name (optional, used for display purposes)
- Data Restriction activation
- SMDR activation
- Call Coverage path (redirection criteria can be Don't Answer and Cover All Calls)
- Extension numbers of voice terminals assigned to PCOL group (up to four terminals can share a PCOL)
- PCOL button (per terminal assigned to the PCOL group)

- Exclusion button (optional on a per-terminal basis)
- Remote Message Waiting Indicator (one allowed per PCOL group)

The following items can be administered for the CO, FX, or WATS trunk used for the PCOL:

- Circuit pack port number
- Trunk type
- Trunk name (for display purposes)
- Trunk access code (nondialable, used to identify the trunk for SMDR)
- Outgoing dialing type
- CO disconnect timing
- Terminating area code
- Prefix for code conversion
- Toll table index for code conversion
- Prefix 1 (needed for CO and FX trunks if the prefix 1 is needed for toll calls)

Hardware and Software Requirements

- Requires one port on a TN747 CO Trunk circuit pack for each CO, FX, or WATS trunk assigned as a PCOL.
- No additional software is required.

Interactions

The following features cannot be used with the PCOL feature:

- Abbreviated Dialing
- Automatic Route Selection
- Call Forwarding All Calls
- Ringback Queuing

POWER FAILURE TRANSFER

Description

Provides service to and from the local telephone company central office (CO) if power fails.

Considerations

Only local CO trunks can be used for Power Failure Transfer.

The 500-type (rotary dial) or 2500-type (touch-tone) voice terminals must be used for Power Failure Transfer. Rotary dialing must be used if the local CO accepts dial pulses only. Rotary dial, however, cannot be used except when the system is in the Power Failure Transfer mode.

From 6 to 42 (maximum) voice terminals can be connected to from 6 to 42 CO trunks for the Power Failure Transfer feature. The Power Failure Transfer feature is available in multiples of six.

Hardware and Software Requirements

- Requires one emergency transfer unit for every six trunks assigned to Power Failure Transfer.
- If CO trunks are ground start, requires a ground start key for each assigned voice terminal to originate calls.
- No additional software is required.

Interactions

During the Power Failure Transfer mode, no other system features can be activated.

PRIORITY CALLING

Description

Provides a special form of call alerting between internal voice terminal users. The called voice terminal user receives a distinctive 3-burst alerting signal.

An active single-line voice terminal user who receives a Priority Calling call will hear a distinctive 3-burst priority Call Waiting tone.

A multi-appearance voice terminal user receives the Priority Calling call on an idle call appearance. If all call appearances are active, the caller receives busy tone.

Considerations

Call Coverage Consult calls and callback calls from Automatic Callback and Ringback Queuing are Priority Calling calls.

Priority Calling is assigned to a voice terminal through the administered Class of Service.

The feature access code for Priority Calling can be administered.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Alerting

  Single-line voice terminals (2500-series) can be administered so that distinctive signals are not provided.

- Automatic Callback and Ringback Queuing

  Callback calls do not redirect, do not forward, and cannot be picked up by a Call Pickup group member.

- Call Coverage

  Priority Calling calls do not redirect to coverage unless the caller activates Go to Cover. If the call redirects, it remains a Priority Call, and the covering user receives a distinctive 3-burst alerting signal.

- Call Forwarding All Calls

  Priority Calling calls (except callback calls) will forward, and the forwarded call remains a Priority Calling call.

- Call Waiting Termination

  A Priority Calling call will wait on an active single-line voice terminal even if the Terminal Call Waiting feature is not assigned to the voice terminal.

- Dial Access to Attendant

  A Priority Calling call cannot be originated to the attendant. However, the attendant can originate Priority Calling calls.

PRIVACY—ATTENDANT LOCKOUT

Description

Prevents an attendant from reentering a multiple-party connection held on the console unless recalled by a voice terminal user.

Considerations

None.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

Privacy—Attendant Lockout does not function when a call using the Trunk-to-Trunk Transfer feature is held on the console

PRIVACY—MANUAL EXCLUSION

Description

Allows multi-appearance voice terminal users to keep other users with appearances of the same extension number from bridging onto an existing call.

Considerations

Exclusion is activated by pressing the Exclusion button on a per-call basis. The Privacy—Manual Exclusion feature is automatically deactivated when the Exclusion button is pressed a second time or when the existing call is transferred, parked, or placed on hold.

Privacy—Manual Exclusion can be used with the Personal Central Office Line and Terminating Extension Group features.

An Exclusion feature button with associated status lamp must be administered to the multi-appearance voice terminal.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

RECALL SIGNALING

Description

Allows a single-line voice terminal user, active on a call, to place the party on hold and obtain recall dial tone by pressing the Recall button or by flashing the switchhook. The user can then place another call or activate a feature, and return to the held party by pressing Recall twice or by flashing the switchhook again.

Considerations

Recall Signaling cannot be used to answer a waiting call.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

RECORDED TELEPHONE DICTATION ACCESS

Description

Permits voice terminal users, including Remote Access and incoming tie trunk users, to access dictation equipment.

Considerations

The system provides access to dictation equipment.

The start and stop functions of the dictation equipment can be voice- or dial-controlled.

Attendants cannot access the dictation equipment.

The following items can be administered:

- One port on an Analog Line circuit pack (per dictation machine) and an extension number

  or

- One port on an Auxiliary Trunk circuit pack and a trunk access code.

Hardware and Software Requirements

Requires telephone dictation machines and, depending on the type of machine, one port on a TN742 Analog Line circuit pack or one port on a TN763 Auxiliary Trunk circuit pack for each machine assigned. No additional software is required.

Interactions

The Recorded Telephone Dictation Access feature cannot be used with the following features:

- Attendant Conference
- Automatic Route Selection
- Terminal Conference

REMOTE ACCESS

Description

Permits callers from the public network to access the system and then use its features and services.

Remote Access users can dial into the system using central office, foreign exchange, or 800 Service trunks. The Remote Access feature is assigned an extension number, like any voice terminal. When a call is received on a trunk group dedicated to Remote Access, the system routes the call to the assigned extension number. If Direct Inward Dialing (DID) is provided and if the Remote Access number is within the range of numbers that can be accessed by DID, then the Remote Access feature can be accessed through the DID feature.

After access to the feature, the user hears system dial tone, and, for system security, may be required to dial a Barrier code. If a valid Barrier code is dialed, the user again hears dial tone, and can place calls the same as an on-premises user. If a Barrier code is not required, the user immediately dials the desired code or number.

The destination of incoming, non-DID, trunk calls can be an attendant or an extension number. The destination is specified on each individual trunk group. When the trunk group is dedicated to Remote Access, the Remote Access extension number is specified. In this case, the user does all dialing. If an attendant is needed on a call, the user dials the public network telephone number assigned, the Barrier code, if required, and "0," the attendant access code. To provide attendant-assisted calling, service can be arranged so the attendant handles calls during the day, but Remote Access applies after normal business hours. This is accomplished by setting the trunk group destination as "0" (the attendant), and specifying the Remote Access extension number as the Night Station number. Incoming calls route to the attendant unless the Night button on the primary console is pressed. When Night Service is in effect, incoming calls route to Remote Access.

Considerations

Ten Barrier codes, each with a different Class of Restriction (COR), can be administered. The Barrier codes can be from four to seven digits, but all codes must be the same length. Barrier codes not only provide system security but also define the calling privileges through the administered COR. Use of Barrier codes is optional on a per-system basis.

Ringback Queuing cannot be used on a Remote Access call since the system does not have access to the calling (outside) number.

Any feature requiring recall dial tone (for example, Hold and Transfer) cannot be accessed remotely.

The Remote Access caller must use a touch-tone voice terminal or equivalent.

The following items can be administered:

- Extension number
- Barrier code length (from four to seven digits)
- Barrier codes
- COR (per Barrier code)
- Trunk groups

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Night Service—Night Station Service

  The Remote Access extension number can be specified as the Night Station extension number on an incoming, non-DID, trunk group.

RESTRICTION—CONTROLLED

Description

Allows the attendant to activate and deactivate the following restrictions for an individual voice terminal or a group of voice terminals:

- Outward—The voice terminal(s) cannot be used for placing calls to the public network. Such call attempts receive intercept tone.
- Total—The voice terminal(s) cannot be used for placing or receiving calls. Direct Inward Dialing calls are routed to the attendant or a recorded announcement. All other calls receive intercept tone.

The desired Controlled Restriction is activated when the attendant dials the feature access code, 1 for Outward or 2 for Total, and the voice terminal extension number (Attendant Control—Extension) or the Class of Restriction (COR) for a group of voice terminals (Attendant Control—COR).

Considerations

All voice terminals with the same COR are affected by a group restriction.

Separate activate and deactivate feature access codes can be administered.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

Controlled Restriction overrides restrictions assigned by a COR.

RESTRICTION—MISCELLANEOUS TERMINAL

Description

Restricts callers at specified voice terminals from accessing certain other voice terminals.

Considerations

The Miscellaneous Terminal Restriction is controlled by the Class of Restriction (COR) assigned to the calling voice terminal user and to the voice terminal being called. Any COR can be administered to allow or deny access to any other COR. Restricted calls are routed to intercept tone.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

A voice terminal user with authorization to access an Abbreviated Dialing Privileged Group Number List or Privileged System Number List can place calls to any number on that list. COR assignments are not checked.

RESTRICTION—MISCELLANEOUS TRUNK

Description

Restricts users at specified voice terminals from accessing certain trunk groups, such as Wide Area Telecommunications Service (WATS).

Considerations

The Miscellaneous Trunk Restriction is controlled by the Class of Restriction (COR) assigned to the calling voice terminal user and to the trunk group being accessed. Any COR can be administered to allow or deny access to any other COR. Restricted calls are routed to intercept tone.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Abbreviated Dialing

  A voice terminal user with authorization to access an Abbreviated Dialing Privileged Group Number List or a Privileged System Number List can place calls to any number on that list. COR assignments are not checked.

- Automatic Route Selection

  This feature overrides the Miscellaneous Trunk Restriction feature. Permission or denial of Automatic Route Selection calls is determined by the Facility Restriction Level.

RESTRICTION—TOLL/CODE

Description

Restricts users at specified voice terminals from placing public network calls to certain numbers within the local area code, to certain foreign (nonlocal) area codes, and to service codes (such as 411 for directory assistance and 911 for emergency service).

Considerations

Nontoll calls within the local area code are always allowed.

Toll Restriction applies as follows:

- When a toll-restricted system user accesses a trunk group and dials a number string containing a 0 or 1 as the first or second digit, the system checks the Allowed Calls List. If the number is found, the call is permitted. If not, the caller receives intercept tone.
- When any system user accesses a toll-restricted trunk group and dials a number string containing a 0 or 1 as the first or second digit, the system checks the Allowed Calls List. If the number is found, the call is permitted. If not, the caller receives intercept tone.

The Allowed Calls List can include up to 10 central office codes (that is, the first three digits of a 7-digit number), area codes, and/or service codes that toll-restricted system users will be permitted to access.

Code Restriction applies when a code-restricted system user accesses a code-restricted trunk group and dials a number string. The system checks the code-restriction tables. If the number is found, the call is permitted. If not, the caller receives intercept tone.

Two code-restriction tables are established. One table lists certain toll central office codes within the local area code and the other lists certain foreign area codes and service codes. Code-restricted users are permitted to access the codes listed in the code-restriction tables.

Toll or Code Restriction (but not both) is administered to each foreign exchange and central office trunk on a trunk group basis.

Toll or Code Restriction (but not both) is administered to the following by the Class of Restriction:

- Attendant consoles as a group
- Incoming tie trunks on a trunk group basis
- Voice terminals on a per-terminal basis

Other items that can be administered are as follows:

- Allowed Calls List containing up to 10 codes that toll-restricted users will be permitted to access
- Code-restriction table listing central office codes within the local area code that code-restricted users will be permitted to access
- Code-restriction table listing foreign area codes and service codes that code-restricted users will be permitted to access

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

The Automatic Route Selection feature overrides Toll and Code Restriction. Permission or denial of Automatic Route Selection calls is determined by the Facility Restriction Level.

RESTRICTION—VOICE TERMINAL—INWARD

Description

Restricts callers at specified voice terminals from receiving public network, attendant-originated, and attendant-extended calls. A denied call is routed to intercept tone, a recorded announcement, or the attendant.

Considerations

Inward Restriction is administered to voice terminals by the Class of Restriction (COR).

Calls can redirect to an inward-restricted voice terminal. The COR of the originally called extension number is the only one checked.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Controlled Restriction

  Restrictions activated by the attendant override restrictions assigned by the COR.

- Night Service

  The Trunk Answer From Any Station and Night Station Service features, if assigned to an inward-restricted voice terminal, override the Inward Restriction.

- Tie Trunk Access

  Incoming dial repeating tie trunk calls can be completed directly to an inward-restricted extension number. However, such calls cannot be extended by an attendant to an inward-restricted voice terminal.

- Transfer

  Incoming trunk calls can be transferred from an unrestricted extension number to an inward-restricted extension number.

RESTRICTION—VOICE TERMINAL—MANUAL TERMINATING LINE

Description

Restricts callers at specified voice terminals from receiving calls other than those from an attendant. All other calls are routed to intercept tone, a recorded announcement, or an attendant. The voice terminal user can originate calls and activate features.

Considerations

The Manual Terminating Line Restriction feature is administered to voice terminals by the Class of Restriction (COR).

Calls can redirect to a voice terminal assigned this feature. The COR of the originally called extension number is the only one checked.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Controlled Restriction

  Restrictions activated by the attendant override restrictions assigned by the COR.

- Night Service

  The Trunk Answer From Any Station or Night Station Service feature, if assigned to a restricted voice terminal, overrides Manual Terminating Line Restriction.

RESTRICTION—VOICE TERMINAL—ORIGINATION

Description

Restricts callers at specified voice terminals from originating calls. Voice terminal users can receive calls.

If a voice terminal user attempts to place a call, intercept tone is received.

Considerations

The Origination Restriction feature is administered to voice terminals by the Class of Restriction (COR).

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

The Controlled Restriction overrides restrictions assigned by the COR.

RESTRICTION—VOICE TERMINAL—OUTWARD

Description

Prevents specified voice terminal users from activating the Public Network Access feature. Calls can be placed to other voice terminal users, to the attendant, and to tie trunks.

Considerations

The attendant or an unrestricted voice terminal user can extend a call to an outside number for the outward-restricted voice terminal user.

The Outward Restriction feature is administered to voice terminals by the Class of Restriction (COR).

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

The Controlled Restriction overrides restrictions assigned by the COR.

RESTRICTION—VOICE TERMINAL—TERMINATION

Description

Restricts voice terminal users on specified extension numbers from receiving any calls. Voice terminal users can originate calls.

Considerations

The Termination Restriction feature is administered to voice terminals by the Class of Restriction (COR).

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

The Controlled Restriction overrides restrictions assigned by the COR.

RINGBACK QUEUING

Description

Places outgoing calls in an ordered queue (first-in, first-out) when all trunks are busy. The voice terminal user is automatically called back when a trunk becomes available. The voice terminal receives a distinctive 3-burst alerting signal (Priority Calling) when called back.

When an all-trunks-busy condition exists within a trunk group, a multi-appearance voice terminal user receives reorder (fast-busy) tone after dialing is complete. To access Ringback Queuing, the user presses the Automatic Callback button. The system acknowledges the availability of the queue by returning a confirmation tone (three short bursts of tone).

Ringback Queuing is automatic for a single-line voice terminal. After dialing is complete, the user hears confirmation tone if the queue is available. No action is required by the voice terminal user. The user hangs up and awaits callback.

The callback call is automatically placed to the terminal when a trunk becomes available. When the user answers the callback call, the original call automatically continues. Redialing is not required.

Considerations

The system allows a maximum of 100 calls in queue for all combined trunk groups.

A single-line voice terminal can have only one call waiting at a time; therefore, Ringback Queuing is denied to these voice terminals if a call is already waiting.

A multi-appearance voice terminal can have one callback call associated with each Automatic Callback button assigned to the terminal.

A queue request will be canceled for the following reasons:

- A trunk is not available within 30 minutes.
- The voice terminal user does not answer the callback call within the administered interval (2 to 9 alerting cycles).
- The voice terminal is busy when the callback call is attempted.
- The voice terminal user dials the Ringback Queuing cancellation code or presses the Automatic Callback button associated with the queued call.

Queuing can reduce the number of trunks required.

The following items can be administered:

- Callback call no-answer time-out (from 2 to 9 alerting cycles)
- Automatic Callback button (per multi-appearance voice terminal)
- Ringback Queuing cancellation code

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

If Ringback Queuing is provided, Automatic Callback must also be provided. Automatic Callback is administered through the Class of Service.

Ringback Queuing affects the following features:

- Call Coverage

  Callback calls do not redirect even if Send All Calls is activated.

- Call Forwarding All Calls

  Callback calls are not forwarded.

- Call Pickup

  Callback calls cannot be picked up.

- Conference or Transfer

  A single-line voice terminal cannot receive a callback call while it has a call on hold and can have only one active call at a time.

- Remote Access

  A callback call cannot be made to a Remote Access user because the system does not know the calling number.

Description

Station Message Detail Recording (SMDR) records detailed call information on all incoming and outgoing calls on specified trunk groups and sends this information to an SMDR output device. Internal voice terminal calls are not recorded. The SMDR output device provides a detailed printout which can be used by the System Manager to compute costs, allocate charges, analyze calling patterns, and keep track of unnecessary calls.

Call detail information is provided on trunk groups, loudspeaker paging, and code calling access administered for SMDR. SMDR provides detailed call information for the following types of calls:

- Outgoing Calls—Calls originated by a system voice terminal user or attendant going out on a trunk group
- Incoming Calls—Calls incoming on a trunk group and terminating at a system voice terminal or attendant console
- Tandem Calls—Calls incoming on a trunk group and outgoing on another trunk group
- Ineffective Call Attempt—Calls originated by a system voice terminal user blocked because the user did not have sufficient calling privileges or because all outgoing trunks were busy
- Calls made using the Loudspeaker Paging Access and Code Calling Access features

SMDR Data Formats

This part covers the two formats sent to the SMDR output device, call detail and date record formats.

Call Detail Record Format

The call detail record format provides detailed information concerning an incoming or outgoing call. Call detail records are generated during call processing and are sent to the SMDR output device in American Standard Code for Information Interchange (ASCII). SMDR data transmitted to the Applications Processor (AP) is not included because the System 75 link to the AP uses BX.25 protocol and transmits more than SMDR data.

The following list describes the SMDR data collected for each call and the number of digits in each field. All information is right adjusted in the respective field, unless otherwise indicated. The list describes the data fields associated with an SMDR output device such as TELESEER* unit, printer, 94A Local Storage Unit (LSU), COMM-STOR† II unit, or client provided equipment.

- Access Code Dialed (up to 3 digits)

  This field is used only for outgoing calls. This field can be the Automatic Route Selection access code or the access code of a specific trunk group.

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* Trademark of AT&T Information Systems, Inc.

† Registered Trademark of Sykes Datatronics, Inc.

- Access Code Used (up to 3 digits)

  This field is used only for outgoing calls and only when the trunk group used is different from the trunk group access code dialed. This field contains the access code of the actual trunk group that the call was routed over. When the dialed and used access code is the same, this field will be blank.

- Account Code (up to 5 digits)

  This field is optional but can contain a number to associate call information with projects or account numbers. Account Codes must be prefixed with a fixed digit or a series of digits. The access code is administrable on the Feature Access Code form. On outgoing calls, the access code must be dialed before the trunk access code or Automatic Route Selection code. Information in this field is left adjusted. These account codes allow the System Manager associate calling information with projects or account numbers. The access code is not recorded because it is not part of the SMDR account code.

- Calling Number (up to 4 digits)

  This field contains the extension number of the originating voice terminal user or the trunk group access code used for an incoming or tandem call. Information in this field is left adjusted.

- Condition Code (1 character)

  These codes reflect special events relating to the call. There are two sets of condition codes. The first set applies to the printer, TELESEER unit, and 94A LSU. These condition codes are listed and defined in Table 5-A. The second set of codes applies to the COMM-STOR II unit. Mapping the COMM-STOR II unit condition code to the condition codes previously identified is done as follows: 1=A, 4=D, 7=G, 9=I, A=blank, C=L, and E=N.

  When two condition codes apply on the same call, one will override the other. The matrix in Table 5-B defines the overrides.

TABLE 5-A. Condition Codes

CONDITION CODES	DESCRIPTION
1	Identifies an attendant-handled call or an attendant-assisted call (except conference calls).
4	Identifies a call of about 10 hours. On such a call, a call record with this condition code and a duration entry of 9 hours, 59 minutes, and 1 to 9 tenths of a minute is produced after the first period. A similar call record with this condition code is produced after each succeeding 10-hour period. When the call does terminate, a final call record with a different condition code identifying the call type is produced.
7	Identifies calls served by the Automatic Route Selection feature.
8	Identifies calls which have been served on a delayed basis via the Ringback Queuing feature.
9	Identifies an incoming or tandem call.
A	Identifies an outgoing call.
C	Identifies a conference call. A separate call record with this condition code is produced for each incoming or outgoing trunk serving the conference connection. The only voice terminal recorded for a conference call is the conference call originator.
E	Identifies an ineffective call attempt due to facilities not being available such as all trunks are busy and no queuing exists or the queue is full on an outgoing call, or the called voice terminal is busy or unassigned for an incoming call attempt.
F	Identifies an ineffective call attempt due to insufficient calling privileges of the originator (assigned per the Class of Restriction).

Note: When more than one condition applies to a call, the overriding code is shown in Table 5-B.

TABLE 5-B. Condition Code Override Matrix

	CONDITION CODE								
	1	4	7	8	9	A	C	E	F
1	NA	4	1	NA	9	1	C	E	NA
4	4	NA	4	4	4	4	4	NA	NA
7	1	4	NA	7	9	7	C	E	F
8	NA	4	7	NA	NA	8	C	E	NA
9	9	4	9	NA	NA	NA	C	E	F
A	1	4	7	8	NA	NA	C	E	F
C	C	4	C	C	C	C	NA	NA	NA
E	E	NA	E	E	E	E	NA	NA	NA
F	NA	NA	F	NA	F	F	NA	NA	NA

- Dialed Number (up to 15 digits)

  This field contains the outside number dialed by a system user.

- Duration (4 digits)

  All calls are timed. The timing is measured in hours (0 through 9), minutes (00 through 59), and to the nearest tenth of a minute (00 through 09).

- Facilities Restriction Level (FRL) (1 digit)

  FRLs, numbered 0 through 7, are associated with the Automatic Route Selection feature and define calling privileges. This field contains the originating voice terminal user's FRL for outgoing calls or the FRL assigned to the incoming trunk group for incoming and tandem calls.

- Incoming Circuit Identification (2 digits)

  This field contains the member number of a trunk within a trunk group used for an incoming call.

  This field does not appear for the COMM-STOR II unit records.

- Outgoing Circuit Identification (2 digits)

  This field contains the member number of the trunk within a trunk group used for an outgoing call.

  This field does not appear for the COMM-STOR II unit records.

The call detail information sent to the TELESEER unit is shown in Figure 5-3.

The call detail information sent to the printer is shown in Figure 5-4.

The call detail information sent to the 94A Local Storage Unit (LSU) is shown in Figure 5-5.

The call detail information sent to the COMM-STOR II unit is shown in Figure 5-6.

ASCII Character Position	Data Field Description
00-03	Space
04	Time Hour- (tens)
05	Time Hour- (units)
06	Time Minute (tens)
07	Time Minute (units)
08	Space
09	Duration Hour
10	Duration Minute (tens)
11	Duration Minute (units)
12	Duration Minute (tenths)
13	Space
14	Condition Code*
15	Space
16-18	Access Code #1†
19-21	Access Code #2†
22	Space
23-37	Dialed Number†
38	Space
39-42	Calling Number†
43	Space
44-48	Account Code†
49-60	Space
61	FRL
62-64	Space
65-66	Incoming Circuit ID‡
67-69	Space
70-71	Outgoing Circuit ID‡
72-76	Space
77	Carriage Return
78	Line Feed
79-81	null

* Refer to Table 5-A.
† Data is right justified and padded with blanks.
‡ Data is right justified and padded with Os.

Figure 5-3. SMDR Data Format—TELESEER Unit

ASCII Character Position	Data Field Description
00-02	Space
03	Time Hour- (tens)
04	Time Hour- (units)
05	Time Minute (tens)
06	Time Minute (units)
07	Space
08	Duration Hour
09	Duration Minute (tens)
10	Duration Minute (units)
11	Duration Minute (tenths)
12	Space
13	Condition Code*
14	Space
15-17	Access Code #1†
18	Space
19-21	Access Code #2†
22	Space
23-37	Dialed Number†
38	Space
39-42	Calling Number†
43	Space
44-48	Account Code†
49-60	Space
61	FRL
62-64	Space
65-66	Incoming Circuit ID‡
67-69	Space
70-71	Outgoing Circuit ID‡
72	Space
73	Carriage Return
74	Line Feed

* Refer to Table 5-A.
† Data is right justified and padded with blanks.
‡ Data is right justified and padded with 0s.

Figure 5-4. SMDR Direct Output Format From the System to the Printer

ASCII Character Position	Data Field Description
01	Time Duration-Hours
02	Time Duration-Minutes (tens)
03	Time Duration-Minutes (units)
04	Time Duration-Minutes (tenths)
05	Condition Code*
06-08	Access Code #1†
09-11	Access Code #2†
12-26	Dialed Number†
27-30	Calling Number†
31-35	Account Code†
36-44	Space
45	FRL
46	Space
47-48	Incoming Circuit ID
49	Space
50-51	Outgoing Circuit ID
52-54	Space
55	Carriage Return
56	Line Feed
57-59	null

* Refer to Table 5-A.
† Data is right justified and padded with blanks.

Figure 5-5. SMDR Direct Output Format From the System to the 94A Local Storage Unit System

ASCII Character Position	Data Field Description
00-02	Space
03	Time Hour- (tens)
04	Time Hour- (units)
05	Time Minute (tens)
06	Time Minute (units)
07	Space
08	Duration Hour
09	Duration Minute (tens)
10	Duration Minute (units)
11	Duration Minute (tenths)
12	Space
13	Condition Code*
14	Space
15-17	Access Code #1†
18-20	Access Code #2†
21	Space
22-36	Dialed Number†
37	Space
38-41	Calling Number†
42	Space
43-47	Account Code†
48-56	Space
57	FRL
58	Carriage Return
59	Line Feed
	null
	null
	null

* Refer to Table 5-A.
† Data is right justified and padded with blanks.

Figure 5-6. SMDR Direct Output Format From the System to the COMM-STOR II Unit

Date Record Format

Three formats are available for date records, one for the 94A LSU (Figure 5-7), one for the COMM-STOR II unit, printer, and AP (Figure 5-8), and one for the TELESEER unit (Figure 5-9). The records sent to the TELESEER, COMM-STOR II, and printer contain the date only while the records sent to the 94A LSU contain both date and time.

ASCII Character Position	Data Field Description
01-02	Hour*
03	Colon (:)
04-05	Minute*
06	Blank
07-08	Month*
09	Slash (/)
10-11	Day*
12	Carriage Return
13	Line Feed
14-16	Null

* Leading zero added if needed.

Figure 5-7. Date Record Format to 94A LSU

ASCII Character Position	Data Field Description
01-02	Month*
03	Space
04-05	Day*
06	Carriage Return
07	Line Feed
08-10	Null

* Leading zero added if needed.

Figure 5-8. Date Record Format to COMM-STOR II Unit, Printer, and AP

ASCII Character Position	Data Field Description
00-01	Month*
02-03	Day
04	Carriage Return
05	Line Feed
06-08	Null

* Leading zero added if needed.

Figure 5-9. Date Record Format to TELESEER Unit

Set Time and Date

The System 75 clock must be set for daylight savings time. Changing the time and date assures that SMDR records have the correct date and time for the records being kept. The time and date can be changed using the System Access Terminal.

SMDR Output Devices

SMDR data is collected by the system and is continually sent to an output device for processing. An output device could be a TELESEER unit, printer, a 94A LSU, a COMM-STOR II unit, AP, host computer, or client provided equipment. Only one type of output device can be used for a given system.

The System 75 can store up to 305 SMDR records which are sent to the output devices at the rate of one record per second.

A 1200 baud rate should be used over the cable for the TELESEER unit, 94A LSU, COMM-STOR II unit, or printer. The link to the AP should operate at 9600 baud.

The following paragraphs give a brief description of each output device.

TELESEER Unit

The TELESEER unit is an output device that stores two types of System 75 information regarding each call record: call record details and summary totals. Call details consist of the following:

- Time of call
- Duration of call
- Account code
- Type of call
- Extension of call
- Dialed number
- Date of call

Summary totals are running totals of the call records that fall into the following categories:

- Time of day (on an hourly basis)
- Cost (eight ranges)
- Duration (seven time ranges)
- Date
- Department by cost center and extension
- Call type
- Account code
- Access code/trunk number/trunk group number
- Printed call categories
- Recorded call categories

The TELESEER unit can store up to 28,000 call records, 500 extension numbers, and 2000 account codes. SMDR records sent to the TELESEER unit are 80 bytes or 640 characters long.

The TELESEER unit provides four different types of reports: Summary, Account Code Detail, Activity, and Selection.

The Summary Report provides a condensed listing of the number, duration, and cost of calls. This report provides a general overview of voice terminal activity. The following information can be taken from the report:

- Large departmental voice terminal costs
- Large costs attributed to specific extension numbers
- Improper use of WATS lines
- Voice terminal usage for specific account codes
- Lengthy voice terminal conversations

An example of a Summary Report is shown in Figure 5-10.

The Account Code Detail Report lists call record details for each call record that contains an account code. This report is helpful in tracking calls from specific users. In addition, this report can be used for user billback or cost allocation by account code. The Account Code Report provides duration of calls, number dialed, type of calls made, account codes, and cost for each call. An example of an Account Code Detail Report is shown in Figure 5-11.

An Activity Report lists call record details for each extension number assigned to the System 75. The Report provides time, date, type of call, account codes, and cost of each call. An example of a Unit Activity Report is shown in Figure 5-12.

A Selection Report allows the System Manager to specify the type of information to be printed in a report. All call record details stored in the TELESEER unit that pertain to parameters selected are printed. Any or all of the following data can be specified:

- Time of day
- Date for each
- Cost for each call
- Duration of each call measured in hours, minutes, and seconds

- Extension number that originated the call
- Trunk Number/Access Code
- Account code number used for the call
- Dialed number
- Type trunk used for the call
- Department

An example of a Unit Selection report is shown in Figure 5-13.

Printer

An 80- or 132-column (character) printer can be connected as an SMDR output device. The printer prints SMDR records in a 1-line format. No data processing or reports are provided. SMDR records sent to the printer are 80 bytes or 640 characters long.

94A LSU

The 94A LSU collects and stores Message Detail Records (MDR) data from System 75. The 94A LSU stores MDR for Electronic Tandem Network (ETN) clients or multi-location clients served by other System 75s. SMDR records sent to the 94A LSU are 59 bytes or 470 characters long. The 94A LSU can handle up to 14,600 call records per hour, store up to 16,000 records, and transmit up to 7200 calls per hour over a 1200 baud link to a 93B Centralized Message Detail Recorder (CMDR).

COMM-STOR II Unit

The COMM-STOR II unit records all information and generates report summaries by date and time of call, call duration, dialed numbers, calling voice terminal, account codes, department, dial-access code, and cost. SMDR records sent to the COMM-STOR II unit are 80 bytes or 640 characters long. The COMM-STOR II unit can handle up to 2000 voice terminals and can operate from 300 to 4800 baud.

Applications Processor (AP)

The AP Call Detail Recording and Reporting (CDRR) collects and formats switch generated SMDR. CDRR provides comprehensive and customized call detail reporting.

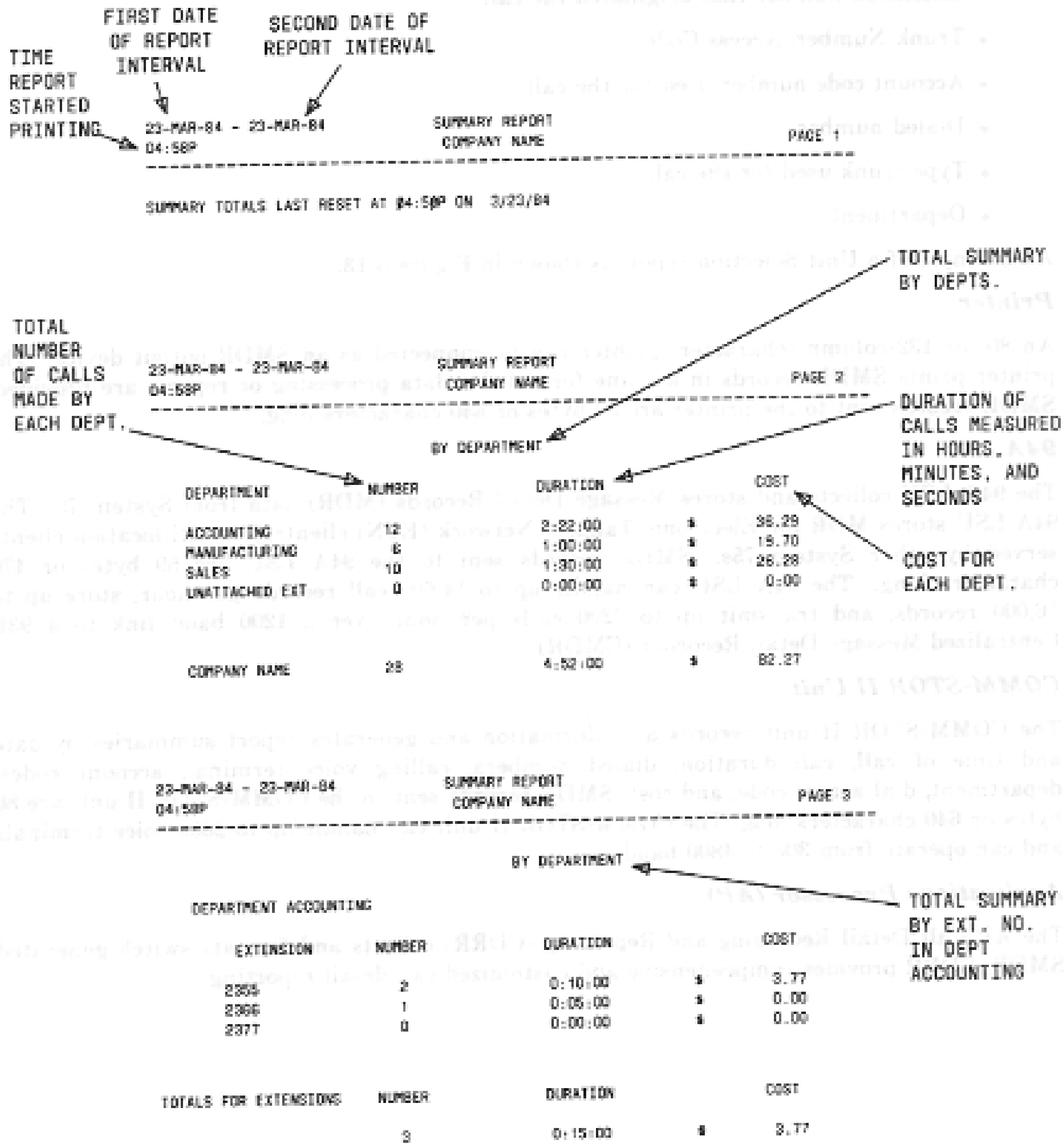

Figure 5-10. Example of a TELESEER Unit Summary Report

THIS IS THE TRUNK ACCESS CODE, TRUNK NUMBER, OR TRUNK GROUP NUMBER

THIS INDICATES THE TYPE OF CALL
IDD=INTERNATIONAL
OST=INTERSTATE LONG DISTANCE
OCC=OTHER COMMON CARRIERS

```
23-MAR-84 - 23-MAR-84                    ACCOUNT CODE REPORT
0:12P                                       COMPANY NAME                        PAGE 1

03/23 09:43AM      3:00   101    7        454-1362      FX 556432                 1.05
03/23 01:01PM      4:00   530    8        473-1502     LOC 556432                 0.08


      NUMBER                  DURATION                    COST
        2                      0:07:00               $     1.13


ACCOUNT CODE DETAIL REPORT FOR ACCOUNT CODE 88644
 DATE   TIME   DURATION  EXT  ACC DIALED DIGITS         TYP ACCOUNT CODE      COST $
 ----   ----   --------  ---  --- ------ ------         --- ------- ----      -------
03/23 11:31AM     8:00  1766    9 1-416-324-5012        IDD 88644                2.55
03/23 01:16PM    30:00  2388    9 1-206-888-4587        OST 88644               13.20


      NUMBER                  DURATION                    COST
        2                      0:38:00               $    15.75

ACCOUNT CODE DETAIL REPORT FOR ACCOUNT CODE 95643
 DATE   TIME   DURATION  EXT  ACC DIALED DIGITS         TYP ACCOUNT CODE      COST $
 ----   ----   --------  ---  --- ------ ------         --- ------- ----      -------
03/23 11:29AM    11:00  2400    9 9872011222323413128312459
                                                       OCC 95643                 3.74


      NUMBER                  DURATION                    COST
        1                      0:11:00               $     3.74

ACCOUNT CODE DETAIL REPORT FOR ACCOUNT CODE 12367
 DATE   TIME   DURATION  EXT  ACC DIALED DIGITS         TYP ACCOUNT CODE      COST $
 ----   ----   --------  ---  --- ------ ------         --- ------- ----      -------
03/23 10:10AM     5:00  2388   6T                       --- 12367                 0.00


      NUMBER                  DURATION                    COST
        1                      0:05:00               $     0.00

ACCOUNT CODE DETAIL REPORT FOR ACCOUNT CODE 41363
 DATE   TIME   DURATION  EXT  ACC DIALED DIGITS         TYP ACCOUNT CODE      COST $
 ----   ----   --------  ---  --- ------ ------         --- ------- ----      -------
03/23 11:01AM    12:00  3011    4 663-2828               FX 41364                 5.01
03/23 01:15PM     5:00  3011    7 288-5454               FX 41364                 1.77
03/23 02:13PM     3:00  2400    6 1-417-987-3498        WTS 41363                 0.70


      NUMBER                  DURATION                    COST
        3                      0:20:00               $     7.40
```

Figure 5-11. Example of a TELESEER Unit Account Code Detail Report

FIRST DATE
OF REPORT
INTERVAL

TIME
REPORT
STARTED
TO PRINT

```
23-MAR-84 - 23-MAR-84              ACTIVITY REPORT
Ø5:ØTP                              COMPANY NAME                        PAGE 1
DEPARTMENT SALES

ACTIVITY REPORT FOR EXTENSION 4150
 DATE   TIME   DURATION  EXT  ACC DIALED DIGITS          TYP ACCOUNT CODE     COST $
 ----   ----   --------  ---  --- ------ ------          --- ------- ----    -------
03/23 10:10AM      5:00 4155  68                          FX                    1.77
03/23 02:53PM     12:00 4155    6 1-206-324-5151         WST                    2.86
03/23 03:12PM      5:00 4155    6 1-312-654-7829         WST                    1.18

                                                                                5.81
TOTALS            22:00

TOTAL CALLS 3

ACTIVITY REPORT FOR EXTENSION 4366
NO RECORDS STORED

ACTIVITY REPORT FOR EXTENSION 4444
NO RECORDS STORED

23-MAR-84 - 23-MAR-84              ACTIVITY REPORT
05:07P                              COMPANY NAME                        PAGE 2
COST CENTER 516

ACTIVITY REPORT FOR EXTENSION 4355
 DATE   TIME   DURATION  EXT  ACC DIALED DIGITS          TYP ACCOUNT CODE     COST $
 ----   ----   --------  ---  --- ------ ------          --- ------- ----    -------
03/23 10:12AM      2:00 4355    6 1-408-454-1362         WST 88267              0.46
03/23 01:10PM     30:00 4355    9 1-206-999-5478         OST 54321             13.20

                                                                               13.66
TOTALS            32:00

TOTAL CALLS 2

ACTIVITY REPORT FOR EXTENSION 4455
NO RECORDS STORED

ACTIVITY REPORT FOR EXTENSION 45ØØ
 DATE   TIME   DURATION  EXT  ACC DIALED DIGITS          TYP ACCOUNT CODE     COST $
 ----   ----   --------  ---  --- ------ ------          --- ------- ----    -------
03/23 06:05AM     11:00 4500    9 1-419-643-7474         OST 34671              3.02

                                                                                3.02
TOTALS            11:00

TOTAL CALLS 1

ACTIVITY REPORT FOR EXTENSION 4622
 DATE   TIME   DURATION  EXT  ACC DIALED DIGITS          TYP ACCOUNT CODE     COST $
 ----   ----   --------  ---  --- ------ ------          --- ------- ----    -------
03/23 00:05AM     10:00 4622    9 1-419-643-7474         OST 54321              1.49
03/23 02:02PM      1:00/4622    7 344-7542                FX 88267              0.33

                                                                                1.82
TOTALS            11:00

TOTAL CALLS 2

23-MAR-84 - 23-MAR-84              ACTIVITY REPORT
05:07P                              COMPANY NAME                        PAGE 3
COST CENTER 52Ø

ACTIVITY REPORT FOR EXTENSION 53ØØ
 DATE   TIME   DURATION  EXT  ACC DIALED DIGITS          TYP ACCOUNT CODE     COST $
 ----   ----   --------  ---  --- ------ ------          --- ------- ----    -------
03/23 10:42AM     10:00 5300  84                         WST                    1.79
03/23 01:01PM      4:00 5300    9 473-1502               LOC 55643              0.08

                                                                                1.87
TOTALS            14:00

TOTAL CALLS 2
```

Figure 5-12. Example of a TELESEER Activity Report

SELECTION REPORT
COMPANY NAME

DURATION(10:00 - 18:00:00)

DATE	TIME	DURATION	EXT	ACC	DIALED DIGITS	TYP	ACCOUNT CODE	COST $
03/23	10:42AM	10:00	5300	64		WST		1.79
03/23	11:01AM	12:00	3011	4	003-2028	FX	41363	5.01
03/23	11:15AM	10:00	1122	9	1-315-681-0846	IST	335678	0.83
03/23	11:29AM	11:00	2400	9	98720112222334131293124S9			
						OCC	95643	3.74
03/23	11:35AM	22:00	1011	9	1-213-324-5012	OST	12345	9.76
03/23	00:05AM	10:00	4622	9	1-419-643-7474	OST	54321	1.49
03/23	06:05PM	11:00	4500	9	1-818-643-7474	OST	34671	3.02
03/23	01:16PM	30:00	4355	9	1-206-383-5478	OST	54321	13.20
03/23	01:16PM	30:00	2388	9	1-206-888-4587	OST	88644	13.20
03/23	02:35PM	15:00	1122	9	9871011278585618158832200			
						OST		4.60
03/23	02:53PM	12:00	4155	6	1-206-324-5151	WST		2.88
03/23	03:02PM	35:00	2388	8		---		6.30
03/23	03:22PM	25:00	2400	9	98720112834576131536572l9			
						OCC		1.72
TOTALS		3:53:00						67.62

TOTAL CALLS 13

TOTAL TIME SPENT ON 13 CALLS: 3 HOURS AND 53 MINUTES

TOTAL COST FOR 13 CALLS WHICH LASTED FOR 3 HOURS AND 53 MINUTES

Figure 5-13. Example of a TELESEER Unit Selection Report

Considerations

When a voice terminal user wants an SMDR record generated for a particular account number, the SMDR access code (normally *6) and the account number must be dialed before the Automatic Route Selection or trunk access code and called number are dialed.

The originally dialed extension number on an incoming call, or the originator's extension number on an outgoing call, is always recorded for SMDR even if the call is transferred to another voice terminal.

On an attendant-assisted call, whether the attendant dials the outside number or allows Through Dialing, the extension number of the requesting user will be recorded for SMDR. However, the attendant must dial an account code, if provided, before dialing the trunk access code.

If the attendant is extending a call to a voice terminal, an account code can be dialed before the extension number is dialed.

Voice terminal users cannot dial an account code when extending a call to another voice terminal. However, a voice terminal user extending a call to a trunk can dial an account code before dialing the Automatic Route Selection or trunk access code.

SMDR Administration

To activate SMDR for the system, the following items can be administered.

System Parameters

- The type of SMDR output device to be used.
- The extension number assigned to the output device.
- The printer paper width (80 or 132 columns) if a printer is the output device.
- The SMDR account code length number (from 1 to 5), the system defaults to 2 digits.
- The SMDR can be suppressed for Ineffective Call Attempts or for All Calls Excluding Outgoing Calls; system defaults to no. Ineffective call attempts are calls originated by a voice terminal user blocked because the user did not have sufficient calling privileges or because all outgoing trunks were busy.

Trunks, Loudspeaker Paging, and Code Calling Access

SMDR can be assigned to all trunk groups, Loudspeaker Paging Access trunks, and Personal Central Office Line trunks. The system defaults to yes for SMDR. The System Manager must determine which types of trunks will be assigned SMDR.

Feature Access Codes

Assign SMDR account code access code. The system defaults to *6.

Modules and Modems

The SMDR output device can be connected to a Processor Data Module (PDM), Trunk Data Module, or a Modem. The following items must be administered if SMDR is not connected to an AP.

- A data-channel number (01 to 04) must be assigned using the 700A data module form and entering data-channel for the type. This channel provides a path for SMDR data from the Switch Processing Element to the time-division bus.
- If the SMDR output device is connected to a PDM, administer a 700A PDM form.
- If the SMDR output device is connected to Trunk Data Module, administer a 700B Trunk Data Module form.

- If the SMDR output device is connected to a 212A type modem, a 2500 Voice Terminal and Pooled Modem form must be completed. This allows circuit switched data connections between digital data communications equipment and analog data communications equipment (modems).

If the SMDR output device is connected to an AP, the following forms must be administered.

- Data Module form administered as an interface module. This is used with the TN719 Interface 3 circuit pack.
- 700A Processor Data Module.

The modules or modems used to administer the TELESEER unit or printer are also used to administer an SMDR output device connected to client provided equipment such as a personal computer or tape unit.

The modules or modems used to administer the COMM-STOR II unit and 94A LSU are also used to administer an SMDR output device connected to Data Communications Equipment (DCE).

The System Parameters form must be administered regardless of what type SMDR output device is used.

Hardware and Software Requirements

Requires an output device (such as a printer), a Processor Data Module, Trunk Data Module, or a 212A type modem to interconnect the output device. The Trunk Data Module is also used to connect SMDR to a Host Computer. The Host Computer must be connected over a private line terminated at the System 75 using a Trunk Data Module. The Host Computer is used to process SMDR information sent from the system.

If SMDR is used with the AP, a TN716 Interface 1, TN720 Interface 2, and a TN719 Interface 3 circuit pack must be installed and connected.

The PDM or Trunk Data Module requires one port on a TN754 Digital Line circuit pack.

If the output device is connected to a 212A type modem, a TN742 Analog Line circuit pack and a TN758 Pooled Modem circuit pack must be installed and connected. A 212A type modem is also required.

No additional software is required.

Interactions

The following interaction discussions assume SMDR is activated.

- Abbreviated Dialing

  Outgoing calls made by this feature are recorded just as if the stored number had been manually dialed.

- Attendant Console

  If an attendant-assisted call involves an outgoing trunk, the primary extension of the voice terminal user which requested attendant service is recorded as the calling number, even if the attendant dialed the outside number. Condition Code 1 indicates the call was assisted by the attendant.

  If the attendant allows through dialing, the primary extension of the voice terminal user who dialed the number is recorded as the calling party. Condition Code 1 indicates the call was assisted by the attendant.

On attendant-assisted calls that require an account code, the account code must be entered before the trunk access code.

If the attendant is redirecting an incoming call to a voice terminal, the attendant may dial an account code before dialing the extension number.

- Automatic Route Selection

  SMDR records the following information for Automatic Route Selection (ARS):

  - The fact an ARS call was made
  - The calling extension number
  - The Facility Restriction Level of the calling extension
  - The called number
  - The type of trunk group used for the ARS call
  - The time of call completion
  - The call duration (how long the parties talked)

  If SMDR is suppressed for the trunk group actually used on an ARS call, an SMDR record is not generated; otherwise, Condition Code 7 applies. The ARS access code is recorded in the Access Code Dialed field and the trunk access code for the trunk group actually used is recorded in the Access Code Used field.

- Call Coverage

  When an incoming call is answered by a covering voice terminal, the extension number dialed by the originating party is recorded as the dialed number.

- Call Forwarding All Calls

  When a call is forwarded to another voice terminal, the extension number dialed by the calling party is recorded as the dialed number.

- Call Pickup

  When the call is answered by another voice terminal user in the pickup group, the extension number dialed by the calling party is recorded as the dialed number.

- Call Waiting Termination

  Call duration timing starts when the voice terminal answers an incoming call.

- Central Office Trunks

  All incoming and outgoing calls on a CO trunk group will be recorded.

- Conference

  A call is considered a Conference Call if it contains at least one trunk which is eligible for SMDR recording plus two or more nonattendant parties. Condition Code C applies to each SMDR record made for a Conference Call.

  For a conference call, a separate SMDR is produced for each outgoing/incoming trunk serving the conference call.

  For the outgoing portion of a conference call involving multiple voice terminals, the voice terminal which requested outside dial tone to bring an outside party into the conference is recorded as the calling party.

  For the outgoing/incoming portion of a conference call, the call duration in SMDR reflects the entire time from when the first party was connected until all parties are disconnected.

- Direct Department Calling (DDC) and Uniform Call Distribution (UCD)

  The DDC or UCD group extension number is recorded as the called number.

- Direct Inward Dialing (DID)

  All incoming calls on the DID trunk group will be recorded.

- Foreign Exchange (FX) Trunks

  All calls made on an FX trunk group will be recorded.

- Intercept Treatment

  If an outgoing or tandem call is routed to Intercept Treatment, the number dialed by the calling party is recorded as the dialed number and Condition Code F applies.

- Loudspeaker Paging and Code Calling Access

  All calls made to Loudspeaker Paging and Code Calling equipment will be recorded.

- Manual Originating Line Service

  If an attendant establishes an outgoing call for a voice terminal, designated as a Manual Originating Line, the SMDR record for the call will be the same as for any attendant-assisted outgoing call. The calling voice terminal extension number is recorded as the calling number, and Condition Code 1 applies.

- Multiple Listed Directory Numbers (LDNs)

  If incoming call information is recorded, Condition Code 1 applies and the called number recorded for LDN calls is the extension number or trunk group access code to which the attendant completes the call. If the call terminates at the attendant console only, the called number recorded is 0, which is used to identify the attendants.

- Night Service—Night Station

  The extension number assigned to the attendants (0) is recorded as the dialed number.

- Night Service—Trunk Answer From Any Station

  The extension number assigned to the attendants (0) is recorded as the dialed number.

- Off-Premises Station

  SMDR data is recorded if the voice terminal is involved in an outgoing/incoming trunk call.

- Personal Central Office Line Groups (PCOLG)

  An outgoing PCOLG call will be recorded as a call from the originating extension number via the trunk group associated with the PCOLG. The answering voice terminal's primary extension is recorded as the called number if incoming calls are recorded.

- Private Network Access

  Private Network Access calls will be recorded.

- Remote Access

  Remote Access calls will be recorded if Remote Access is provided on a per trunk group basis.

If a voice terminal receives a call on an incoming trunk and then transfers the call, the voice terminal user who received the call will be recorded as the dialed number.

- Ringback Queuing

  Condition Code 8 is recorded for an outgoing call which is queued for a trunk before completion. The length of time the call is queued will not be recorded.

- Tandem Tie Trunk Switching

  The calling party on an incoming trunk can dial the SMDR account code. The calling number field in SMDR is the trunk access code for the incoming trunk group, the called number is the number dialed.

- Temporary Bridged Appearance

  An SMDR record is not affected by any second or subsequent voice terminal bridging a call.

- Tie Trunk Access

  Tie trunk calls will be recorded.

- Transfer

  If a user originates a call on an outgoing trunk and then transfers the call to another voice terminal, the originating voice terminal will be recorded as the calling party.

- Wide Area Telecommunications Service (WATS) and 800 Service

  All calls made on a WATS or 800 trunk group will be recorded.

STRAIGHTFORWARD OUTWARD COMPLETION

Description

Allows an attendant to complete an outgoing trunk call for a voice terminal user.

Considerations

The attendant determines which calls should be allowed and selects the trunk group used for the call.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

SYSTEM MEASUREMENTS

Description

Provides reports on trunk group usage, hunt group usage and efficiency, attendant group activity and efficiency, and security violations.

Group reports are all on a clock-hour basis. Reports available are yesterday's peak usage, today's peak usage, and last hour's usage. The peak usage hour is simply the clock-hour the group received the most usage. This hour can be different for different groups, such as hunt group 15 and hunt group 11. Conversely, today's peak hour usage may be the same as the last hour's usage.

Individual reports are available for each of the following:

- Trunk group—Yesterday's peak usage
- Trunk group—Today's peak usage
- Trunk group—Last hour's usage
- Hunt group—Yesterday's peak usage
- Hunt group—Today's peak usage
- Hunt group—Last hour's usage
- Attendant group—Yesterday's peak, today's peak, and last hour's usage are all one report

Security violations are attempts to access the system via an invalid login or Remote Access barrier code. These violations are accumulated from the time at which the count was reset. (The System Manager performs this function.)

All reports are on-demand reports. None are given automatically. Reports are available on the System Access Terminal (SAT) or a remote SAT. The reports can be printed if a printer is associated with the SAT.

Considerations

Reports provided by System Measurements contain data that is useful to determine group efficiency. Details of specific items on the reports, as well as guidelines to use the data provided, are given in the *AT&T System 75 Administration Manual—System Management*, 999-700-221IS.

Traffic measurements are automatically accumulated by the system and are available on demand. However, reports are not archived. If needed, reports must be requested periodically. Obtaining a printed copy can aid in maintaining a history of the system traffic.

Detailed information of each call handled by a specific trunk group, if required, must be provided by the Station Message Detail Recording (SMDR) feature. Processed SMDR data can also provide detailed information on trunk group usage. However, if individual call details are not required for bill-back or cost-allocation, System Measurements should be considered as the means to determine and maintain trunk group efficiency.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

TEMPORARY BRIDGED APPEARANCE

Description

Allows multi-appearance voice terminal users who share an extension number to bridge onto an existing call. Also allows a called party to bridge onto a call that redirects to coverage before the called party can answer it.

Considerations

A bridged call can occur with a Terminating Extension Group or a group sharing a Personal Central Office Line. A temporary bridged call occurs on a call that was answered (picked up) by a Call Pickup group member or that was redirected by a feature such as Call Coverage.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

Privacy—Manual Exclusion, when activated, prevents other users from bridging onto a call. A user who attempts to bridge onto a call with the Privacy—Manual Exclusion feature active will receive an intercept tone.

TERMINATING EXTENSION GROUP

Description

Allows an incoming call to alert (ring) as many as four voice terminals at one time. Any of the voice terminal users can answer the call.

Any voice terminal can be administered as a Terminating Extension Group (TEG) member; however, only a multi-appearance voice terminal can be assigned a TEG button with associated status lamp. The TEG button allows the user to select a TEG call appearance for answering or for bridging onto an existing call but not for call origination.

When an incoming call is answered by a TEG member, a Temporary Bridged Appearance is maintained at the multi-appearance voice terminals in the group. However, the Temporary Bridged Appearance is not visible on a call appearance. Any of the TEG members can bridge onto the call by pressing the TEG button, if assigned. For example, suppose an incoming call has been answered by a certain TEG member, and this TEG member does not have the needed information. If another member has the needed information, that member needs only to bridge onto the call to provide the information.

The Privacy—Manual Exclusion feature can be assigned to any or all of the multi-appearance voice terminals in a TEG. This allows the answering TEG member, by pressing the Exclusion button, to prohibit bridging by other group members. Pressing the button again reestablishes the bridging capability.

Considerations

A single-line voice terminal administered as a TEG member is alerted for a TEG call if it is idle or if Call Waiting is available.

The system allows for as many as 32 TEGs with up to 4 members each. A voice terminal user can be a member of more than one TEG, but can have only one TEG button for each group.

A TEG is established by associating the individual member's extension number with a TEG extension number. The members have call placing and receiving privileges for their individual extension numbers, as defined by the assigned Class of Restriction (COR). Each TEG is also assigned a COR. The group COR overrides an individual member's COR on calls to the group. Thus, the members could be Termination Restricted, but still receive TEG calls.

Calls to a TEG can be redirected to alternate answering positions whenever the Call Coverage feature is assigned and no group member is available to answer the call. If any member of a TEG is active on a TEG call, all subsequent TEG calls redirect to coverage. However, a TEG cannot serve as an alternate answering position. In other words, a TEG can have a Call Coverage path assigned, but cannot be a point in a Call Coverage path.

A Send Term button for the TEG can be assigned to any or all group members who have multi-appearance voice terminals. When the Send Term button is pressed, all calls to that TEG redirect to coverage. The associated status lamp lights on the activating voice terminal and all other voice terminals with a Send Term button. Any member with a Send Term button can deactivate Send Term by pressing the button. The Send Term status lamp then goes dark on all voice terminals. Incoming calls are again directed to the group.

Leave Word Calling messages can be stored for a TEG and can be retrieved by a member of the group, a covering user of the group, or a systemwide message retriever. The Voice Terminal Display feature and proper authorization must be assigned to the message retriever. Also, a Remote Message Waiting Indicator button can be assigned to a group member to provide a visual indication that a message has been stored for the group. One Remote Message Waiting Indicator is allowed per TEG.

The following items can be administered:

- Group number (from 1 to 32)
- Extension number for the group
- Group name (for display purposes)
- Call Coverage path number
- Group COR
- Up to four group member extension numbers

The following items can be administered to multi-appearance voice terminal TEG members:

- TEG button with associated status lamp.
- Exclusion button associated with the TEG extension number. (Keeps other group members from bridging onto an existing call.)
- Send Term button for the TEG extension number.
- Remote Message Waiting Indicator (one per TEG extension number).

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Automatic Callback

  This feature cannot be activated for a TEG.

- Call Forwarding All Calls

  A TEG call cannot be forwarded.

- Call Park

  A TEG call cannot be parked on the group extension number; however, a group member answering a call can park such a call on his or her own extension number.

- Direct Department Calling and Uniform Call Distribution

  A TEG cannot be a member of a Direct Department Calling or Uniform Call Distribution group.

- Call Coverage

  A TEG can have a call coverage path assigned, but cannot be a point in a coverage path. A Send Term button can be assigned to each multi-appearance voice terminal in the group. When activated at any TEG voice terminal, it is activated at all TEG voice terminals. Likewise, when deactivated at any TEG voice terminal, it is deactivated at all TEG voice terminals.

- Leave Word Calling

  Leave Word Calling messages can be stored for a TEG and can be retrieved by a member of the group, a covering user of the group, or a systemwide message retriever. The Voice Terminal Display feature and proper authorization must be assigned to the message retriever. Also, a Remote Message Waiting Indicator can be assigned to a group member to provide a visual indication that a message has been stored for the group. One indicator is allowed per TEG.

- Temporary Bridged Appearance

  At multi-appearance voice terminals in the TEG, a Temporary Bridged Appearance is maintained after a call is answered. This allows other members of the group to bridge onto the call.

  The Privacy—Manual Exclusion feature, when activated, prevents other TEG members from bridging onto a call. A TEG member who attempts to bridge onto a call with Privacy—Manual Exclusion activated will receive an intercept tone.

THROUGH DIALING

Description

Allows the attendant to select an outgoing trunk for a voice terminal user. The attendant then releases from the connection, and the user completes the call.

Considerations

The attendant can select a trunk by dialing an access code or by pressing a Trunk Group Select button. Also, the attendant can dial the Automatic Route Selection feature access code prior to releasing from the call.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

TIMED REMINDER

Description

Automatically alerts the attendant after a predetermined time for the following types of calls:

- Extended calls waiting to be answered or waiting to be connected to a busy single-line voice terminal.
- One-party incoming calls placed on hold on the console.
- Incoming calls answered by a voice terminal user, but which are unanswered after being transferred

The attendant can reenter the call and decide whether to terminate the call or permit the waiting to continue.

Considerations

The Timed Reminder intervals for calls waiting for connection and for calls placed on hold are assigned separately. Each interval can be from 10 seconds to 17 minutes.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Attendant Call Waiting

  An attendant-extended call to a busy single-line voice terminal will return to an attendant console if the Timed Reminder Interval expires before the call is answered, or redirects to coverage.

- Call Coverage

  After a voice terminal user transfers a call to an on-premises voice terminal, the call, if unanswered at the expiration of the Timed Reminder Interval, redirects to an attendant console. Redirection to an attendant occurs even if the call has redirected via Call Coverage or Call Forwarding from the transferred to voice terminal.

  An attendant-extended call redirects to coverage instead of returning to an attendant console, if the coverage criteria are met before the Timed Reminder Interval expires. However, unanswered calls return to a console at the expiration of the Timed Reminder Interval.

TOUCH-TONE CALLING SENDERIZED OPERATION

Description

Reduces the time necessary to place calls to distant locations equipped to receive touch-tone signals and allows end-to-end signaling to remote computer equipment.

The number dialed and end-to-end signaling digits from voice terminals and trunks are detected by the system and regenerated for transmission over outgoing trunks. The distant end associated with the trunk must be equipped to receive touch-tone signals.

Considerations

None.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

TOUCH-TONE DIALING

Description

Provides quick and easy pushbutton dialing. Touch-Tone Dialing is always provided with the system. In addition to the 0 through 9 buttons, the * and # buttons have special functions, such as forming a part of a feature access code. A distinctive tone is generated when each button is pressed.

If a distant switching system can accept only dial pulse signals, the system converts the touch-tone signals to the required dial pulses for transmission to the distant end.

Considerations

None.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

TRANSFER

Description

Allows voice terminal users to transfer trunk or internal calls to other voice terminals within the system without attendant assistance.

Single-line voice terminal users momentarily flash the switchhook or press the Recall button, dial the desired extension number, and hang up.

Multi-appearance voice terminal users press the Transfer button, dial the desired extension number, and press the Transfer button again.

Considerations

Multi-appearance voice terminals must have an idle call appearance in order to transfer a call.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

TRUNK GROUP BUSY/WARNING INDICATORS TO ATTENDANT

Description

Provides the attendant with a visual warning that the number of busy trunks in a group has reached an administered level. A visual indication is also provided when all trunks in a group are busy.

Considerations

The Trunk Group Busy and the Trunk Group Warning Indicators are particularly useful when the Attendant Control of Trunk Group Access feature is provided. The indicators alert the attendant that control of access to trunk groups is necessary.

Each attendant console has 12 Trunk Group Busy and 6 Trunk Group Warning Indicators.

If Trunk Group Select buttons are assigned for Loudspeaker Paging Access zones, Trunk Group Busy Indicators will provide a visual indication of the busy or idle status of the zones.

The warning threshold is administrable per trunk group and indicates the number of busy trunks in that trunk group.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

TRUNK-TO-TRUNK TRANSFER

Description

Allows the attendant or voice terminal user to connect an incoming or outgoing trunk call to an outgoing trunk.

Considerations

An attendant-assisted call connecting an outgoing trunk to an outgoing trunk must be held on the console. The system does not allow the attendant to release such a call. The attendant, however, can use the Forced Release button and disconnect the call.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

The Attendant Lockout feature does not function on Trunk-to-Trunk Transfer.

UNIFORM CALL DISTRIBUTION

Description

Allows direct inward access to an answering group other than the attendant even if the system does not have the Direct Inward Dialing (DID) feature.

One extension number is assigned to all voice terminals in a group or department, that is, to a set of voice terminals that serve the same function and require call distribution among the members of the group. Incoming calls to a Uniform Call Distribution (UCD) group can be internal or external. An incoming call will alert the voice terminal in the group that has not received a UCD group call for the longest period of time (the most-idle terminal). In other words, incoming calls to a UCD group extension number will be distributed evenly among the group members.

When UCD [or the similar Direct Department Calling (DDC) feature] is not provided, incoming Listed Directory Number (LDN) calls, foreign exchange calls, 800 service calls, and automatic tie trunk calls are normally directed to an attendant, who must extend the call. When UCD is provided, on an incoming trunk group, calls are automatically directed to the desired UCD group by the System 75 switch. Attendant intervention is not required.

A queue can be established for a UCD group. When all voice terminals within the group are active, the queue allows incoming calls to await an idle terminal. When a call enters the queue, a delay announcement interval is started. This interval (1 to 999 seconds) indicates how long a call will remain in queue before the call is connected to a recorded announcement. If Call Coverage is provided, the Don't Answer Interval (2 to 9 alerting cycles) also begins when the call enters the UCD group queue. After these intervals have begun, one of the following occurs:

- If the Coverage Don't Answer Interval expires before the delay announcement interval expires, the call is redirected to coverage. If no coverage point is available to handle the call, the call remains in queue and may then be connected to delay announcement.
- If the delay announcement interval expires before the Coverage Don't Answer interval, the call is connected to a delay recorded announcement, if available. If the announcement is already in use, the delay announcement interval is reset. This process continues until the call is answered, goes to coverage, is connected to a delay announcement, or the calling party hangs up.

Calls connected to a delay recorded announcement remain in queue while the announcement is heard by the caller. If the call has not been answered by the time the announcement is over, the call is connected to music (if provided) or there will be silence, as long as the call remains in queue.

The queue length can be set from 0 (no queue) to 35 calls. If queuing is not provided, if the queue is full, or if all group members have activated the Make Busy option (discussed later), calls to a busy group receive busy tone or redirect via the Call Coverage feature. Lamp indicators may be used to give a warning when the number of calls waiting in the queue reaches a predetermined limit (queue warning limit). The queue warning level can be 0 to 35, however, it cannot exceed the queue length. Although it is possible, the queue warning level should not be set to 0, as this would result in the indicator lamp lighting at all times.

When the queue warning level is reached, the indicator lamp lights and remains lighted until the calls waiting in queue are fewer than the queue warning level. A queue warning level lamp may be provided for each UCD group queue. The lamp can be installed at any location convenient for the group.

As an example of queue warning level and delay announcement operation, assume that a call is incoming to a UCD group with the following parameters.

- Queue length is 10 calls.
- Queue warning level is 5 calls.
- Recorded announcement delay is 20 seconds.

Also assume the following:

- All UCD group voice terminals are busy.
- The call is the fifth call in the queue.

Since all voice terminals in the UCD group are busy, the incoming call enters the queue. The incoming call, being the fifth call in the queue, causes the queue warning level to be reached. This causes the queue warning level lamp to light.

From the indicator lamp, the UCD group members know the queue warning level has been reached and try to complete their present calls. Meanwhile, the incoming call has been in the queue for 20 seconds and hears the delay recorded announcement. The caller may decide to hang up or may decide to remain in the queue. Assume the caller remains in the queue. When a UCD group member becomes idle, the longest queued call is directed to that voice terminal. The queue warning level lamp may or may not be lighted at that time depending on the number of other calls that have been queued. Also, the first four calls in the queue will have heard the delay announcement after being queued for 20 seconds. The queue warning level and delay announcement capabilities are independent of each other.

Considerations

UCD is particularly useful when the answering group assigned receives a high volume of incoming calls. Call completion time is minimized and attendant assistance is not required. This feature can also minimize the use of DID trunks.

Calls incoming on a non-DID trunk group can route to a UCD group instead of to an attendant. Calls incoming on any non-DID trunk group can have only one primary destination; therefore, the trunk group must be dedicated to the UCD group.

If DID is provided and the UCD group extension number is within the range of extension numbers that can be dialed directly, then the group can be called the same as any voice terminal.

Any voice terminal can be administered to one or more UCD groups. Each voice terminal in a group also has its own unique extension number and can be alerted individually. Multi-appearance voice terminals can have an assigned status lamp that identifies an incoming UCD call. However, the voice terminal must be idle (not active on any call appearance) before a group call will be directed to the terminal.

If UCD and DDC groups are both used in the system, the number of combined groups and the number of voice terminals per group are determined by the size of the system and call traffic requirements. A maximum of 32 groups with up to 32 members per group can be provided. The system maximum, however, is 448 group members.

The UCD group queue is optional on an individual group basis, and the maximum queue size is 35. If a queue is not assigned or if an assigned queue is full, incoming calls can be redirected through the Call Coverage feature.

An optional queue warning level lamp can be provided for each UCD group. The warning level can be administered for 0 to 35 calls waiting in queue. The lamp lights when the warning level is reached. Each queue warning level lamp requires a port on an Analog Line circuit pack. The lamp can be placed at a location convenient to the group.

The system provides access to music sources and delay announcement equipment, if used.

Each System 75 can contain up to ten different recorded announcements. Each group queue can be assigned one of these announcements as a delay announcement. A delay announcement can be shared among the UCD groups, DDC groups, or a combination of these groups. However, only one caller can be connected to an announcement at any one time. Callers are always connected at the beginning of the announcement. More efficient use of the announcements is realized if the announcements are brief.

If a delay announcement is used, answer supervision is sent to the distant office when the caller is connected to the announcement. Charging for the call, if applicable, begins when answer supervision is returned.

Multi-appearance voice terminals can receive only one UCD call at a time. A voice terminal is idle for a UCD call only if all call appearances are idle.

A Make Busy option can be administered for the system. When a voice terminal user dials the Make Busy activation code followed by the UCD group extension number or presses the Make Busy button, the terminal appears busy to the UCD group. This effectively removes the terminal from the group until the user dials the Make Busy cancellation code or presses the button again. The Make Busy button can be assigned to multi-appearance voice terminals only.

The last available member of a DDC group cannot activate the Make Busy option if any calls are remaining in the queue. An attempt by the last available group member to activate the Make Busy option results in the following:

- New calls to the DDC group either receive busy tone or redirect to coverage.
- Calls already in the queue continue to route to the last available voice terminal until the queue is empty.
- At the last available voice terminal, the status lamp associated with the Make Busy button, if provided, flashes until the queue is empty. When no more calls remain in the queue, Make Busy is activated and the status lamp, if provided, lights steadily. (The same sequence applies when Make Busy is dial activated instead of button activated, except there is no status lamp.)

Leave Word Calling messages can be stored for a UCD group and can be retrieved by a member of the UCD group, a covering user of the group, or a systemwide message retriever. The Voice Terminal Display feature and proper authorization must be assigned to the message retriever. Also, a Remote Message Waiting Indicator can be assigned to a group member to provide a visual indication that a message has been stored for the group. One Remote Message Waiting Indicator is allowed per UCD group. The status lamp associated with this button informs the user that at least one message has been left for the group.

Hardware and Software Requirements

Each queue warning level lamp requires one port on a TN742 Analog Line circuit pack. A 21C-49 indicator lamp may be used as a queue warning level lamp. This lamp is approximately 2 inches in diameter and has a clear beehive lens. The lamp operates on ringing voltage and can be mounted at a location convenient to the group.

Each delay announcement requires announcement equipment and one port on a TN742 Analog Line circuit pack. If music is to be heard after the delay announcement, a music source and a port on a TN763 Auxiliary Trunk circuit pack is required. Announcement equipment and music sources are not provided by the system.

No additional software is required.

Interactions

- Attendant Call Waiting

  An attendant can originate or extend a call to a UCD group. Attendant Call Waiting cannot be used on such calls. However, such calls can enter the group queue, if provided.

- Call Coverage

  Calls can redirect to or from a UCD group.

  When the Send All Calls button is pressed, the associated voice terminal appears busy to the UCD group until the button is pressed again. Activating Send All Calls also lights the lamp associated with the Make Busy button, if provided. Likewise, deactivating the Send All Calls function deactivates the Make Busy function. However, activating or deactivating the Make Busy function does not activate or deactivate the Send All Calls function.

  For a call to a UCD group to be directed to Call Coverage, each voice terminal in the group must be active on at least one call appearance and the queue, if there is one, must be full. If the queue is not full, a call will enter the queue when no voice terminal is available. Queued calls remain in queue for a time interval equal to the Coverage Don't Answer Interval before redirecting to coverage. If any voice terminal in the group is idle, the call directs to that voice terminal.

- Call Forwarding All Calls

  When activated, the activating voice terminal appears busy to the UCD group.

- Multi-Appearance Preselection and Preference

  All assigned call appearances must be idle before a UCD group call is directed to a voice terminal.

- Music-on-Hold Access

  A call placed in a UCD group queue can receive a delay announcement followed by music.

- Priority Calling

  A priority call directed to a UCD group is treated the same as a non-priority call, except that the distinctive 3-burst alerting is heard.

- Terminating Extension Group

  A Terminating Extension Group cannot be a member of a UCD group.

- Voice Terminal Display

  On calls dialed directly to a UCD group extension number, the UCD group's identity is displayed at the calling extension.

UNIFORM NUMBERING

Description

Allows the same extension number to be used for incoming Private Network Access and Public Network Access calls.

Considerations

None.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

None.

VOICE TERMINAL DISPLAY

Description

Provides multi-appearance voice terminal users with updated call and message information. This information is displayed on a 40-character alphanumeric display module or on the screen of a 515 Business Communications Terminal. The information displayed depends upon the display mode selected by the user.

Several modes can be assigned to buttons and then selected by pressing the assigned button. All buttons are located on the display module or voice terminal. All buttons are administrable.

- Normal Mode

  Displays call-related information for the active call appearance. This display includes information identifying the call appearance, calling or called party, and calling or called number.

- Inspect Mode

  Displays call-related information for an incoming call when the user is active on a different call appearance. This button is pressed when the user is active on one call appearance and receives a call on another appearance.

- Stored Number Mode

  Displays the last number the user dialed (Last Number Dialed feature), the number stored in an Abbreviated Dialing button administered to the voice terminal, a number stored in an Abbreviated Dialing list, or a number assigned to a button administered through the Facility Busy Indication feature.

- Date and Time Mode

  Displays the current date and time of day.

- Elapsed Time Mode

  Displays elapsed time in hours, minutes, and seconds. The timing starts or stops when the button is pressed. This button can be pressed at any time.

- Integrated Directory Mode

  Turns off the touch-tone signals and allows the touch-tone buttons to be used to key in the name of a system user. After a name is keyed in, the display shows that name and associated extension number. (Refer to the Integrated Directory feature for complete details.)

- Message Retrieval Mode

  Retrieves messages for voice terminal users. If no messages are stored, display shows NO MESSAGES.

- Coverage Message Retrieval Mode

  Retrieves messages for voice terminal users who do not have a display module assigned to their voice terminal. Retrieval permission must be administered for a user to retrieve another user's messages. Messages can be retrieved at any time. The retriever does not need to lift the receiver to retrieve messages. Also, messages can be retrieved even if the retriever is active on a call.

The Message Retrieval, Coverage Message Retrieval, or Integrated Directory buttons have three other associated buttons:

- Next Message

  Retrieves the next message or displays END OF FILE, PUSH Next TO REPEAT when in the retrieval mode. Displays the next name in the alphabetical listing when in the Integrated Directory mode. This button must be assigned when a Retrieval button is assigned.

- Delete

  Deletes the currently displayed message. This button must be assigned when a Retrieval button is assigned.

- Return Call

  Automatically returns the call requested by the currently displayed message or the currently displayed name and extension number.

The system provides the following call-related information:

- Call Appearance Identification

  The call appearance buttons are designated on the display by a lower-case letter; for example, a, b, and c. The display shows a= for a call incoming on the first call appearance button, b= for a call incoming on the second call appearance button, and so on.

- Calling Party Identification

  When the incoming call is internal, the identification consists of the caller's name as administered on the individual voice terminal form. If the name field is blank, the identification consists of the caller's extension number. When the incoming call is external, the display shows the name of the trunk group as administered on the trunk group form.

- Called Party Identification

  When a call is placed to an internal user, the display shows the called party's name or the called extension number if the name is not established in the system. When a call is placed using an outgoing trunk, the display shows the name of the trunk group as administered on the trunk group form.

- Call Purpose

  This identifies the reason for an incoming call or a redirected call. (A normal incoming call is not identified by a call purpose.) The following call purpose identifiers can be displayed:

  f—Call Forwarding—Indicates that another user has forwarded calls to this voice terminal.

  s—Send All Calls—Indicates that the called user is temporarily sending all calls to coverage, and that the call has been redirected to this voice terminal.

  c—Go to Cover—Indicates that the calling user has sent the call to coverage, and that the call has been redirected to this voice terminal.

  b—Busy—Indicates that the called user is active on a call, and that the call has been redirected to this voice terminal.

  d—Don't Answer—Indicates that the called user is not available, and that the call has been redirected to this voice terminal.

  callback—Indicates that the call is an Automatic Callback call from the system.

  icom—Indicates that the incoming call is an Intercom call.

  park—Indicates that the user parked a call.

  pickup—Indicates that the user answered a Call Pickup group member's call.

  priority—Indicates that the incoming call has priority status.

Some typical displays are as follows.

- Internal call:

```
a=3602
```

or

```
a=        TOM  BROWN
```

- Outgoing trunk call:

```
b=87843541
```

Where 8 is the trunk access code and 784-3541 is the number dialed,

then

```
b=    OUTSIDE  CALL
```

or

```
b=              WATS
```

- Incoming trunk call:

```
a=    OUTSIDE CALL
```

- Conference call:

```
b=    CONFERENCE 4
```

Where 4 is the number of conferees. The number does not include the conference call originator.

- Internal call redirected to coverage:

```
b=         3174  to  3077              c
```

or

```
b=     BOB SMITH  to  JOYCE THOMAS     c
```

Where c indicates that Go to Cover was activated by the calling voice terminal user.

- Incoming trunk call redirected to coverage:

```
a= OUTSIDE CALL  to  DON SMITH         s
```

Where s indicates that Send All Calls was activated by the called voice terminal user.

- Message retrieval:

```
IN PROGRESS
```

then

```
MESSAGES FOR BETTY R. SIMS
```

then

```
   JOE JONES 10/16 11:40a 2 CALL 3124
```

This message means that Joe Jones called Betty Sims the morning of October 16. The second message was stored at 11:40. Joe wants Betty to call his extension number, 3124.

- Integrated Directory mode:

```
   CARTER, ANN             3408            3
```

This display shows the name and extension number as administered in the system. The 3 indicates that three buttons were pressed to reach this particular display.

Considerations

Certain voice terminals and the attendant group can be designated for systemwide message retrieval. Users of these voice terminals or consoles can retrieve Leave Word Calling and Call Coverage messages for other voice terminal users including Direct Department Calling groups, Uniform Call Distribution groups, Personal Central Office Line groups, and Terminating Extension Groups. Selected users cannot retrieve messages for other selected users. Up to ten voice terminals, or up to nine voice terminals and the attendant group, can be designated for systemwide message retrieval. Systemwide retrieving voice terminals or consoles are assigned when the system is implemented.

If the following conditions are met, messages for a voice terminal user can be retrieved at selected terminals or any attendant console:

- The retriever must be in the user's Call Coverage path.
- Permission to retrieve messages must be assigned for the user's voice terminal.

If permission is granted, any voice terminal with a display module or the attendant group in the user's Call Coverage path can retrieve messages for that user.

Up to 62 display modules can be provided per system.

When all messages have been displayed and deleted for an extension number, the Message lamp on the voice terminal and any associated Remote Message Waiting Indicator, if assigned, go dark.

The display module used with voice terminals is similar to the attendant console display. However, the display module has an On-Off button, and can be turned off when not in use. The display module can be used only with a Model 7405D voice terminal.

The following buttons can be administered:

- Normal
- Inspect
- Stored Number
- Date and Time
- Elapsed Time
- Integrated Directory
- Message Retrieval
- Coverage Message Retrieval
- Next Message (must be assigned with either Retrieval button)
- Delete (must be assigned with either Retrieval button)
- Return Call (optional with either Retrieval button or the Integrated Directory button)

Hardware and Software Requirements

Requires a Model 7405D voice terminal, a Digital Display Module, and one port on a TN754 Digital Line circuit pack. No additional software is required.

Interactions

None.

DATA MANAGEMENT

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DATA MANAGEMENT OVERVIEW

System 75 is a private digital switching system that permits connections with a variety of data equipment. Data terminals, printers, graphics and facsimile equipment, computers, and virtually all data equipment with an Electronic Industries Association (EIA) 25-pin RS-232C interface can be connected to the switch. The physical connection can be through a digital data module or analog modem.

The family of data modules includes a Processor Data Module (PDM), a Digital Terminal Data Module (DTDM), and a Trunk Data Module. The data modules are generally more versatile than modems, operate at faster data rates, and provide additional features. The 515 Business Communications Terminal (BCT) consolidates the capabilities of a data terminal, data module, 7403D voice terminal, and a Digital Display Module into a single unit. The 515 BCT provides an all-digital interface with the system. Through its built-in RS-232C interface, the 515 BCT can connect to other equipment.

In summary, System 75 supports data calls between digital-to-digital, digital-to-analog, analog-to-digital, and analog-to-analog terminal devices. Terminal devices are the points where a user can access the system. These are called endpoints, and can be either digital or analog interfaces. Digital data endpoints are data modules and associated equipment, 515 BCTs, data channels (used for remote System Access Terminals [SATs] and Station Message Detail Recording [SMDR]), and/or the Applications Processor (AP) interface (internal to the system). Analog data endpoints are modems (or acoustic coupled modems) and associated equipment connected to the system through analog lines or trunks.

System 75 supports a Digital Communications Protocol (DCP). This protocol provides framing, control, and signaling for each of two information channels. Only one channel is used for voice-only or data-only applications. Both channels are used for simultaneous voice and data transmission. Simultaneous voice and data information can be transmitted on calls to or from a 515 BCT and a 7403D or 7405D voice terminal with a DTDM. Other configurations are either voice-only or data-only.

A typical data communications configuration for the system is shown in Figure 6-1.

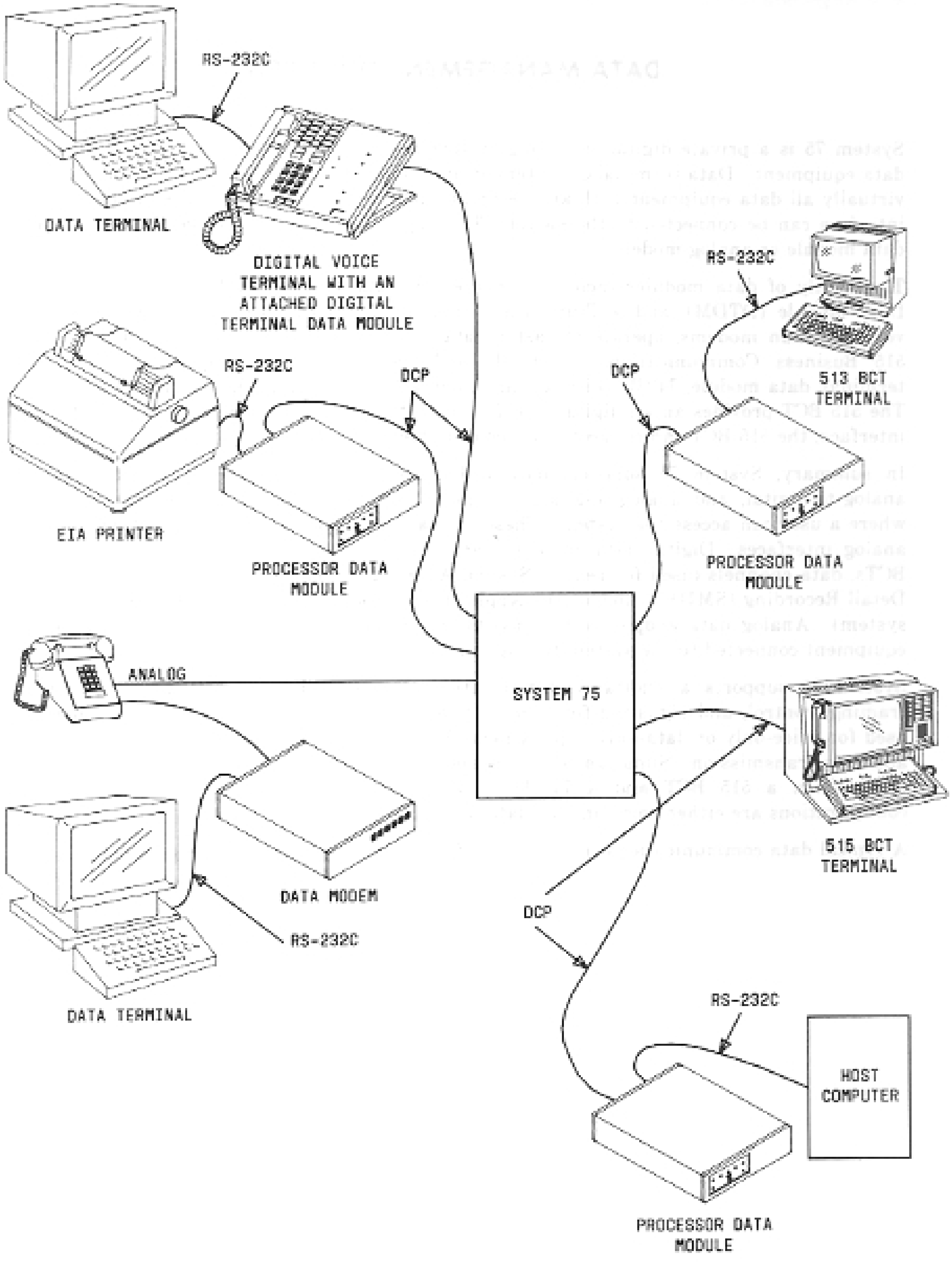

Figure 6-1. System 75 Data Communications Configuration

DATA NETWORKING

Data networking connects two or more data endpoints. System 75 is a highly reliable centralized node that provides switched access between the endpoints.

Switched access allows one terminal to connect to any number of devices. Therefore, more effective use of data equipment is obtained than with dedicated (hard-wired) links. Switched access also reduces the need for duplicated (dedicated) equipment.

The system uses twisted-pair standard building wiring and 8-pin modular wall jacks. Each wall jack is a single outlet that can handle simultaneous voice and data information.

The digital switch, data modules, DCP, twisted-pair wiring, modular wall jacks, and switched data features give the system its unique capabilities. These capabilities merge the business office data processing and telecommunications functions into a single system.

Generally, data networks are either local area networks, extended networks, or combinations of both. The two networks are defined as follows:

- **Local Area Networks**

  System 75 provides this capability by connecting communication devices that are physically located within a local-area or campus-like environment. These include conventional, semi-intelligent, and intelligent data terminals, personal computers, host computers, and virtually any device with the proper communications interface.

  The centralized network provides circuit switched paths using twisted-pair building cable that extends to the endpoints. Since the business office equipment can access multiple data systems, the data equipment and applications can be used more productively. The system also provides several data-related features that are easy to use and that contribute toward expedient use of the system and its networking capabilities.

- **Extended Networks**

  Extended networks mainly provide connections between the system and other distant switching nodes, including remote access facilities. Through use of remote access facilities, a local terminal can access remote host computers. Also, remote terminals can access either local computer facilities or other remote computer facilities. Extended networks are constructed of analog or digital facilities and can be either public or private. Typical networking configurations are shown in Figure 6-2.

  Public networks include:

  - Local central office (CO) switching extended through direct distance dialing
  - Foreign exchange (FX) central office trunking
  - Wide Area Telecommunications Service (WATS)

Private networks include:

- Tandem tie trunk networks
- Enhanced Private Switching Communications Service (EPSCS)
- Electronic Tandem Networks (ETN)
- DATAPHONE* Digital Service
- Private line (PL) networks

* Service Mark of AT&T.

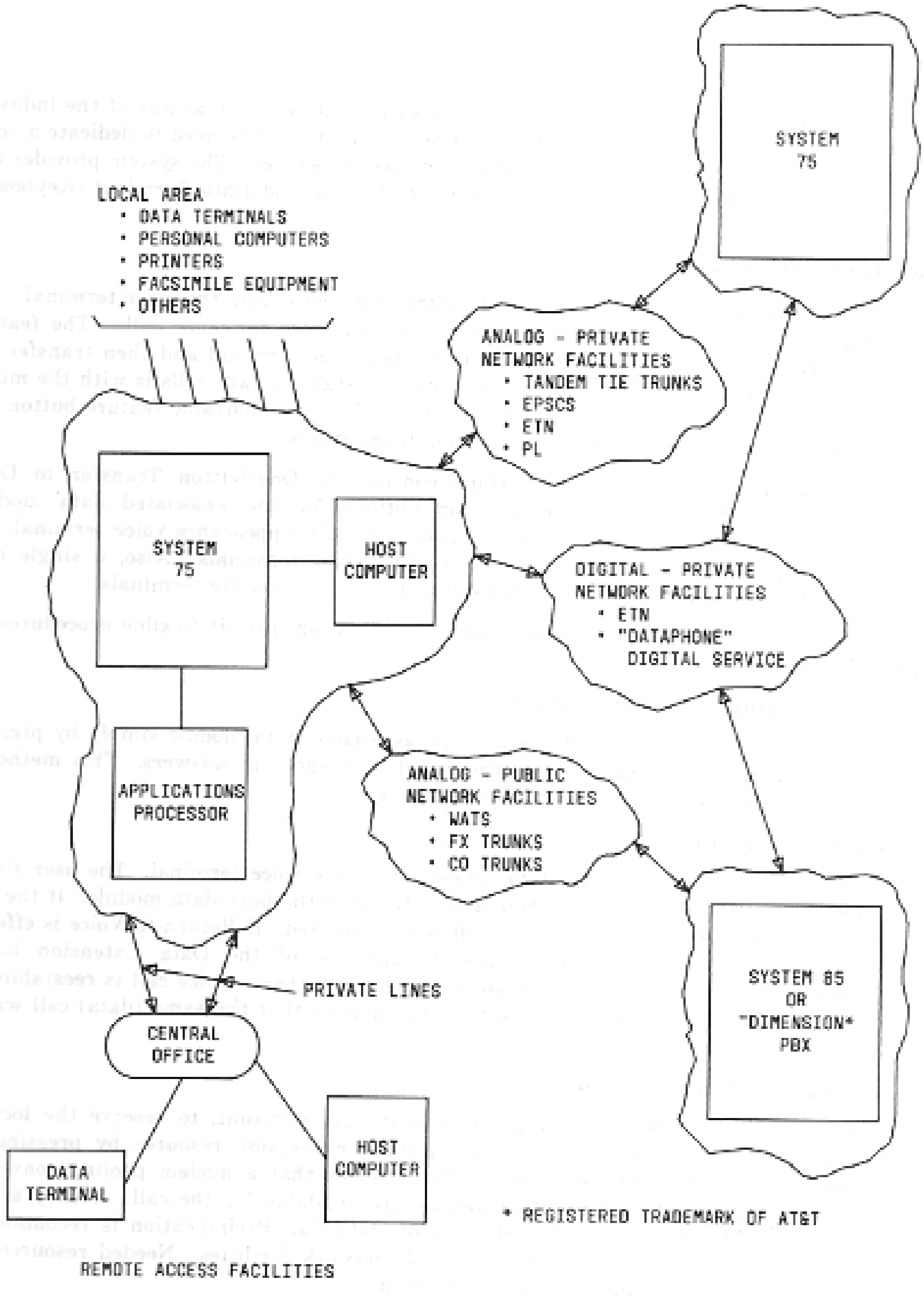

Figure 6-2. System 75 Networking Configurations

DATA CALL SETUP

Description

Enables system users to establish data calls. Data calls can be made at any of the industry standard data rates up to 19.2 Kbps. Data Call Setup eliminates the need to dedicate a voice terminal to a data call, although such arrangements are supported. The system provides two methods for establishing data calls, Voice Terminal Dialing and Data Terminal (Keyboard) Dialing.

Voice Terminal Dialing

Allows voice terminal users to originate and control data calls from the voice terminal. The Transfer feature functions the same for data calls as it does for voice calls. The feature permits a user to set up a call using any unrestricted voice terminal and then transfer the call to a data endpoint. However, the primary way to establish data calls is with the multi-appearance voice terminal Data Extension button(s). Any administrable feature button can be assigned as a Data Extension button in system translations.

The voice terminal Data Extension buttons control the One-Button Transfer to Data, Return-to-Voice, and Data Call Preindication options for the associated data module. Multiple Data Extension buttons can be assigned to a multi-appearance voice terminal, and that voice terminal can set up data calls for other data terminals. Also, a single data endpoint can be controlled from Data Extension buttons on four voice terminals.

The following attributes, either singularly or in combination, permit flexible procedures for establishing data calls:

- **One-Button Transfer to Data**

  Allows a user to transfer the call to the associated data module simply by pressing the Data Extension button after the called data endpoint answers. This method is recommended for voice terminal data call setup.

- **Return-to-Voice**

  Allows a user to return the data connection to the voice terminal. The user simply presses the Data Extension button associated with the busy data module. If the user hangs up following the return, the call is disconnected. If Return-to-Voice is effected by two voice terminal users, each through use of the Data Extension button associated with the two data endpoints of the call, then a voice call is reestablished. Return of a data call to the voice terminal implies that the same (data) call will be continued in the voice mode.

- **Data Call Preindication**

  Allows the user, before dialing the distant data endpoint, to reserve the local or near-end data module and a modem pooling conversion resource by pressing the associated Data Extension button. This ensures that a modem pooling conversion resource, if needed, and the data module are available for the call. If they are not available, Preindication is denied. Use of Data Call Preindication is recommended when establishing data calls that use toll network facilities. Needed resources are reserved before any toll charges are incurred.

Data Terminal (Keyboard) Dialing

Allows a user to set up and break down data calls directly from a data terminal. A voice terminal is not needed. The voice terminal functions of switchhook, touch-tone or rotary dialing, and the audible call progress tones are replaced with keyboard commands and text prompts known as call progress messages. Data terminal users can receive and respond to call progress messages while a call is being placed (digits are being manually entered). Table 6-A lists the call progress messages.

TABLE 6-A. Call Progress Messages for Keyboard Dialing

Displayed Message	Application	Meaning
DIAL:	Placing a call	Equivalent to dial tone. Enter the desired number or feature access code followed by a carriage return or a line feed.
TRY AGAIN	Placing a call	Equivalent to reorder tone. System facilities are currently not available.
BUSY	Placing a call	Equivalent to busy tone. Called number is in use.
DENIED	Placing a call	Equivalent to intercept tone. Call cannot be placed as dialed.
CHECK OPTIONS	Placing a call	Notifies calling terminal that data module options are incompatible.
CONFIRMED	Activating or deactivating a feature	Equivalent to confirmation tone. Feature request is accepted.
RINGING	Placing a call	Equivalent to alerting tone. Called terminal (far-end) is alerting.
INCOMING CALL-	Receiving a call	Equivalent to alerting.
PLEASE ANS-	Receiving a call	Being alerted due to 1-button transfer. Call originated from a voice terminal. Far-end data module has answered. Originating terminal user has transferred call to near-end data module using One-Button Transfer to Data (One-Button Transfer Alert).
FORWARDED	Call has been forwarded	Equivalent to redirection notification signal. Called terminal has activated Call Forwarding All Calls and a call has been forwarded.
ANSWERED	Placing or receiving a call	Notifies calling and called terminal users that call has been answered.
DISCONNECTED	Call is terminated	Call or call attempt is disconnected from system.
-TRANSFER	Call is transferred to voice	Notifies calling terminal when Data Call Transfer to Voice occurs.
WAIT	Placing a call	Notifies calling terminal that a trunk is seized for an outgoing call (Pause Dialing only).

Considerations

To originate and disconnect a call using Data Terminal Dialing, the user presses the BREAK key on the terminal. (This is equivalent to a voice terminal user lifting the receiver [call origination] or hanging up [call disconnect].) If the terminal being used does not generate a 2-second continuous break signal, the user can either turn the terminal off, then back on, or press the ORIGINATE/DISCONNECT button on the data module.

In addition to Data Terminal Dialing and Voice Terminal Dialing, the system accepts calls from other devices, such as Automatic Calling Units. Computer-generated dialing requires special programming that must be provided by the client. Computer-generated dialing must follow the Data Terminal Dialing protocol.

To reserve a modem pooling conversion resource, the user can dial a Data Origination feature access code assigned in the system. This manual reservation, however, is only needed on a call originated by a local analog data endpoint to a digital data endpoint.

Data Call Preindication is in effect until the associated Data Extension button is pressed again (with or without a connected call). There is no time-out.

The following items can be administered:

- Data Origination dial access code
- Data Extension buttons (per multi-appearance voice terminal)

The number of assigned Data Extension buttons per voice terminal is not limited. However, only four voice terminals can be assigned buttons that access the same data module.

When multiple Data Extension buttons control a single data module, the control is shared except for Data Call Preindication. The module is reserved for the preindicating user while Preindication is in effect. After a data call is established, any of the users with an associated Data Extension button could disconnect the call by selecting an idle call appearance, lifting the receiver, pressing Data Extension, and hanging up.

Hardware and Software Requirements

Data Call Setup does not require hardware, but is a means of using equipment to establish data calls. However, for completeness, requirements for data modules, 515 BCTs, and modems are given below.

- ***Data Modules:*** Each data module requires one port on a Digital Line circuit pack. [A Digital Terminal Data Module (DTDM) shares the port with the associated voice terminal.]
- ***515 BCTs:*** Each 515 BCT requires one port on a Digital Line circuit pack for shared use of voice and data.
- ***Modems:*** Each modem requires one port on an Analog Line circuit pack. (Administration designates the modem as a 2500-series voice terminal and an extension number is assigned. A modem is connected to the port instead of a voice terminal. Access is through the assigned extension number.)
- No additional software is required.

Interactions

- Abbreviated Dialing

  This feature can be used on calls to data endpoints.

- Call Forwarding All Calls

  When a data terminal is associated with a data module, calls incoming to the module can be forwarded. That is, calls can be redirected to another data module (and associated data terminal). This feature is activated using Data Terminal (Keyboard) Dialing.

- Modem Pooling

  This feature is available on data calls, either by feature access code dialing or automatically when the system can ascertain the need for a conversion resource and one is available. Data Call Preindication can be used to verify availability. Preindication is not needed to insert the conversion resource into the connection. The system does this.

- Uniform Call Distribution (UCD)

  UCD can provide a pool of data modules or modems for answering calls to facilities, such as computer ports, connected to the data modules or modems. Similarly, UCD can provide a pool of 515 BCTs.

DATA-ONLY OFF-PREMISES EXTENSIONS

Description

Allows users to establish data calls involving data terminal or data communications equipment that is located remotely from the System 75 site using DATAPHONE Digital Service or other private line data facilities. A Data-Only Off-Premises Extension uses a Trunk Data Module located on-premises. The communication with the remote data terminal or data communications equipment is accomplished through the private line or DATAPHONE Digital Service facility linking the on-premises Trunk Data Module and the remote data terminal or data communications equipment.

The Trunk Data Module and data terminal or data communications equipment constitute a digital data endpoint. Data calls to this type of data endpoint can be placed using Voice Terminal Dialing or Keyboard Dialing. Data calls can be originated from the remote data terminal using Keyboard Dialing only. If computer-generated dialing is used on calls, it must follow the Keyboard Dialing protocol. (Computer-generated dialing requires special programming that must be provided by the client.)

Considerations

Multiplexing data communication signals can be engineered with System 75. Multiplexing data module signals requires the use of data modems when private line service is used, or data service units when a digital data system is used.

Data-Only Off-Premises Extensions provide digital data endpoints located off-premises through a Trunk Data Module located on-premises. Communications to or from this Trunk Data Module (and the associated off-premises equipment) must be through an on-premises Processor Data Module or Digital Terminal Data Module. Communications between Trunk Data Modules are not supported. Likewise, Modem Pooling, which is conceptually similar to a Trunk Data Module, cannot be used on calls to or from a Data-Only Off-Premises Extension.

Hardware and Software Requirements

Requires a Trunk Data Module and one port on a Digital Line circuit pack. No additional software is required.

Interactions

- Voice Terminal Dialing

  A multi-appearance voice terminal may have a Data Extension button associated with the Trunk Data Module used for a Data-Only Off-Premises Extension. The on-premises voice terminal user and the remote data terminal user (if there is one) share control of the data module. Actions of the user at the voice terminal may affect the remote user.

  - One-Button Transfer to Data

    The voice terminal user can transfer a call to the Data-Only Off-Premises Extension. The Call in Progress lamp on the data module lights during an established data call.

— Return-to-Voice

If a data call has already been established using a Data Extension button, the voice terminal user may press this busy Data Extension button to transfer the call to the voice terminal. The data module associated with the Data Extension button is disconnected from the call. The Call in Progress lamp on the data module goes dark.

— Data Call Preindication

The multi-appearance voice terminal user presses the idle associated Data Extension button to reserve the data module. The data module is then busy to all users except the Preindicating user, including the remote user. When the data module is reserved, the lamp associated with the Data Extension button winks at the preindicator's voice terminal and lights at any other associated voice terminals. A remote data terminal user receives the message BUSY when attempting to originate a call.

DATA PRIVACY

Description

Protects calls from being disturbed by any of the system's overriding or alerting features. Data Privacy denies the system the ability to gain access to, or to superimpose tones onto, the protected call. To activate this feature, the user dials the activation code at the beginning of the call.

Considerations

Connections involving a digital data endpoint (data module) are automatically protected from receiving system-generated tones. In this case, the Data Privacy feature is not needed.

Data Privacy, when activated, applies to both voice and data calls. The feature can be activated on Remote Access calls, but not on other incoming trunk calls. Data Privacy is canceled if the call is transferred, added to a conference call, or disconnected by the activating user. Data Privacy can be activated on calls originated from attendant consoles.

Data Privacy is administered individually to voice terminals through the assigned Class of Service.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Attendant Call Waiting and Call Waiting Termination

  If Data Privacy is activated, Call Waiting is denied.

- Priority Calls

  If Data Privacy is activated, Priority Calls to the activating extension number are denied.

DATA RESTRICTION

Description

Protects calls from being disturbed by any of the system's overriding or alerting features. Data Restriction denies the system the ability to gain access to, or to superimpose tones onto, the protected call. This feature is administered at the system level to selected analog and multi-appearance voice terminals and trunk groups. Once administered, the feature is active on all calls to or from the associated terminal or trunk group.

Considerations

Data Restriction applies to both voice and data calls.

Connections involving a digital data endpoint (data module) are automatically protected from receiving system-generated tones. In this case, the Data Restriction feature is not needed.

Data Restriction cannot be assigned to attendant consoles.

Hardware and Software Requirements

No additional hardware or software is required.

Interactions

- Attendant Call Waiting and Call Waiting Termination

  If Data Restriction is activated, Call Waiting is denied.

- Priority Calls

  Priority Calls to a data-restricted extension number are denied.

MODEM POOLING

Description

Provides switched connections between digital data endpoints (data modules) and analog data endpoints (modems). The analog data endpoint can be either a trunk or line circuit. Data transmission between a digital endpoint and an analog endpoint requires a conversion resource. The conversion resource translates data between the Digital Communications Protocol (DCP), used by the data modules, and the modulated signals used by analog modems.

The Modem Pooling feature provides a pool of conversion resources to perform these translations. The conversion resource (pool) is an integral part (circuit pack) of the system. Each Pooled Modem circuit pack provides two conversion resources that are independently known as pooled modems.

Each pooled modem is functionally similar to a Trunk Data Module connected in series with a 212AR modem. Conversion resources cannot be directly dialed by the user. The system automatically seizes a resource from the pool when it has been identified that one is needed. Data calls originated from a local analog data endpoint to a digital data endpoint require that the user indicate the need for a conversion resource. This is done by dialing the Data Origination access code before dialing the digital data endpoint. In all other cases, the system can detect the need for a conversion resource.

Once it has been determined that a pooled modem is needed, the system queries the data module that will be in the call to determine if its options are compatible with those supported by the modem pool. If the data module options are not compatible, the originating user receives intercept treatment. At data connection time, the user receives reorder tone if a pooled modem is not available. The data call will be automatically disconnected within 15 seconds (handshake time-out) if the data call is not successfully established.

Considerations

Data Call Preindication is recommended for off-premises data calls involving toll charges.

On data calls between a data module and an analog facility, Return-to-Voice releases the pooled modem and returns it to the resource pool. The voice user is then connected to the analog facility.

Hardware and Software Requirements

The following modem options are supported by the pool:

- Receiver Responds to Remote Loop
- Loss of Carrier Disconnect
- Send Space Disconnect
- Receive Space Disconnect
- CF-CB Common

System 75 supports up to a maximum of 32 modem pool conversion resources (16 Pooled Modem circuit packs, each containing 2 conversion resources). All conversion resources have the same characteristics and operate full-duplex, either asynchronously (at 0-300, 300, and 1200 bps) or synchronously at 1200 bps. The mode of operation is automatically determined during call setup.

No additional software is required.

Interactions

- Data-Only Off-Premises Extensions

  Modem Pooling is not available on calls to or from a Data-Only Off-Premises Extension.

- Data Call Setup

  Data calls to a Trunk Data Module cannot use Modem Pooling.

UNIFORM CALL DISTRIBUTION (UCD)

Description

Provides switched access to a group of data modules or modems by either lines or trunks. Access to a group of like resources, such as modems, is through a single group extension number, which minimizes the dialing of a busy resource. The system selects the resource that has not received a UCD group call for the longest period of time (the most-idle resource) and provides the connection. In other words, incoming calls to a UCD group extension number are evenly distributed among the group members.

A queue can be established for a UCD group. When all members of a group are busy, the queue allows the incoming call to await an idle resource.

Considerations

Calls incoming on any non-Direct Inward Dialing (DID) trunk group can have only one primary destination. Therefore, any non-DID trunk groups routing calls to a UCD group must be dedicated. However, more than one non-DID trunk group can route to the same UCD group. Likewise, that same UCD group can also receive DID and internal calls.

Resources in a UCD group should be of the same type and serve the same function. Either data modules or modems can be used in a UCD group, not a mixture of the two, and the group must be dedicated to a specific, intended use.

Since any member of a UCD group can be used on a given call, option settings should be the same for all group members. This minimizes call setup failures because of incompatible options between the originating data module or modem and the UCD group data module or modem selected for the call.

A Data Extension button can be used to access the associated data module, even if the module is in a UCD group.

Each UCD group and each individual UCD member is assigned a Class of Restriction (COR). Miscellaneous Restrictions described in Section 5 can be used to prohibit selected users from accessing certain UCD groups. Either Miscellaneous Restrictions or restrictions assigned through the COR can be used to prohibit the group members from being accessed individually. Unless such restrictions are administered, each group member can be accessed individually as well as through the group.

The UCD group queue is optional on an individual group basis. The maximum queue size is 35. An optional queue warning lamp can be provided for each UCD group. The warning level can be administered for 1 to 35 calls waiting in queue. The lamp lights when the warning level is reached. Each queue warning level lamp requires a port on an Analog Line circuit pack.

The system is limited to a maximum of 32 UCD and Direct Department Calling (DDC) groups combined. Each UCD and DDC group used for voice and each UCD group used for data count as one of the 32 available groups. Each group can have up to 32 members. The system maximum, however, is 448 members for all groups combined.

Hardware and Software Requirements

UCD requires one port on an Analog Line circuit pack for each queue warning lamp provided. (The hardware requirements for data modules, 515 BCTs, and modems are given with the Data Call Setup feature.)

No additional software is required.

Interactions

- Data Call Setup

  Voice Terminal Dialing or Data Terminal (Keyboard) Dialing can be used on calls to a UCD group.

DATA COMMUNICATIONS PROTOCOLS AND INTERFACES

Overview

A protocol is a set of conventions or rules that govern how data is transmitted and received. The rules generally cover such subjects as the following:

- Physical interface
- Mechanical interface
- Electrical interface
- Framing
- Error detection and control

Communications protocols are designed to meet the transmission requirements for specific data exchange and data communications equipment. These communications protocols are sponsored by a national or international organization or a major corporation. The System 75 equipment, Applications Processor (AP), and communications processing software provide the following protocols:

- RS-232C
- RS-449
- RS-366
- Standard Serial Interface (SSI)
- Teletypewriter (TTY) Modes
- X.25 Packet Switching
- Digital Communications Protocol (DCP)
- Switched Digital Communications Protocol Interface (SDCPI)
- BX.25 Packet Switching
- Binary Synchronous Communications (Bisync)

Electronic Industries Association (EIA)

RS-232C

This protocol is widely used for short distance and low-speed applications such as data terminals and modems connecting data terminals. The data link consists of a 25-conductor cable. The conductors are used for data-link control and timing, as well as to transmit and receive signals. Data-link control is accomplished by handshake signaling between the transmit and receive devices. Data speeds are limited to 20 Kbps or less.

The RS-232C protocol provides two interface connectors. The female side connector is known as data communications equipment (DCE). The male side connector is known as data terminal equipment (DTE). Data equipment manufacturers design either the DCE or DTE interface into their products. Products such as modems, data service units (DSUs), Digital Terminal Data Modules (DTDMs), and Processor Data Modules (PDMs) have a built-in DCE interface. Products such as some types of multiplexers, data terminals, printers, computer ports, and Trunk Data Modules have a built-in DTE interface.

The maximum cable length recommended by EIA for the RS-232C protocol is 50 feet (15 meters). However, practical applications have shown that the cable length can be much greater. Factors limiting cable length include transmission speed, cable capacitance, and nearness of noise sources, such as fluorescent lights or electric generators. Each application should be considered separately.

RS-449

This protocol allows longer cables than the RS-232C. Maximum cable length for various data speeds are as follows:

- 19.2 Kbps—200 feet (61 meters)
- 9.6 Kbps—400 feet (122 meters)
- 4.8 Kbps—800 feet (244 meters)
- 2.4 Kbps—1600 feet (488 meters)

The RS-449 protocol is provided as a communications line interface on the AP. This standard uses a 37-conductor cable. The AP RS-449 interface contains unbalanced driver/receivers that also permit interconnection to the RS-232C interface when accompanied with a 37- to 25-pin cable adapter. Since the AP RS-449 interface is compatible with the RS-232C protocol, it also is limited to the same maximum data rate of 20 Kbps.

RS-366

The RS-366 communication protocol specifies the standards for interfacing computers to Automatic Call Units (ACUs). This enables a computer to originate data calls over a switched telephone network. The AP provides one RS-366 interface for each six RS-232C interface ports.

AT&T Teletype Corporation

SSI

The SSI communications protocol was developed by AT&T Teletype Corporation and is used with the 500 Business Communications Terminal (BCT) and 400-series printers. The interface operates full-duplex, in synchronous mode, at 56 Kbps, and over 24-gauge standard building cable at distances up to 5000 feet (1524 meters). Cable connections are made through the 8-pin modular-type connectors.

Teletypewriter (TTY) Modes

The AP EIA RS-232C interface ports support the TTY protocol. This protocol is implemented as software within the AP's EIA terminal or port subsystem. The protocol permits each port to operate in either the transparent or TTY mode.

- **Transparent Mode**

  When operating in the transparent mode, the ports pass ASCII characters between the AP and terminal device unchanged. Incoming characters can be echoed back to the terminal device as they are received. However, no recognition of control characters is provided. The BREAK character is the only special character that can be recognized. The following options are available:

  - Parity (enable and disable, even and odd)
  - Data rate—less than 300 up through 19,200 bps
  - Stop bits—1, 1-1/2, or 2 bits
  - Local—assume line with or without modem control
  - Character size—5, 6, 7, or 8 bits plus parity bit
  - Echo—on and off

  Some of these options are pre-coded by the applications software and cannot be changed by the terminal user.

- **TTY Mode**

  When operating in the TTY mode, the EIA interface port acts as both the pre-processor and post-processor between the terminal and the AP applications software. In addition to all options listed under the transparent mode, several ASCII control characters are recognized.

  A variety of control (delay) options are available to interface with different types of EIA-compatible printers. The ASCII characters DEL and NUL are used for fill (delay) characters. Termination options are provided for line control of modems and ACUs. The TTY mode also provides several mapping options.

  The AP applications software determines the mode (transparent or TTY) and the options within each mode that are implemented per EIA channel. The methods for selecting EIA channel parameters are provided through option designation display forms or by default. When display forms are provided, they are an integral part of the applications software.

International Telegraph and Telephone Consultative Committee (CCITT)

X.25 Packet Switching Protocol

The CCITT is one of three divisions of the International Telecommunications Union, an agency of the United Nations. The standards set by the CCITT generally deal with public networks. Two series of standards or recommendations specifically deal with data transmission:

- The V-series provides recommendations for data transmission over analog or voice telephone networks.
- The X-series provides recommendations for data transmission over digital networks.

The X.25 protocol is the CCITT recommendation for implementing the International Standards Organizations Reference Model of Open Systems Interconnection which is the international model for packet switching networks. This is a bit-oriented, layered-type protocol. The transport, network, data link, and physical layers (levels) are defined functionally by the CCITT.

The X.25 protocol specifies network requirements and procedures to provide the user interface for a packet switching network (such as Net 1000). Typically, users generate low-speed asynchronous data. The X.25 software segments this data into packets, adds framing and routing information, and queues the packets into a buffer memory. User data packets, along with the added framing bits, are then transmitted over high-speed carriers. This permits efficient and dynamic sharing of these high-speed data links.

The X.25 protocol provides the communications links between multiple APs.

Digital Communications Protocol (DCP)

The DCP is used by the system's digital switch, digital voice terminals, data modules, and the 515 BCT. This protocol enables simultaneous voice and data over the same communications link to the switch. It is upward compatible with the emerging CCITT integrated voice and data standard.

The DCP consists of a 160-Kbps, 4-wire serial data link that operates full-duplex over standard twisted-pair building cable. For data-only transmission, the maximum cable length is 5000 feet (1524 meters). When voice and data transmission is carried over the same data link, as when a 515 BCT or a DTDM is used, the cable length is limited by the voice transmission distance.

The DCP sends digitized voice and digital data in frames. Each frame consists of four fields or channels (see Figure 6-3). The first field is a unique 3-bit framing pattern that defines the frame boundary. The second field is a 1-bit control or signaling channel between the digital switch and digital data endpoint. The third and fourth fields are two independent information (I) channels. The information channels are 8 bits each and are used to send digitized voice or digital data.

Figure 6-3. Digital Communications Protocol Frame Structure

There are 8000 frames per second. Therefore, the bit rate available is 8 Kbps for the signaling channel and 64 Kbps for the information channel. The digital switch routes each information channel independently so that simultaneous voice and data can be completed to different destinations.

The full capacity of the information channels (64 Kbps) is available for digitized voice. Data terminals typically operate at speeds from below 300 bps up to 19.2 Kbps, asynchronous or synchronous. The DCP uses data modules to map the data terminal functions into a 64-Kbps information channel.

The framing rate of 8000 per second and 8 bits per information channel is consistent with other telecommunication systems such as the T1 carrier. This minimizes potential conversion problems when interfacing to different digital facilities.

Switched Digital Communications Protocol Interface (SDCPI)

The SDCPI is used between the AP and Processor Data Modules. This protocol operates at 64 Kbps and uses a 25-conductor cable that is limited to a maximum length of 12 feet (3.7 meters).

BX.25 Packet Switching Protocol

The BX.25 protocol implements the international standard for packet switching. It is a multilayered protocol. (Layering is a structuring of specific protocol functions [for example, error detection and correction] that are grouped together as a unique layer or level.) The BX.25 protocol is similar to the CCITT X.25 protocol and, from a user perspective, is compatible with the standard. The Application, Presentation, and Session layers (see Table 6-B) are defined in the Transaction Oriented Protocol (TOP) of the BX.25. The X.25 protocol does not provide specifications for these layers. Differences between X.25 and BX.25 are as follows:

- The X.25 protocol specifies network standards only; the BX.25 protocol places requirements on the user interface as well.
- The X.25 protocol provides for datagram services while the BX.25 protocol does not. Datagram service has not been implemented within the continental United States.
- The X.25 protocol leaves the users in a point-to-point environment to develop their own solutions to the following areas of potential conflict, while the BX.25 protocol provides solutions:
    - Link layer addressing
    - Logical channel selection
    - Call collision

Basic elements of the Application and Presentation layers must be user defined under both protocols. Table 6-B shows the relationship and similarity between the BX.25 and X.25 protocols.

The BX.25 protocol is used in the system to provide communications between the switch and the AP when the AP provides the switch-related features.

TABLE 6-B. Packet Switching Protocols

<table>
<tr><th colspan="2">BX.25 Protocol Layers</th><th>X.25 Protocol Layers</th></tr>
<tr><td>Top Application</td><td rowspan="2">User Defined</td><td rowspan="3">Not Specified by Protocol</td></tr>
<tr><td>Top Presentation</td></tr>
<tr><td colspan="2">Top Session</td></tr>
<tr><td colspan="2">Packet</td><td>Packet</td></tr>
<tr><td colspan="2">Link</td><td>Link</td></tr>
<tr><td colspan="2">Physical</td><td>Physical</td></tr>
</table>

International Business Machines

Binary Synchronous Communications (Bisync)

Bisync is a character-oriented protocol that provides data transfer, error detection, and error correction. It is widely used for interactive data communications networks.

This is a multilayered protocol. Layering is a structuring of specific protocol functions (for example, error detection and correction) that are grouped together as a unique layer or level.

The protocol is implemented partly in hardware and partly in software. The physical or hardware level consists of the AP and its associated communications line controller. The line controller has an RS-232C and an RS-449 communications port. Both ports can be used for connection to Bisync-type networks.

The Bisync protocol can be used in either point-to-point or multipoint data-link configurations. These network configurations can be either switched or dedicated lines. Generally, the data link operates in half-duplex, synchronous mode at 2.4, 4.8, or 9.6 Kbps. Either the American Standard Code for Information Interchange (ASCII) or the Extended Binary Coded Decimal Interchange Code (EBCDIC) can be used.

The AP uses the Bisync protocol in providing 2780/3780 and 3270 terminal emulation features.

GLOSSARY

Access Code

A 1-, 2-, or 3-digit dial code used to activate or cancel a feature. The star (*) and pound (#) can be used as the first digit of an access code.

Administer

To access and change the parameters associated with the services or features of the system.

Answer-Back Code

A code dialed to retrieve a parked call.

Appearance

See Call Appearance.

Asynchronous Data Transmission

A scheme for sending and receiving data where data elements may occur at irregular times.

Attendant

The operator of the console.

Applications Processor

A minicomputer used to support several user-controlled applications such as traffic analysis and electronic documentation.

Attendant Console

An electronic call-handling position with pushbutton control. Used by attendants to answer and place calls and to manage and monitor some of the system operations.

Automatic Trunk

A trunk that does not require the sending or receiving of digits. The destination is predetermined. A request for service on the trunk (called a seizure) is sufficient to route the call. The normal destination of an automatic trunk is the system attendant group.

Barrier Code

A security code used with the Remote Access feature to prevent unauthorized access to the system.

Bit (Binary Digit)

One unit of information in binary notation (having two possible states or values, zero or one).

Bridge (Bridging)

The connection of one or more calls onto an existing connection without interrupting the connection.

Bridged Appearance

A call appearance on a voice terminal that matches a call appearance on another voice terminal for the duration of a call.

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

One or more conductors used as a path to transmit information.

Bus, Time Division Multiplex

See Time Division Multiplex Bus.

Business Communications Terminal

An advanced series of semi-intelligent terminals.

Byte

A unit of information containing eight bits of data.

Call Appearance, Attendant Console

Six buttons, labeled a through f, used to originate, receive, and hold calls. Each button has two associated lamps 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. A lamp next to the button shows the status of the call appearance.

Callback Call

A call that is automatically returned to a voice terminal user who activated the Automatic Callback or Ringback Queuing feature.

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 number. These codes are numbered from 200 through 999.

Central Office Trunk

Access from the system to the public network through the local central office.

Channel

A communications path for transmitting voice and data.

Class of Restriction (COR)

A number (0 through 63) that specifies the restrictions assigned to voice terminals, voice terminal groups, data modules, and trunk groups.

Class of Service (COS)

A number (0 through 15) that specifies if voice terminal users can activate the Automatic Callback, Call Forwarding All Calls, Data Privacy, or Priority Calling features.

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 Answer Group

A group of up to eight voice terminals that alert (ring) simultaneously when a call is redirected by Call Coverage. Any one of the group can answer the call.

Coverage Call

A call that is redirected from the called party's extension number 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 Department Calling group, Uniform Call Distribution group, Coverage Answer Group, a voice terminal, or Message Center designated as an alternate answering position in a coverage path.

Covering User

The person at an alternate answering position who answers a redirected call.

Data Channel

A communications path between two points used to transmit digital signals.

Data Communications Equipment (DCE)

Any equipment that connects to a data service using an EIA RS-232C interface.

Data Terminal Equipment (DTE)

A data endpoint such as a data terminal, host computer, or printer.

Delay-Dial 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.

Designated Voice Terminal

The specific voice terminal to which calls, originally directed to a certain extension number, are redirected. Commonly used to mean the "forwarded-to" terminal when Call Forwarding All Calls is active.

Dial Repeating Tie Trunk

A telecommunications channel between two private switching systems. The number dialed is repeated or dialed-in at the distant end.

Digital Communications Protocol (DCP)

Defines the capability for providing simultaneous voice and data transmission over the same channel.

Digital Terminal Data Module (DTDM)

An adjunct to a 7400-series voice terminal. Provides the required interface between the system and a data terminal such as a 513 Business Communications Terminal.

Digital Trunk

A circuit in a telecommunications channel designed to handle digital data.

Direct Extension Selection (DXS)

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.

Electronic Tandem Network (ETN)

A private telecommunications network that automatically switches calls over specific tie trunks.

End-to-End Signaling

The transmission of signals generated by touch-tone dialing from a voice terminal user to remote computer equipment. A connection must first be established over an outgoing trunk from the calling party to the computer equipment. Then additional digits can be dialed to transmit information to be processed by the computer equipment.

Enhanced Private Switched Communications Service (EPSCS)

A private telecommunications network that provides advanced voice and data telecommunications services to companies with many locations.

Extension Number

A 2-, 3-, or 4-digit number assigned to each voice terminal, certain groups, data modules, and 515 Business Communications Terminals within the system.

External Call

A connection to or from a system user and a party on the public telephone network or on a tie trunk.

Facility

A general term used for the telecommunications transmission pathway and associated equipment.

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

An area code other than the local area code. The foreign area code must be dialed to call outside the local area code.

Ground-Start Trunk

On outgoing calls, System 75 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 System 75 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 ring lead is sufficient to cause the call to route to a predetermined destination, normally the system attendant group. No digits are received.

Handshaking Logic

Logic circuits used to establish a data connection between two devices.

Home Numbering Plan Area Code

The local area code. The area code does not have to be dialed to call numbers within the home area code.

Immediate-Start 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 multi-appearance voice terminal that lights to show when a call appearance is active.

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.

Link

A transmitter-receiver channel or system that connects two locations.

Loop-Start Trunk

After establishing a connection with the distant switching system for an outgoing call, System 75 waits for a signal on the loop formed by the trunk leads 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.

Message Center

An answering service for calls that might otherwise go unanswered; accepts and stores messages for later retrieval. (Requires an Applications Processor.)

Message Center Agent

A person within the Message Center who takes and retrieves messages for voice terminal users.

Modem Pooling

Provides shared-use conversion resources that eliminate the need for a dedicated modem when a data module accesses, or is accessed by, an analog line or trunk.

Multi-Appearance 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.

Multiplexer

A device for simultaneous transmission of two or more signals over a common transmission medium.

Network

An arrangement of inter and/or intra location circuits designed to perform specific functions.

Paging Trunk

A telecommunications channel used to access an amplifier for loudspeaker paging.

Pickup Group

A group of individuals authorized to answer any call directed to an extension number within the group.

Port

A designation of the location of a circuit that provides an interface between the system and lines and/or trunks.

Principal (User)

With Call Coverage, a person for whom a call was originally intended.

Private Network

A network used exclusively for handling the telecommunications needs of a particular customer.

Processor Data Module (PDM)

Provides the required interface between the system and a computer or data terminal.

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 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.

Remote Home Numbering Plan Area Code

A foreign numbering plan area code that is treated as a home area code by the Automatic Route Selection feature. Calls can be allowed or denied based on the area code and the dialed central office code rather than just the area code. If the call is allowed, the Automatic Route Selection pattern used for the call is determined by these six digits.

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 Terminals

Voice terminals served by a single-line tip and ring circuit (Models 2500, 2554, 7101A, and 7103A).

Software

A set of computer programs that accomplish one or more tasks.

Special Audible Alerting Tone

A low-pitched tone identical to the audible alerting tone except the tone decreases the last 0.2 second. This tone notifies the attendant that the Attendant Call Waiting feature has been activated and that the called user is aware of the waiting call.

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.

Standard Serial Interface (SSI)

A communications protocol developed by AT&T Teletype Corporation for use with the 500-series Business Communications Terminals and the 400-series printers.

Status Lamp

A green lamp that shows the status of a call appearance or a feature button by the state of the lamp (lighted, flashing, fluttering, or dark).

Switchhook

The button(s) on a voice terminal located under the receiver.

Synchronous Data Transmission

A scheme for sending and receiving data, where data elements may occur only at regular specified times. Sending and receiving devices must operate in step with each other.

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).

Tandem Through

The switched connection of an incoming trunk to an outgoing trunk without human intervention.

Tandem Tie Trunk Network

A private network that interconnects several customer switching systems by dial repeating tie trunks. Access to the various systems is dictated by codes that must be individually dialed for each system.

Tie Trunk

A telecommunications channel that directly connects two private switching systems.

Time Division Multiplex Bus

This bus has special electrical characteristics permitting it to traverse almost 15 feet (4.5m) for a 5-carrier system. It may be driven from any of the approximately 80 circuit pack slots, and the signal can be received by any circuit pack.

Tone Ringer

A device with a speaker, used in electronic voice terminals to alert the user.

Trunk

A telecommunications channel between two switching systems.

Trunk Data Module

Provides the required interface between the system and a data set (modem) or data service unit connected to a private or switched data line.

Trunk Group

Telecommunications channels assigned as a group for certain functions.

Voice Terminal

A single-line or multi-appearance voice instrument that replaces the telephone.

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 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 process of 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.

INDEX

GENERAL

This Appendix contains information on new features, hardware, software, and services that will be available with Release 1 Version 2 of System 75. The primary purpose of most of these System 75 enhancements is to provide for communications among sites in a network. The information is presented in a format similar to that of Section 5.

All Release 1 Version 2 System 75s will incorporate, at minimum, the following hardware and software differences from Release 1 Version 1:

- A new Processor circuit pack (TN711B replacing TN711)
- An Expansion Memory circuit pack (TN743)
- New Release 1 Version 2 voice applications software
- In an upgrade of an existing Release 1 Version 1 system having an Applications Processor connected to the switch processing element, a new Interface 2 circuit pack (TN738 replacing TN720)

Additional requirements are summarized in Table A and discussed on a feature-by-feature basis in the following sections.

The Distributed Communications System (DCS) service extends the capability and versatility of System 75. DCS is discussed in detail in its own section in this Appendix. The following features related to voice management are significantly affected by DCS and receive separate coverage in this Appendix:

- DCS Alphanumeric Display for Terminals
- DCS Applications Processor Oriented Voice Services
- DCS Call Waiting—Origination
- DCS Distinctive Alerting
- DCS Leave Word Calling
- DCS Uniform Dial Plan

Some voice features operate virtually the same in a DCS as they do within a System 75 and are discussed briefly in the DCS section:

- Abbreviated Dialing
- Automatic Callback
- Call Coverage
- Call Forwarding
- Conference/Transfer
- Priority Calling

The impact of DCS on the following attendant features is also covered in the DCS section:

- Alphanumeric Display
- Attendant Call Waiting
- Attendant Control of Trunk Group Access
- Automatic Circuit Assurance Referral

- Busy Verification of Terminal Lines
- Direct Trunk Group Selection
- Trunk Group Busy/Warning Indication
- Trunk Verification by Customer

The following enhancement applies to a network of System 75s and other switching systems:

- Centralized Attendant Service

Four additional enhancements involve new System 75 hardware, software, and related services:

- Digital Multiplexed Interface (using DS-1 trunk)
- Digital Tie Trunk Interface (using DS-1 trunk)
- EIA Interface
- Automatic Route Selection—Enhanced

TABLE A. System 75 Release 1 Version 2 Hardware/Software Requirements

Feature	Required Hardware	Required Software
Distributed Communications System (DCS)	New Processor circuit pack (CP)(TN711B replacing TN711) Expansion Memory CP(TN743) New Interface 2 CP (TN738 replacing TN720) Tie Trunk CP(s) (TN760 or TN760B) or DS-1 Tie Trunk CP(s) (TN722) Tone/Clock CP (TN741) (needed only when DS-1 Tie Trunk is used)	Release 1 Version 2 voice applications software DCS software
Centralized Attendant Service (CAS)	New Processor CP (TN711B replacing TN711) Expansion Memory CP (TN743) Release Link Trunk CP(s) (TN760B)	Release 1 Version 2 voice applications software CAS software
Digital Multiplexed Interface (DMI)	New Processor CP (TN711B replacing TN711) Expansion Memory CP (TN743) DS-1 Tie Trunk CP(s)(TN722)	Release 1 Version 2 voice applications software
Digital Tie Trunk Interface	New Processor CP (TN711B replacing TN711) Expansion Memory CP (TN743) DS-1 Tie Trunk CP(s) (TN722) Tone/Clock CP (TN741)	Release 1 Version 2 voice applications software

TABLE A. System 75 Release 1 Version 2 Hardware/Software Requirements (Contd)

Feature	Required Hardware	Required Software
EIA Interface	New Processor CP (TN711B replacing TN711) Expansion Memory CP (TN743) Data Line CP(s) (TN726) Asynchronous Data Unit(s)	Release 1 Version 2 voice applications software
Enhanced Automatic Route Selection (ARS)	New Processor CP (TN711B replacing TN711) Expansion Memory CP (TN743) New Tone Detector CP (TN748B replacing TN748)	Release 1 Version 2 voice applications software and ARS software
Applications Processor (AP) with Release 1 Version 2	New Interface 2 CP (TN738 replacing TN720)	None

DISTRIBUTED COMMUNICATIONS SYSTEM

Description

A Distributed Communications System (DCS) is a network of two or more switches, each with its terminals and trunks, configured to function as a single large system. The switches in the DCS are interconnected by tie trunks for voice communications and data links for control and feature information. Figure 1 shows a typical DCS in simplified form.

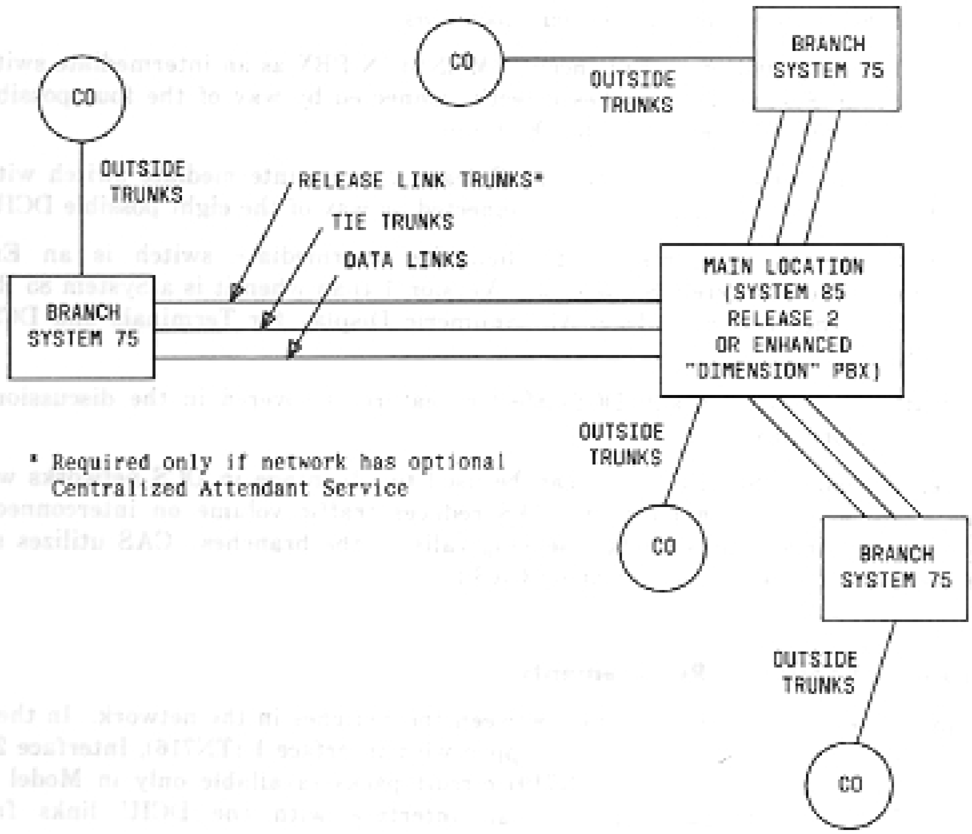

Figure 1. Typical Distributed Communications System

Some of the applications of the DCS configuration are as follows:

- At a single site where multiple communications subsystems are required.
- In a "campus environment" that has two or more separate buildings and the communications subsystems are connected by local cable.
- In a larger area such as a city, several states, or even the entire country, where the communications subsystems are separated by distances too great for local cable and may be connected to different central offices.

A System 75 DCS has the property of "transparency" with respect to internal calling and some features. This greatly enhances the convenience and usability of a multi-switch system. Transparency is the ability of the system, from the user's standpoint, to operate across several switches in the same way it does within the area served by one switch. This allows users to dial from any terminal to any other terminal within the DCS with no regard for which switches are involved. Likewise, transparency allows certain voice features (listed under Interactions) to be used across switch boundaries.

Considerations

A Uniform Dial Plan (UDP) is necessary in a DCS. (See the UDP section.)

For System 75s in DCS networks, a limited number of configurations exist.

- System 75s cannot be connected together, back-to-back; a System 85 Release 2 or Enhanced DIMENSION PBX must be used as the "main" location and intermediate switch between System 75 "branch" locations.
- A DCS can consist of an Enhanced DIMENSION PBX as an intermediate switch with up to four System 75 branches directly connected by way of the four possible Data Communications Interface Unit (DCIU) links.
- A DCS can consist of a System 85 Release 2 as an intermediate switch with up to eight System 75 branches directly connected by way of the eight possible DCIU links.

DCS transparency is more restricted when the intermediate switch is an Enhanced DIMENSION PBX or a System 85 Release 2 Version 1 than when it is a System 85 Release 2 Version 2. (See the section on DCS Alphanumeric Display for Terminals and DCS Leave Word Calling.)

Administration required for any DCS-affected feature is covered in the discussion of the feature in this Appendix.

Centralized Attendant Service (CAS) can be used to advantage in DCS networks where all attendants are at the main location. CAS reduces traffic volume on interconnecting tie trunks caused by incoming attendant-seeking calls at the branches. CAS utilizes software and Release Link Trunks. (See section on CAS.)

Hardware and Software Requirements

DCS requires tie trunks and data links between the switches in the network. In the System 75 switch, the control carrier must be equipped with Interface 1 (TN716), Interface 2 (TN738 replacing TN720), and Interface 3 (TN719) circuit packs (available only in Model 2 and 3 systems). These circuit packs provide an interface with the DCIU links from the intermediate switch.

DCS software is required.

Interactions

The following System 75 voice features have unique aspects in a DCS environment and are described in detail in separate parts of this Appendix.

- Alphanumeric Display for Terminals
- Applications Processor Oriented Terminal Voice Services
- Call Waiting—Origination
- Distinctive Alerting
- Leave Word Calling
- Uniform Dial Plan

The following System 75 voice terminal features have transparency in a DCS configuration.

- Abbreviated Dialing and Last Number Dialed

  Transparency allows any terminal user in the DCS complex to use these related features to call any other terminal on the same or another switch. These features must be coordinated with the DCS numbering plan.

- Automatic Callback

  This feature can be used on calls from any terminal in the DCS complex to any other terminal on the same or another switch. Automatic Callback provides ringback and return calling for calls to busy or unanswered terminals in System 75 and to busy terminals in System 85.

- Call Coverage

  Limited transparency is provided for this feature. In order for Call Coverage to be effective, the covering terminal must be on the same switch as the covered terminal (the principal). When this is the case, calls from any terminal in the DCS complex can be redirected to coverage and the calling terminal receives coverage tone and the caller response interval. The covering user can activate coverage callback.

- Call Forwarding All Calls

  Transparency for this feature allows a terminal user to forward all incoming calls to any other terminal on the same or another switch in the DCS complex.

- Conference/Transfer

  The System 75 terminal user in a DCS can set up conferences with, and transfer calls to, any other terminal on the same or another switch in the DCS complex. Trunk calls can be included in a conference.

- Priority Calling

  Priority calls, with their distinctive alerting, can be placed between any terminals in a DCS network.

An attendant at a branch System 75 switch in a DCS complex can have access to trunks and lines at other switches, as well as locally. However, the following attendant features are transparent only for attendants at the main DCS location (System 85 Release 2 or Enhanced DIMENSION PBX). They are not transparent for any attendants at branch locations.

- Alphanumeric Display

  Incoming call identification for terminal-originated calls and trunk calls at another switch is displayed at the main location attendant console.

- Attendant Call Waiting

  The main location attendant can extend a call to a busy single-line terminal at a remote switch where the call will wait and alert the called user.

- Attendant Control of Trunk Group Access

  The main location attendant has control of access to trunk groups at remote switches. Terminal users who attempt to access a controlled trunk group are routed to the attendant.

- Automatic Circuit Assurance Referral

  This feature monitors call durations on assigned trunks at remote switches and generates a referral call to the main location attendant when software detects a possible malfunction.

- Busy Verification of Terminal Lines

  The main location attendant can verify the busy/idle state of terminal lines at remote switches.

- Direct Trunk Group Selection

  The main location attendant can select a trunk group at another switch.

- Trunk Group Busy/Warning Indication

  This information is sent from a remote switch to the main location attendant.

- Trunk Verification by Customer

  The main location attendant can be connected to trunks at remote switches for testing.

DCS ALPHANUMERIC DISPLAY FOR TERMINALS

Description

In a DCS, calls to or from terminals equipped with alphanumeric displays have some transparency with respect to Calling Name Display and Called Name Display.

Calling Name Display is the presentation, on the *called* terminal's alphanumeric display, of the name of the party who originated the call. Called Name Display is the presentation on the *originating* terminal's display of the name of the party to whom the call is directed. Both displays provide more useful and precise information than such general identifiers as a trunk group name or an extension number.

The transparency allows calling and called name information, plus miscellaneous identifiers (ids) to be sent from a terminal on one switch to a terminal on another switch. Transparency in this area is limited by the type of switching systems at the calling and called ends and the type of intermediate switching system, if any, that the call is routed through.

Considerations

On the 7405D Voice Terminal, alphanumeric display is implemented by the addition of a Digital Display Module. Alphanumeric display is an integral feature of the Model 515 Business Communications Terminal (BCT).

Calls to and from a System 75 in a DCS network have Calling/Called Name Display transparency under the following conditions:

- The other party is at another System 75 and the intermediate switch is a System 85 Release 2 Version 2.
- The other party is at a System 85 Release 2 Version 2.
- The call is not routed through an intermediate System 85 Release 2 Version 1 or Enhanced DIMENSION PBX switch. (Such calls will display only the extension number of the calling or called party.)

On outgoing DCS calls, display of the called name may be delayed for a few seconds until the required information arrives from the remote switch.

Within the same System 75 switch in a DCS, complete transparency of Calling and Called Name Display exists.

DCS tie trunk groups between switches must be administered with the Outgoing Display disabled. This enables the called party's name to be displayed at the calling terminal.

Hardware and Software Requirements

No additional hardware is required. DCS software is required.

Interactions

The following DCS configurations provide transparency of alphanumeric display information:

- Networks of two or more System 75s with a System 85 Release 2 Version 2 as an intermediate switch
- A System 75 connected to a System 85 Release 2 Version 2

Configurations in which System 75s are connected to or through a System 85 Release 2 Version 1 or an Enhanced DIMENSION PBX are not covered because these switches do not provide display transparency.

The following features have transparency with respect to Calling and Called Name Display and miscellaneous id. If the display for a DCS call differs at all from the display for a call between terminals at the same System 75, the difference is noted. Refer to the *User's Guide AT&T System 75 Voice Terminal Operations*, 999-700-212IS, for detailed descriptions of call information displays.

- Automatic Callback

  Complete display transparency.

- Call Coverage

  At the calling terminal, the miscellaneous id "cover" is not displayed.

- Call Forwarding

  When a System 75 user calls a party on a different switch in the DCS and the call is forwarded, the miscellaneous id "forward" is not displayed. At the covering (forwarded-to) user's terminal, only the calling party's name is shown; the called party's name is not displayed.

- Call Park

  When a DCS call between a local System 75 user and a user on another switch is parked by the remote user, the miscellaneous id "park" is not displayed at the local terminal.

- Call Pickup

  When a DCS call from a System 75 user to another switch is answered by way of Call Pickup, the miscellaneous id "cover" is not displayed at the caller's terminal.

- Call Waiting

  When a DCS call from a System 75 user to another switch is waiting at the called terminal, the miscellaneous id "wait" is not displayed at the caller's terminal.

- Centralized Attendant Service

  When a user dials the extension for Centralized Attendant Service, a Release Link Trunk (RLT) is seized or the caller is queued for an RLT. The caller's terminal will display the trunk group identifier, such as OPERATOR or OUTSIDE CALL.

- Conference

  When a DCS call is conferenced either at a remote switch or at the local System 75, all DCS Calling and Called Name Display transparency is lost to local System 75 users. If all parties drop out except for a local user and another DCS user, the local user's terminal will display the trunk group identifier.

- Direct Department Calling (DDC)/Uniform Call Distribution (UCD)

  Complete display transparency.

- Internal Terminal-to-Terminal Calling

  Complete display transparency.

- Transfer

  When a DCS call is transferred at a remote switch to a user on any switch other than the local System 75, all DCS Calling and Called Name Display transparency is lost to users on the local System 75.

DCS APPLICATIONS PROCESSOR ORIENTED VOICE SERVICES

Description

The Applications Processor (AP) oriented terminal features that will operate transparently in a DCS environment are Message Center and Directory. System 75 Release 1 Version 2 will support AP Release 2 Version 1.

- Message Center

  Message Center service requires an AP at each DCS switch. Message Center agents receive calling party identification for calls from terminals within the DCS complex.

  Message Center operation is transparent for all terminals in the DCS that have been assigned this feature capability. A terminal user is able to leave a message for the called user on a different switch within the DCS complex without having to provide the agent with his/her identity or extension number. A terminal user is able to retrieve messages from an agent from anywhere in the DCS complex by dialing a unique number. The number corresponds to a Message Center UCD group associated with the AP on which the user's message file resides.

- Directory

  Transparent operation of Directory service requires an AP at each switch in the DCS complex, with the entire Directory on each AP. If the complex size exceeds the capacity of Directory data base, Directory service loses transparency and provides information for the local extensions only.

  AP-based Directory service in a DCS is transparent for authorized users. Users perceive the same service in a DCS with several APs as if there were only a single AP.

Considerations

- Message Center User Interfaces

  For Message Center agents in a DCS, the display of information about a redirected call is the same as for the single switch case. Terminal users in a DCS have all the Message Center functionality for both direct and redirected calls as a user in the single switch configuration.

- Directory User Interfaces

  From any AP at any switch, a user can query the entire Directory data base. Directory updates are made to the master copy of the Directory data base.

- System Management

  Measurements for the DCS transparent Directory and Message Center services are the same as provided for the stand-alone configuration.

Hardware and Software Requirements

An AP is required for Message Center and Directory services at each DCS switch. The same AP can provide both services. DCS software is required.

Interactions

None.

DCS CALL WAITING—ORIGINATION

Description

Call Waiting—Origination enhances the Call Waiting—Termination voice feature, which operates transparently across switches in a DCS network as well as within the System 75. Call Waiting—Termination is described in Section 5 of this manual.

Call Waiting—Origination is not a feature of System 75, but will be supported in a DCS complex for calls into System 75 from switches that do provide it. Call Waiting—Origination is, in effect, a combination of the present Call Waiting—Termination and Priority Calling. When activated before a call, it alerts an idle single-line terminal with 3-burst priority ringing. If the called party is busy, however, the call waits and the busy party hears 3-burst alerting through the receiver. The waiting party hears special audible alerting instead of busy tone.

The unique aspect of Call Waiting—Origination is that it can also be activated *after* the caller receives busy tone from the busy single-line terminal and wants to wait. After activation the call waits, the busy party hears 3-burst alerting through the receiver, and busy tone changes to special audible alerting tone.

Considerations

Call Waiting—Origination can only be received in System 75, not activated.

Hardware and Software Requirements

No additional hardware is required. DCS software is required.

Interactions

Call Waiting—Origination is denied when the following features are activated at the single-line voice terminal:

- Automatic Callback (to or from the voice terminal)
- Data Privacy
- Data Restriction

DCS DISTINCTIVE ALERTING

Description

The Distinctive Alerting feature activates the alerting, or ringing, device of a called terminal so that the user is aware of the type of incoming call before answering it. Distinctive Alerting functions in a DCS environment as it does within a System 75.

Considerations

When DCS transparency is lost for any reason, terminal-to-terminal calls made between switches produce 2-burst alerting instead of the usual 1-burst. Loss of transparency may occur when the data link between switches is down or when data transmission delay exceeds the trunk signaling time.

Hardware and Software Requirements

No additional hardware is required. DCS software is required.

Interactions

- Intercom

  This feature and its distinctive alerting are not provided between switches in a DCS.

- Manual Signaling

  This feature and its distinctive alerting are not provided between switches in a DCS.

- Tie Trunk Access

  In a DCS, tie trunk groups can be administered as either internal or external tie trunk groups. Calls from internal tie trunk groups are treated as terminal-originated calls and receive 1-burst alerting. Calls from external tie trunk groups are treated as calls originated external to the system and receive 2-burst alerting.

DCS LEAVE WORD CALLING

Description

The Leave Word Calling (LWC) feature enables System 75 terminal users to leave "call me" messages at other terminals. Messages can be left by calling, called, or covering users.

LWC transparency in a DCS configuration allows messages from a System 75 to another switch, depending on the storage capability of the remote switch.

Considerations

Both System 75 and System 85 Release 2 Version 2 can store LWC messages on the switch itself, without being connected to an AP. However, both System 85 Release 2 Version 1 and Enhanced DIMENSION PBX must be connected to an AP in order to store LWC messages.

The LWC feature cannot be successfully activated toward any system that is not capable of storing the message on the switch itself or in an associated AP.

Messages from one switch, through an intermediate switch, to a remote destination switch in the same DCS do not require storage capability at the intermediate switch.

The following configurations have LWC transparency in a DCS network:

- From System 75 to System 85 Release 2 Version 2
- From System 75 through any intermediate switch to another System 75 or a System 85 Release 2 Version 2
- To System 75 from any other switch

Retrieval of LWC messages is permitted only at a terminal on the same switch where the messages are stored.

Hardware and Software Requirements

No additional hardware is required. DCS software is required.

Interactions

- DCS Conference/Transfer

  Activation of LWC is denied after a DCS call has been conferenced or transferred.

DCS UNIFORM DIAL PLAN

Description

A Uniform Dial Plan (UDP) assigns a unique extension number to each terminal in a DCS configuration. Within any one DCS, all the UDP extension numbers have either four digits or five digits. The UDP feature enables a terminal user at any switch to call any other terminal on any switch in the DCS complex using only the 4- or 5-digit extension. The unique extension numbers are also used in feature transparency messages between switches to refer to specific terminals. In any UDP, the first two digits of the extension determine the particular switch to which a call is directed.

Considerations

All of the extension numbers on a switch are not necessarily part of the UDP. Any that do not belong to the UDP are handled by a regular, non-UDP dial plan.

Hardware and Software Requirements

No additional hardware is required. DCS software is required.

Interactions

None.

CENTRALIZED ATTENDANT SERVICE

Description

Centralized Attendant Service (CAS) allows all services performed by attendants in a private network of switching systems to be concentrated at a central, or main, location. Each branch in a CAS has its own listed directory number (LDN). Incoming trunk calls to the branch, as well as attendant-seeking terminal calls, are routed to the centralized attendants over release link trunks (RLTs).

RLTs can be seized only from the branch switch and are used only for CAS calls and CAS signaling. After processing by a centralized attendant, CAS calls are extended back over the same RLT to, for example, the requested extension number or outgoing trunk. The RLT is then dropped and becomes available for other calls toward the centralized attendants.

System 75 can only be a branch location in a CAS network.

Considerations

A network with CAS can also be a Distributed Communications System, but this association is not required.

In a CAS network, System 75s can function only as branches; the main location, where the centralized attendants reside, must be a system capable of providing attendant concentration. From the standpoint of System 75, the feature is called CAS-Branch.

When all RLTs are busy, CAS attendant-seeking calls are placed in a CAS queue.

Backup service provides for all CAS calls being sent to a backup extension in the local branch if all RLTs are maintenance busy or out-of-service.

A System 75 branch can have a local attendant. Access to the local attendant must be by way of a unique code other than 0. Incoming trunk calls in a CAS network bypass local attendants but can be routed back to them by the centralized attendant.

A CAS call from a branch can be put on Remote Hold by the CAS attendant. The branch holds the call and drops the RLT. After a time-out (same as the timed reminder for an attendant-held call), the branch automatically seizes an RLT and routes the call back to the attendant.

The System 75 branch in a CAS network generates information tones and transmits them to the CAS attendant by way of the RLT. These tones indicate to the attendant the type of call coming from the branch or the status of a call extended to or held at the branch. The attendant hears these tones in the console receiver.

- Incoming trunk call: 480 Hz (100 ms), 440 Hz (100 ms), 480 Hz (100 ms) in sequence; heard immediately after attendant lifts receiver
- Call from branch terminal to attendant or transferred by branch terminal to attendant: 440 Hz (100 ms), silence (100 ms), 440 Hz (100 ms) in sequence; heard immediately after attendant lifts receiver
- Call extended to idle station or recall on don't answer: audible alerting for 300 ms followed by connection to normal alerting cycle
- Call extended to busy terminal—automatically waiting or recall on attendant call waiting: 440 Hz (100 ms)
- Call extended to busy terminal—waiting denied or not provided: busy tone

- Remote hold or remote hold recall: a series of four to six cycles of 440 Hz (50 ms), silence (50 ms)

In a CAS network the following items are administrable:

- Branch access to CAS
- Branch attendant access code
- RLT group
- RLT group queue length
- CAS backup extension
- Terminal lamps for CAS backup notification
- Extension permitted to put system into night service
- Recall time-out values
- Remote hold access code

Hardware and Software Requirements

CAS can be implemented by replacing the TN760 Tie Trunk circuit packs with the new TN760B in each System 75 switch. The 760B will serve all other tie trunk applications in addition to CAS. As an alternative, the DS-1 Tie Trunk circuit pack (TN722) can be used for the release link trunks of the CAS network.

CAS software is required.

Interactions

The following System 75 voice features operate the same for CAS calls extended from the centralized attendant by way of RLTs as they do for non-CAS calls:

- Call Coverage (including Send All Calls)
- Call Forwarding
- Call Pickup

Calls do not go to CAS attendants when Night Service has been activated at the branch.

DIGITAL MULTIPLEXED INTERFACE

Description

Digital data transmission at the DS-1 rate supplies a high speed, high capacity communications trunking service. Its format consists of twenty-four 64 Kbps digital trunks multiplexed onto a single 4-wire 1.544 Mbps line. Digital Multiplexed Interface (DMI) provides the System 75 switch with data communications by way of DS-1 digital facilities. Its primary use is high speed data transmission over a single line between System 75 and a computer on the same or a remote switch. DMI has 23 channels with up to 64 Kbps data speed and a 24th channel for signaling.

Considerations

DMI is designed to be compatible with the international Integrated Services Digital Network (ISDN) standards now under development.

The host computer must multiplex and demultiplex the 1.544 Mbps data stream from System 75. Several major manufacturers of computers are providing DMI compatibility in their equipment.

Only bit-oriented signaling is supported in System 75 Release 1 Version 2.

Hardware and Software Requirements

The use of DMI in System 75 requires the installation of DS-1 Tie Trunk circuit packs (TN722) in port slots of the switch. The packs must be administered for DMI. No additional software is required.

Interactions

None.

DIGITAL TIE TRUNK INTERFACE

Description

Digital data transmission at the DS-1 rate supplies a high speed, high capacity communications service. Its format consists of twenty-four 64 Kbps digital trunks multiplexed onto a single 4-wire 1.544 Mbps line. The Digital Tie Trunk Interface feature, using DS-1 transmission facilities, provides digital tie trunks for voice and data transmission. It is an economical alternative to the standard 4-wire analog E&M tie trunks.

Two types of DS-1 Digital Tie Trunk Interface are available:

- Voice

  The voice interface provides 24 voice grade channels for communications between System 75 and other switches. For the user, a voice DS-1 digital tie trunk operates exactly like an analog trunk.

- Alternate Voice/Data (AVD)

  AVD also provides 24 channels, of which one is dedicated to trunk supervisory signaling. The other 23 channels can be used for voice or data. Procedures for voice calls and data call setup are exactly the same as those for analog tie trunks.

Considerations

- Voice

  It is not required that all channels be assigned to the same trunk group.

  Voice DS-1 tie trunks can be used in any of the following tie trunk applications:

  - Tandem Tie Trunk Network (TTTN)
  - Main/Branch tie trunks used to access Electronic Tandem Network (ETN) tandem switches
  - Advanced Private Line Termination (APLT) tie trunks used to access Enhanced Private Switched Communication Service (EPSCS) and Common Control Switching Arrangement (CCSA)
  - Release Link Trunks (RLTs) for Centralized Attendant Service

- AVD

  AVD tie trunks can only be used between multiple System 75s and/or System 85s.

  AVD tie trunks can be used in the same applications as voice DS-1 tie trunks.

  Modem pooling is not required for data calls.

Hardware and Software Requirements

Both types of interface require the installation of DS-1 Tie Trunk circuit packs (TN722) in port slots of the System 75 switch. The packs must be administered for DS-1. AVD interface also requires the installation of a Tone/Clock circuit pack (TN741) in the switch. No additional software is required.

Interactions

Administrative care should be taken to prevent overflow from a voice DS-1 tie trunk to an AVD tie trunk if modem pools are present in the network. Modem pooling support is not provided for AVD tie trunks.

EIA INTERFACE

Description

The EIA Interface makes it possible to directly connect RS-232C data equipment and host ports, including Information Systems Network, to ports in the System 75 switch. It is an economical alternative to the Digital Terminal Data Module, the Processor Data Module, and the Trunk Data Module for asychronous applications. Data Terminal Equipment (DTE), such as remote and local data terminals and host computers, can communicate with System 75 without going through external modules.

The EIA Interface provides for full duplex, asynchronous switching of data at rates up to 19,200 bps. The following maximum cable distances (24-gauge wire) are possible:

Speed (bps)	Distance (ft.)
19,200	2,000
9,600	5,000
4,800	7,000
2,400	12,000
1,200	20,000
300	40,000

Considerations

None.

Hardware and Software Requirements

The use of the EIA Interface requires the installation of Data Line circuit packs (TN726) in the System 75 switch. Each pack contains eight interface ports. A Data Line pack can be placed in any port slot of the control carrier or a port carrier.

An additional device, the Asynchronous Data Unit (ADU), must be installed at the terminal equipment. It has an RS-232C plug mounted on one end of the unit for connection to the terminal equipment. At the other end of the ADU is a receptacle for the standard 8-pin mounting cord from the building information outlet. The ADU can be powered from the terminal to which it is connected or by a 2012D plug-in transformer. An optional external originate/disconnect switch is available for use with the ADU.

No additional software is required.

Interactions

None.

AUTOMATIC ROUTE SELECTION—ENHANCED

Description

Enhanced Automatic Route Selection (ARS) allows totally transparent access to tie, FX, WATS, and DDD facilities, as does standard ARS; it also provides the same capability with respect to interexchange carriers (IXCs). The number of 6-digit translation tables and the number of toll lists are increased from 4 to 64. The number of routing patterns is increased from 16 to 254.

Considerations

See Section 5, Voice Management, Automatic Route Selection to evaluate the impact of the ARS enhancements.

Hardware and Software Requirements

Enhanced ARS requires the installation of a new Tone Detector circuit pack (TN748B replacing TN748) in the control carrier. No additional software is required.

Interactions

See Section 5, Voice Management, Automatic Route Selection. Interactions are not affected by the ARS enhancement.