OVERVIEW

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

This manual supports Release 1, Version 2, and Version 3 of System 75 XE. The features associated with Version 3, along with the Version 3 hardware, are identified throughout the manual with a (V3) or (Version 3) notation.

Purpose

This manual, along with the *AT&T System 75—Feature Description*, 555-200-201, is intended to serve as an overall reference for the planning, operation, and administration stages of the system.

Organization

This manual is divided into 13 sections. The remaining sections are as follows:

- SECTION 2—FEATURES AND SERVICES
- SECTION 3—FUNCTIONAL DESCRIPTION
- SECTION 4—HARDWARE DESCRIPTION
- SECTION 5—SOFTWARE DESCRIPTION
- SECTION 6—SYSTEM ADMINISTRATION
- SECTION 7—SYSTEM MAINTENANCE
- SECTION 8—UPGRADE PROCESS
- SECTION 9—TECHNICAL SPECIFICATIONS
- SECTION 10—ENVIRONMENTAL REQUIREMENTS
- SECTION 11—REFERENCES
- SECTION 12—GLOSSARY
- SECTION 13—INDEX

An individual Table of Contents is provided for Sections 2 through 10 of this manual.

Introduction to System 75 XE

The System 75 XE is a user-oriented advanced business communications system. The system's digital switch, using a time division circuit switching technique, provides voice communications, data communications, and integrated messaging capabilities.

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 transmit digitized voice or digital data. The digital switch routes each information channel independently so that simultaneous voice and data connections can be completed to different locations.

The System 75 XE is arranged for touch-tone and rotary dialing. The system also automatically converts touch-tone signals to dial pulses on trunks requiring such conversion.

The system interfaces an Applications Processor (3B2 AP used as a messaging server) to provide Message Center service.

The following optional services are available with Version 3 of the system:

- Hospitality Services for the Lodging and Health Services environment, with the capability of interfacing the Property Management System (PMS)—the PMS allows a customer to control certain features used in both a hospital and a hotel/motel environment.
- Call Management Services, supporting the Automatic Call Distribution (ACD) function, with the capability of interfacing with Call Management Systems (CMS)—the CMS is an adjunct that collects and processes ACD data on the status of agents, splits, and trunks.
- Audio Information Exchange (AUDIX) Interface that allows AUDIX message alerting—AUDIX provides a voice interface to both system users and outside callers to leave and retrieve AUDIX messages.

Call-Handling Capabilities

The System 75 XE switch can be arranged as a stand-alone system or can be an integral part of a private network. The system can serve as:

- Tandem or end location in a Tandem Tie Trunk Network (TTTN)
- Main or Tributary location in an Electronic Tandem Network (ETN) [or Enhanced Private Switched Communications Service (EPSCS)/Common Control Switching Arrangement (CCSA) network]
- Main or Satellite location in a Main/Satellite configuration
- Endpoint in a Distributed Communications System (DCS)
- Branch location for Centralized Attendant Service (CAS)
- Tandem in an ETN
- Tandem in a DCS (V3).

The system can provide the following:

- Up to 600 lines that support digital, hybrid, and analog terminals and equipment
- Data switching capacity of up to 400 digital data endpoints* and/or 160 integrated and combined pooled modems with optional single-button access to the pooled facilities
- Up to 200 trunks including central office (CO), Direct Inward Dialing (DID), tie, foreign exchange (FX), Wide Area Telecommunications Service (WATS), 800 Service trunks, DS1 (Data Services Level 1) tie trunks, and release link trunks (RLTs).

The limits listed for each of these three items probably cannot be achieved simultaneously in any one system. Allowable limits are determined according to expected call usage. See *AT&T System 75 XE—Planning/Configuration*, 555-200-600, and *AT&T System 75 and System XE—Administration*, 555-200-500, for details on determining the allowable limits.

System Configuration

The System 75 XE can be housed in a 1-cabinet, 2-cabinet, 3-cabinet, or 4-cabinet (growth) configuration depending on the capacity requirements of the system. Figure 1-1 shows a typical 3-cabinet system configuration.

* Digital data endpoints are defined in the Glossary—Section 12.

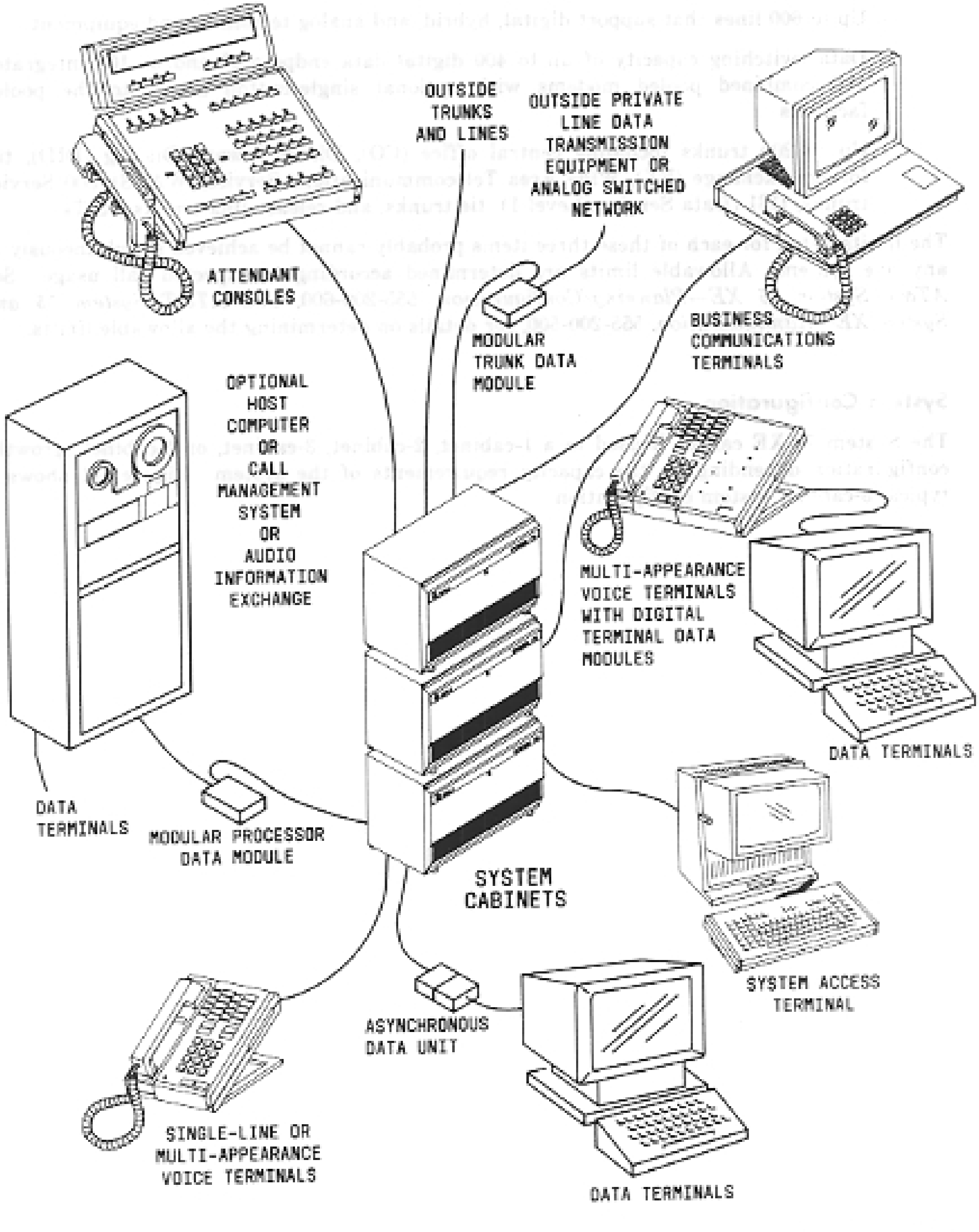


Figure 1-1. Typical System Components

FEATURES AND SERVICES

CONTENTS

LIST OF TABLES

FEATURES AND SERVICES

General

This section contains lists of all system features. The features are first listed alphabetically within the following major feature groups:

- Voice Management
- Data Management
- Network Services
- System Management
- Hospitality Services (V3)

All the features are then listed alphabetically with a concise definition or description without regard to the major feature groups.

Table 2-A at the end of this section lists all of the features and indicates the major group of each feature.

For more detailed information on individual features, refer to the *AT&T System 75—Feature Description*, 555-200-201.

Voice Management

This group of features includes all of the voice communications capabilities available with the system. Each voice capability is designed to improve a particular part of business communications. Many voice features interact with each other.

The following features are associated with Voice Management:

Abandoned Call Search (V3)
Abbreviated Dialing
Agent Call Handling (V3)
AP Demand Print (V3)
Attendant Auto-Manual Splitting
Attendant Call Waiting
Attendant Control of Trunk Group Access
Attendant Direct Extension Selection With Busy Lamp Field
Attendant Direct Trunk Group Selection
Attendant Display
Attendant Recall
Attendant Release Loop Operation
Audio Information Exchange (AUDIX) Interface (V3)
Authorization Codes (V3)
Automatic Callback
Automatic Call Distribution (V3)
Automatic Incoming Call Display
Bridged Call Appearance
Busy Verification of Terminals and Trunks
Call Coverage
Call Forwarding All Calls

Call Park
Call Pickup
Call Waiting Termination
Centralized Attendant Service
Class of Restriction
Class of Service
Code Calling Access
Conference—Attendant
Conference—Terminal
Consult
Coverage Callback
Coverage Incoming Call Identification
Dial Access to Attendant
Dial Plan
Direct Department Calling and Uniform Call Distribution
Direct Inward Dialing
Direct Outward Dialing
Distinctive Ringing
Emergency Access To the Attendant (V3)
Facility Busy Indication
Forced Entry of Account Codes
Go To Cover
Hold
Hot Line Service
Hunting
Individual Attendant Access
Integrated Directory
Intercept Treatment
Intercom—Automatic
Intercom—Dial
Inter-PBX Attendant Calls
Intraflow and Interflow (V3)
Last Number Dialed
Leave Word Calling
Line Lockout
Loudspeaker Paging Access
Manual Message Waiting
Manual Originating Line Service
Manual Signaling
Multi-Appearance Preselection and Preference
Multiple Listed Directory Numbers
Music-on-Hold Access
Night Service—Hunt Group (V3)
Night Service—Night Console Service
Night Service—Night Station Service
Night Service—Trunk Answer From Any Station
Night Service—Trunk Group (V3)
Personal Central Office Line
Personalized Ringing
Power Failure Transfer
Priority Calling
Privacy—Attendant Lockout

Privacy—Manual Exclusion
Queue Status Indications (V3)
Recall Signaling
Recorded Announcement
Recorded Telephone Dictation Access
Remote Access
Restriction—Controlled
Restriction—Miscellaneous Terminal
Restriction—Miscellaneous Trunk
Restriction—Toll/Code
Restriction—Voice Terminal—Inward
Restriction—Voice Terminal—Manual Terminating Line
Restriction—Voice Terminal—Origination
Restriction—Voice Terminal—Outward
Restriction—Voice Terminal—Termination
Ringback Queuing
Rotary Dialing
Send All Calls
Senderized Operation
Service Observing (V3)
Single Digit Dialing and Mixed Station Numbering (V3)
SMDR Account Code Dialing
Straightforward Outward Completion
Temporary Bridged Appearance
Terminating Extension Group
Through Dialing
Timed Reminder
Touch-Tone Dialing
Transfer
Trunk Group Busy/Warning Indicators To Attendant
Trunk Identification By Attendant
Trunk-To-Trunk Transfer
Voice Message Retrieval
Voice Terminal Display

Data Management

This group of features includes all of the data communications and management capabilities available within the system. Data communications is the process of transferring data from one point (usually a data base) to another where it can be used. Data management is the sum total of the planning and control measures applied to effectively use and protect data.

The following features are associated with Data Management:

Data Call Setup
Data Hot Line
Data-Only Off-Premises Extensions
Data Privacy
Data Restriction
Digital Multiplexed Interface
DS1 Tie Trunk Service
EIA Interface

Information System Network (ISN) Interface
Modem Pooling
Permanent Switched Calls

Network Services

This group of features includes all of the capabilities that assure efficient interconnection of the private network. This network is the web of trunk and switching facilities dedicated for use by a business or organization. Similar to the public network, a large private network can make use of intermediate (tandem) switches to complete call connections. By concentrating and distributing call traffic, tandem switches and their available features offer a cost-effective alternative to large numbers of direct trunk groups.

The following features are associated with Network Services:

- AAR/ARS Partitioning (V3)
- Automatic Alternate Routing
- Automatic Circuit Assurance
- Automatic Route Selection
- Distributed Communication System (DCS)
  - DCS Alphanumeric Display for Terminals
  - DCS Attendant Control of Trunk Group Access
  - DCS Attendant Direct Trunk Group Selection
  - DCS Attendant Display
  - DCS Automatic Callback
  - DCS Automatic Circuit Assurance
  - DCS Busy Verification of Terminals and Trunks
  - DCS Call Forwarding All Calls
  - DCS Call Waiting
  - DCS Distinctive Ringing
  - DCS Leave Word Calling
  - DCS Multi-Appearance Conference/Transfer
  - DCS Trunk Group Busy/Warning Indication
- Facility Restriction Levels and Traveling Class Marks
- Network Access—Private
- Network Access—Public
- Off-Premises Station
- Subnet Trunking
- Uniform Dial Plan

System Management

This group of features includes support services that provide maintenance and administration for the system. Maintenance and administration systems are used to administer features, perform maintenance, analyze traffic, and take corrective action.

The following features are associated with System Management:

- Customer-Provided Equipment (CPE) Alarm
- Facility Test Calls
- Move Agent From CMS (V3)
- Station Message Detail Recording
- System Measurements

System Status Report

Dialup Administration: Allows an off-premises data terminal user to remotely access the system and perform administrative tasks. All administrative commands used by the System Manager are available to the remote user.

Initialization and Administration System (INADS): Allows administration and maintenance from a remote location. INADS allows the user to access the system and perform administrative tasks assigned to the System Manager. The administrative commands used by the System Manager are also available to the INADS users. INADS can also be used to perform maintenance routines.

Remote Administration: Allows the system to be administered from a remote terminal located on or off the customer's premises. A local System Access Terminal (SAT) is located on-premises within 50 feet of the system cabinet. A terminal located more than 50 feet from the system cabinet is considered remote. A remote administration terminal can be on the same premises as the local SAT, or can be off-premises. The remote terminal performs the same functions as the local SAT.

System Administration: Allows the user to implement (initialize) and administer the voice terminal and system features. System Administration allows the following:

- Initializing the system
- Managing system and voice terminal features on a day-to-day basis
- Performing system backup procedures
- Monitoring, detecting, and determining system performance
- Maintaining system security

Hospitality Services (V3)

This group of features includes support services for the lodging and health industries. These industries such as hotels, motels, and hospitals use the features to better manage their property and provide services to their guest/patients.

The following features are associated with Hospitality Services:

Automatic Wakeup
Do Not Disturb
Property Management System Interface
  Call Rating
  Check-In/Check-Out
  Controlled Restriction
  Housekeeping Status
  Message Waiting Notification
  Room Change/Room Swap
Single Digit Dialing and Mixed Station Numbering

See *AT&T System 75—Feature Description*, 555-200-201, for details on these features and *AT&T System 75—Hospitality Operation*, 555-200-723, for operation information.

AT&T QUOTE Service, a telephone billing information system, is available for organizations that need to bill back or allocate long-distance charges among callers or customers. The customers must make arrangements with their AT&T Account Executive to provide the service. In addition, the customer must arrange for the installation of a dial-up line through the local telephone company.

Call Management System (V3)

This group of features supports services for industries such as airlines and travel agencies that have a large number of similar calls and allows balanced call distribution to a group of voice terminals.

The following features are associated with the Call Management System:

- Abandoned Call Search
- Agent Call Handling
- Automatic Call Distribution (ACD)
- Intraflow and Interflow
- Move Agent From CMS
- Queue Status Indications
- Service Observing

See *AT&T System 75—Feature Description*, 555-200-201, for details on these features and *AT&T System 75—Hospitality Operation*, 555-200-723, for operation information.

Feature Descriptions

The following is a list of all the system features listed alphabetically with a concise definition or description without regard to the major feature groups.

AAR/ARS Partitioning (V3): Provides for the Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) services to be partitioned among as many as four different groups of users within a single System 75 XE.

Abandoned Call Search (V3): Provides identification of abandoned calls. When the calling party on an Automatic Call Distribution (ACD) call abandons (drops) the call while waiting to be connected to an agent, the call is not connected to the agent, and the call is reported to the Call Management System (CMS) as being abandoned.

Abbreviated Dialing: Provides lists of stored numbers that can be accessed to place local, long-distance, and international calls; to activate features; or to access remote computer equipment. Stored numbers can be accessed by voice terminal users and data terminal users. Certain stored numbers can also be accessed by attendants.

Agent Call Handling (V3): Provides Automatic Call Distribution (ACD) agents with the various capabilities required to answer and process ACD calls.

AP Demand Print (V3): Allows the voice terminal user to print his or her own undelivered messages without calling the AP-based Message Center.

Attendant Auto-Manual Splitting: Allows the attendant to announce a call or consult privately with the called party without being heard by the other party on the call.

Attendant Call Waiting: Allows an attendant originated or extended call to a busy single-line voice terminal to wait at the called terminal. The attendant is free to handle other calls.

Attendant Control of Trunk Group Access: Allows the attendant to control trunk groups, and prevents voice terminal users from directly accessing a controlled trunk group.

Attendant Direct Extension Selection With Busy Lamp Field: 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.

Attendant Direct Trunk Group Selection: Allows the attendant direct access to an idle outgoing trunk by pressing the button assigned to the desired trunk group.

Attendant Display: Shows call-related information that helps the attendant to operate the console more efficiently. Also shows personal-service and message information. Information is shown on the 40-character alphanumeric display on the attendant console.

Attendant Recall: 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.

Attendant Release Loop Operation: 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.

Audio Information Exchange (AUDIX) Interface (V3): Provides a communications link between the System 75 XE and the Audio Information Exchange (AUDIX) Interface. AUDIX allows both System 75 XE users and outside callers to write, edit, send, and forward voice messages to other users. In addition, System 75 XE users can also receive and store incoming voice messages from others.

Authorization Codes (V3): Provides the means for extending control of system users' calling privileges.

Automatic Alternate Routing: Provides alternate routing choices for private on-network calls. Also provides digit modification to allow on-network calls to route through the public network when on-network routes are not available.

Automatic Callback: 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.

Automatic Call Distribution (V3): Provides automatic connection of incoming calls to specific splits (hunt groups). Calls to a specific split are automatically distributed among the agents (hunt group members) assigned to that split. Automatic Call Distribution (ACD) data, transmitted from the switch to the Call Management System (CMS), is used to generate various reports on the status of ACD agents, splits, and trunks.

Automatic Circuit Assurance: Assists users in identifying possible trunk malfunctions. The system maintains a record of the performance of individual trunks relative to short and long holding time calls. The system automatically initiates a referral call to an attendant or display-equipped voice terminal user when a possible failure is detected.

Automatic Incoming Call Display: Provides display-equipped voice terminal users, who are already active on a call, with the identity of a second or subsequent caller. The identity is displayed on the terminal's alphanumeric display.

Automatic Route Selection: Routes calls over the public network based on the preferred (normally the least expensive) route available at the time the call is placed.

Automatic Wakeup (V3): Allows attendants, front desk users, and guests to request that a wakeup call be placed automatically to a certain extension number at a later time. Wakeup requests may be placed from 5 minutes to 23 hours and 55 minutes in advance of the call.

Bridged Call Appearance: Allows multi-appearance voice terminal users to have an appearance of another user's primary extension number. The bridged call appearance can be used to originate, answer, and bridge onto calls to or from the other user's primary extension number.

Busy Verification of Terminals and Trunks: Allows attendants and specified multi-appearance voice terminals to make test calls to trunks, voice terminals, and hunt groups [Direct Department Calling (DDC) and Uniform Call Distribution (UCD) groups]. These test calls are used to determine the status of the called voice terminal, hunt group, or trunk.

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

Call Forwarding All Calls: Allows all calls to an extension number to be forwarded to a selected internal extension number, external (off-premises) number, the attendant group, or a specific attendant. This feature is activated or deactivated by dial access code or by a Call Forwarding button.

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

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

Call Waiting Termination: Provides for calls to busy single-line voice terminals to wait and sends a distinctive call waiting tone to the called party.

Centralized Attendant Service: Allows services performed by attendants in a private network of switching systems to be concentrated at a central, or main, location. Each branch in a Centralized Attendant Service (CAS) has its own listed directory number (LDN). Incoming trunk calls to the branch, as well as attendant-seeking voice terminal calls, are routed to the centralized attendants over release link trunks (RLTs).

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

Class of Service: Defines whether or not voice terminal users may access the following features and functions:

- Automatic Callback
- Call Forwarding All Calls
- Data Privacy
- Priority Calling
- Off-Hook Alert
- Console Permission
- Client Room

Code Calling Access: Allows attendants, voice terminal users, and tie trunk users to page with coded chime signals.

Conference—Attendant: 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.

Conference—Terminal: 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.

Consult: Allows a covering user, after answering a coverage call, to call the principal (called party) for private consultation.

Coverage Callback: Allows a covering user to leave a message for the principal (called party) to call the calling party.

Coverage Incoming Call Identification: Allows multi-appearance voice terminal users without a display in a Coverage Answer Group to identify an incoming call to that group.

Customer-Provided Equipment (CPE) Alarm: Provides the customer with an indication that a system alarm has occurred and that the system has attempted to contact a preassigned service organization about the problem. A customer-provided device, such as a lamp or a bell, is used to indicate the alarm situation.

Data Call Setup: Provides three methods to set up a data call: Data Terminal (keyboard) Dialing, Voice Terminal Dialing, or dedicating a voice terminal for data calls. Typically, when a data terminal is available, keyboard dialing is more convenient and requires less steps; therefore, it should be used whenever possible.

Data Hot Line: Provides for automatic nondial placement of a data call to a digital data endpoint, upon receiver pickup.

Data-Only Off-Premises Extensions: Allows users to establish data calls involving data communications equipment (DCE) or Data Terminal Equipment (DTE) that is located remotely from the System 75 XE site using DATAPHONE* data communications terminal or other private line data facilities. A Data-Only Off-Premises Extension uses a Modular Trunk Data Module located on-premises. Communication with the remote data equipment is accomplished through the private line facility linking the on-premises Modular Trunk Data Module and the remote data equipment.

Data Privacy: Protects analog data calls from being disturbed by any of the system's overriding or ringing features. Data Privacy, when activated by a user, denies the system the ability to gain access to, or to superimpose tones onto, the protected call.

Data Restriction: Protects analog data calls from being disturbed by any of the system's overriding or ringing features. Data Restriction, when administered to an extension number or trunk group, denies the system the ability to gain access to, or to superimpose tones onto, the protected call.

DCS Alphanumeric Display for Terminals: Allows calls to or from terminals equipped with alphanumeric displays to have transparency with respect to the display of call-related information.

DCS Attendant Control of Trunk Group Access: Allows an attendant at any node in the DCS to exercise control over an outgoing trunk group at a different node in the cluster.

DCS Attendant Direct Trunk Group Selection: Allows attendants at one node to have direct access to an idle outgoing trunk at a different node in the DCS.

DCS Attendant Display: Provides some transparency with respect to the display of call-related information.

DCS Automatic Callback: Allows a user at one node to make an automatic callback call to a user at another node in the DCS.

DCS Automatic Circuit Assurance: Allows a voice terminal user or attendant at a System 75 XE endpoint node to activate or deactivate Automatic Circuit Assurance (ACA) referral calls for the entire DCS network. This transparency also allows the referral calls to be generated at a node other than the node that detects the problem.

DCS Busy Verification of Terminals and Trunks: Allows attendants and multi-appearance voice terminal users to make test calls to voice terminals and trunk groups that are located at other nodes within the DCS.

DCS Call Forwarding All Calls: Allows all calls to an extension number to be forwarded to a selected extension number within the DCS network, or to an external (off-premises) number. This feature is activated or deactivated by dial access code or by a Call Forwarding button. The feature can be activated or deactivated only by voice terminal users within the DCS.

DCS Call Waiting: Allows calls to busy single-line voice terminals to wait until the called party is available to accept the call.

DCS Distinctive Ringing: Activates ringing device of a called terminal so that the user is aware of the type of incoming call before answering it. Distinctive Ringing functions in a DCS environment as it does within a System 75 XE.

DCS Leave Word Calling: Enables System 75 XE terminal users to leave preprogrammed "call me" messages at other terminals in the DCS network. Messages can be left by calling, called, or covering users.

DCS Multi-Appearance Conference/Transfer: Provides transparency of conference calls and the transfer of calls within a DCS network. A user in the DCS can make conference calls or transfer calls originated from any extension in the DCS network to another extension within the DCS.

DCS Trunk Group Busy/Warning Indication: Provides attendants with a visual warning that the number of busy trunks in a remote group has reached an administered level. A visual indication is also provided when all trunks in a trunk group are busy.

Dial Access to Attendant: 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.

Dial Plan: 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.

Digital Multiplexed Interface: Supports two signaling techniques: Bit Oriented Signaling and Message Oriented Signaling for direct connection to host computers.

Direct Department Calling and Uniform Call Distribution: 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.

Direct Inward Dialing: Connects calls from the public network directly to the dialed extension number without attendant assistance.

Direct Outward Dialing: Allows voice terminal users to access the public network without attendant assistance.

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

Do Not Disturb (V3): Allows guests, attendants, and authorized front desk voice terminal users to request that no calls, other than priority calls, terminate at a particular extension number until a specified time. At the specified time, the system automatically deactivates the feature and allows calls to terminate at the extension normally.

DS1 Tie Trunk Service: Provides for two types of digital tie trunk interfaces: Voice-Grade DS1 and Alternate Voice/Data (AVD) DS1 tie trunks. The Voice-Grade DS1 tie trunks are an alternative to 4-wire analog E&M tie trunks and may be used to interface with other properly-equipped switching systems. AVD DS1 tie trunks permit alternate voice and data calling between System 75s and System 85.

EIA Interface: Provides a lower cost alternative to Digital Terminal Data Modules (DTDMs) and Modular Processor Data Modules (MPDMs), within the system hardware, for interconnection between RS-232 compatible Digital Terminal Equipment (DTE) and the system. The EIA Interface consists of a Data Line circuit pack port and an Asynchronous Data Unit (ADU).

Emergency Access To the Attendant (V3): Provides for emergency calls to be placed to the attendants automatically by the system or dialed by system users, and allows for such calls to receive priority handling by the attendants.

Facility Busy Indication: Provides multi-appearance voice terminal users with a visual indication of the busy or idle status of an extension number, a trunk group, terminating extension group, a hunt group (Direct Department Calling or Uniform Call Distribution 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.

Facility Restriction Levels and Traveling Class Marks: Provides up to eight levels of restriction for users of the Automatic Alternate Routing (AAR) and/or Automatic Route Selection (ARS) features.

Facility Test Calls: Provides a voice terminal user with the capability of making test calls to access specific trunks, touch-tone receivers, time slots, and system tones. The test call is used to make sure the facility is operating properly. 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.

Forced Entry of Account Codes: Requires users to dial an account code when making certain types of outgoing calls. The conditions under which dialing of account codes is required depends on system administration.

Go to Cover: Allows users, when making a call to another internal extension, to send the call directly to coverage.

Hold: Allows voice terminal users to disconnect from a call temporarily, use the voice terminal for other call purposes, and then return to the original call.

Hot Line Service: Allows single-line voice terminal users, by simply lifting the handset, to automatically place a call to a preassigned extension number, public or private network telephone number, or feature access code.

Hunting: Checks for the active or idle status of extension numbers in one or more ordered groups. If all members of a group are active, the call can route to another group through Call Coverage or can wait in a queue for an available group member, if a queue is provided.

Individual Attendant Access: Allows users to access a specific attendant console. Each attendant console can be assigned an individual extension number, to provide access to each individual attendant.

Information System Network (ISN) Interface: The AT&T ISN is a packet switched local area network that links mainframe computers, minicomputers, word processors, storage devices, personal computers, printers, terminals, and communications processors into a single system. The interface to System 75 XE is via an Asynchronous Data Unit (ADU). A Modular Processor Data Module (MPDM) may be used but the ADU is more economical. Also, future versions of the ISN will have integrated ADUs.

Integrated Directory: Allows internal system users with display-equipped terminals 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.

Intercept Treatment: 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.

Intercom—Automatic: Provides a talking path between two voice terminal users. Calling users press the Automatic Intercom button and lift the handset, or vice versa. The called user receives a unique intercom ringing signal, and the status lamp associated with the Dial or Automatic Intercom button, if provided, flashes.

Intercom—Dial: 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 handset, press the Dial Intercom button, and dial the 1- or 2-digit code assigned to the desired party. The called user receives ringing tone, and the status lamp associated with the Intercom button, if provided, flashes.

Inter-PBX Attendant Calls: Allows attendant positions for more than one branch location to be concentrated at one central, or main, location. Each branch location has its own Listed Directory Number (LDN). Incoming trunk calls to the branch location, as well as attendant-seeking voice terminal calls, are routed over tie trunks to the attendants at the main location.

Intraflow and Interflow (V3): Allows Automatic Call Distribution (ACD) calls to be redirected from one split to another split under busy or unanswered conditions. Intraflow provides redirection of ACD calls to other splits within the system. Interflow allows redirection of ACD calls to an external location.

Last Number Dialed: Automatically redials the last number dialed when users press the Last Number Dialed button or dial the Last Number Dialed feature access code.

Leave Word Calling: Allows internal system users to leave a short preprogrammed message for other internal users. Users can activate Leave Word Calling (LWC) at any time during a call attempt.

Line Lockout: 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.

Loudspeaker Paging Access: Provides attendants and voice terminal users dial access to voice paging equipment.

Manual Message Waiting: 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.

Manual Originating Line Service: Connects users to attendant automatically when the user lifts the handset.

Manual Signaling: Allows a voice terminal user to signal another voice terminal user. The receiving voice terminal user hears a 2-second burst of tone.

Modem Pooling: Allows switched connections between digital data endpoints (data modules) and analog data endpoints, and acoustic coupled modems. The analog data endpoint can be either a trunk or line circuit.

Moves Agent From CMS (V3): Allows a Call Management System (CMS) user to move agents from one split to another via the screen on the CMS terminal. This feature gives the user of the CMS screen some of the same capabilities that the System Manager has with the System Access Terminal (SAT). The user of the CMS screen can, with a single request, move one agent or multiple agents from the same split to another split.

Multi-Appearance Preselection and Preference: Provides multi-appearance voice terminal users with options for placing or answering calls on selected appearances.

Multiple Listed Directory Numbers: Allows a publicly published number for each incoming and two-way (incoming side) foreign exchange (FX) and local central office (CO) trunk group assigned to the system. Also allows up to eight Direct Inward Dialing (DID) numbers to be treated as Listed Directory Numbers (LDNs).

Music-on-Hold Access: 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.

Network Access—Private: Allows calls to be connected to the following types of networks:

- Common Control Switching Arrangement (CCSA)
- Electronic Tandem Network (ETN)
- Enhanced Private Switched Communications Service (EPSCS)
- Tandem Tie Trunk Network (TTTN)

Network Access—Public: Provides voice terminal users and attendants with access to and from the public network.

Night Service—Hunt Group (V3): Allows an attendant or a split supervisor to individually assign a hunt group or split to the night service mode. All calls terminating on the hunt group or split in the night service mode will be redirected to the hunt group/split's designated Night Service Extension (NSE).

Night Service—Night Console Service: Directs all calls for the primary and daytime attendant consoles to a night console.

Night Service—Night Station Service: Redirects incoming attendant-seeking trunk calls to designated extension numbers whenever the system is placed in Night Service.

Night Service—Trunk Answer From Any Station: Allows voice terminal users to answer all incoming attendant-seeking calls when the attendant(s) is not on duty and when other voice terminals have not been designated to answer the calls.

Night Service—Trunk Group (V3): Allows an attendant or a designated voice terminal user to individually assign a trunk group or all trunk groups to the night service mode. Specific trunk groups (individually) assigned to Trunk Group Night Service are in the "Individual Trunk Night Service Mode." In this mode, incoming calls made on a specific trunk group will be redirected to its designated Night Service Extension (NSE). Incoming calls on the trunk groups not assigned to Trunk Group Night Service will be processed normally. The specific trunk groups can be assigned to Trunk Group Night Service by pressing the individual Trunk Night Service button(s) on the attendant console or a voice terminal.

Off-Premises Station: Allows a voice terminal located outside the building where the switch is located to be connected to the system. If central office (CO) trunks are used, the voice terminal must be analog and must be FCC-registered.

Permanent Switched Calls: Maintains a call between two data endpoints that should always be connected while the system is active. The specified calls are automatically placed when the system is started or restarted, and remain active until the system becomes inactive.

Personal Central Office Line: Provides a dedicated trunk for direct access to or from the public network for multi-appearance voice terminal users.

Personalized Ringing: Allows users of certain voice terminals to uniquely identify their own calls. Each user can choose one of a number of possible ringing patterns.

Power Failure Transfer: Provides service to and from the local telephone company central office (CO), including incoming Wide Area Telecommunications Service (WATS), during a power failure.

Priority Calling: Provides a special form of ringing between internal voice terminal users. The called voice terminal user receives a distinctive 3-burst ringing signal.

Privacy—Attendant Lockout: Prevents an attendant from reentering a multiple-party connection held on the console unless recalled by a voice terminal user.

Privacy—Manual Exclusion: Allows multi-appearance voice terminal users to keep other users with appearances of the same extension number from bridging onto an existing call.

Property Management System Interface (V3): Provides a communications link between the System 75 XE and a customer-owned Property Management System (PMS). The PMS allows a customer to control certain features used in both a hospital-type and a hotel/motel-type environment.

Queue Status Indications (V3): Provides indications of queue status for Automatic Call Distribution (ACD) calls based on the number of calls in queue and time in queue. These indications are provided via lamps assigned to the terminals or consoles of split agents or supervisors. In addition, an auxiliary warning lamp can be provided to track queue status based on time in queue. Also, display-equipped voice terminals and consoles can display the time in queue of a split's oldest call and the number of calls in that split's queue.

Recall Signaling: Allows a single-line voice terminal user, who is 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 twice.

Recorded Announcement: Provides a recorded announcement to the following types of calls:

- Direct Inward Dialing calls that cannot be completed as dialed
- Incoming Private Network Access calls that cannot be completed as dialed
- Direct Department Calling and Uniform Call Distribution calls that have been in queue for an assigned interval
- Automatic Call Distribution (ACD) calls that have been in queue for an assigned interval (V3 only)
- Any call whose destination is a Recorded Announcement (V3 only)

Recorded Telephone Dictation Access: Permits voice terminal users, including Remote Access and incoming tie trunk users, to access dictation equipment.

Remote Access: Permits callers from the public network to access the system and then use its features and services.

Restriction—Controlled: 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.
- Station-to-Station (V3)—The voice terminal cannot receive or place station-to-station calls. Such call attempts receive intercept treatment.
- Termination (V3)—The voice terminal cannot receive any calls. Incoming calls are routed to the attendant, are redirected via Call Coverage, or receive intercept treatment.

Restriction—Miscellaneous Terminal: Restricts callers at specified voice terminals from accessing certain other voice terminals.

Restriction—Miscellaneous Trunk: Restricts users at specified voice terminals from accessing certain trunk groups, such as Wide Area Telecommunications Service (WATS).

Restriction—Toll/Code: 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).

Restriction—Voice Terminal—Inward: 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.

Restriction—Voice Terminal—Manual Terminating Line: 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.

Restriction—Voice Terminal—Origination: Restricts callers at specified voice terminals from originating calls. Voice terminal users can receive calls.

Restriction—Voice Terminal—Outward: Prevents specified voice terminal users from placing calls to the public network. Calls can be placed to other voice terminal users, to the attendant, and over tie trunks.

Restriction—Voice Terminal—Termination: Restricts voice terminal users on specified extension numbers from receiving any calls. The restricted users can, however, originate calls.

Ringback Queuing: 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 ringing signal (Priority Calling) when called back.

Rotary Dialing: Allows rotary dialing voice terminal to be used with a System 75 XE.

Send All Calls: Allows users to temporarily direct all incoming calls to coverage regardless of the assigned Call Coverage redirection criteria. Send All Calls also allows covering users to temporarily remove their voice terminals from the coverage path.

Senderized Operation: 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.

Service Observing (V3): Allows a specified user, such as a supervisor, to observe a call that involves other users while the call is in progress. While observing a call, the specified user can toggle between a listen-only and a listen/talk connection to the call.

Single Digit Dialing and Mixed Station Numbering (V3): Allows easy access to internal hotel/motel services and provides the capability to associate room numbers with guest room voice terminals.

SMDR Account Code Dialing: Allows certain calls to be associated with a particular project or account number. This is accomplished by dialing specified account codes before making outgoing calls. This information is recorded by the Station Message Detail Recording (SMDR) feature and can be used later for accounting and/or billing purposes.

Station Message Detail Recording: Records detailed call information on all incoming and outgoing calls on specified trunk groups and sends this information to a Station Message Detail Recording (SMDR) output device. Internal calls are not recorded. The SMDR output device provides a detailed printout which can be used by the System Manager to compute call costs, allocate charges, analyze calling patterns, and keep track of unnecessary calls.

Straightforward Outward Completion: Allows an attendant to complete an outgoing trunk call for a voice terminal user, without requiring the voice terminal user to hang up.

Subnet Trunking: Provides modification of the dialed number so an Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS) call can route over trunk groups that terminate in switches with different dial plans.

System Measurements: Provides reports on trunk group usage, hunt group usage and efficiency, attendant group activity and efficiency, and security violations.

System Status Report: Allows the user to view data associated with attendants, major and minor alarms, and traffic measurements. The information is displayed on the System Access Terminal (SAT) and presents a basic picture of the System 75 XE condition. The report can only be displayed by the System Manager and maintenance personnel.

Temporary Bridged Appearance: Allows multi-appearance voice terminal users in a Terminating Extension Group or Personal Central Office Line Group to bridge onto an existing group call. If a call has been answered using the Call Pickup feature, the originally called party can bridge onto the call. Also allows a called party to bridge onto a call that redirects to coverage before the called party can answer it.

Terminating Extension Group: Allows an incoming call to ring as many as four voice terminals at one time. Any user in the group can answer the call.

Through Dialing: 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.

Timed Reminder: Automatically rings 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

Touch-Tone Dialing: 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.

Transfer: Allows voice terminal users to transfer trunk or internal calls to other voice terminals within the system without attendant assistance.

Trunk Group Busy/Warning Indicators to Attendant: Provides the attendant with a visual indication 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.

Trunk Identification By Attendant: Allows an attendant or display-equipped voice terminal user to identify a specific trunk being used on a call. This capability is provided by assigning a Trunk ID button to the attendant console or voice terminal.

Trunk-to-Trunk Transfer: Allows the attendant or voice terminal user to connect an incoming trunk call to an outgoing trunk.

Uniform Dial Plan: Provides a common 4- or 5-digit dial plan that can be shared among a group of switches. Interswitch dialing and intraswitch dialing both require 4- or 5-digit dialing. The Uniform Dial Plan (UDP) is used with Main/Satellite/Tributary and Distributed Communications System (DCS) configurations. Additionally, UDP can be used alone to provide uniform 4- or 5-digit dialing between two or more private switching systems without Main/Satellite/Tributary or DCS configurations.

Voice Message Retrieval: Allows attendants, voice terminal users, and remote access users to retrieve Leave Word Calling (LWC) and Call Coverage messages in the form of a voice output.

Voice Terminal Display: Provides multi-appearance voice terminal users with updated call and message information. This information is displayed on a display-equipped terminal. The information displayed depends upon the display mode selected by the user.

For more detailed information on the individual features, refer to the *AT&T System 75—Feature Description*, 555-200-201.

TABLE 2-A. Features Available

FEATURE	FEATURE GROUP*
AAR/ARS Partitioning	NS
Abandoned Call Search	VM
Abbreviated Dialing	VM
Agent Call Handling	VM
AP Demand Print	VM
Attendant Auto-Manual Splitting	VM
Attendant Call Waiting	VM
Attendant Control of Trunk Group Access	VM
Attendant Direct Extension Selection With Busy Lamp Field	VM
Attendant Direct Trunk Group Selection	VM
Attendant Display	VM
Attendant Recall	VM
Attendant Release Loop Operation	VM
Audio Information Exchange (AUDIX) Interface	VM
Authorization Codes	VM
Automatic Alternate Routing (AAR)	NS
Automatic Callback	VM
Automatic Call Distribution	VM
Automatic Circuit Assurance	NS
Automatic Incoming Call Display	VM
Automatic Route Selection	NS
Automatic Wakeup	HS
Bridged Call Appearance	VM
Busy Verification of Terminals and Trunks	VM
Call Coverage	VM
Call Forwarding All Calls	VM
Call Park	VM
Call Pickup	VM
Call Waiting Termination	VM

* Legend
HS — Hospitality Services
NS — Network Services
VM — Voice Management

TABLE 2-A. Features Available (Contd)

FEATURE	FEATURE GROUP*
Centralized Attendant Service (CAS)	VM
Class of Restriction	VM
Class of Service	VM
Code Calling Access	VM
Conference—Attendant	VM
Conference—Terminal	VM
Consult	VM
Coverage Callback	VM
Coverage Incoming Call Identification	VM
Customer Provided Equipment Alarm	SM
Data Call Setup	DM
Data Hot Line	DM
Data-Only Off-Premises Extensions	DM
Data Privacy	DM
Data Restriction	DM
DCS Alphanumeric Display for Terminals	NS
DCS Attendant Control of Trunk Group Access	NS
DCS Attendant Direct Trunk Group Selection	NS
DCS Attendant Display	NS
DCS Automatic Callback	NS
DCS Automatic Circuit Assurance	NS
DCS Busy Verification of Terminals and Trunks	NS
DCS Call Forwarding All Calls	NS
DCS Call Waiting	NS
DCS Distinctive Ringing	NS
DCS Leave Word Calling	NS
DCS Multi-Appearance Conference/Transfer	NS
DCS Trunk Group Busy/Warning Indication	NS

* Legend
- DM — Data Management
- NS — Network Services
- SM — System Management
- VM — Voice Management

TABLE 2-A. Features Available (Contd)

FEATURE	FEATURE GROUP*
Dial Access to Attendant	VM
Dial Plan	VM
Dialup Administration	SM
Digital Multiplexed Interface (DMI)	DM
Direct Department Calling and Uniform Call Distribution	VM
Direct Inward Dialing	VM
Direct Outward Dialing	VM
Distinctive Ringing	VM
Do Not Disturb	HS
DS1 Tie Trunk Service	DM
EIA Interface	DM
Emergency Access to the Attendant	VM
Facility Busy Indication	NS
Facility Restriction Levels and Traveling Class Marks	NS
Facility Test Calls	SM
Forced Entry of Account Codes	VM
Go to Cover	VM
Hold	VM
Hot Line Service	VM
Hunting	VM
Individual Attendant Access	VM
Information System Network (ISN) Interface	DM
Initialization and Administration System (INADS)	SM
Integrated Directory	VM
Intercept Treatment	VM
Intercom—Automatic	VM
Intercom—Dial	VM
Inter-PBX Attendant Calls	VM
Intraflow and Interflow	VM
Last Number Dialed	VM
Leave Word Calling	VM

* Legend
- DM — Data Management
- HS — Hospitality Services
- NS — Network Services
- SM — System Management
- VM — Voice Management

TABLE 2-A. Features Available (Contd)

FEATURE	FEATURE GROUP*
Line Lockout	VM
Loudspeaker Paging Access	VM
Manual Message Waiting	VM
Manual Originating Line Service	VM
Manual Signaling	VM
Modem Pooling	DM
Move Agents From CMS	VM
Multi-Appearance Preselection and Preference	VM
Multiple Listed Directory Numbers	VM
Music-on-Hold Access	VM
Network Access—Private	NS
Network Access—Public	NS
Night Service—Hunt Group	VM
Night Service—Night Console Service	VM
Night Service—Night Station Service	VM
Night Service—Trunk Answer From Any Station	VM
Night Service—Trunk Group	VM
Off-Premises Station	NS
Permanent Switched Calls	DM
Personal Central Office Line (PCOL)	VM
Personalized Ringing	VM
Power Failure Transfer	VM
Priority Calling	VM
Privacy—Attendant Lockout	VM
Privacy—Manual Exclusion	VM
Property Management System Interface	HS
Queue Status Indications	VM
Recall Signaling	VM
Recorded Announcement	VM
Recorded Telephone Dictation Access	VM

* Legend
- DM — Data Management
- HS — Hospitality Services
- NS — Network Services
- VM — Voice Management

TABLE 2-A. Features Available (Contd)

FEATURE	FEATURE GROUP*
Remote Access	VM
Remote Administration	SM
Restriction—Controlled	VM
Restriction—Miscellaneous Terminal	VM
Restriction—Miscellaneous Trunk	VM
Restriction—Toll/Code	VM
Restriction—Voice Terminal	VM
Ringback Queuing	VM
Rotary Dialing	VM
Send All Calls	VM
Senderized Operation	VM
Service Observing	VM
Single Digit Dialing and Mixed Station Numbering	HS
SMDR Account Code Dialing	SM
Station Message Detail Recording (SMDR)	VM
Straightforward Outward Completion	VM
System Administration	SM
System Measurements	SM
System Status Report	SM
Temporary Bridged Appearance	VM
Terminating Extension Group	VM
Through Dialing	VM
Timed Reminder	VM
Touch-Tone Dialing	VM
Transfer	VM
Trunk Group Busy/Warning Indicators to Attendant	VM
Trunk Identification by Attendant	VM
Trunk-to-Trunk Transfer	VM
Uniform Dial Plan	NS
Voice Message Retrieval	VM
Voice Terminal Display	VM

* Legend
HS — Hospitality Services
NS — Network Services
SM — System Management
VM — Voice Management

FUNCTIONAL DESCRIPTION

CONTENTS

LIST OF FIGURES

FUNCTIONAL DESCRIPTION

This part briefly describes the system operation. A general description of the individual circuits is provided under the heading ***Port Circuits*** in the ***Switching Network*** part of this section.

Switch Hardware

Figure 3-1 shows the system communications switch. The basic switch hardware consists of the following:

- Switching network
- Switch processing element (SPE)
- Distributed Communication System (DCS) Interface
- Call Management System (CMS) Interface (V3)
- Audio Information Exchange (AUDIX) Interface (V3)
- Property Management System (PMS) Interface (V3)
- Processor/Maintenance circuit
- Power system

Switching Network

The System 75 XE uses distributed processing techniques to provide switched voice and data services. The switching network consists of the following:

- Time division multiplex (TDM) bus
- Port circuits
- Service circuits

The port circuits connect terminals and external communications facilities to the TDM bus. The TDM bus connects the port circuits to each other and to the SPE through the network control circuit. The service circuits provide tone sources, receivers, detectors, pooled modems, speech synthesis, and announcements. The SPE provides high level control of the port and service circuits.

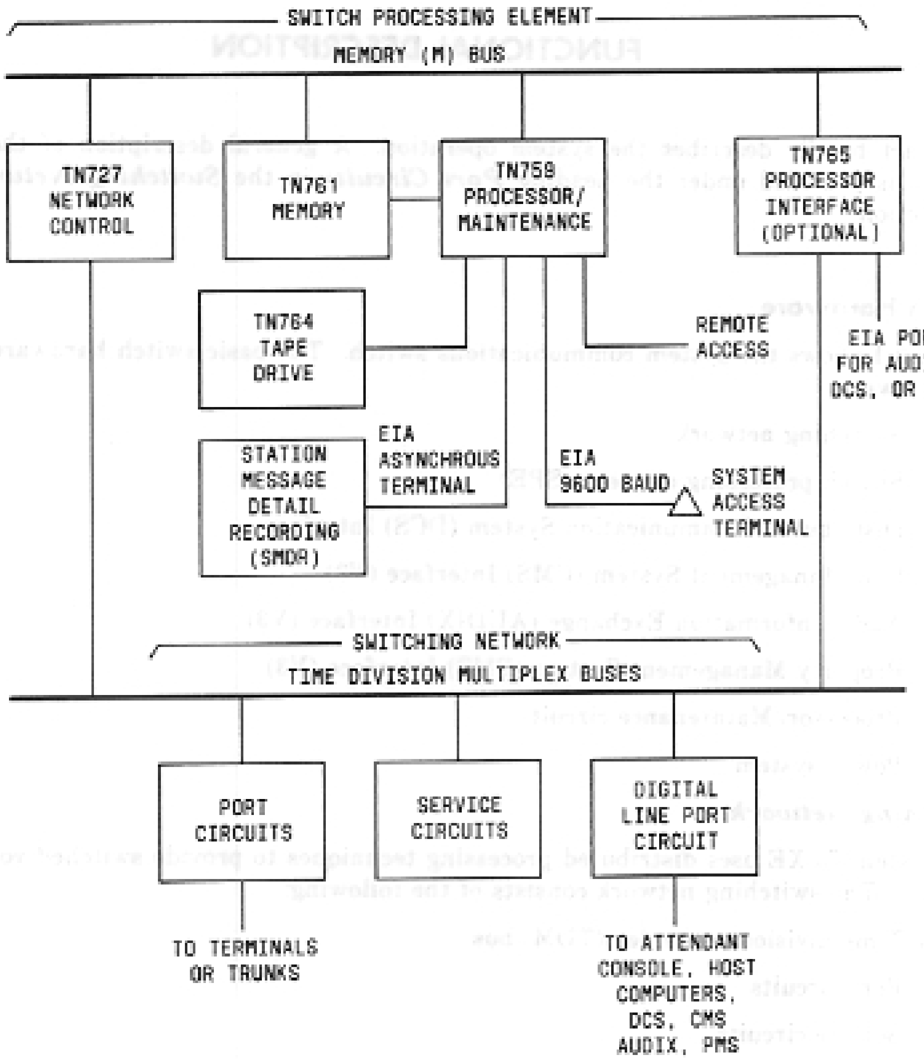


Figure 3-1. Communications Switch

Time Division Multiplex Bus

The TDM bus consists of two identical 8-bit buses that are referred to as Bus A and Bus B. The port circuit packs place digitized voice [pulse code modulated (PCM)] signals on the bus. The TDM bus carries control channel information. The Common Channel Message Set (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 each bus to provide a total of 512 time slots for voice conversation, data calls, and control information. Each time slot is 488 nanoseconds wide. Time slots are generated as shown in Figure 3-2.

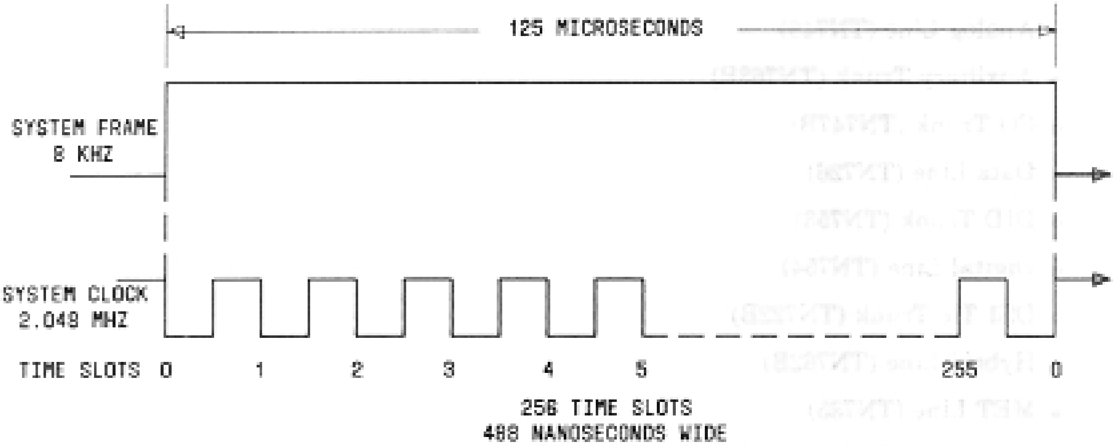


Figure 3-2. TDM Bus Time Slot Generation

Two time slots are required for a 2-party conversation. Each party transmits (talks) on one time slot and receives (listens) on another. The 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 241 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. Seven slots are reserved for future use.

Physical Characteristics: The TDM bus runs continuously through all the cabinets as shown in Figure 3-3. The total length is about 15 feet for a 4-cabinet system. The bus is driven from any of the circuit pack slots in the cabinets. Similarly, a signal on the bus can be received by any circuit pack.

Within a cabinet, the bus is printed on one side of the backplane; the other side is solid ground. A ground plate is required between cabinets to maintain ground integrity and to stablize the cabinets. Shielded ribbon cables are used between cabinets 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 cabinets and the shielded ribbon 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.

Port Circuits

The following port circuits provide the link between trunks or external communications equipment and the TDM buses (see Figure 3-4):

- Analog Line (TN769) (V3)
- Analog Line (TN742)
- Analog Line (TN746)
- Auxiliary Trunk (TN763B)
- CO Trunk (TN747B)
- Data Line (TN726)
- DID Trunk (TN753)
- Digital Line (TN754)
- DS1 Tie Trunk (TN722B)
- Hybrid Line (TN762B)
- MET Line (TN735)
- Tie Trunk (TN760B)—Used for Release Links

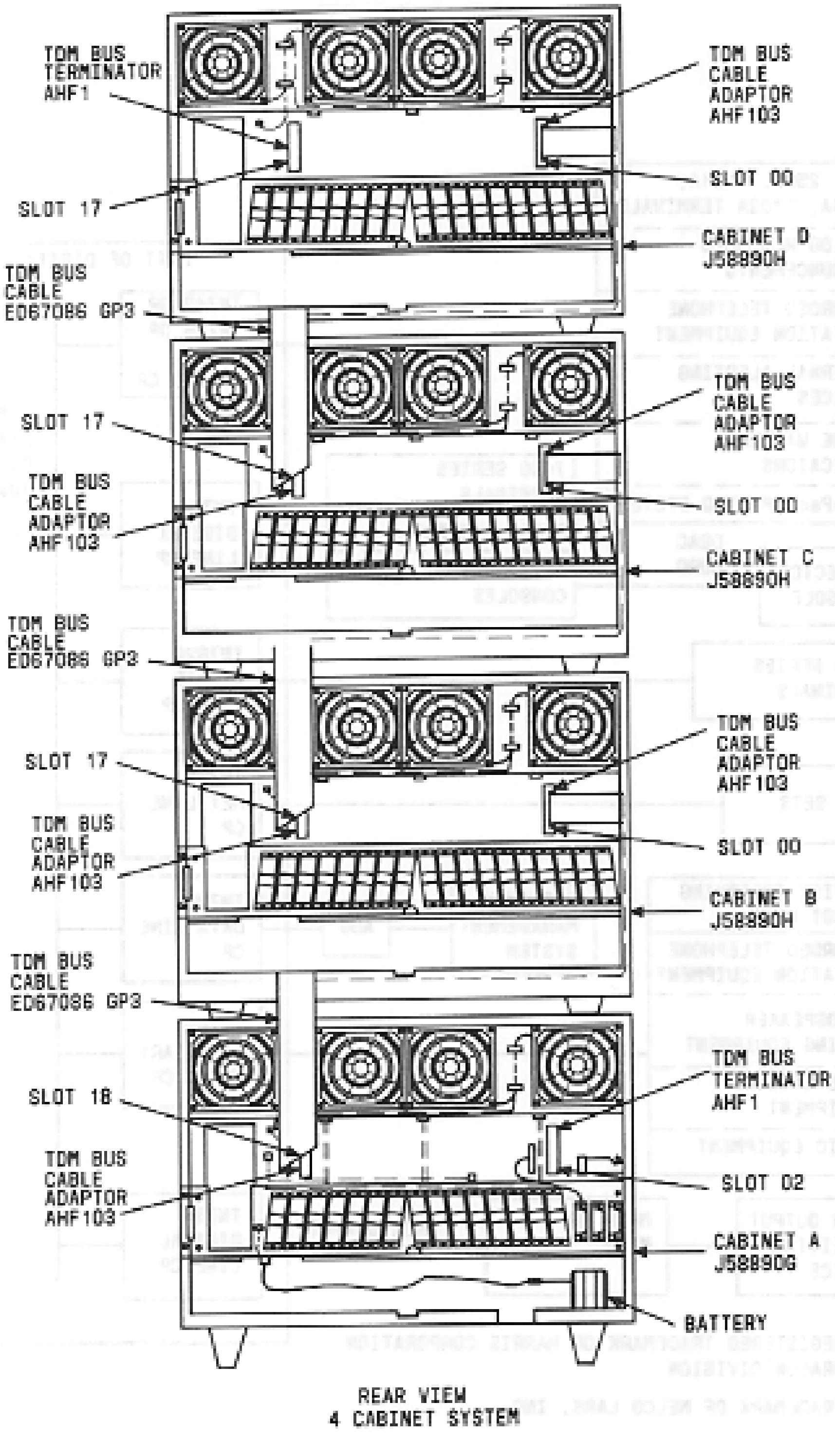


Figure 3-3. TDM Bus Wiring Diagram—Fully Loaded Configuration

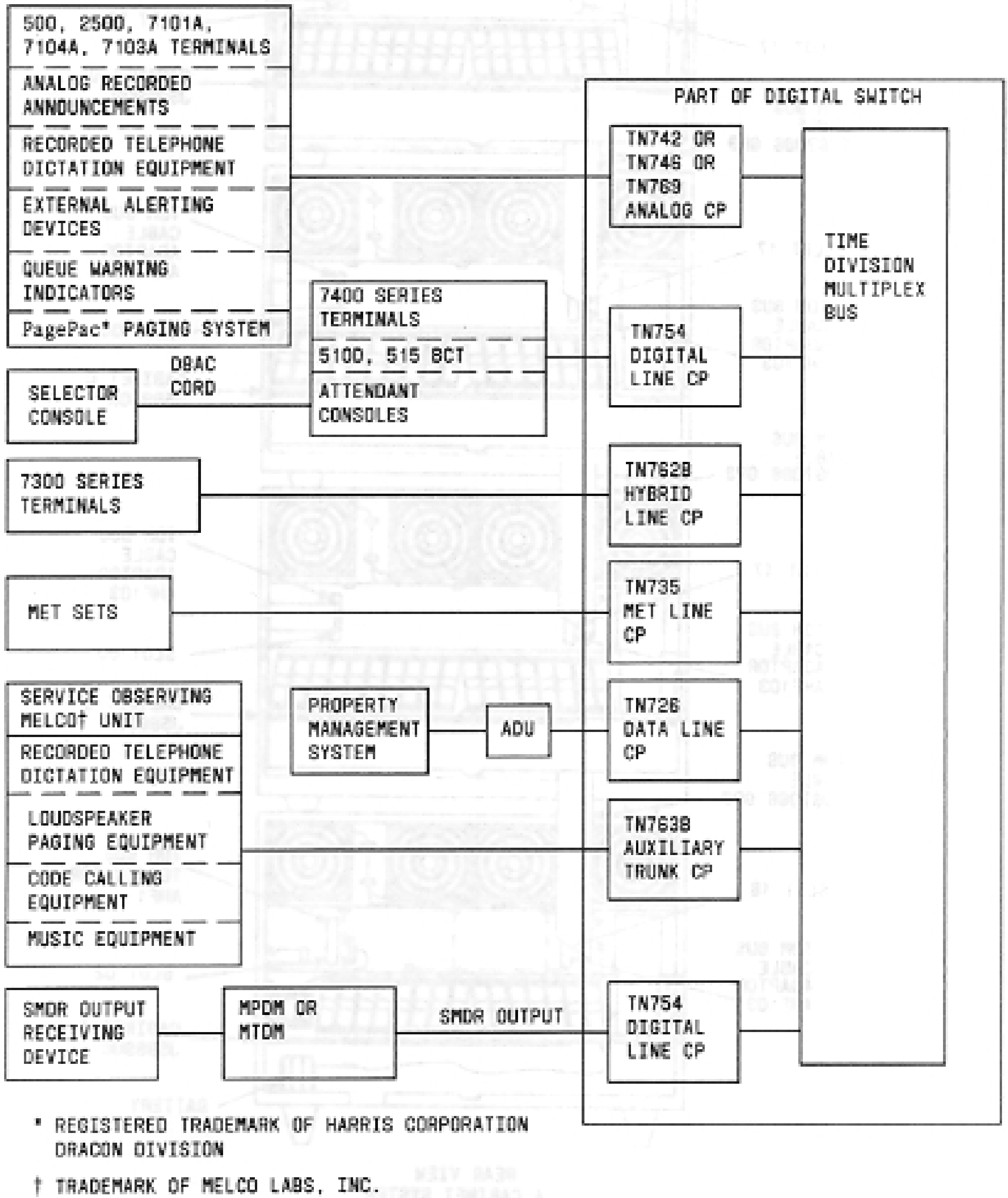


Figure 3-4. Equipment Connected to the Digital Switch by Port Circuits (Sheet 1 of 4)

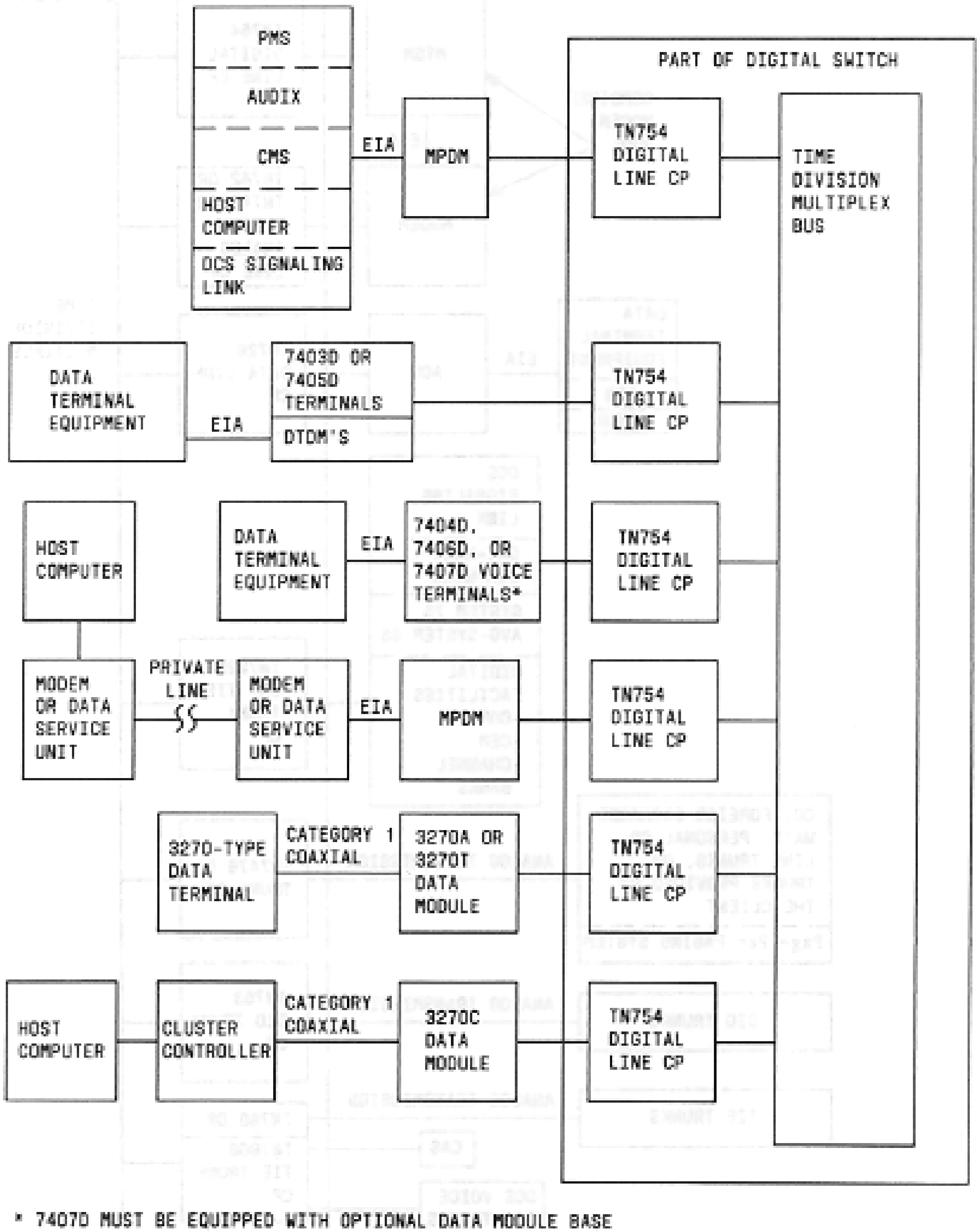


Figure 3-4. Equipment Connected to the Digital Switch by Port Circuits (Sheet 2 of 4)

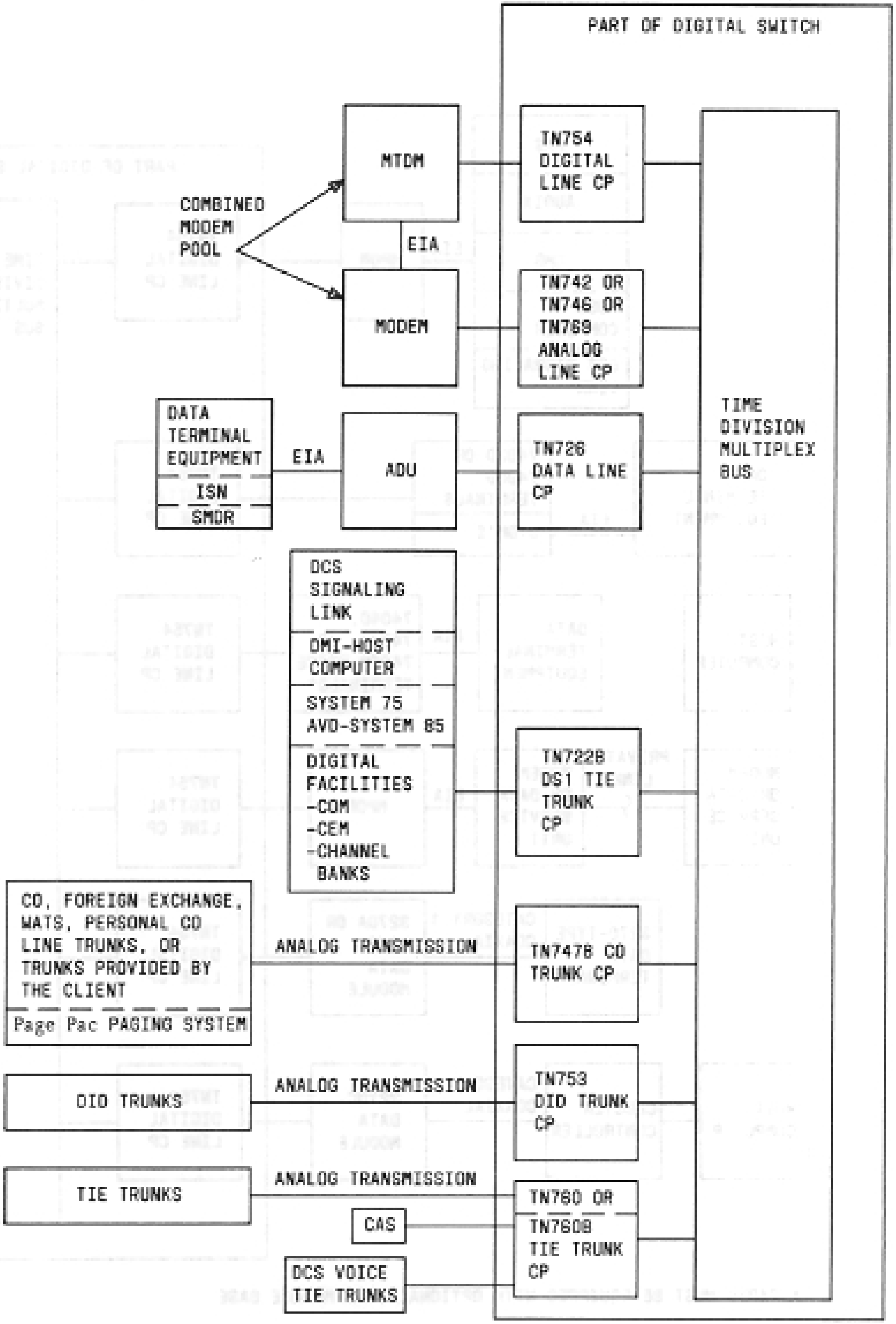


Figure 3-4. Equipment Connected to the Digital Switch by Port Circuits (Sheet 3 of 4)

LEGEND:

ADU - ASYNCHRONOUS DATA UNIT
AUDIX - AUDIO INFORMATION EXCHANGE
BCT - BUSINESS COMMUNICATIONS TERMINAL
CAS - CENTRALIZED ATTENDANT SERVICE
CDM - CHANNEL DIVISION MULTIPLEXING
CMS - CALL MANAGEMENT SYSTEM
CO - CENTRAL OFFICE
CP - CIRCUIT PACK
DCS - DISTRIBUTED COMMUNICATIONS SERVICE
DID - DIRECT INWARD DIALING
DMI - DIGITAL MULTIPLEXED INTERFACE
DTDM - DIGITAL TERMINAL DATA MODULE
EIA - ELECTRONICS INDUSTRIES ASSOCIATION
[CONNECTIONS SHOWN ARE 50 FT (15.3M) MAX.]
ISN - INFORMATION SYSTEMS NETWORK
MET - MULTIBUTTON ELECTRONIC TELEPHONE
MPDM - MODULAR PROCESSOR DATA MODULE
MTDM - MODULAR TRUNK DATA MODULE
PMS - PROPERTY MANAGEMENT SYSTEM
PT - PERSONAL TERMINAL
SMDR - STATION MESSAGE DETAIL RECORDING
SSI - STANDARD SERIAL INTERFACE
WATS - WIDE AREA TELECOMMUNICATIONS SERVICE

Figure 3-4. Equipment Connected to the Digital Switch by Port Circuits (Sheet 4 of 4)

Port Circuits (Contd)

Each of the System 75 XE port circuit packs contains a number of elements common to all port circuits (see Figure 3-5). 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)

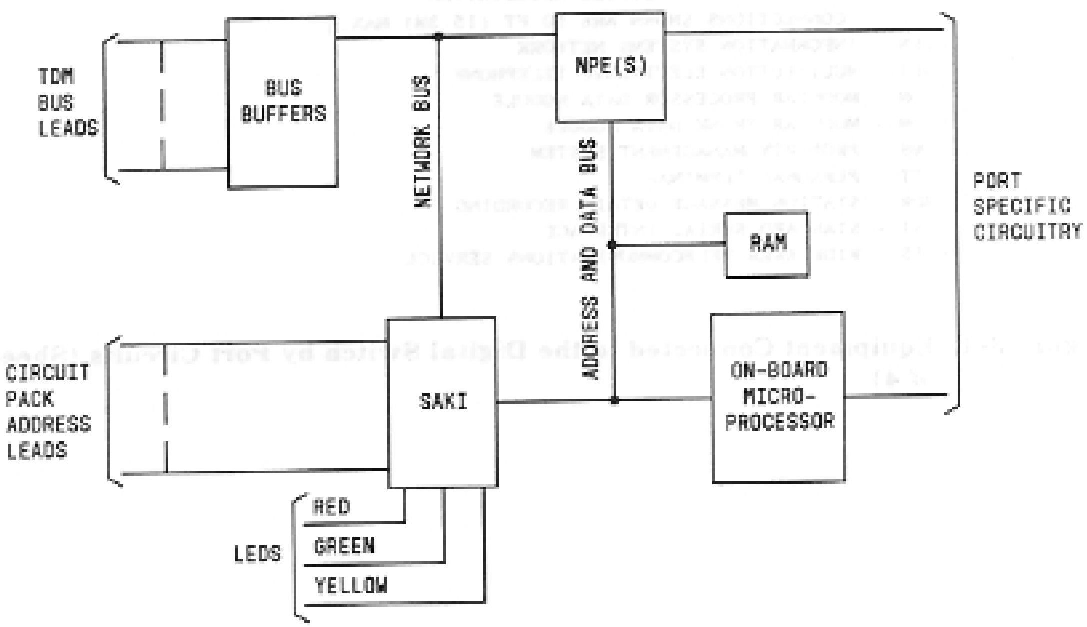


Figure 3-5. Port Circuit Pack Common Elements

Bus Buffers: The bus buffers are the digital interface between the backplane TDM bus wires and the on-board circuitry. They receive or transmit on either of the two 8-bit TDM buses.

SAKI: The SAKI is the circuit pack's interface to control channel information on the TDM bus. The SAKI picks off the TDM bus common channel information intended for the circuit pack and gives it to the on-board microprocessor. When the on-board microprocessor has control channel information to transmit, it gives it to the SAKI which transmits it over the TDM bus.

The SAKI also does the following:

- 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 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. Some port circuit packs contain additional microprocessors.

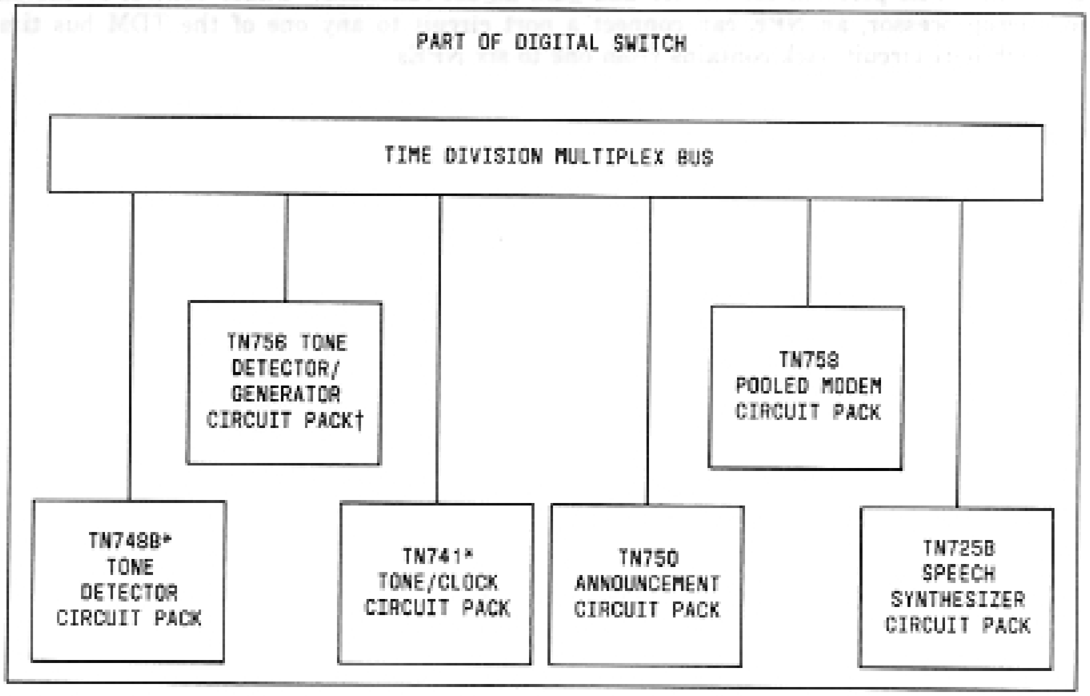
NPEs: The NPEs perform conference and gain-adjust functions. Under control of the on-board microprocessor, an NPE can connect a port circuit to any one of the TDM bus time slots. Each port circuit pack contains from one to six NPEs.

Service Circuits

The service circuits that connect to the TDM bus are as follows (see Figure 3-6):

- Tone Detector/Generator (TN756)
- Tone Detector (TN748B)
- Tone/Clock (TN741)
- Pooled Modem (TN758)
- Announcement (TN750) (V3)
- Speech Synthesizer (TN725B)

As shown in Figure 3-6, the service circuits do not connect to any outside equipment. The service circuit packs contain basically the same common elements as the port circuit packs. These common elements perform similar functions for the service circuit packs. The port and service circuit packs are located in the TDM bus slots.



* REQUIRED WHEN OPTIONAL DS1 TIE TRUNK (TN722B) IS PROVIDED.
† NORMALLY PROVIDED IN SYSTEM 75 XE.

Figure 3-6. Service Circuits

Switch Processing Element

The main components of the switch processing element (SPE) are as follows (see Figure 3-1).

- Processor (TN759)
- Memory (TN761)
- Network Control (TN727)
- Processor Interface (TN765) (Optional)

These components are interconnected by the 16-bit M (memory) bus located on the backplane. The M bus, containing 24 address and 30 interrupt and control lines, provides a communication link among the control circuit packs.

The processor performs high-level call processing by executing programs stored in memory. It monitors and controls port-to-port connections, provides status indications to users, and initiates the operations required to implement system features. In addition, the processor provides the following:

- An interface to the System Access Terminal (SAT) that is used to manage the system
- An interface to a Station Message Detail Recording output device
- An asynchronous 212A serial channel interface that allows Remote Dial-up for system administration and dial-out for alarm reporting
- A tape drive interface that controls the direct data transfer between the memory circuit and the tape drive
- A maintenance circuit that provides alarm LEDs for system status, monitors and controls processor circuit conditions, and originates alarms.

The memory circuit pack contains 4 megabytes of Dynamic Random Access Memory (RAM) that stores the system translations including addresses of equipment connected to the switch through the port circuit packs and call processing software.

The Network Control circuit is the interface to the TDM bus. It monitors the port circuit packs for activity and transfers this information to the processor. Information from the processor circuit is transferred back to the port circuits through the network control.

The optional processor interface is a microprocessor communications interface that supports the physical, link, and packet layers of the BX.25 protocol. It provides four data links to the TDM bus to allow access to 3B2 Applications Processor (AP), Distributed Communications System (DCS), Call Management System (CMS), Property Management System (PMS), and Audio Information Exchange (AUDIX) services.

Tape Drive

The tape drive circuit pack provides a nonvolatile system bootstrap and translation storage device. It emulates both the incremental and streaming operating modes, and the tape cartridge stores up to 18 megabytes of data.

AP, CMS, DCS, PMS, and AUDIX Interface

The interface to the 3B2 AP, the DCS, the CMS, the PMS, and the AUDIX services is the optional processor interface circuit pack. As shown in Figure 3-1, the processor interface connects to the M bus and provides two arrangements for connecting to CMS, DCS, or AUDIX service. With one arrangement, a single EIA port is used to connect directly to CMS, DCS, or AUDIX service. The other arrangement requires a digital line port circuit along with a trunk data module to access a CMS, DCS, or AUDIX application.

Power Supply

A single, plug-in, multi-output power supply, located in a slot position in each cabinet, provides the total power for the System 75 XE. The input to each power supply is a 60 Hertz 120 volts ac source. The power supply output voltages, consisting of ±5 volts dc, -48 volts dc, +12 volts dc, ringing voltage, and battery charging voltage, are distributed on the cabinet backplane to the slot positions (see Figure 3-7). These output voltages are maintained for 250 milliseconds during power interruptions. Batteries, located external to the power supply in the control cabinet, enable the power supply to provide a 2-minute battery reserve holdover to the control circuit packs and fans during power failure beyond 250 milliseconds.

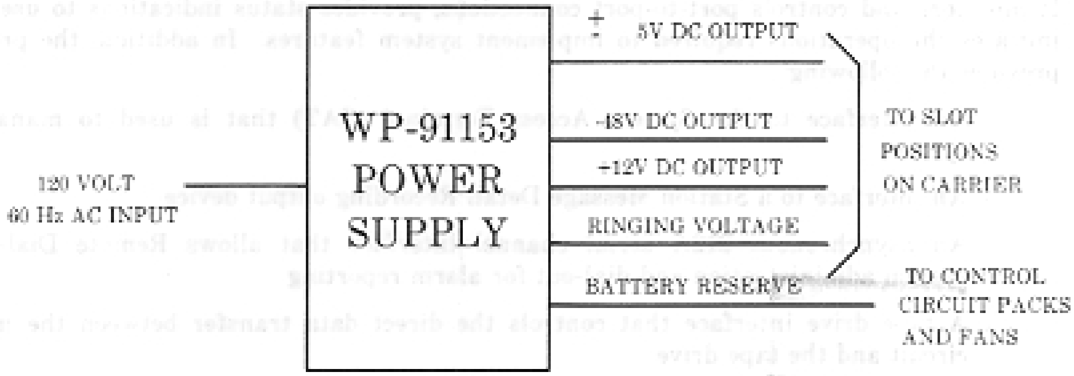


Figure 3-7. Power Supply

Call Processing

The number of calls that can be processed per hour by the system varies according to the mix of voice terminals (analog, hybrid, and digital) in the system. A typical system can process up to 3600 calls per hour.

The following is a step-by-step description of a call originated by a System 75 XE digital voice terminal to another System 75 XE 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 Digital Line port controller detects that the voice terminal user has lifted the handset.
2. The Digital Line port controller sends an off-hook uplink message to Network Control.
3. The Network Control stores the message for the SPE.
4. The SPE software retrieves the message and interprets it as a call origination.
5. The SPE sends downlink messages through 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 caller's terminal.
    b. The second set of messages instructs 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 digit as it is dialed.
6. When the caller dials the first digit, the terminal converts the analog touch-tone signal to a digital signal and the Digital Line port circuit places the digital signal on the previously allocated time slot for that port.
7. The touch-tone detector, listening on the same time slot, interprets the tone. The Tone Detector port controller sends the value of the digit in an uplink message through Network Control to the SPE.
8. The SPE sends a downlink message to the Digital Line port controller instructing it to stop listening to dial tone on the dedicated time slot.
9. The SPE analyzes the first digit, determines that the call is being placed to another System 75 XE extension, and continues collecting digits.
10. When the SPE collects enough digits to identify an extension (as specified in translations), it discontinues collecting digits.
11. 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 dialed extension number 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. When intercept tone is applied, the caller hangs up. Continue with the description from step 19.

12. The SPE determines if the called digital voice terminal has an available call appearance and sends a message to the Tone/Clock port controller to place ringback or busy tone, as appropriate, on the time slot assigned to the originating port. When busy tone is applied, the caller hangs up. Continue with the call processing from step 19.

13. The SPE sends a downlink message through 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 voice terminal.

14. When the called party lifts the handset, the Digital Line port controller sends an off-hook message to the SPE.

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

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

17. 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 caller.

18. 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.

19. When one of the parties hangs up, the associated Digital Line port controller sends an on-hook message through the Network Control to the SPE.

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

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

Private Network Configurations

Included in this section is a general description of private network configurations with four specific examples involving the System 75 XE: Electronic Tandem Network (ETN), Distributed Communications System (DCS), Main/Satellite/Tributary, and Tandem Tie Trunk Network (TTTN).

Do not assume that the system has any capabilities other than those explicitly stated herein. Refer to *AT&T System 75—Network and Data Services*, 555-025-201, for differences between System 75 XE and other AT&T systems. (Check *AT&T System 75—Documentation*, 555-200-010, for the availability of this document.)

A private network is a configuration of trunk and switching facilities dedicated to the use of a business or organization. Private networks benefit customers who have moderate to heavy calling between locations. A private network can have as few as two switches or it can have hundreds of switches located throughout the country. The following configurations make it possible for organizations of all sizes to realize the benefits of a private network.

- DCS—Serves the needs of customers with several locations in a small or large geographic area. A Distributed Communications System appears as a single switch with respect to certain attendant and voice terminal features.
- ETN—Serves the needs of customers with many locations in a large geographic area.
- Main/Satellite/Tributary—Serves the needs of customers with a few locations in a small geographic area.

The System 75 XE also can be used within a TTTN. A TTTN is a nonhierarchical network of tie trunks interconnecting three or more switches. User dialing into each switch in the call's path is required. That is, the user at one switch dials the trunk access code for a tie trunk group to another switch, receives dial tone from that switch, and then dials another trunk access code to reach another switch. When dial tone is received from the final (desired) switch, the user dials the desired extension number.

Distributed Communications System (DCS)

DCS is a cluster of two or more switches. It provides transparency in certain attendant and voice terminal features as if the cluster were a single large switch. Transparency exists when a user at one switch can call or activate a feature towards a user at a different switch without noticing a difference in operation. This provides simplified dialing procedures between locations and also provides the convenience of using some of the system's features between locations.

DCS switches are interconnected by tie trunks for voice communications and data links for control and feature information. The data links, also called DCS signaling links, support the feature transparency. Figure 3-8 shows examples of DCS configurations.

Some of the applications of the DCS configuration are as follows:

- In a campus-type environment that has two or more separate buildings and the switches are connected by local cable
- In a larger area such as a city, several states, or even the entire country, where the switches are separated by distances too great for local cable and may be connected to different central offices

For a detailed description of the DCS, refer to *AT&T System 75—Feature Description*, 555-200-201, and *AT&T System 75—Network and Data Services*, 555-025-201. (Check *AT&T System 75—Documentation*, 555-200-010, for the availability of this document.)

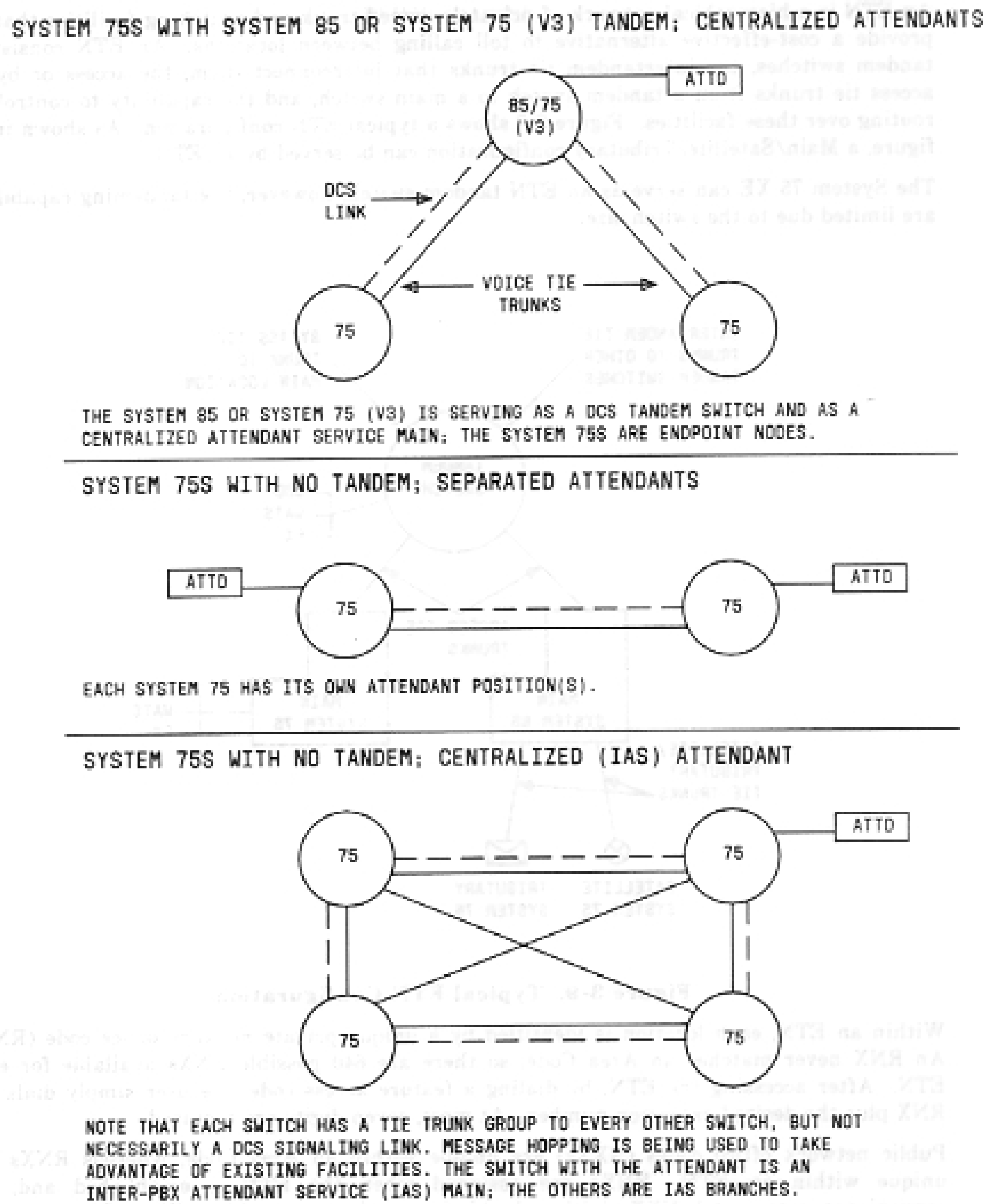


Figure 3-8. Examples of DCS Configurations

Electronic Tandem Network (ETN)

An ETN is a hierarchical network of privately owned trunk and switching facilities that can provide a cost-effective alternative to toll calling between locations. An ETN consists of tandem switches, the intertandem tie trunks that interconnect them, the access or bypass access tie trunks from a tandem switch to a main switch, and the capability to control call routing over these facilities. Figure 3-9 shows a typical ETN configuration. As shown in the figure, a Main/Satellite/Tributary configuration can be served by an ETN.

The System 75 XE can serve as an ETN tandem switch; however, the tandeming capabilities are limited due to the switch size.

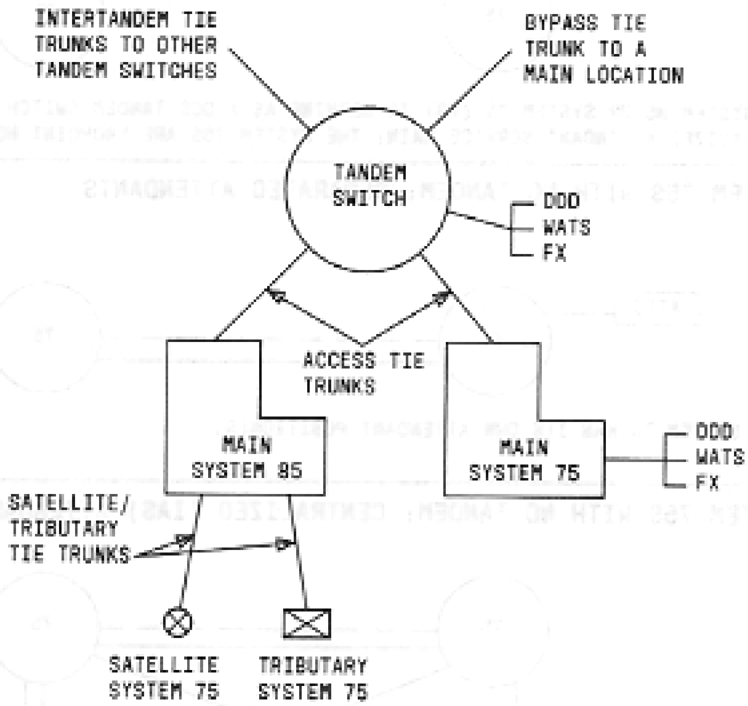


Figure 3-9. Typical ETN Configuration

Within an ETN, each location is identified by a unique private network office code (RNX). An RNX never matches an Area Code, so there are 640 possible RNXs available for each ETN. After accessing the ETN, by dialing a feature access code, the user simply dials the RNX plus the desired extension number. At most, seven digits are required.

Public network office codes (NXXs) are unique within an Area Code, whereas RNXs are unique within an ETN. RNXs are assigned when the ETN is established and, for convenience, may match NXXs (although this is not always possible). When Direct Inward Dialing (DID) is provided by the local central office, the extension numbers (last four digits of the number) will match. Network Inward Dialing (NID) is the ETN equivalent of DID and can be provided without DID.

The software program that controls call routing over an ETN is called Automatic Alternate Routing (AAR). AAR not only determines the route for a call, but, through the Facilities

Restriction Level (FRL) function, defines up to eight levels of calling privileges for users of the ETN. Another function of AAR, Subnet Trunking, can convert an on-network number to a public network or international number. This function is useful when all on-network routes are busy or are not provided. The AAR, FRL, and Subnet Trunking features are described briefly in Section 2 of this manual. For more details, refer to *AT&T System 75—Feature Description*, 555-200-201, and *AT&T System 75—Network and Data Services*, 555-025-201.

Main/Satellite/Tributary

Figure 3-10 shows a Main/Satellite/Tributary configuration. It can function independently or serve as an ETN access arrangement. For a Main/Satellite configuration, attendant positions and public network trunk facilities are concentrated at the Main, and calls to or from satellite locations pass through the Main. To a caller outside the Main/Satellite complex, the system appears to be a single switch with one Listed Directory Number (LDN).

A Tributary location is similar to a satellite location except that it has one or more attendant positions and has its own LDN.

A small business can start with a single Main/Satellite or Main/Tributary complex and add trunk and switching facilities as the business grows. In this situation, tie trunks connect the main locations within an urban area, and intercity traffic is routed over the public network. This arrangement favors a medium-size organization or one that has small isolated locations where the intercity traffic is too light to justify the cost of tie trunks.

The Uniform Dial Plan (UDP) feature enables a terminal user at any switch to call any other terminal on any switch in the UDP complex by using the 4- or 5-digit extension number.

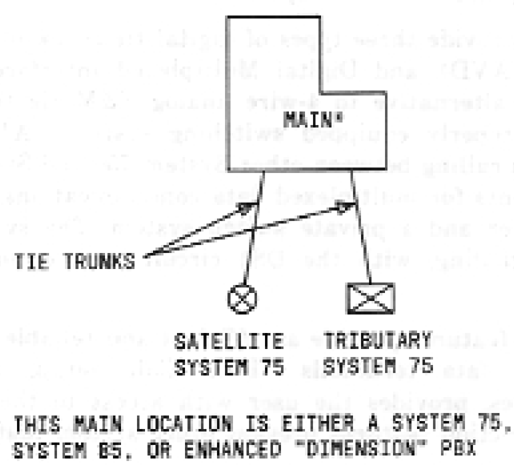


Figure 3-10. Main/Satellite/Tributary Configuration

Data Communications

A variety of data equipment can be connected to System 75 XE. The system connects to data equipment provided with an Electronic Industries Association (EIA) RS-232C interface, or an RS-449, RS-366, V.35, or Category A coaxial interface. Supported types of equipment include the following:

- Data terminals
- Printers
- Graphic and facsimile equipment
- Computers
- Personal Computers

Data modules provide the digital interface between the system switch and data terminals, hosts, and other off-premises facilities. Data equipment can be connected directly to the switch by the EIA Interface [Data Line circuit pack and an Asynchronous Data Unit (ADU)].

The connectivity between System 75 XE and the data endpoints is provided by several Data Management interface features. Modem Pooling provides analog-to-digital and digital-to-analog conversion resources for both on-premises and off-premises connections. The resources are pooled for switched access operation. Remote data equipment can be provided switched access from local data terminals through Off-Premises Data-Only Extensions. Analog or digital private-line facilities, dedicated to data, can be used.

When multiplexing and digital connectivity is advantageous, the DS1 provides a means to replace analog tie trunks with DS1 facilities. This arrangement can be expanded to increase the capacity or add dedicated data channels by using Channel Expansion and Channel Division Multiplexing. The EIA Interface (Data Line circuit pack and ADU) provides the direct connection to on-premises data endpoints.

The DS1 Interface can provide three types of digital tie trunk interfaces: Voice-Grade DS1, Alternate Voice/Data (AVD), and Digital Multiplexed Interface (DMI). The Voice-Grade DS1 tie trunks are an alternative to 4-wire analog E&M tie trunks and may be used to interface with other properly equipped switching systems. AVD DS1 tie trunks permit alternate voice and data calling between other System 75s and System 85. The DMI specifies the interface requirements for multiplexed data communications, over DS1 digital facilities, between a host computer and a private switch system. The system supports the DMI by using Bit Oriented Signaling, with the DS1 circuit pack optioned for Common Channel Signaling.

Data management user features provide an efficient and reliable means of establishing data connections from the data terminals. Data Call Setup, along with several Voice Communications features, provides the user with access to the data network. And Data Privacy and Data Restriction features prevent other system features from interrupting the data transmissions.

For a more detailed discussion of data communications, refer to *AT&T System 75—Feature Description*, 555-200-201.

Information Systems Network (ISN) Interface

The AT&T ISN is a packet switched local area network that links mainframe computers, minicomputers, word processors, storage devices, personal computers, printers, terminals, and communications processors into a single system. The interface to the System 75 XE is through an Asynchronous Data Unit (ADU). A Modular Processor Data Module (MPDM) may be used but the ADU is more economical. The MPDM or ADU connects to an Asynchronous Interface Module (AIM) on the Packet Controller or Terminal Concentrator (see Figure 3-11). This interface allows System 75 XE and the ISN to share data capabilities.

Connectivity between ISN and System 75 XE provides the following major benefits:

- Users on ISN may (in addition to having access to other endpoints directly connected to ISN) have access to any endpoint connected to System 75 XE or addressable from the System 75 XE.
- Users who either connect to or have access to the system may also access endpoints connected to ISN.

For a detailed description of the ISN, refer to *AT&T System 75—Feature Description*, 555-200-201.

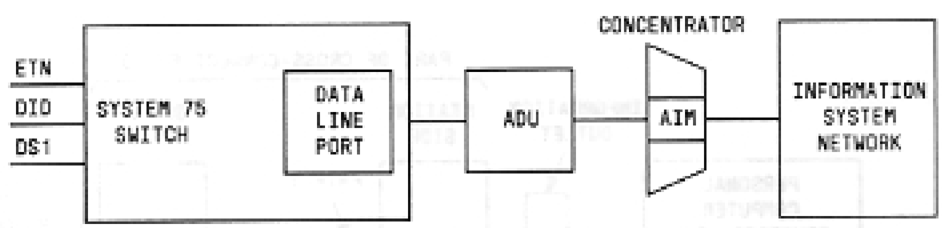


Figure 3-11. System 75 to ISN Connectivity

Star-Based Local Area Network (STARLAN NETWORK)

The STARLAN NETWORK interconnects small numbers of personal computers, data terminals, resource units, and printers. When a STARLAN NETWORK and a System 75 XE are collocated, voice and data can be shared at the same information outlet (see Figure 3-12).

The voice pair that connects to a TN742, TN769, or TN746 Analog Line circuit pack port occupies the first pair of the information outlet. The data pairs that connect to the STARLAN NETWORK occupy the second and third pairs of the information outlet. The voice and data pairs must be separated at the blue or white field located in the equipment room or at the blue field located in a satellite location.

The two major components of STARLAN NETWORK are the Network Access Unit (NAU) and the Network Sharing Unit (NSU). The NAUs are plugged into expansion slots of workstations that enable the workstation to connect to the media and thus access the network. A stand-alone NAU is provided that will attach up to two RS-232C devices to the network. The NAU is then connected to a port of the NSU located in the satellite closet. The network hub function is provided by the NSU. A single NSU can terminate up to 11 NAU links, where each NAU link can have up to 10 NAU-equipped devices connected together in a daisy chain.

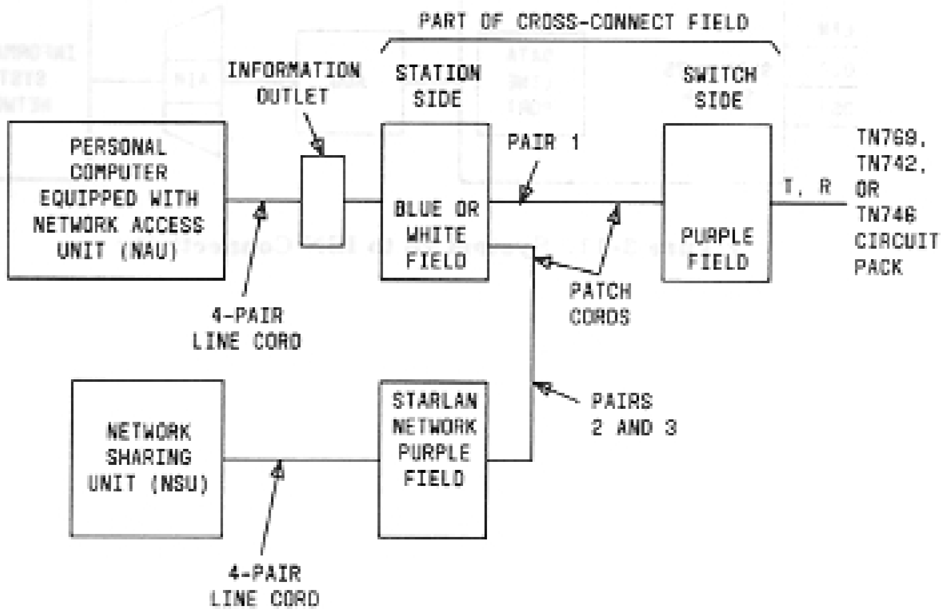


Figure 3-12. System 75 to STARLAN NETWORK Connectivity

HARDWARE DESCRIPTION

CONTENTS

LIST OF FIGURES

LIST OF TABLES

HARDWARE DESCRIPTION

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

- System Cabinets
- Auxiliary Equipment
- Peripheral Equipment

System Cabinets

The System 75 XE switch cabinet is a combined cabinet and carrier unit (see Figure 4-1). Two cabinets are available, a basic control cabinet (J58890G) and an expansion port cabinet (J58890H). Both cabinets are identical in color and size.

The cabinets are cobblestone grey with an off-white removable front cover. A system identification stripe appears across the top front of the cabinet, and a slotted area at the bottom of the cabinet provides air circulation. The cabinets can be located in an office area.

Physical characteristics of each cabinet are as follows:

- Control Cabinet (J58890G)—This cabinet weighs about 130 pounds when fully equipped. It is 20 inches high by 27 inches wide by 22 inches deep including the door.
- Expansion Port Cabinet (J58890H)—A fully equipped cabinet weighs about 125 pounds. It is the same width and depth dimensions as the control cabinet. It is approximately 19 inches high because the cabinet feet are shorter.

A screw-type latch, located below the identification stripe, secures the front door to the cabinet. Turning the screw to the left loosens the latch and releases the door.

Two holes, located in the rear bottom of the cabinet, are available for securing the control cabinet to the floor.

All systems require a control cabinet. An expansion port cabinet supports system requirements that exceed the capacity of the control cabinet. The expansion port cabinet is stacked on top of the control cabinet, and a cabinet clip mechanically fastens the cabinets together (see Figure 4-2). A ground plate, connected between cabinets, provides ground integrity and stabilizes the cabinets. A maximum of three port cabinets can be stacked on the control cabinet to provide a 4-cabinet system.

Each cabinet has a power supply. A power cord, with a 3-prong plug on one end and a single connector on the other end, connects the power supply to a dedicated power source (120-volt ac 20 amp for the multiple cabinets and 120-volt ac 15 amp for a single cabinet). (Power requirements are described in Section 10 of this manual.)

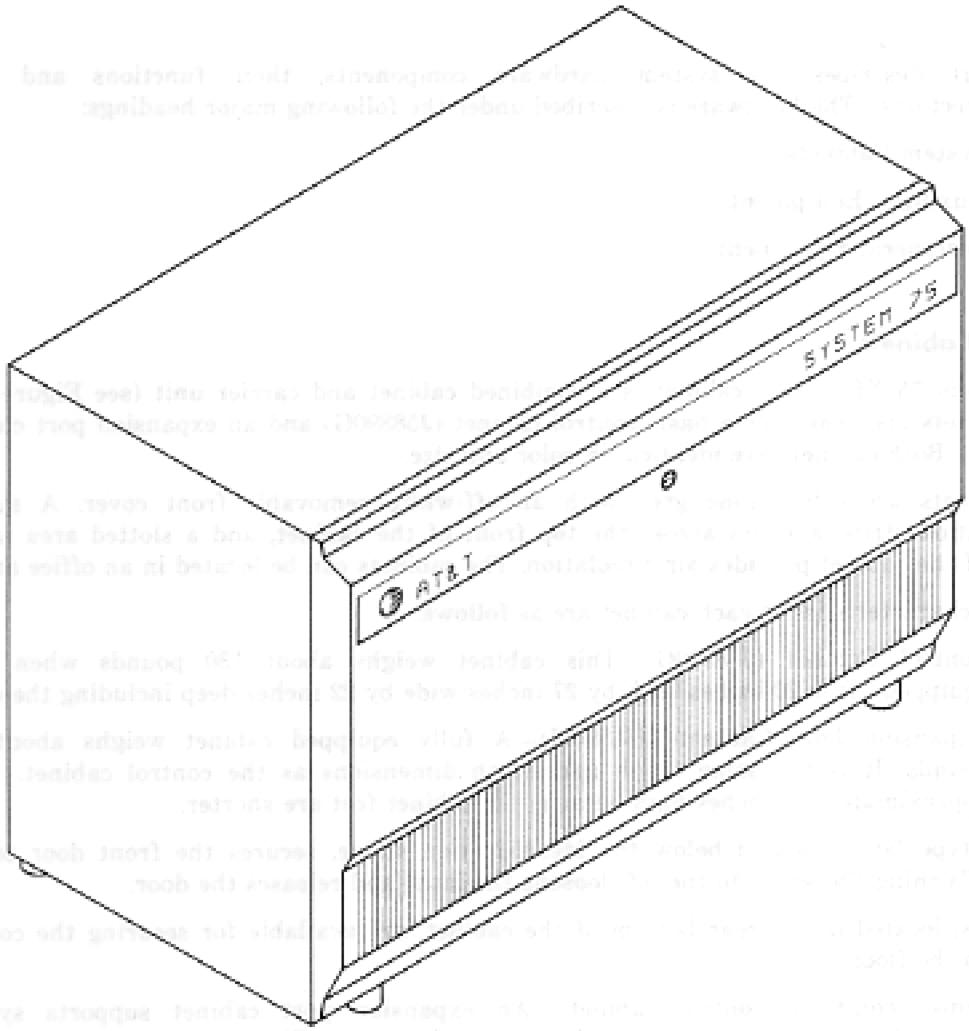


Figure 4-1. System Cabinet (J58890G or J58890H)

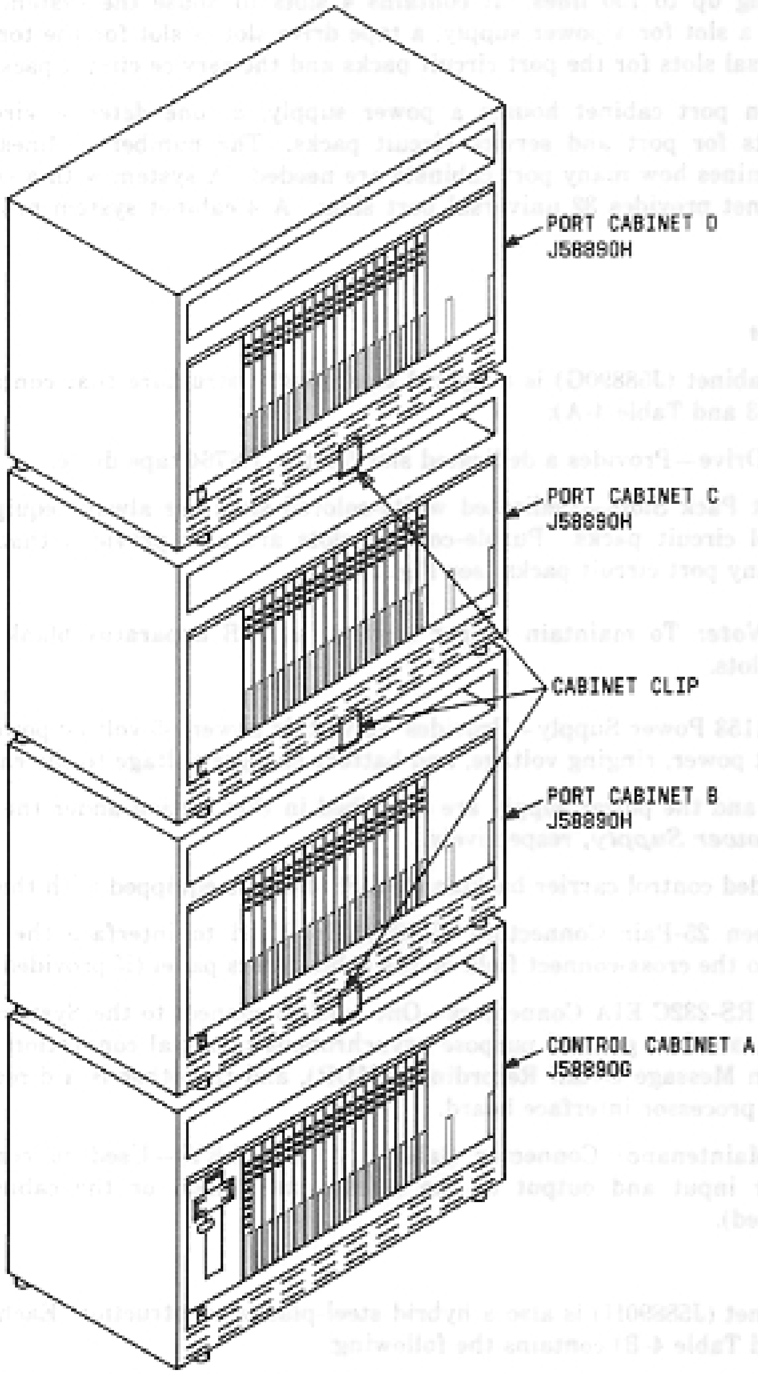


Figure 4-2. Fully Equipped 4-Cabinet System (J58890G and J58890H)—Doors Removed

System Configurations

The control cabinet and the expansion port cabinet house the main hardware of the system. These cabinets can be arranged in a 1-cabinet, 2-cabinet, 3-cabinet, or 4-cabinet system configuration. Figure 4-2 shows a fully equipped 4-cabinet system.

The control cabinet is the basic cabinet and can serve as a single cabinet system accommodating up to 150 lines. It contains 4 slots to house the system logic and control circuit packs, a slot for a power supply, a tape drive slot, a slot for the tone detector circuit, and 13 universal slots for the port circuit packs and the service circuit packs.

The expansion port cabinet houses a power supply, a tone detector circuit pack, and 17 universal slots for port and service circuit packs. The number of lines and trunks in a system determines how many port cabinets are needed. A system with a control cabinet and one port cabinet provides 32 universal port slots. A 4-cabinet system provides 68 universal port slots.

Cabinets

Control Cabinet

The control cabinet (J58890G) is a hybrid steel-plastic structure that contains the following (see Figure 4-3 and Table 4-A):

- Tape Drive—Provides a dedicated slot for the TN764 tape drive.
- Circuit Pack Slots—Dedicated white-colored slots are always equipped with specific control circuit packs. Purple-colored slots are also provided that can be equipped with any port circuit packs (see Figure 4-3).

  Note: To maintain proper airflow, a 158B apparatus blank covers all empty slots.

- WP-91153 Power Supply—Provides +5 volt dc power, -5 volt dc power, -48 volt power, 12 volt power, ringing voltage, and battery charger voltage to the cabinet.

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

The double-sided control carrier backplane (J58890AG) is equipped with the following:

- Fourteen 25-Pair Connectors (J58890AB)—Used to interface the port circuit pack slots to the cross-connect field or the cable access panel (if provided).
- Three RS-232C EIA Connectors—One used to connect to the System Access Terminal (SAT), one is a general purpose asynchronous terminal connection typically used for Station Message Detail Recording (SMDR), and the other is a direct EIA connection to the processor interface board.
- One Maintenance Connector (labeled AUXILIARY)—Used to connect the control carrier input and output to the cross-connect field or the cable access panel (if provided).

Port Cabinet

The port cabinet (J58890H) is also a hybrid steel-plastic construction. Each port cabinet (see Figure 4-4 and Table 4-B) contains the following:

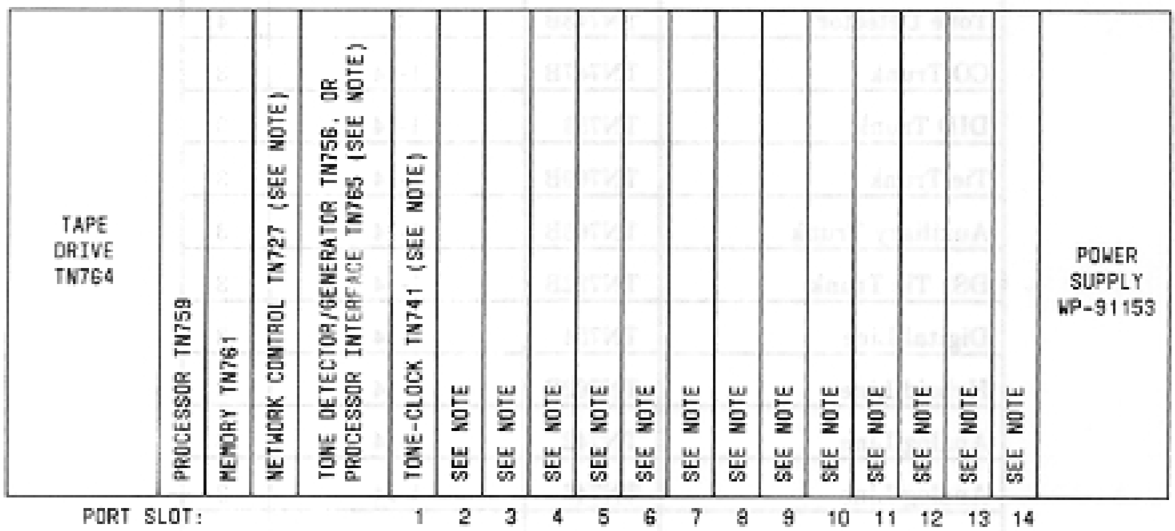
- Port Circuit Pack Slots—Provide 18 slots. The first slot contains a Tone Detector circuit pack and the remaining 17 slots can be equipped with any type of trunk or line circuit pack. If a Power Unit (TN755) is required, the power unit occupies slot position 1 and the Tone Detector resides in slot position 2.

> ***Note:*** To maintain proper airflow, a 158B apparatus blank covers all empty slots.

- WP-91153 Power Supply—Provides +5 volt dc power, -5 volt dc power, -48 volt power, 12 volt power, and ringing voltage to the cabinet.

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

The port carrier backplane (J58890BD) is equipped with eighteen 25-pair connectors used to interface the circuit pack slots with the cross-connect field or the cable access panel (if provided).



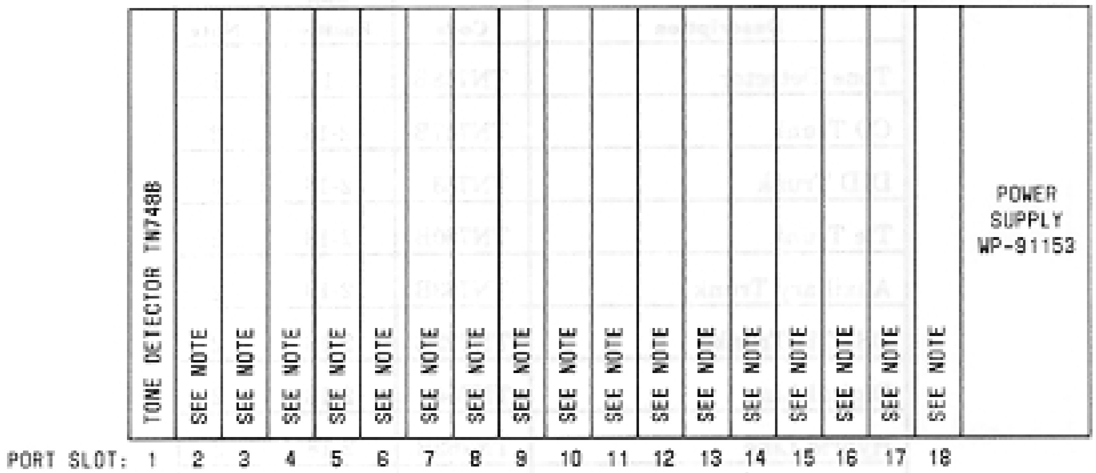
NOTE:
SEE TABLE 4-A FOR CIRCUIT PACK SLOT POSITIONS.

Figure 4-3. Control Cabinet (J58890G)

TABLE 4-A. Control Cabinet Circuit Pack Locations

Description	Code	Slot Position	Note
Tape Drive	TN764	TAPE DRIVE	1
Processor	TN759	PROCR	1
Memory	TN761	MEMORY	1
Network Control	TN727	NETCON	1
Tone Detector/Generator	TN756	TONE DET/GEN	1,2
Processor Interface	TN765	PROCR INFC	2,3
Tone-Clock	TN741	TONE CLOCK 1	4
Tone Detector	TN748B	2	4
CO Trunk	TN747B	1-14	3
DID Trunk	TN753	1-14	3
Tie Trunk	TN760B	1-14	3
Auxiliary Trunk	TN763B	1-14	3
DS1 Tie Trunk	TN722B	1-14	3
Digital Line	TN754	1-14	3
Hybrid Line	TN762B	1-14	3
Analog Line	TN742	1-14	3
Analog Line	TN746	1-14	3
Met Line	TN735	1-14	3
Pooled Modem	TN758	1-14	3
Data Line	TN726	1-14	3
Speech Synthesizer	TN725B	1-14	3
Power Unit	TN755	13,14	3

Notes:
1. One always required.
2. TN756 is located in slot position 1 when a TN765 is required.
3. Provided as required.
4. Provided in place of TN756 when DS1 Tie Trunk (TN722B) circuit packs are used.



NOTE:
1. PORT SLOTS 1 THROUGH 18 CAN CONTAIN THE CIRCUIT PACKS SHOWN IN TABLE 4-B.

Figure 4-4. Port Cabinet (J58890H)

TABLE 4-B. Port Cabinet Circuit Pack Locations

Description	Code	Slot Position	Note
Tone Detector	TN748B	1	1
CO Trunk	TN747B	2-18	2
DID Trunk	TN753	2-18	2
Tie Trunk	TN760B	2-18	2
Auxiliary Trunk	TN763B	2-18	2
DS1 Tie Trunk	TN722B	2-18	2
Digital Line	TN754	2-18	2
Hybrid Line	TN762B	2-18	2
Analog Line	TN742	2-18	2
Analog Line	TN746	2-18	2
Analog Line	TN769	2-18	2
MET Line	TN735	2-18	2
Pooled Modem	TN758	2-18	2
Data Line	TN726	2-18	2
Speech Synthesizer	TN725B	2-18	2
Announcement Circuit Pack	TN750	2-18	2
Power Unit	TN755	1	2

Notes:
1. One always required.
2. Provided as required.

Circuit Packs

All system circuit packs have the following features:

- Solid-state logic circuitry mounted on an 8-inch by 14-inch 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)
- Metal latch for grounding

Control Circuit Packs

The following control circuit packs are located in the control cabinet.

- Processor Circuit Pack (TN759)

  Manages control of the entire system and executes stored programs to effect call processing functions. Has 64,000 bytes of Read-Only Memory (ROM)/Erasable Programmable ROM (EPROM), 2,000 bytes of Random Access Memory (RAM), and timers. Provides additional programming in ROM to boot the system from tape in the streaming mode. Provides alarm LEDs for system status; monitors and controls circuit pack conditions; provides direct access to the SAT (System Access Terminal); and originates alarms to the remote maintenance system (INADS—which stands for Initialization and Administration System). This circuit pack has a direct access to the tape drive.

- Memory Circuit Pack (TN761)

  Contains system translations including addresses of equipment connected to the switch through the port circuit packs and call processing software. Provides 4 megabytes of dynamic RAM with a single-bit error correction and double-bit error detection. Contains a memory array, on-board refresh logic, address decode logic, and bus buffers.

- Network Control Circuit Pack (TN727)

  Continuously monitors the port circuit packs for activity and then transfers this information to the Processor circuit pack. Information from the Processor circuit pack is also transferred to the port circuit packs. Contains four separate data channels that transfer information. These data channels connect equipment such as an on-premises remote pooled modem or administration terminal used with a Modular Processor Data Module (MPDM), an off-premises administration terminal used with a Modular Trunk Data Module (MTDM), and an output device for an MPDM or MTDM.

- Tone Detector/Generator Circuit Pack (TN756)

  Provides four touch-tone receiver ports, two general purpose tone detector ports, tone generation, and clock generation. This circuit pack is used instead of the TN741 Tone-Clock circuit pack and the TN748B Tone Detector circuit pack in system configurations that do not require DS1 tie trunks.

- Processor Interface Circuit Pack (TN765)

  Provides four data links to the TDM bus and a link through the memory bus to the processor. This circuit pack provides an interface to the 3B2 Applications Processor (AP), Distributed Communications System (DCS), Call Management System (CMS), Property Management System (PMS), and Audio Information Exchange (AUDIX) Interface service. A single EIA port on this circuit pack allows direct access to one data link.

- Tone-Clock Circuit Pack (TN741)

  Supplies call progress tones, touch tones, answer-back tones, and trunk transmission test tones; provides 2-megahertz (MHz), 160-kilohertz (kHz), and 8-kilohertz (kHz) clocks. The TN741 provides clock signals required for DS1 synchronization for digital tie trunks. This circuit pack is used when TN722B DS1 Tie Trunk circuit packs are required.

- Tone Detector Circuit Pack (TN748B)

  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. TN748B provides additional tone detection capability required for enhanced Automatic Route Selection (ARS), Off-Premises (out of building) Keyboard Dialing, and Off-Premises Abbreviated Dialing. This circuit pack is used when the TN741 Tone-Clock circuit pack is required.

Port Circuit Packs

Universal slots are provided in the port cabinet and control cabinet for the following circuit packs:

- Tone Detector Circuit Pack (TN748B)

  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. TN748B provides additional tone detection capability required for enhanced Automatic Route Selection, Off-Premises Keyboard Dialing, and Off-Premises Abbreviated Dialing.

- CO Trunk Circuit Pack (TN747B)

  Provides eight ports for loop-start or ground-start central office (CO), foreign exchange (FX), or Wide Area Telecommunications Service (WATS) trunks. A port can also be used for a PagePac* Paging System. This circuit pack supports the Abandoned Call Search feature for ACD applications in Version 3. The following lead appearances are provided for each port: T, R.

- DID Trunk Circuit Pack (TN753)

  Provides eight ports for immediate-start or wink-start Direct Inward Dialing (DID) trunks. The following lead appearances are provided for each port: T, R.

- Tie Trunk Circuit Pack (TN760B)

  Provides four ports for Type 1 or Type 5 4-wire E&M lead signaling tie trunks. Trunks can be automatic, immediate-start, wink-start, or delay-dial. Contains option switches on each port for connection to the following signaling formats:

  — Type 1 E&M Compatible (Unprotected)

* Registered trademark of Harris Corporation, Dracon Division

— Type 1 E&M Compatible (Protected)

— Type 5 Simplex

See *AT&T System 75 XE—Upgrades and Additions*, 555-201-106. The TN760B also serves as the release link trunks required for Centralized Attendant Service (CAS). The following lead appearances are provided for each port: T, R, T1, R1, E, M.

- Auxiliary Trunk Circuit Pack (TN763B)

  Provides four ports for on-premises trunk applications such as Music-on-Hold, Loudspeaker Paging, Code Calling, and Recorded Telephone Dictation Access. This circuit pack supports Audichron* announcement equipment in Version 3. The following lead appearances are provided for each port: T, R, SZ, SZ1, S, S1.

- DS1 Tie Trunk Circuit Pack (TN722B)

  Provides connection capability to a 1.544 Mbps DS1 facility as 24 independent trunks. Each trunk can provide 64 kbps data transmission. Three types of digital tie trunk interfaces can be provided: Voice Grade DS1 and Alternate Voice/Data (AVD) DS1 tie trunks and Digital Multiplexed Interface (DMI). The data transmission formats are specified by the DMI. The circuit pack can also provide bit-oriented signaling on a per trunk basis for automatic, immediate-start, delay-dial, or release link trunks. The following lead appearances are provided for the circuit pack: LBACK2, LBACK1, LO, LO (high), LI, LI (high).

- Digital Line Circuit Pack (TN754)

  Provides eight ports for connection to multi-appearance 7400 Series digital voice terminals, attendant consoles, 510D Personal Terminal, 515 Business Communications Terminals, or data modules over DCP links. The following lead appearances are provided for each port: TXT, TXR, PXT, PXR.

- Hybrid Line Circuit Pack (TN762B)

  Provides eight ports for multi-appearance hybrid voice terminals (7300 Series). The following lead appearances are provided for each port: VT, VR, CT, CR, P-, P+.

- Analog Line Circuit Pack (TN742, TN769, or TN746)

  The TN742 and TN769 (Version 3) provides 8 ports and the TN746 (Version 2 and 3) provides 16 ports. Each port provides the following lead appearances: T,R.

  The TN746 provides a 24-volt battery feed circuit that supports on-premises (in building) wiring only for the following voice terminals: AT&T 500 and 2500 terminals (LED or neon message waiting lamps) with no adjuncts or special equipment such as amplifier handsets, speakerphones, answering machines, line status indicators, etc. The TN746 supports only one voice terminal per port.

  The TN742 supports on-premises (in building) or off-premises wiring (out of building only with AT&T certified protection equipment) with either touch-tone or rotary dialing and with or without the LED message waiting indicators. The TN742 does not support neon message waiting indicators.

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* Registered trademark of Audichron Company

The TN755 power unit supports applications requiring neon message waiting lamps.

The TN769 circuit pack has the same applications as the TN742 with the additional capability of supporting neon message indicators for on-premises (out of building only with AT&T certified protection equipment) voice terminals for Version 3.

The TN742 and TN769 circuit packs support a maximum of five voice terminals per port (where the maximum ringer equivalents per port is five and a maximum of two voice terminals per port are off-hook at any given time). The TN742 and TN769 also support the following conditions:

- 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
- PagePac Paging System for Loudspeaker Paging feature
- External alerting devices associated with the Trunk Answer From Any Station feature
- Modems

- MET Line Circuit Pack (TN735)

  Provides four ports for the Multibutton Electronic Telephone (MET) sets. The following lead appearances are provided for each port: T, R, BT, BR, LT, LR.

- 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 160 conversion resources are allowed. A conversion resource can also be a Modular Trunk Data Module and an analog modem combination.

- Data Line Circuit Pack (TN726)

  Provides eight ports for asynchronous equipment with RS-232C serial interfaces. An on-board asynchronous data unit extends the serial communications link out to the customer-provided equipment (CPE) over two pairs of standard, voice grade wire. The following lead appearances are provided for each port: TXT, TXR, PXT, PXR.

- Speech Synthesizer Circuit Pack (TN725B)

  Provides four ports with tone detection capability for retrieval of Leave Word Calling messages.

- Announcement Circuit Pack (TN750) (V3)

  Provides an integrated means for recording announcements that can be played back on demand from call processing as part of a calling feature. Messages can be recorded by customers from their voice terminals, on- or off-premises, and have flexible message lengths. The circuit pack provides 4 minutes of storage at 32 Kbits/second. It has 16 channels, and any announcement can be played on any channel. There are 50 queue slots for the board. Five call connections can listen per channel which results in a total simultaneous call capacity of 80 calls. There is a limit of one TN750 per system. The TN750 cannot be used for the Automatic Wakeup feature.

Power Supply

Each cabinet contains a single power supply (WP-91153). The ac input to the power supply is a nominal 120 volts ac. The power supply outputs are a +5 volt dc power, a -5 volt dc power, a -48 volt dc power, a +12 volt dc power, a ringing voltage, and a battery charge voltage. The power supply provides circuit breakers and EMI filtering.

A 250 millisecond holdover power circuit in the power supply allows the system to operate normally during ac power interruptions. A battery reserve provides power to the memory and processor circuit packs and fans, for 2 minutes, if power fails. The power supply contains a battery charger that charges the holdover batteries located in the bottom of the control cabinet.

Fan Assembly

Four fans are mounted at the top, rear of the cabinet. An air filter is located below the fan assembly and air flows down through the filter over the circuit packs. This filter can be easily removed and cleaned or replaced when the cabinet door is removed. If the cabinet temperature reaches 158 degrees Fahrenheit (70 degrees Celsius), the temperature sensor in the power supply causes the system to shut down automatically.

Auxiliary Equipment

The following auxiliary equipment can be optionally mounted in an equipment rack or on a backboard:

- Cook Electric Model 213300-23016120 or 213400-23016140 Digital Announcer—Provides recorded announcements for Intercept Treatment—Recorded Announcement feature. Model 213300-23016120 provides one channel of voice with a message length of up to 16 seconds. Model 213400-23016140 provides four channels of voice with a message length of up to 16 seconds for each channel. Both models require a Cook Electric Model 213288 ac adapter for 115-volt ac power.

  Note: Models 213300-00016120 (one channel) or 213400-00016140 (four channels) can be ordered when the digital announcer is shelf-mounted.

- PagePac Paging System—Provides an amplifier system for Loudspeaker Paging feature. Three models are available:
  - PagePac 20—Provides a single zone of paging with an input source for music. The unit can also be modified to provide 3, 9, or 39 paging zones.
  - PagePac VS—Provides one to three paging zones. It also permits all zone paging. Two optional feature cards are available to provide music or talkback over paging.
  - PagePac 50/100/200—Provides 1 to 24 paging zones. Optional add-ons are available to provide music or talkback over paging. It is also possible to use a customer-supplied music source.

  All PagePac Paging System models require 115-volt ac power.

- 278A Adapters—Provides an interface, when required, for customer-provided equipment for the Loudspeaker Paging feature. It can be modified for -48 volt dc power with a D-181321 kit of parts. The 278A adapter can also be wall-mounted.

- 36A Voice Couplers—Provides an interface and system protection for customer-provided equipment for Intercept Treatment—Recorded Announcement, Music-on-Hold, and Recorded Telephone Dictation Access features. The 36A voice coupler is powered by a 2012D power transformer (set at -15 dBm). The 36A voice coupler can also be wall-mounted.
- 3270C Data Module—Provides the interface (protocol conversion) to a cluster controller connected to a host computer. One data module can contain up to eight ports. The data module requires 115-volt ac power.
- 71A Multiple Data Mounting—Provides mounting space and power for up to eight processor data modules (PDMs), modular processor data modules (MPDMs), trunk data modules, or modular trunk data modules (MTDMs). The 71A data mounting requires 115-volt ac power.
- J58889B Fan Assembly—Required for cooling when two or more 71A data mountings are provided. The fan assembly requires 120-volt ac power.
- Service Observing Melco Unit—Used to access a particular line. The observing station user depresses the assigned pickup key and dials a 2-digit access code assigned to that line. Nothing can be heard by the line being observed. Three units are available:
  - KM-330A Service Observing Unit—Allows for observation of up to 30 lines.
  - TA-300 Talk Assist Unit—Interfaced with KM-330A, and allows a service observer to enter the conversation on the observed line.
  - KMX-333 Service Observing Expander—Allows the expansion of the KM-330A by up to 30 lines. Two of these units may be used for a total capacity of 90 lines.
- Automatic Wakeup Recorder/Announcer—Provides recorded announcement equipment for the Automatic Wakeup feature. An Audichron Company Model HQD614B Recorder/Announcer and power supply are required. Each recorder/announcer uses four TN763B auxiliary trunk ports that must be located on the same circuit pack. The TN725B speech synthesizer can be used for the Automatic Wakeup feature.
- Two emergency transfer panels are available for connection to any FCC registered terminal:
  - Z1A Panel—Each unit serves up to six Power Failure Transfer terminals. A ground-start key is required at each preselected voice terminal when ground-start trunks are used.
  - PORTA SYSTEMS* Model 574-5 Panel—Each unit serves up to five Power Failure Transfer terminals. The unit provides automatic ground start or loop start.

When Z100-type (modular) hardware is provided, the panels are mounted in (or adjacent to) the cable access panel. When 110-type or 66-type hardware is provided, the panels are mounted adjacent to the cross-connect field.

* Trademark of PORTA SYSTEMS Corporation

Station Message Detail Recording

A Station Message Detail Recording (SMDR) output device can optionally be provided. The SMDR output device provides a detailed printout that can be used to compute costs, allocate charges, analyze calling patterns, and keep track of unnecessary calls. The output device may be a TELESEER® SMDR unit, printer, 94A Local Storage Unit, or other customer-provided equipment.

Optional SMDR Account Code Dialing software is available to associate certain calls with a particular project or account number for accounting and/or billing purposes.

Peripheral Equipment

Peripheral equipment is any equipment that can be connected to the system switch. The following equipment is described in detail in the *AT&T System 75 and System 85—Terminals and Adjuncts*, 555-015-201.

- Attendant Console
- Voice Terminals
- Voice Terminal Adjuncts
- Data Modules
- Business Communications Terminals

The wiring required to connect the peripheral equipment is briefly discussed in Section 10 of this manual. For detailed information, see the *AT&T System 75—Wiring*, 555-200-111.

Attendant Console

The attendant console (Figure 4-5) is a digital, call-handling position with pushbutton control used to answer incoming calls, place outgoing calls, and manage and monitor some of the system operations.

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 night-only console can be provided. If a seventh console is provided, it cannot operate when the other six consoles operate. However, it is not required that an attendant console be provided.

The attendant console is available, with or without a selector console, in black and burgundy. The *AT&T System 75—Console Operations*, 555-200-700, provides detailed information and complete operating instructions for both consoles.

Basic Attendant Console

The basic attendant console (see Figure 4-5) is 11 inches wide by 10-1/2 inches deep by 7-1/2 inches high. It can be used on a table, a desk, or any other flat surface. Cabling distances are explained in Section 9 of this manual.

Three basic consoles, with or without a selector console, are available. A separate 48 volt power supply provides power for the console. If a selector console is also used, a 329A or 346A 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, 346A, or 329A Power Unit. The KS-22911 List 1 or 329A Power Unit plugs directly into a 115-volt ac receptacle. The 346A Power Unit is powered by a 346A1 Power Panel that plugs into a 115-volt ac receptacle. Each 346A Power Unit provides four outputs. As many as three 346A Power Units can be mounted in one 346A1 Power Panel. Two outputs must be parallel to power the basic attendant console equipped with a selector console. For Version 3, the attendant console has a special alerting tone for Emergency Access to the Attendant.

The basic console includes the following:

- Two Handset or Headset Jacks—These jacks, located on both the left and the right sides of the console, connect a handset or a headset. If a handset is used, a handset adapter is required. The handset adapter is available in two colors: 854-03 (black) or 854-143 (burgundy). An adapter is not required for a headset. The handset cradle can be moved from one side to the other simply by unscrewing the knob and moving the cradle to the opposite side. The handset or headset jack not being used by the attendant may be used by another person for service observing in Version 3.

- DXS Jack—This jack, located on the bottom of the console, connects the optional selector console.

- Line Jack—This jack, located on the bottom of the console, connects the console to the information outlet (modular wall jack).

- Lamp Test Switch—This switch tests the lamps on the basic console and the optional selector console.

- Three Audible Tone Volume Controls—These slide controls adjust the volume of the ringing, 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 2 of this manual.)

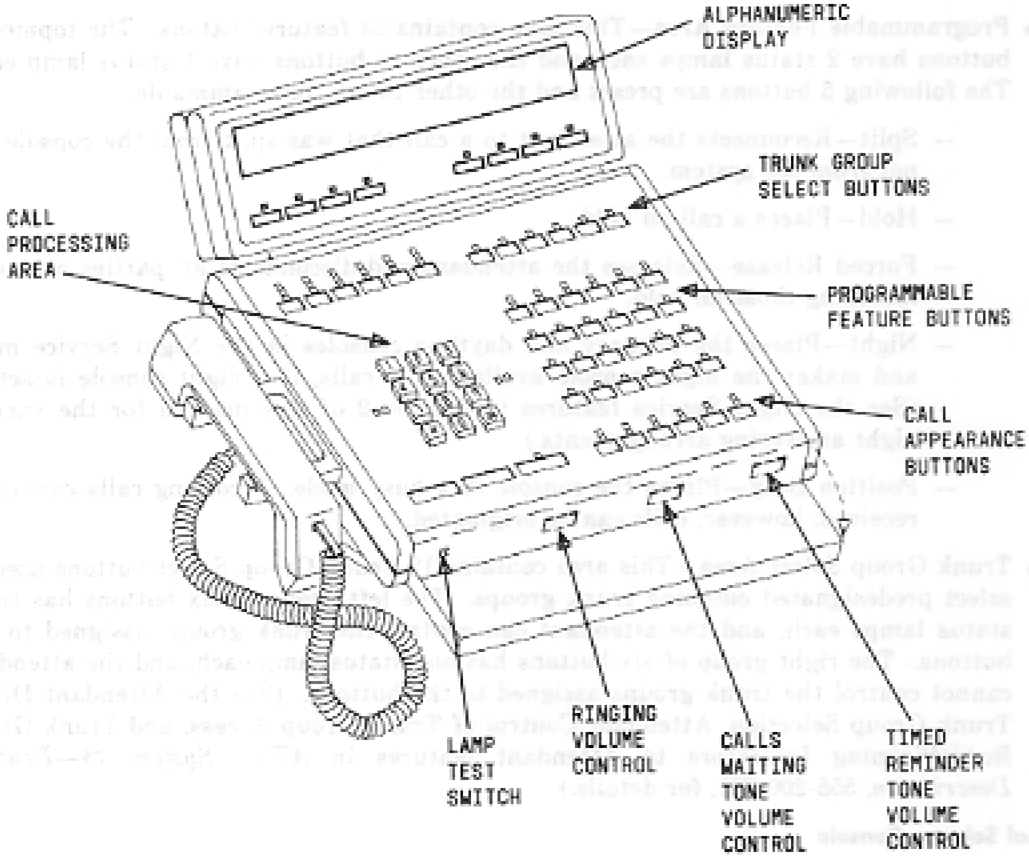


Figure 4-5. Basic Attendant Console

- 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 customer 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.

- 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. 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 2 of this manual 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 *AT&T System 75—Feature Description*, 555-200-201, for details.)

Optional Selector Console

The selector console (see Figure 4-6) is 8-1/2 inches wide by 8-3/4 inches deep by 4-3/4 inches 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 2 in 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.

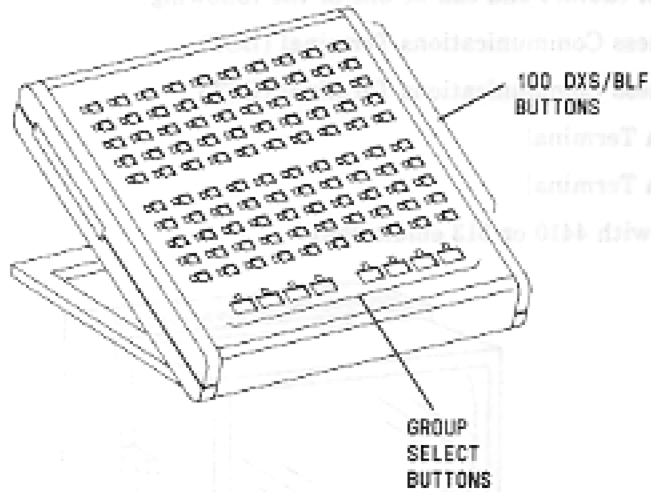


Figure 4-6. Selector Console

System Access Terminal

The system access terminal (SAT) is a general purpose asynchronous data terminal that provides a standard EIA RS-232C interface for administration and maintenance functions. The SAT, consisting of a video display and a keyboard (see Figure 4-7), is located within 50 feet of the control cabinet and can be one of the following:

- 513 Business Communications Terminal (BCT)
- 610 Business Communications Terminal (BCT)
- 4410 Data Terminal
- 4425 Data Terminal
- PC 6300 (with 4410 or 513 emulation).

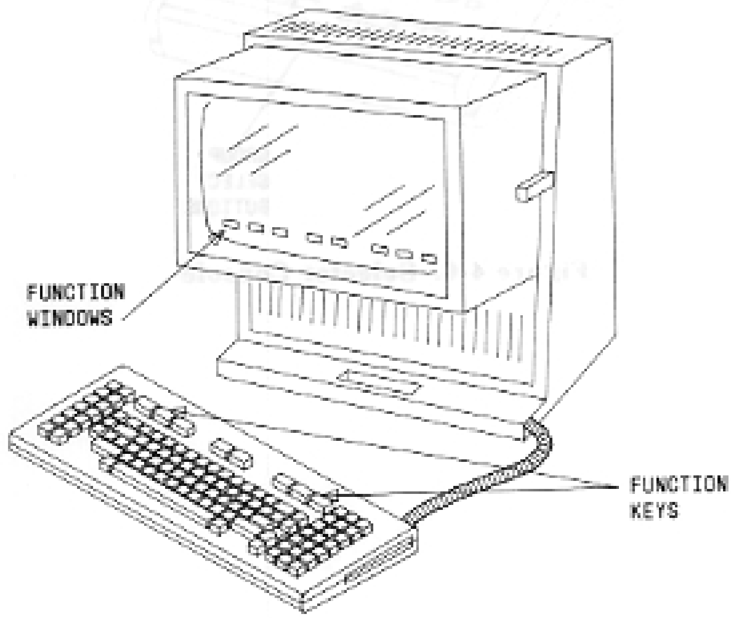


Figure 4-7. 513 BCT With Optional 103-Key Keyboard

Refer to the following manuals for additional information on using these terminals:

- *513 Business Communications Terminal (BCT)*, 999-700-486
- *610 Business Communications Terminal (BCT)*, 999-300-270
- *DATASPEED® 4410 Display Terminal*, 999-300-180
- *DATASPEED® 4425 Display Terminal*, 999-310-181

On systems without available data channels, the SAT connects to the processor through a direct 9600 baud EIA connection on the back of the control cabinet. If a data channel is available, the SAT can be connected through a Digital Terminal Data Module, a Processor Data Module, or an Asynchronous Data Unit used in conjunction with a TN726 Data Line circuit pack.

The SAT requires 120-volt 60-Hertz commercial power from a 3-wire grounded outlet. AC power requirements for the SAT are explained in Section 9 of this manual.

The dedicated SAT must be located in the same equipment room as the switch cabinet(s) and can be used on a table or desk. It requires approximately 3 square feet of space.

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 system features.

Table 4-C lists the various System 75 XE voice terminals. Terminals not supported by screen forms must be implemented as other type terminals. See *AT&T System 75 and System 75 XE—Implementation Release 1 Version 2*, 555-200-651, or *AT&T System 75 and System 75 XE—Implementation Release 1 Version 3*, 555-200-652.

TABLE 4-C. System Voice Terminals

TERMINAL TYPE	MODEL
Single-Line Analog	2500 2500 With Message Waiting Adjunct
Multi-Appearance Hybrid	7303S 7305S
Multi-Appearance Digital	7404D 7405D 7406D With Display 7407D

Table 4-D lists the voice terminals used in other systems that can be reused in the System 75 XE switch. Terminals not supported by screen forms must be implemented as other type terminals. See *AT&T System 75 and System 75 XE—Implementation Release 1 Version 2*, 555-200-651, or *AT&T System 75 and System 75 XE—Implementation Release 1 Version 3*, 555-200-652.

TABLE 4-D. Reusable Voice Terminals

TERMINAL TYPE	MODEL
Single-Line Analog	500 7101A 7103A Programmable
Multi-Appearance Hybrid	7302H 7303H 7305H01B 7305H02B 7305H03B
Multi-Appearance Digital	7403D
Multi-Button Electronic Telephone (MET) Sets	10 Button 10 Button With Built-In Speakerphone 20 Button 30 Button 7203M (12 Button)

Data Modules

Data modules provide an interface between the digital switch, Data Terminal Equipment (DTE), and Data Communications Equipment (DCE). DTE is equipment that provides the data source, termination, or both—a host computer or a data terminal is an example of DTE. DCE is equipment that provides the functions required to establish, maintain, and terminate 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. The digital switch provides a Digital Communications Protocol (DCP) interface to the data module.

The DCE and DTE interconnect through an RS-232C, V.35, RS-449, RS-366, or Category A coaxial interface. The interface is transparent to the code being used.

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

- Half- or full-duplex operation
- Standard data rates of 300 bps, 1.2 kbps, 2.4 kbps, 4.8 kbps, 9.6 kbps, 19.2 kbps, 56 kbps, and 64 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)
- Built-In 7404D Data Module Base (asynchronous only)
- Z702AL1-DSU Data Module Base (Optional base for 7407D voice terminal) (asynchronous only)
- Z703AL1-DSU Data Modular Base (Optional base for 7406D voice terminal) (asynchronous only)
- Processor Data Module (PDM)
- Trunk Data Module
- Modular Processor Data Module (MPDM)
- Modular Trunk Data Module (MTDM)
- 3270 Data Module
- Asynchronous Data Unit (ADU)

SOFTWARE DESCRIPTION

CONTENTS

SOFTWARE DESCRIPTION

General

The System 75 XE switch software consists of the switched services software, administrative 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 Switch Processing Element (SPE), the Network Control circuit, the Processor Interface 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 and the 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.

The software in the Processor Interface circuit board provides the protocol support for the communications link between the 3B2 Applications Processor (AP), Distributed Communications System (DCS), the Call Management System (CMS), the Property Management System, or the Audio Information Exchange (AUDIX) and the SPE. This software also translates between the system commands and the format the DCS, CMS, PMS, and AUDIX accepts.

Administrative Software

The administrative software provides the control for system rearrangement and change through 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 System Access Terminal (SAT) or by the Initialization and Administration System (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 errors such as assigning 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 (saved), on command, to the tape located in the tape drive assembly. The download operation can also be administered so that it automatically occurs 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 maintenance software resides entirely in the SPE.

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 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 hardware and an interface for the customer or an AT&T technician to do the periodic tests on demand
- Maintenance Load Regulation—Ability to reduce the amount of periodic testing when a large amount of call processing is 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.

Memory Allocation

The system software, like the hardware, is identified by release number and by version. Each version pertains to a particular memory configuration for the release number. Main memory is located in the SPE. The main memory contains the operating system, call processing, system data, system translations, and other related programs.

Real-Time Constraints

Real-time constraints are a function of the speed of the SPE and the traffic load. The switch is designed so that many time-consuming and repetitious functions are performed by processors in the port and service circuit packs, thus relieving the SPE.

Traffic load, defined as the sum of static and dynamic loads, is a function of the number of features that are executed, the frequency with which they are executed, the customer configuration, and the instantaneous (peak) call processing load. The configuration contribution to load is known as dynamic load. For additional information concerning traffic engineering, refer to the *AT&T System 75—Administration*, 555-200-500.

Tape Cartridge

Overview

The tape drive provides a nonvolatile system bootstrap and translation storage device. The software is contained on a tape cartridge cassette. The cassette provides a flexible and convenient way to transport software from the factory to the field. Therefore, software updates or revisions are easily incorporated into existing systems.

The tape drive has extensive error detection and correction capabilities that minimize the need to have multiple copies of data on the tape. The drive can read or write in both the forward or reverse tape directions.

Organization

Data is recorded on the tape cartridge in 1024 word blocks using a 6-track format. Data is organized in a logically ordered format that assures the quickest reload of the system in case power fails.

Data is written sequentially, a block at a time, on each of the tracks on the tape cartridge. The tracks are arranged in serpentine fashion. Subsequent access for reading or editing can be done randomly over the previously written blocks.

The tape recorder assembly records data on the tape in the incremental mode for all operations but emulates the streaming mode during system boot. The tape cartridge stores up to 18 megabytes of data.

The high density digital tape in the cartridge is 183 m long. During read, write, and search operations, the tape speed is 198 cm per second and during rewind the tape speed is 229 cm per second.

Configuration

The tape cartridge is available in a generic configuration. The generic cartridge contains the program instructions with default values for some of the administrable translations.

SYSTEM ADMINISTRATION

CONTENTS

- i -

SYSTEM ADMINISTRATION

Screen Forms

The system is administered using screen forms (see Figure 6-1) displayed on the System Access Terminal (SAT). The forms are used to display and list data, and to add, change, and remove system and voice terminal features.

Copies of screen forms are provided in *AT&T System 75 and System 75 XE—Implementation Release 1 Version 2*, 555-200-651, or *AT&T System 75 and System 75 XE—Implementation Release 1 Version 3*, 555-200-652. Procedures and requirements for entering the data are provided in *AT&T System 75 Administration*, 555-200-500.

Commands

Screen forms are accessed by entering valid system commands at the SAT. These commands are standard English words and phrases that instruct the system to do a specific function.

```
                                                         Page 1 of 1
                                  STATION

Extension: ____

     Type: 7103A      Lock Messages: n      COR: 1   Room: _______

     Port: ______________  Security Code: ____   COS: 1   Jack: _______

     Name: ______________  Coverage Path: ___   Tests? _  Cable: _______

FEATURE OPTIONS

               LWC Reception? y     Coverage Msg Retrieval Permission? y

              LWC Activation? y                    Data Restriction? n

       Redirect Notification? y              Call Waiting Indication? y

         Off Premise Station? n             Distinctive Audible Alert? y

ABBREVIATED DIALING

     List1: _______          List2: _________          List3: _________

HOT LINE DESTINATION

        Abbreviated Dialing List Number (From above 1, 2 or 3): _

                                               Dial Code: __
```

Figure 6-1. Typical Defaulted Screen Form

SYSTEM MAINTENANCE

CONTENTS

SYSTEM MAINTENANCE

This part provides general information on system maintenance. For more details, see *AT&T System 75 XE—System Maintenance*, 555-201-105.

The primary objective of system maintenance is to detect, report, and clear troubles as quickly as possible and with minimum disruption of normal service. This goal is supported by periodic tests, automatic software diagnostic programs, and fault detection hardware. System design allows most troubles to be reduced to the circuit pack level.

The system hardware is maintained as a group of independent units or maintenance objects, as far as possible. Each maintenance object is normally a separately replaceable unit. These units include circuit packs, power units, fans, the tape unit, voice terminals, lines, and trunks.

The two general categories within system maintenance are: system-alarmed troubles and user-reported troubles. For alarmed troubles, both the remote maintenance facility Initialization and Administration System (INADS) and the local attendant console are automatically alerted. Most alarms are also reported by light-emitting diodes (LEDs) on the circuit packs. User-reported troubles usually result from service problems at individual voice and data terminals and are often related to alarmed conditions. The major part of the maintenance effort is directed toward system-alarmed troubles. The system itself detects and reports most problems automatically.

The system automatically retires alarms. After an alarmed trouble has been cleared, the system retests the previously faulty area. When the trouble is no longer detected, the alarm is removed. It is not necessary for maintenance personnel to retire alarms after a problem has been fixed.

Maintenance Hardware

The following hardware is provided for speed and accuracy in fault detection diagnosis and repair.

- Maintenance Circuit (Processor)—Functions as follows:
  - Originates alarm information to the attendant
  - Provides alarm LEDs for system status
  - Provides emergency transfer switch and emergency transfer control
  - Monitors and controls the reset condition and sanity of the switch processing element (SPE)
  - Monitors the power units
  - Provides direct access to System Access Terminal (SAT)
  - Supports asynchronous maintenance and administration commands on the INADS link.
- System Access Terminal—Provides a maintenance interface for the maintenance technician.

- Attendant Console LEDs—Provides two red LEDs, labeled Alm and Ack. The left LED lights steadily when there is a major or minor alarm at the switch cabinet. The right LED lights steadily if the alarm has been successfully reported to INADS. If the system is unable to report the alarm to INADS, the right LED flashes; this is a signal for the attendant to call INADS and report the alarm.
- Multifunction Voice Terminals—Major, minor, and warning buttons may be administered.
- Circuit Pack LEDs—Indicate the following when lighted:
  - Red (alarm)—The system has detected a fault in this circuit pack.
  - Green (test)—The system is running tests on this circuit pack.
  - Amber (busy)—This circuit pack is in use.
- In-Line Error Detection Circuitry—Checks for correct operation each time a maintenance object is used.

Maintenance Tests

The maintenance tests can be divided into two groups, periodic and demand. The periodic tests are run automatically at fixed intervals on a specific schedule. The short tests are run hourly and the long tests are run every 24 hours. Heavy call processing may push these tests out.

Demand tests are run by the system when it detects a need for them or by maintenance personnel when required during trouble clearing activities. Demand tests include the periodic tests, plus others that are required only when trouble occurs. Some of the nonperiodic tests may be disruptive to system operation. Using the SAT, maintenance personnel can initiate the same tests that the system initiates, and the results are displayed on the SAT screen.

Maintenance Procedures

Alarm Reporting

If a maintenance object in the system fails some of the periodic tests a preset number of times, the system automatically generates an alarm. This alarm alerts the maintenance personnel that action is required to restore the system to a normal condition. The system supports three levels of alarms:

- Major Alarms—Failures that cause critical degradation of service and require immediate attention.
- Minor Alarms—Failures that cause marginal degradation of service while not rendering a crucial portion of the system inoperable. This condition requires action, but its consequences are not immediate. Problems might be impairing service to a few trunks or stations or interfering with one feature across the entire system.
- Warning Alarms—Failures that cause no noticeable degradation of service. Warning Alarms are not reported to the attendant console or INADS.

A customer provided equipment (CPE) alarm is provided by the system to a customer device such as a lamp, an automatic dialer, a bell, or other customer provided equipment. The CPE Alarm Activation Level field on the System-Parameters Maintenance form must be administered to indicate which level of alarm (major, minor, warning, or none) activates the CPE device.

Logs of Errors and Alarms

The system produces a software record of every error detected in the system. This record, the Error Log, can be displayed on the SAT by maintenance personnel. It can be useful in analyzing problems that have not caused an alarm or when alarms cannot be retired by replacement of maintenance objects.

When errors result in alarms, the alarms are listed on another software record, the Alarm Log. This can also be displayed at the SAT. If a number of alarms are active, the Alarm Log can be used to determine which alarms should be cleared first.

Both the Alarm Log and the Error Log are historical. They list current conditions that have not been resolved as well as past alarms and errors to provide a profile of system maintenance. The Error Log is saved on tape after a major system failure or restart.

Local and Remote Testing

The SAT or the remote administration terminal located at INADS can be used to do the following:

- Display error and alarm logs
- Test circuit packs
- Test other system functions
- Busyout and release system equipment
- Reset the system

Port Circuit Pack Replacement and Verification Testing

A port circuit pack can be replaced without turning off power or interrupting service except in the area directly affected by the replacement. In most cases, verification tests are automatically run on the circuit pack as soon as it is plugged in.

UPGRADE PROCESS

CONTENTS

UPGRADE PROCESS

A system upgrade is the process of transforming the hardware and software of a previously installed system to that of a later version system. This upgrade is performed when increased call processing demands, need for greater feature capabilities, and other changes in customer requirements justify such an action.

For details on the upgrade process and administration required, see *AT&T System 75 XE—System Upgrades and Additions*, 555-201-106, and *AT&T System 75 and System 75 XE Administration*, 555-200-500.

TECHNICAL SPECIFICATIONS

CONTENTS

TECHNICAL SPECIFICATIONS

General

This section provides information on the overall characteristics and capacities of the system. Detailed specifications for system features are contained in the *AT&T System 75—Feature Description*, 555-200-201.

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

Hardware and Software Parameters

A list of system maximum parameters for hardware and software items are listed on the following pages. These parameters apply to a 4-cabinet system.

ITEM	R1V2	R1V3
Abbreviated Dial Lists:	802	802
Personal Lists	800	800
Max Entries	10	10
Per Extension Number	3	3
Group Lists	102	102
Max Entries	90	90
Per Extension Number	3	3
System List	1	1
Max Entries	90	90
Enhanced List	-	1000
Size of List Entry	24	24
Total Entries	4010	4010
Attendant Consoles:		
Daytime Consoles	6	6
Emergency Access Queue Slots	-	50
Night-Only Console	1	1
AUDIX Systems	-	1
Authorization:		
Authorization Codes	-	5000
Length of Authorization Codes	-	4-7
Barrier Codes (Remote Access)	10	10
Length of Barrier Code	4-7	4-7
Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS):		
Partitions	-	4
Patterns	254	254
Trunk Groups per Pattern	6	6
Toll Lists	32	32
3-Digit Translation Tables (per Partition)	2	2
6-Digit Translation Tables (per Partition)	32	32
Automatic Callback Calls	80	80
Automatic Wakeup		
Wakeup Requests per Extension	-	1
Max Wakeup Requests	-	800
Max Wakeup Requests in any 15-min Interval	-	200
Advance Wakeup Request Time	-	23 hrs 55 min
Attendant Consoles and/or Front Desk Terminals in Display Mode	-	10

ITEM	R1V2	R1V3
Bridged Call Appearances	500	800
Cabinets:		
Control	1	1
Port	3	3
Call Coverage:		
Coverage Paths per System	400	400
Coverage Points per Path	3	3
Coverage Answer Groups	200	200
Members per Coverage Answer Group	8	8
Max number of Coverage Paths in a Coverage Path List	1	4
Call Park		
Attendant Group Common Shared Extension Numbers	10	10
Call Pickup Groups	400	400
Members per Group	50	50
Total Members	800	800
Centralized Attendant Service		
Release Link Trunks	16	16
Classes of Restriction	64	64
Classes of Service	16	16
Code Calling Ids	125	125
Communications Interface Links	4	4
Conference Parties	6	6
Digital Data Endpoints (Defined in the Glossary—Section 12).	400	400
Do Not Disturb		
Maximum Do Not Disturb Requests	-	800
Attendant Consoles and/or Front Desk Terminals in Display Mode	-	10
DS1 Circuit Packs	20	20
Extension Numbers	1200	1200
Max Digits in Extension	5	6*
Facility Busy Indicators	1600	1600
Buttons per Tracked Resource	100	100
Hunt Groups (DDC and UCD Combined)	32	32
ACD Agents per System		200
ACD Supervisors per System	-	32
Members per Group	32	100
Members per System	448	448
Queue Slots per Group	35	100
Queue Slots per System	-	1000
Announcements per Group	1	2

* A prefixed extension number can be six digits.

ITEM	R1V2	R1V3
Intercom Groups (Automatic and Dial Combined)	32	32
Members per Group	32	32
Members per System	128	128
Integrated Directory Entries	800	800
Size of List Entry	15	15
Leave Word Calling (Switch-Based, No AP):		
Messages Stored	2000	2000
Messages per User	125	125
Systemwide Message Retrievers	10	10
Remote Message Waiting Indicator:		
Per Extension Number	1	80
Per System	80	80
Loudspeaker Paging Zones*	9	9
Move Agents from CMS		
Max Agents Moved per Request	-	32
Multiple Listed Directory Numbers	50	50
DID Numbers	8	8
Personal Central Office Lines	40	40
Pooled Modems:		
Groups	5	5
Members per Group	32	32
Integrated	160	160
Combined	160	160
Port Circuit Packs (excluding Tone Detectors)	66	66
Recorded Announcements (Integrated and Analog Line)	10	64
Integrated Announcement Circuit Packs	-	1
Announcement Circuit Pack Capacity	-	4 min 16 sec
Analog Line Announcement Queue Slots per System	50	150
Analog Line Announcement Queue Slots per Announcement	5	150
Integrated Announcement Queue Slots per System	-	50
Calls Connected per Announcement	1	5

* These maximum parameters do not apply if PagePac Paging Systems with zoning capability are used.

ITEM	R1V2	R1V3
Remote Access Barrier Codes	10	10
Restriction—Toll/Code		
Allowed Calls List Codes	10	10
Digit Absorption Lists	5	5
Ringback Queue Slots	120	120
Speech Synthesis Circuit Packs	6	6
Terminating Extension Groups	32	32
Members per Group	4	4
Time Slots:		
Total	512	512
Call Switching	483	483
Simultaneous Conversations	241	241
Tone Detectors:		
Call Progress	10	10
Touch-Tone	20	20
Traffic Handling Capability [in Hundred Call Seconds (CCS)]	8670	8670
Call Attempts/Busy Hour	3600	3600
Trunk Access Codes	118	157
Trunks	200	200
Trunk Groups	60	99
Trunks per Group	60	60
Voice Terminals (Also includes 515 or 510D terminals, external alerts, and announcement machines).	650	650
Max Button Modules (Terminal Modules [adjuncts] Terminals with more than 10 assignable buttons)	450	450
Max Digital Display Modules	225	225

Power

In order to maintain system integrity, dedicated power feeders must be used. Separate feeder circuits from a dedicated service are sufficient to serve this purpose. The feeders should not be used to power other equipment.

All system cabinets require 95 to 129 volts at 50 to 60 Hz ac service. Fused current drain requirements for a single cabinet are 15 amperes. See Section 10 for a more complete discussion of power requirements.

Cabling Distances

When the system layout is determined, maximum cabling distances to the system cabinet must be considered. The following are the allowable intra-premises cabling distances. In case of mixed wire sizes, use the columns for 26-Gauge Wire.

EQUIPMENT	24-GAUGE WIRE (0.5106-mm)		26-GAUGE WIRE (0.4049-mm)	
	FEET	METERS	FEET	METERS
Attendant Console	2400	732	1500	457
Powered from Switch	350	107	350	107
510D or 515 Terminals	3000	914	2200	670
513 BCT, 4410 or 4425 terminals, 610 BCT (See Data Module or EIA Interface) Max. distance from terminal or BCT to Module or ADU is 50 feet	-	-	-	-
Data Modules:				
Z702AL1-DSU Data Module Base	5000	1524	4000	1219
7404D Data Module	5000	1524	4000	1219
Digital Terminal Data Module	3400	1037	2200	670
Modular Processor Data Module	5000	1524	4000	1219
Modular Trunk Data Module	5000	1524	4000	1219
3270 Data Module	5000	1524	4000	1219
EIA Interface (Data Line Circuit Pack and ADU):				
19.2 kbps	2000	610	2000	610
9.6 kbps	5000	1524	4000	1219
4.8 kbps	7000	2130	6000	1827
2.0 kbps	12000	3654	10000	3050
1.2 kbps	20000	6100	16000	4875
0.3 kbps	40000	12200	30000	9150
Voice Terminals:				
Analog				
8 Port Board (TN742, TN769) (On-Premises or Out-of-Building—Same Premises)				
500 or 2500 Type (See Note 1)	15000	4410	9500	2794
7100 Series	10500	3088	7000	2058
16 Port Board (TN746) (On-Premises Only—no Out-of-Building—no bridging) (See Note 2)				
AT&T 500 or 2500 Type Terminals Without Adjuncts	2000	610	2000	610
Hybrid (TN762)				
7300 Series (Without Aux Power)	1000	305	750	229
7300 Series (With Aux Power)	2000	610	2000	610
Digital (TN754)				
7403D, 7405D, 7404D, 7407D	3000	914	2200	670
MET Sets (TN735)	1000	305	650	198

Notes:

1. Only AT&T 500 or 2500 type terminals can be used off-premises through a Central Office.
2. For detailed limitations, see Analog Line circuit pack description in Section 4.

Tones

The following call progress tones are generated by the system:

TONE	FREQUENCY	PATTERN (In ms)
Ringback Tone	440 Hz + 480 Hz	1000 on, 3000 off; repeated
Bridging Warning Tone*	440 Hz	500 on, 15000 off; repeated
Busy Tone	480 Hz + 620 Hz	500 on, 500 off; repeated
Call Waiting Tones Internal	750 Hz + 20 Hz	300 on; not repeated
External or Handled by Attendant	750 Hz + 20 Hz	100 on, 100 off, 100 on; not repeated
Priority Call	750 Hz + 20 Hz	100 on, 100 off, 100 on, 100 off, 100 on; not repeated
Coverage Tone	440 Hz	600 on, followed by silence; not repeated
Confirmation Tone	350 Hz + 440 Hz	100 on, 100 off, 100 on, 100 off, 100 on followed by silence; not repeated
Dial Tone	350 Hz + 440 Hz	Continuous
Intercept Tone	480 Hz & 620 Hz	250 on (480 Hz), 250 on (620 Hz); repeated
Reorder Tone	480 Hz + 620 Hz	250 on, 250 off; repeated
Call Waiting Ringback Tone	440 Hz + 480 Hz; 440 Hz	1000 on (440 Hz + 480 Hz), 200 on (400 Hz), 2800 off; repeated

* This tone is used with the Busy Verification feature.

The following ringing tone patterns are generated by the system:

RINGING TONE	PATTERN (In ms)
1	1200 on, 4000 off; repeated
2	400 on, 200 off, 600 on, 4000 off; repeated
3	200 on, 100 off, 200 on, 100 off, 600 on, 4000 off; repeated.

Indicator Lamp Signals

The following lamp signals are generated by the system for the attendant console and multi-appearance voice terminals.

LAMP SIGNAL	PATTERN (In ms)
Dark	Off
Lighted	On
Flashing	500 on, 500 off; repeated
Fluttering	50 on, 50 off; repeated
Broken Flutter	5 cycles of 50 on, 50 off, followed by 500 off; repeated
Wink	350 on, 50 off; repeated

Protocols

The various protocols used in the system are listed below with system application and maximum limitations.

PROTOCOL	APPLICATIONS	MAXIMUM DATA RATE	MAXIMUM DISTANCE
DCP	Digital Switch to Data Endpoints	160 kbps*	5000 ft (1524 m) for data 3400 ft (1036 m) for voice
RS-232C	Switch to SAT PDM to Host Computer MPDM to Printer	19.2 kbps	50 ft (15.2 m)
	MTDM for Downloading and High-Speed Data Transfer	64 kbps	17 ft (5.9 m)
	EIA Interface (Data Line to ADU)	19.2 kbps 9.6 kbps 4.8 kbps 2.4 kbps 1.2 kbps 0.3 kbps	2000 ft (610 m) 5000 ft (1524 m) 7000 ft (2130 m) 12000 ft (3654 m) 20000 ft (6100 m) 40000 ft (12200 m)
X.25	Communications Interface to DCS, CMS, or AUDIX	9.6 kbps	(see Note)
RS-366	Host Computer to ACU MTDM to ACU	 64 kbps	50 ft (15.2 m) 17 ft (5.9 m)
V.35	MPDM to Data Endpoints	56 kbps	50 ft (15.2 m)
Category A Coaxial	3270 Data Modules to 3270-Type Terminals or Cluster Controller	9.6kbps	500 ft (152 m)

* The DCP sends digitized voice and digital data in frames. Each frame consists of four fields or channels. The first field is a unique 3-bit framing pattern (24 kbps) that defines the frame boundary. The second field is a 1-bit control or signaling channel (8 kbps) between the digital switch and digital data endpoint. The third and fourth fields are two independent information (I) channels (64 kbps each).

Note: Data endpoint determines distance limitation.

Trunk Specifications

The specifications for the various trunk-type circuit packs are as follows:

TRUNK TYPE	CIRCUIT PACK	SPECIFICATIONS
Central Office	TN747B	Capacity: 8 Circuits Transmission: 1-Way In, 1-Way Out, or 2-Way 2-Wire 600 Ohms or RC Balance Network Signaling: Ground Start or Loop Start
Auxiliary Trunk	TN763B	Capacity: 4 Circuits Transmission: 1-Way In, 1-Way Out, or 2-Way 2-Wire Signaling: Loop Start on Tip and Ring; Two Additional Pairs Provide Seizure and Answer Supervision and/or Make Busy Information
Direct Inward Dialing	TN753	Capacity: 8 Circuits Transmission: 1-Way Incoming Fixed Impedance to DID Trunk Signaling: Wink or Immediate Start Accepts Touch-Tone Dialing
Tie Trunk	TN760B	Capacity: 4 Circuits Transmission: 4-Wire Tip and Ring Signaling: E & M. 760B supports Type 1 or Type 5 E&M Signaling
DS1 Trunk	TN722B	Capacity: 24 Trunks for Voice Grade Service. 23 Trunks for Alternate Voice/Data Service or DMI, One Trunk Used for Signaling. Mode: Multiplexes 24 or 23 Trunks onto 1 Facility and Demultiplexes 1 Facility into 24 or 23 Trunks Speed: Trunks at 64 kbps, 1 Facility at 1.544 Mbps Signaling: DS1 Over 4-Wire

Analog Transmission Characteristics

Frequency Response:
(Station-to-Station or Station-to-CO-Trunk, relative to loss at 1 kHz)

FREQUENCY	LOSS
60 Hz	>20 dB
200 Hz	<5 dB
300—3000 Hz	<1 dB
3200 Hz	<1.5 dB
3400 Hz	<3 dB

Insertion Loss:

CONNECTION TYPE	LOSS
On-Premises Station to On-Premises Station	6 dB
On-Premises Station to Off-Premises Station	3 dB
Off-Premises Station to Off-Premises Station	0 dB
Station-to-Trunk	0 dB
Trunk-to-Trunk	0 dB

Overload Level: +3 dBm0

Crosstalk: -70 dB

Intermodulation Distortion:

FOUR TONE METHOD	
Second Order Tone Products	>45 dB
Third Order Tone Products	>53 dB

Quantization Distortion:

SIGNAL LEVEL	DISTORTION LEVEL
+2 to -30 dBm0	35 dB
-40 dBm0	29 dB
-45 dBm0	25 dB

Sampling Rate: 8 kHz

Terminating Impedance: 600 ohms

Trunk Balance Impedance: 600 ohms or Complex Z (selectable)

Echo Return Loss:

The echo return loss of the switching equipment is infinite. The echo return loss of the station equipment can be engineered for greater than 18 dB over the range of 500 Hz to 2500 Hz.

Loop Resistance:

- TN742 or TN769—Loop resistance of up to 1300 ohms, including the station
- TN746—Loop resistance of up to 100 ohms, not including the station

Connection Bandwidth: 64 Kbits

Steady State Noise Level:

The steady state noise level presented to any busy path does not exceed 23 dBmc during the busy hour.

Impulse Noise:

The impulse noise is 0 count (hits) in 5 minutes at +55 dBmc during the busy hour.

Single Frequency Return Loss (Talking State):

Station to station—exceeds 12 dB
Station to 4-wire trunk connection—exceeds 14 dB
Station to 2-wire trunk connection—exceeds 12 dB

Peak Noise Level:

Analog to analog—20 dBmc
Analog to digital—19 dBmc
Digital to analog—13 dBmc

Service Codes and Facility Interface Codes

To help administer the Registration Program with respect to Private Line services, the FCC has established a requirement that Service Codes and Facility Interface Codes (FICs) be provided by the customer/vendor to the telephone company to technically identify the service requested. These codes are a shorthand way to describe the technical information necessary to design and implement a connection request.

Service Codes

Service Codes are issued by the FCC to equipment manufacturers/registrants that denote the type of registered terminal equipment and the protective characteristics with respect to the premises wire of the terminal equipment ports.

Private Line Service Codes are as follows:

7.0Y — Totally Protected Private Communications (microwave) Systems
7.0Z — Partially Protected Private Communications (microwave) Systems
8.0X — Port for Ancillary Equipment
9.0F — Fully Protected Terminal Equipment
9.0P — Partially Protected Terminal Equipment
9.0N — Unprotected Terminal Equipment
9.0Y — Totally Protected Terminal Equipment

Service Code Example

The system product line Service Code is 9.0F which indicates it is terminal equipment that has fully protected premises wire at the Private Line ports.

Facility Interface Codes (FICs)

An FIC does not relate to a particular item of terminal equipment but is simply a method to provide the technical information to order a specific Private Line circuit. In effect, it identifies registered Private Line port interfaces. These FICs consist of five alphanumeric characters constructed as follows:

First Character: Identifies the type of service

A = Automatic Identification Outward Dialing, or
M = Message Registration, or
O = Off-Premises Station, or
T = Tie Trunk

Second Character: Defines the transmission parameters

C = Conventional Term Set, or
L = Lossless Interface, or
X = Reserved

Third Character: Defines the number of conductors

1 = Type I Transmission Interface (2-wire) or
3 = Type III Transmission Interface (4-wire)

Fourth Character: Determines the signaling type

1 = Type I E&M Signaling Interface, or
2 = Type II E&M Signaling Interface, or
3 = Loop Signaling Interface, or
4 = Reserved, or
5 = Simplex Signaling

Fifth Character: Describes signaling arrangement and class of OPS Port

A = Registered Class A OPS Port (see Note)
B = Registered Class B OPS Port (see Note)
C = Registered Class C OPS Port (see Note)
E = Registered equipment provides ground on E lead to originate
M = Registered equipment provides battery on M lead to originate
X = Reserved

Note: Each type of Port (A, B, or C) indicates the capability of operation over loops with resistances in the following ranges:

Class A: 0—199 ohms
Class B: 200—899 ohms
Class C: 900 ohms or more

Facility Interface Code Example:

System Tie Trunks—FIC TL31M which indicates it is:

T — Tie Trunk
L — Lossless
3 — 4 Wire
1 — Type I E&M
M — Battery on the M Lead

System Service Connection Information

1. Trunk Type: Either ground start or loop start.
2. Signaling Arrangement Type (OPS): Type C Ringing Frequency 20 Hz.
3. Incoming Call Control: Wink Start, DID, Delay, Immediate are all available.

Private Line Connections

SERVICE	PE CODES	REGIST. CIRCUIT PACKS	FACILITY INTERFACE CODE	SERVICE CODE	JACK USOC	GRANDFATHERED TECHNICAL DESCRIPTION
Tie Trunks	63140	TN760B	TL31M	9.0F	RJ2GX	
OPS	63111	TN742	OL13C	9.0F	RJ2GX RJ21X RJ11X	

System Availability

The system is designed to provide continuous service with a small predicted outage time per year. System availability can be significantly affected by commercial power outages or adverse environmental conditions. Power for the common control is held over with batteries for 2 minutes to minimize the effect of these outages.

Outages of less than 250 milliseconds cause no service interruptions. During commercial power outages lasting more than 250 milliseconds and less than 2 minutes, service is restored automatically within a minute after power is restored, so there is no need to reload the programs or translations from the memory tape. During an extended power failure, customer-designated lines are automatically transferred to central office trunks. Long-term holdover (engineered to customer needs, typically 8 hours) is also available as an option.

ENVIRONMENTAL REQUIREMENTS

CONTENTS

Page

LIST OF FIGURES

LIST OF TABLES

ENVIRONMENTAL REQUIREMENTS

Main Equipment Room Requirements

This part provides information on the floor and wall space required for system equipment and associated peripheral equipment installed in the equipment room. Also included are specifications for temperature, humidity, air purity, lighting levels, and grounding.

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. A typical floor plan is shown in Figure 10-1.

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 cross-connect field and the cable access panel (when provided). Also, room for system growth should be considered.

Floor Loading

The floor must have a commercial floor loading code of at least 50 pounds per square foot. A single cabinet system weighs about 130 pounds, a fully loaded 3-cabinet system weighs about 330 pounds, and a 4-cabinet system weighs about 500 pounds. Thus, a free maintenance area of at least 8 square feet for a 3-cabinet system 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 10-2 shows the zones in the continental United States where bracing may be desirable.

Floor Space

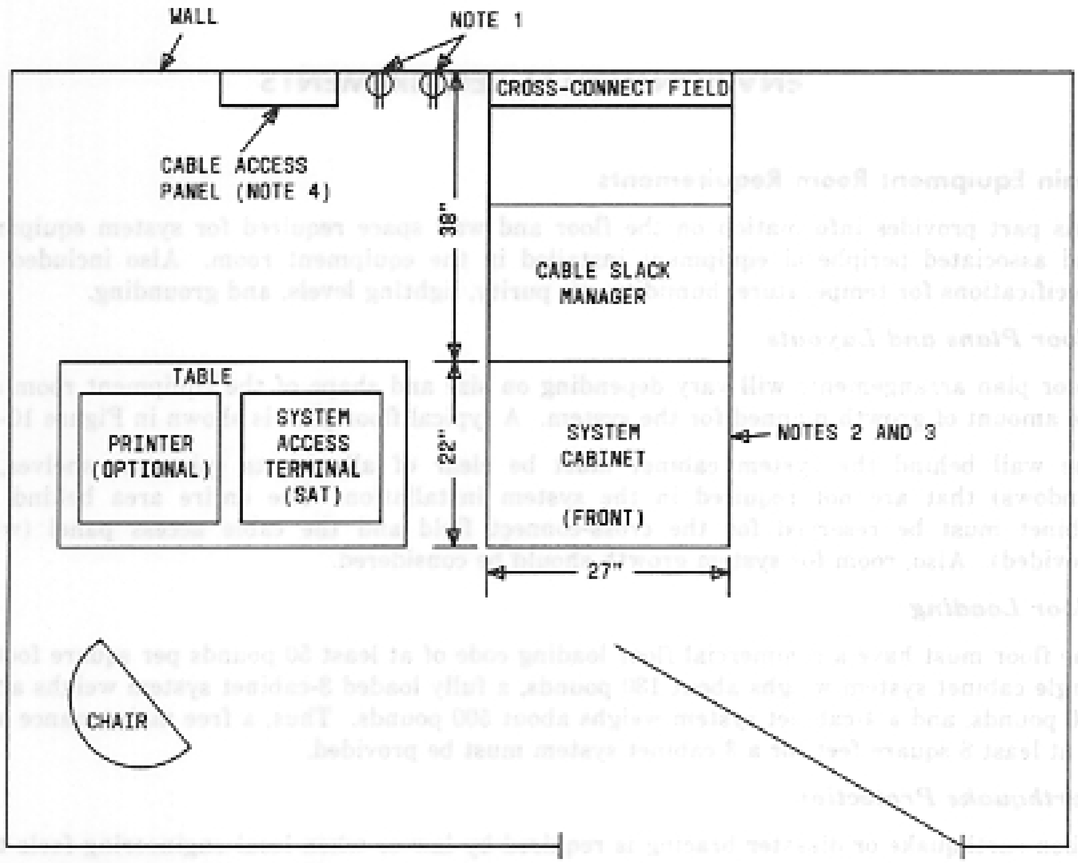
The following system equipment and optional peripheral equipment occupies floor space in the equipment room:

- System Cabinet and Cable Slack Manager—The system cabinet is 27 inches wide and 22 inches deep. A single cabinet system is about 20 inches high, a 2-cabinet system is 39 inches high, a 3-cabinet system is 58 inches high, and a 4-cabinet system is 77 inches high. The cable slack manager requires 38 inches between the cabinet and wall. The cabinet and cable slack manager occupy about 8 square feet of floor space.

References for Optional Equipment Requiring Floor Space

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

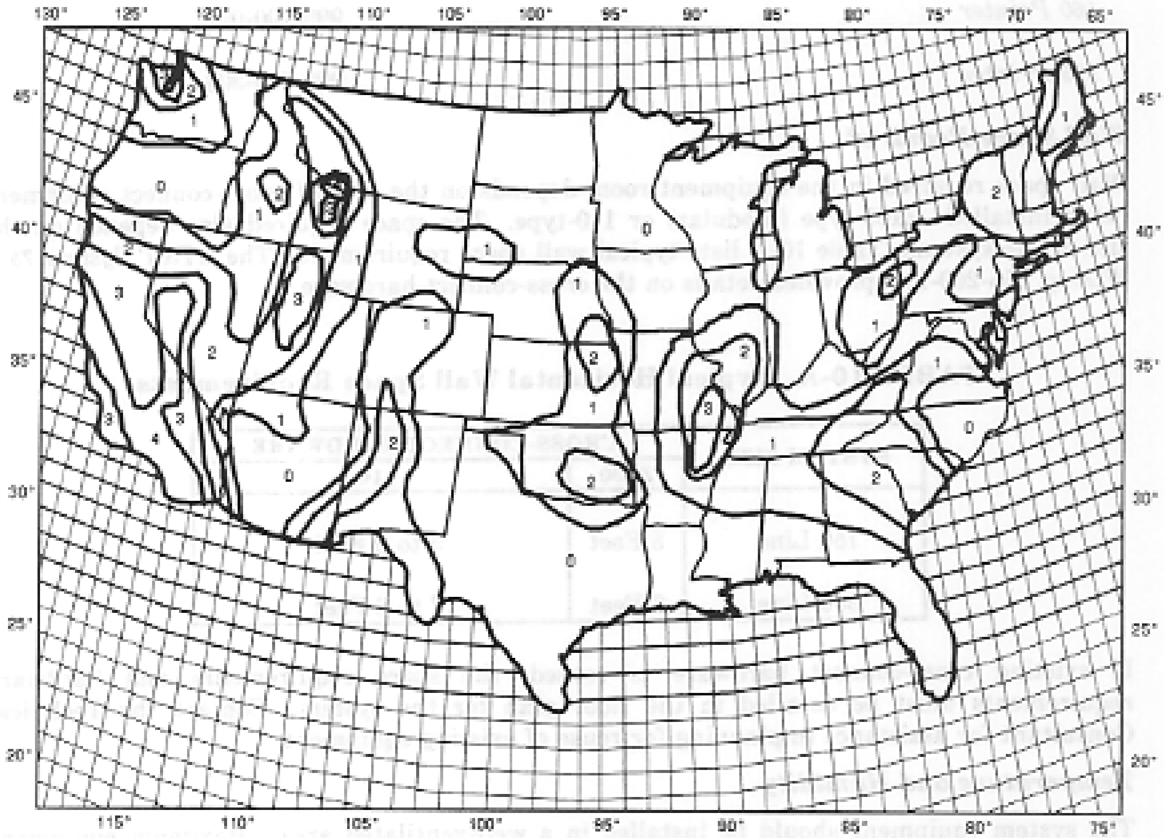
445 Printer 999-700-023



NOTES:

1. POWER OUTLETS SHOULD BE LOCATED OUTSIDE THE CROSS-CONNECT FIELD AREA. POWER OUTLET(S) MUST NOT BE UNDER SWITCH CONTROL AND MUST NOT BE SHARED WITH OTHER EQUIPMENT.
2. SYSTEM MUST BE GROUNDED BY ONE OF THE APPROVED METHODS LISTED IN THIS SECTION.
3. EARTHQUAKE PROTECTION MAY BE REQUIRED.
4. CABLE ACCESS PANELS ARE NOT PROVIDED WHEN 110-TYPE HARDWARE IS USED.

Figure 10-1. Typical Floor Plan



ZONES ARE CLASSIFIED IN INCREASING
EARTHQUAKE SUSCEPTIBILITY FROM 0 TO 4

Figure 10-2. Earthquake Environment (Continental USA)

Desk-Top Space

The 510D Personal Terminal and 513, 515, 4410, or 4425 Terminals can be located in the equipment room and require space on a desk or table. The 513, 515, 4410, and 4425 terminals occupy approximately 3.2 square feet of space. The 510D with optional keyboard occupies approximately 2.1 square feet 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:

3B2 AP	585-205-110
500 Business Communications Terminal	999-700-021
DATASPEED® 4410 Display Terminal	999-300-180
DATASPEED® 4425 Display Terminal	999-310-181
443 Printer	999-700-024
450 Printer	999-700-025

460 Printer 999-700-022

475 Printer 999-700-303

Wall Space Required

Wall space required in the equipment room depends on the type of cross-connect equipment being installed—Z100-type (modular) or 110-type. The space required also depends on the size of the system. Table 10-A lists typical wall space requirements. The *AT&T System 75—Wiring*, 555-200-111, provides details on the cross-connect hardware.

TABLE 10-A. Typical Horizontal Wall Space Requirements

SYSTEM SIZE	CROSS-CONNECT HARDWARE	
	Z100	110
150 Line	5 Feet	5 to 6 Feet
300 Line	8 Feet	7 to 8 Feet

If existing cross-connect hardware is reused, the space requirements and hardware requirements must be detailed in the floor plan for the system. Contact the Technical Consultant for assistance in planning for reuse of existing equipment.

Temperature and Humidity

The system equipment should be installed in a well-ventilated area. Maximum equipment performance is obtained at an ambient temperature between 40 and 110 degrees Fahrenheit (5 and 43 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

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 (RF Interference)

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 wavelength (150 meters for a frequency of 1000 kHz).

AC Power Requirements

Each cabinet requires a separate power outlet as shown in Figure 10-3. These outlets must not be shared with other equipment, must not be under switch control, and should be located outside the cross-connect field area, if possible. Any available power source can be used, as long as the phase or leg provides 115-ac volts at the required drain.

The System Access Terminal (SAT) should be connected to the power outlet as shown in Figure 10-3.

A system cabinet is UL-listed at 10 amperes 120 volts, or 1200 watts per cabinet. Therefore, the power for a 2-cabinet system is 2400 watts, the power for a 3-cabinet system is 3600 watts, and the power for a 4-cabinet system is 4800 watts.

Figure 10-4 depicts a typical power and grounding layout for the system cabinet(s).

* Trademark of Holaday Industries

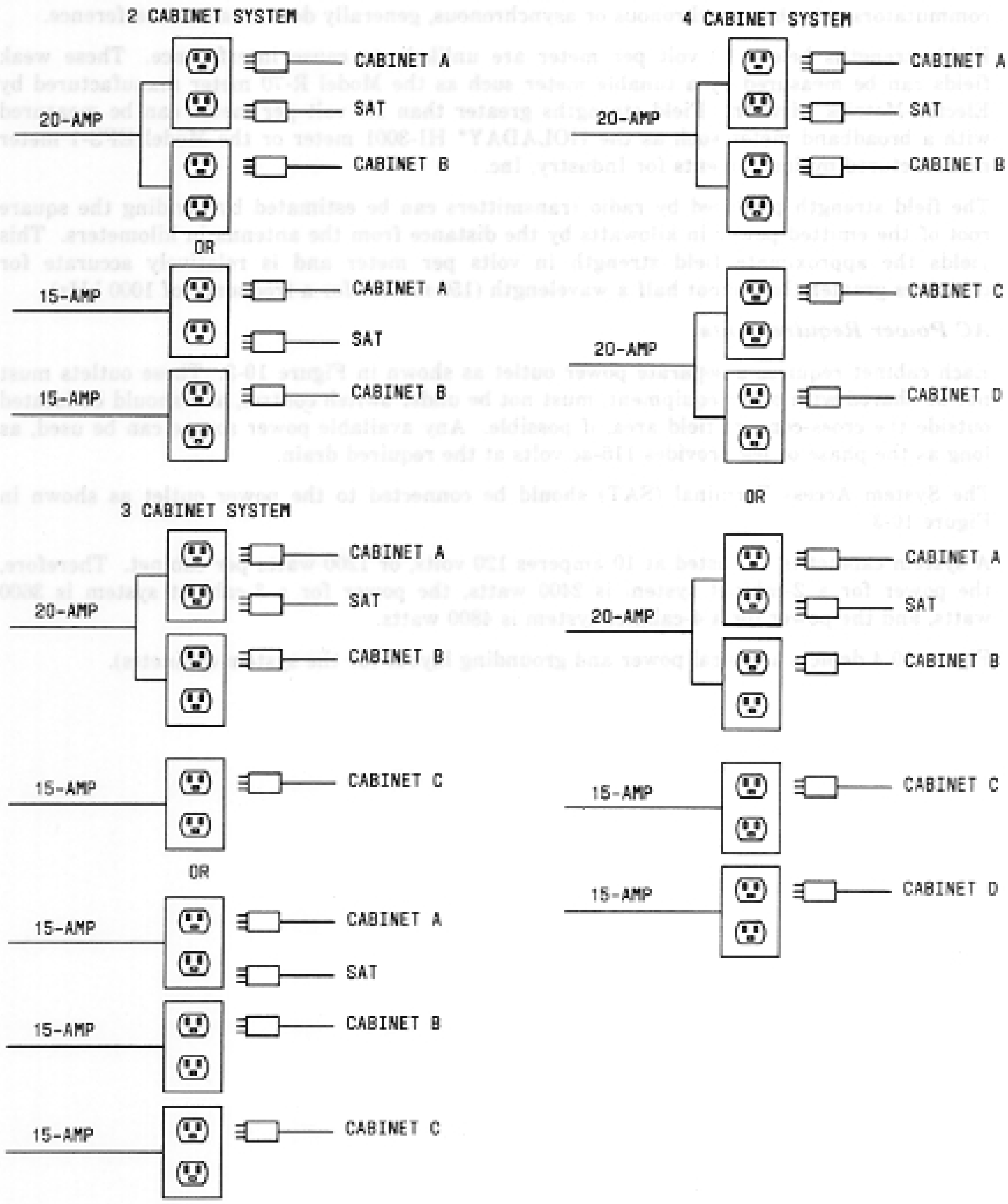


Figure 10-3. AC Power Requirements for Multiple Cabinet Systems

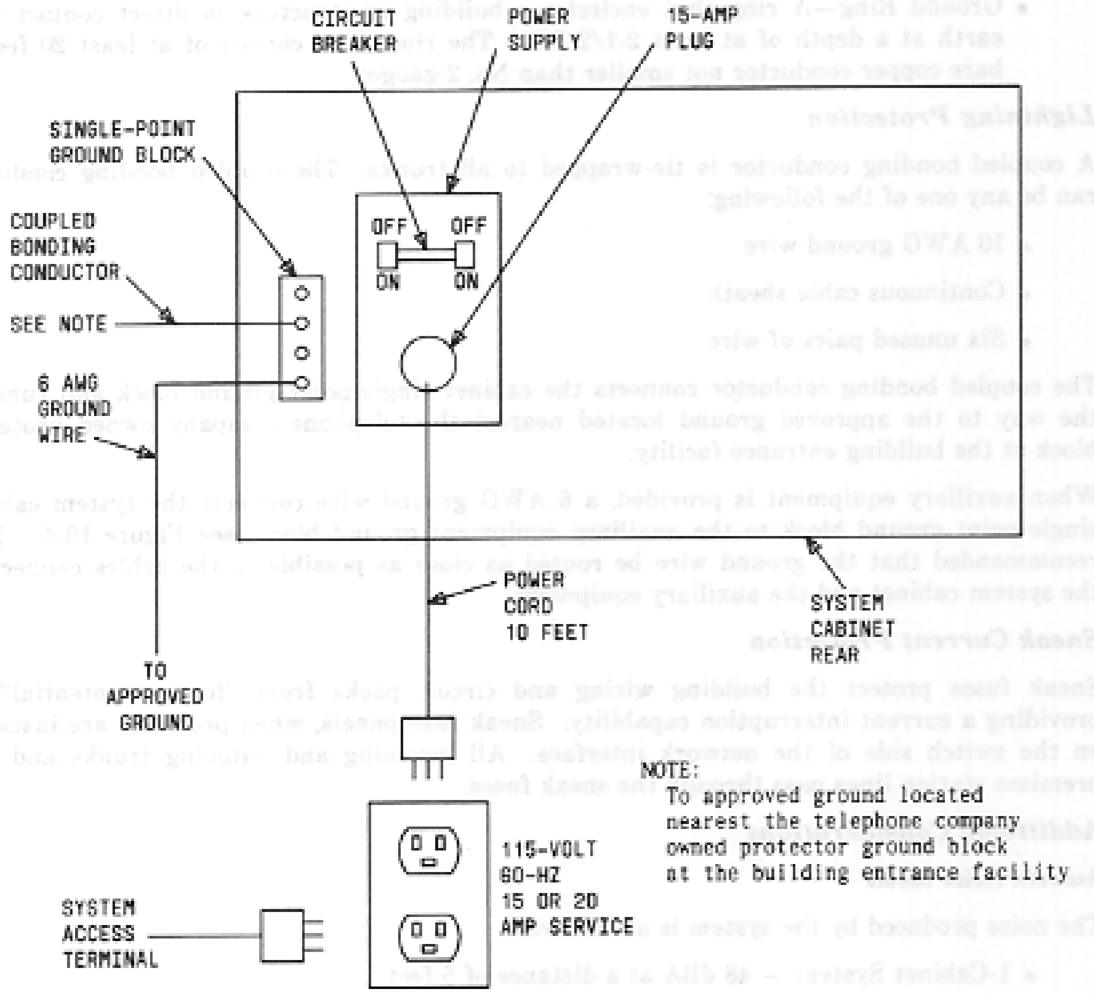


Figure 10-4. Typical Power and Grounding Layout

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 in diameter, that is connected to an underground metal water pipe that is in direct contact with earth for 10 feet or more.
- Concrete-Encased Ground—An electrode encased by at least 2 inches of concrete and located within and near the bottom of a concrete foundation or footing in direct contact with the earth. The foundation must be at least 20 feet of one or more steel

reinforcing bars or rods of not less than 1/2 inch in diameter, or at least 20 feet of bare, solid copper wire not smaller than No. 4 gauge.

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

Lightning Protection

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

- 10 AWG 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 approved ground located nearest the telephone company owned protector block at the building entrance facility.

When auxiliary equipment is provided, a 6 AWG ground wire connects the system cabinet single-point ground block to the auxiliary equipment ground block (see Figure 10-4). It is recommended that the ground wire be routed as close as possible to the cables connecting the system cabinet and the auxiliary equipment.

Sneak Current Protection

Sneak fuses protect the building wiring and circuit packs from "foreign potential" by providing a current interruption capability. Sneak fuse panels, when provided, are installed on the switch side of the network interface. All incoming and outgoing trunks and off-premises station lines pass through the sneak fuses.

Additional Considerations

Acoustic Noise Levels

The noise produced by the system is as follows:

- 1-Cabinet System — 48 dBA at a distance of 5 feet
- 2-Cabinet System — 50 dBA at a distance of 5 feet
- 3-Cabinet System — 52 dBA at a distance of 5 feet
- 4-Cabinet System — 53 dBA at a distance of 5 feet

Note: If the system cabinet door is open, there is an additional 1 dBA of noise. The tape recorder also causes additional noise. When the tape recorder is reading data, there is an additional 2 dB of noise. When the tape recorder is rewinding or fast winding, there is an additional 4 dBA of noise.

Heat Dissipation

A fully-loaded 4-cabinet system dissipates approximately 6700 BTUs per hour. However, the typical average for a 1-cabinet system is a dissipation of 1700 BTUs per hour.

Standby Power System

The power units provide a 250-millisecond power holdover for the system. A battery reserve holds the switch processing elements and fans for 2 minutes. If additional holdover time is required, an external Uninterruptible Power Supply (UPS) can be provided as an alternate source of power during a commercial power failure. Any UPS that meets the requirements given in Table 10-B can be used with the system.

TABLE 10-B. UPS Power Requirements

NO. OF CABINETS	VOLT-AMPERE RATING	FREQUENCY HERTZ	POWER FACTOR	MAXIMUM CUT-IN TIME (Milliseconds)
1	1200	60±5%	0.6	200
2	2400	60±5%	0.6	200
3	3600	60±5%	0.6	200
4	4800	60±5%	0.6	200

House Wiring

House wiring includes all on-premises wiring on the customer side of the cross-connect field. The cross-connect field can be either Z100-type (modular jacks) or 110-type hardware.

Wiring is distributed from the cross-connect field by 25-pair cables. The 25-pair cables are either connected directly to terminal wall jacks using 258-type adapters or to satellite locations. Satellite locations are used when already present or when required by the length of the wiring runs from the switch to the terminals. The 25-pair cables can be divided into either 4-pair or 3-pair wiring groups (4-pair wiring groups are recommended). From the satellite locations, 4-pair D-inside cables connect the satellite locations to information outlets (modular wall jacks).

The *AT&T System 75—Wiring*, 555-200-111, provides details on the cross-connect hardware and wiring distribution.

REFERENCES

The following is an abbreviated listing of System 75 XE documents. Included is a brief description of each document in the list. For a complete listing of System 75 documents, refer to the *AT&T System 75—Documentation*, 555-200-010.

AT&T System 75 and System 85—Terminals and Adjuncts 555-015-201

Provides concise physical and functional descriptions of the peripheral equipment that can be used with System 75 and System 85. It is intended as an aid for both AT&T and customer personnel in selecting appropriate components for these systems and in training and management.

AT&T System 75—Wiring Guide 555-200-111

Provides the information necessary for installing inside wiring for the system.

AT&T System 75 and System 75 XE—Feature Description 555-200-201

Provides a technical description of the Release 1 Version 1, Release 1 Version 2, Release 1 Version 3, and System 75 XE features and parameters.

AT&T System 75 and System 75 XE—Administration 555-200-500

Describes the management of the system's administration and operation. Includes the guidelines for initialization, reconfiguration, backup procedures, monitoring system performance, and maintaining system security. Includes a description of the tasks that can be performed via the System Access Terminal and the prerequisites for completion.

AT&T System 75 and System 75 XE—Planning/Configuration 555-200-600

Provides a method for defining the customer's system requirements and for collecting the information used to estimate system hardware requirements.

AT&T System 75 and System 75 XE—Implementation—Release 1 Version 2 555-200-651

Provides the procedures and associated forms for collecting system and terminal software information. This information is used to initialize the system using the System Access Terminal.

AT&T System 75 and System 75 XE—Implementation—Release 1 Version 3 555-200-652

Provides the procedures and associated forms for collecting system and terminal software information. This information is used to initialize the system using the System Access Terminal.

AT&T System 75—Console Operations 555-200-700

Provides "how-to-operate" instructions for the attendant console. Serves as a reference when defining the console control keys and Incoming Call Identification requirements.

AT&T System 75—Voice Terminal Operations 555-200-701

Describes all the voice features and provides the "how-to-operate" instructions for each voice terminal. Serves as a reference when defining user requirements.

AT&T System 75—Automatic Call Distribution (ACD)—Agent Instructions 555-200-722

Provides information for use by agents after training is completed. The various ACD features are described and the procedures for using them are provided in this document. The information in this document applies only to Release 1 Version 3 systems.

AT&T System 75—Hospitality Operations 555-200-723

Contains the procedures for using the Hospitality Services of AT&T System 75, Release 1 Version 3. These services include a group of System 75-based features that support the lodging industry. Hotels and motels use the features to improve their property management and to provide assistance to their employees and clients.

AT&T System 75—Automatic Call Distribution (ACD)—Supervisor Instructions 555-200-724

Provides information for use by supervisors after training is completed. The various ACD features are described and the procedures for using them are provided in this document. The information in this document applies only to Release 1 Version 3 systems.

AT&T System 75 XE—Switch Installation and Test 555-201-104

Provides the information necessary to perform the tasks of installing and testing the system's common equipment. Includes a description of the necessary tools and equipment.

AT&T System 75 XE—System Maintenance 555-201-105

Provides the information necessary for monitoring, testing, and maintaining the AT&T System 75 XE. It is intended to cover many of the faults and troubles that can occur in the system.

AT&T System 75 XE—System Upgrades and Additions 555-201-106

Provides procedures and information for upgrading or making additions to an operational system after the initial switch installation.

AT&T Telecommunication Electrical Protection 350-060

Provides practical, functional information and application detail combined with training material for telecommunication engineers in the electrical protection field.

User instructions are also available for all terminals.

GLOSSARY

Access Code

A 1-, 2-, or 3-digit dial code used to activate or cancel a feature or access an outgoing trunk. The star (*) and pound (#) can be used as the first digit of an access code.

Access Tie Trunks

Tie trunks used to handle normal ETN calls between Main and Tandem switches.

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.

Applications Processor (AP)

A minicomputer used to support a Message Center service.

Asynchronous Data Transmission

A scheme for transmitting data where each character is preceded by a start bit and followed by a stop bit, thus permitting data elements to occur at irregular intervals. This type transmission is advantageous when transmission is not regular (characters typed at a keyboard).

Asynchronous Data Unit (ADU)

A data communications equipment (DCE) type device that allows direct connection between RS-232C equipment and the system digital switch.

Attendant

The operator of the console.

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.

Audio Information Exchange (AUDIX)

A unit that provides voice mail service to users.

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 appearance of a voice terminal's extension at one or more other voice terminals.

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

A multi-conductor electrical path used to transfer information over a common connection from any of several sources to any of several destinations.

Bus, Time Division Multiplex

See Time Division Multiplex Bus.

Business Communications Terminal

An advanced series of semi-intelligent terminals.

Bypass Tie Trunks

One-way, outgoing tie trunks from a Tandem switch to a Main switch in an ETN. These trunks, provided in limited quantities, are used as a "last-choice" route when all trunks to another Tandem switch are busy. Bypass tie trunks are used only if all applicable intertandem trunks are busy.

Byte

A sequence of bits, 8 bits long, that is usually shorter than a word. A word is 16 bits long.

Call Appearance, Attendant Console

Six buttons, labeled a through f, used to origin. .e, 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. Two lamps next to the button show the status of the call appearance or status of the call.

Call Management System

An adjunct processor that collects data from an ACD and generates reports to be stored or displayed concerning status of agents, splits, and trunks.

Callback Call

A call that is automatically returned to a voice terminal user who activated the Automatic Callback or Ringback Queuing feature.

Call Waiting Ringback Tone

A low-pitched tone identical to the ringback 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.

Central Office

The location housing telephone switching equipment that provides local telephone service and access to toll facilities for long-distance calling.

Central Office Codes

The first three digits of a 7-digit public network telephone number. These codes are numbered from 200 through 999.

Central Office Trunk

A telecommunications channel that provides access from the system to the public network through the local central office.

Channel

A communications path for transmitting voice and data.

Class of 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 ring simultaneously when a call is redirected to it by Call Coverage. Any one of the group can answer the call.

Coverage Call

A call that is automatically redirected from the called party's extension number to an alternate answering position when certain coverage 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 extension, or Message Center Hunt Group designated as an alternate answering position in a coverage path.

Covering User

The person at an alternate answering position who answers a coverage call.

Data Channel

A communications path between two points used to transmit digital signals.

Data Communications Equipment (DCE)

The equipment on the network side of a communication link that provides all the functions required to make the binary serial data from the source or transmitter compatible with the communications channel.

Data Terminal Equipment (DTE)

Equipment comprising the source or sink of data, or both, that also provides communication control functions (protocol). DTE is any piece of equipment at which a communications path begins or ends.

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.

Distributed Communications System (DCS)

A network of two or more switches, each with its terminals and trunks, configured to function as a single large system.

Digital Data Endpoints

In System 75 XE, digital data endpoints include the following:

- 510D Personal Terminal or 515-Type Business Communications Terminal
- 7404D Terminals
- 7407D Equipped With Optional Data Module Base
- Asynchronous Data Units
- Digital Terminal Data Modules
- Modular Processor Data Modules
- Modular Trunk Data Modules
- 3270 Data Modules
- Internal Data Channels

Digital Multiplexed Interface (DMI)

Specifies the remote interface requirements for multiplexed data communications between a host computer and a private switching system.

Digital Terminal Data Module (DTDM)

An adjunct to Model 7403D or 7405D voice terminals. Provides the required interface between the system and a data terminal such as a 513 Business Communications Terminal.

A circuit in a telecommunications channel designed to handle digital voice and 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 special tandem tie trunk network that has automatic call routing capabilities based on the number dialed and most preferred route available at the time the call is placed. Each switch in the network is assigned a unique private network office code (RNX) and each voice terminal is assigned a unique extension number.

End-to-End Signaling

The transmission of touch-tone signals generated by 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 1- to 5-digit number assigned to each voice terminal, certain system groups, data modules, 510D Personal Terminal, or 515 Business Communications Terminal within the system. A 1- or a 5-digit extension number is available for Version 2 and Version 3.

External Call

A connection between 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 geographical area.

Ground-Start Trunk

On outgoing calls, System 75 XE 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 XE 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

A format used to initiate a data connection between two data module devices.

Home Numbering Plan Area Code

The local area code. The area code does not have to be dialed to call numbers within the local geographical area.

Immediate-Start Tie Trunk

After establishing a connection with the distant switching system for an outgoing call, the system waits a nominal 65 milliseconds before sending the digits of the called number. This allows time for the distant system to prepare to receive the digits. Similarly, on an incoming call, the system has less than 65 milliseconds to prepare to receive the digits.

Information Exchange

The exchange of data between users of two different systems (System 75 XE and host computer) over a local area network.

In-Use Lamp

A red lamp on a multi-appearance voice terminal that lights to show which call appearance will be selected when the handset is lifted or which call appearance is active when a user is off-hook.

Intercept Tone

An alternating high and low tone; indicates a dialing error or denial of the service requested.

Interface

A common boundary between two systems or pieces of equipment.

Internal Call

A connection between two users within the system.

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 XE 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.

Main/Satellite/Tributary

A Main switch provides: interconnection, via tie trunks, with one or more subtending switches, called Satellites; all attendant positions for the Main/Satellite configuration; and, access to and from the public network. To a user outside the complex, a Main/Satellite configuration appears as a single switch, with a single Listed Directory Number (LDN). A Tributary is a switch, connected to the Main via tie trunks, but which has its own attendant position(s) and its own LDN.

Message Center

An answering service for calls that might otherwise go unanswered; an agent 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.

Modular Processor Data Module

See Processor Data Module.

Modular Trunk Data Module

See Trunk Data Module.

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)

In terms of 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.

Private Network Office Code (RNX)

The first three digits of a 7-digit private network number. These codes are numbered 220 through 999, excluding any codes that have a 0 or 1 as the second digit.

Processor Data Module (PDM)

Provides the required interface between the system and an EIA computer or data terminal.

Property Management System (PMS)

A stand-alone computer which Lodging and Health Services organizations use for services such as reservations, housekeeping, billing, etc.

Protocol

A set of conventions or rules governing the format and timing of message exchanges to control data movement and correction of errors.

Public Network

The network that can be openly accessed by all customers for local or long-distance calling.

Queue

An ordered sequence of calls waiting to be processed.

Queuing

The process of holding calls in order of their arrival to await connection to an attendant, to an answering group, or to an idle trunk. Calls are automatically connected in first-in, first-out sequence.

Random Access Memory (RAM)

A storage arrangement whereby information can be retrieved at a speed independent of the location of the stored information.

Read Only Memory (ROM)

A storage arrangement primarily for information retrieval applications.

Recall Dial Tone

Three short bursts of tone followed by steady dial tone; indicates the system has completed some action (such as holding a call) and is ready to accept dialing.

Redirection Criteria

The information administered for each voice terminal's coverage path that determines when an incoming call is redirected to coverage.

Remote Home Numbering Plan Area Code (RHNPA)

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 500, 2500, 7101A, 7103A, and 7104A).

Software

A set of computer programs that accomplish one or more tasks.

Split

A condition whereby a caller is temporarily separated from a connection with the attendant. This split condition automatically occurs when the attendant, active on a call, presses the Start button.

Standard Serial Interface (SSI)

A communications protocol developed by AT&T Teletype Corporation for use with the 500 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, broken flutter, 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 Switch

A switch within an ETN that provides the logic to determine the best route for a network call, possibly modifies the digits outpulsed, and allows or denies certain calls to certain users.

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.

Tape Drive Assembly

A tape storage device that stores the software information for the system.

Tie Trunk

A telecommunications channel that directly connects two private switching systems.

Time Division Multiplex Bus

A special bus that is time shared by preallocating short time slots to each transmitter on a regular basis. In a PBX, all port circuits are connected to the time division multiplex bus permitting any port to send a signal to any other port.

Tone Ringer

A device with a speaker, used in electronic voice terminals to alert the user.

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 (telephone).

Uniform Dial Plan

A feature that allows a unique 4- or 5-digit number assignment for each terminal in a multi-switch configuration, such as a Distributed Communications System (DCS) or Main/Satellite/Tributary configuration.

Wide Area Telecommunications Service (WATS)

A service that allows calls to a certain area or areas for a flat-rate charge based on expected usage.

Wink-Start Tie Trunk

After establishing a connection with a distant switching system for an outgoing call, the system waits for a momentary signal (wink) before sending the digits of the called number. Similarly, on an incoming call, the system sends the wink signal when ready to receive digits.

Write Operation

The 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

AT&T SYSTEM 75 AND SYSTEM 75 XE   Date: ________________
SYSTEM DESCRIPTION
555-201-200   Issue 1

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2/5/88

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			COVERAGE			DIAL & BUTTON ACCESSED					BUTTON ONLY								TRUNKING / NETWORK											
			BTN ONLY																CLASS OF RESTRICTION											
CUG ANS GRP		CONSULT	CUG CALLBACK	GO TO CUG	SEND ALL CALLS		CALL PARK	LWC/LWCC	NIGHT SVC		BSY IND-STA	BSY IND-FACILITY	INTERCOM	MAN-MSG WAIT	MANUAL SIGNAL	PRIVACY	BRIDGING		CO.	WATS	TIE	FX	APLT	PAGING	COR ASSIGNMENT		ARS	FRL		POWER FAILURE

ENTRY REQUIRED IF FEATURE IS TO BE ASSIGNED

						DIAL & BUTTON ACCESSED											BTN ONLY			GROUPS				
						CLASS OF SERVICE							ABBR DIAL											
USER NAME	2500 OR 7406 SYSTEM 75 TERMINAL TYPE	ADJUNCT TYPE	ADJUNCT TYPE	Port EXISTING EXTENSION NUMBER	SYSTEM 75 EXTENSION NUMBER	AUTO CLBK	RCBK QUE	CALL FORWARD	DATA PRVCY	PRIORITY	COS ASSIGNMENT		PERSONAL	GROUP	SYSTEM	LAST DIALED			PKU GROUP	[illegible]	UDC/DDC		COVER GROUP
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- This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause interference to radio communications.
- It has been tested and found to comply with the limits for a Class A computing device pursuant to Subpart J of Part 15 of FCC Rules, which are designed to provide reasonable protection against such interference when operated in a commercial environment.
- Operation of this equipment in a residential area is likely to cause interference in which case the user at his or her own expense will be required to take whatever measures may be required to correct the interference.
- The voice terminals described in Section 4 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	1.0A
Network Interface	RJ21X* or RJ2GX

*Ground start trunks are recommended.

TAB PLACEMENT

Place tabs as follows:

- OVERVIEW tab before page 1-1
- FUNCTIONAL DESCRIPTION tab between page 1-1 and FUNCTIONAL DESCRIPTION page i
- FEATURE DESCRIPTIONS tab between page 2-29 and FEATURE DESCRIPTIONS page i
- SYSTEM PARAMETERS tab between page 3-296 and SYSTEM PARAMETERS page i
- GLOSSARY tab between page 4-9 and page 5-1
- INDEX tab between page 5-12 and page 6-1

OVERVIEW

This manual provides a technical description of the system features and parameters.

Purpose

This manual, along with the *AT&T System 75 Reference Manual—System Description*, 555-200-200, 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*, 555-200-650 for V1 and 555-200-651 for V2, for software initialization and subsequent changes in feature assignments.

Organization

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

- SECTION 2—FUNCTIONAL DESCRIPTION—Provides a general description of the functions and services provided with System 75. These functions and services are divided into four groups. The four groups are Voice Management, Data Management, Network Services, and System Management. Each group of functions and services is described separately and includes a list of associated features. The listed features are fully described in Section 3 of this manual.
- SECTION 3—FEATURE DESCRIPTIONS—Provides a detailed description of the features associated with Voice Management, Data Management, Network Services, and System Management. These feature descriptions are arranged in alphabetical order, regardless of function area.
- 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—GLOSSARY—Provides a glossary for the entire manual.
- SECTION 6—INDEX—Provides a permuted index for the entire manual.

An individual Table of Contents is provided for Sections 2 through 4 of this manual.

All features described in this manual apply to both Release 1 Version 1 and Release 1 Version 2 systems, unless otherwise noted as "V1" or "V2."

FUNCTIONAL DESCRIPTION

CONTENTS

LIST OF FIGURES

LIST OF TABLES

OVERVIEW

This section describes the features, functions, and services provided with System 75. These features, functions, and services are divided into four groups: Voice Management, Data Management, Network Services, and System Management. Each group of functions and services is described separately and the description includes a list of associated features. The listed features are fully described in Section 3 of this manual.

VOICE MANAGEMENT OVERVIEW

The many Voice Management features available with the system allow the individual needs of everyone in the system to be met. As the individual needs change, the assigned features can also be changed. The Voice Management features provide many important services, such as saving time and making calling more convenient.

Voice Management Features

The following features are associated with Voice Management.

Abbreviated Dialing
AP Demand Print
Attendant Auto-Manual Splitting
Attendant Call Waiting
Attendant Control of Trunk Group Access
Attendant Direct Extension Selection With Busy Lamp Field
Attendant Direct Trunk Group Selection
Attendant Display
Attendant Recall
Attendant Release Loop Operation
Automatic Callback
Automatic Incoming Call Display (V2)
Bridged Call Appearance
Busy Verification of Terminals and Trunks (V2)
Call Coverage
Call Forwarding All Calls (V1)
Call Forwarding All Calls (V2)
Call Park
Call Pickup
Call Waiting Termination
Centralized Attendant Service (V2)
Class of Restriction
Class of Service
Code Calling Access
Conference—Attendant
Conference—Terminal
Consult
Coverage Callback
Coverage Incoming Call Identification
Dial Access to Attendant
Dial Plan
Direct Department Calling and Uniform Call Distribution
Direct Inward Dialing
Direct Outward Dialing
Distinctive Ringing
Facility Busy Indication
Go To Cover
Hold
Hot Line Service
Hunting
Individual Attendant Access (V2)
Integrated Directory
Intercept Treatment

Intercom—Automatic
Intercom—Dial
Inter-PBX Attendant Calls (V2)
Last Number Dialed
Leave Word Calling
Line Lockout
Loudspeaker Paging Access
Manual Message Waiting
Manual Originating Line Service
Manual Signaling
Multi-Appearance Preselection and Preference
Multiple Listed Directory Numbers
Music-On-Hold Access
Night Service—Night Console Service
Night Service—Night Station Service
Night Service—Trunk Answer From Any Station
Personal Central Office Line
Personalized Ringing (V2)
Power Failure Transfer
Priority Calling
Privacy—Attendant Lockout
Privacy—Manual Exclusion
Recall Signaling
Recorded Telephone Dictation Access
Remote Access
Restriction—Controlled
Restriction—Miscellaneous Terminal
Restriction—Miscellaneous Trunk
Restriction—Toll/Code
Restriction—Voice Terminal—Inward
Restriction—Voice Terminal—Manual Terminating Line
Restriction—Voice Terminal—Origination
Restriction—Voice Terminal—Outward
Restriction—Voice Terminal—Termination
Ringback Queuing
Rotary Dialing (V2)
Send All Calls
Senderized Operation
SMDR Account Code Dialing
Speech Synthesis (V2)
Straightforward Outward Completion
Temporary Bridged Appearance
Terminating Extension Group
Through Dialing
Timed Reminder
Touch-Tone Dialing
Transfer
Trunk Group Busy/Warning Indicators To Attendant
Trunk Identification By Attendant (V2)
Trunk-To-Trunk Transfer
Voice Terminal Display

DATA MANAGEMENT OVERVIEW

System 75 is a private digital switching system that permits connections with a variety of data equipment. Data terminals, printers, graphics, facsimile equipment, and computers can be connected to the switch via the following protocols or interfaces:

- RS-232C
- RS-449
- RS-366
- Standard Serial Interface (SSI)
- Teletypewriter (TTY) Modes
- BX.25 Packet Switching
- Digital Communications Protocol (DCP)
- Binary Synchronous Communications (Bisync)
- Consultative Committee International Telegraph and Telephone (CCITT) V.35

The physical connection can be through a digital data module or analog modem.

System 75 provides the client options for selecting data modules (or data-like devices such as a Data Line Circuit [DLC]) for Terminal Dialing. However, clients can also use data modules without Terminal Dialing for use with host computers, printers, or other such applications. Computer file transfer at a speed rate of 64 kbps is possible with the introduction of the new Modular Processor Data Module (MPDM) and the Modular Trunk Data Module (MTDM).

The family of data modules also includes a Processor Data Module (PDM), a Digital Terminal Data Module (DTDM), a Modular Data Module (MDM), a Trunk Data Module (TDM), and a 3270 Data Module. The data modules are generally more versatile than modems, operate at faster data rates, and provide additional features.

The Model 7404D voice terminal has a built-in data module that allows the voice terminal to control and be connected to a data terminal. Data calls can be originated or disconnected using the key pad of the attached data terminal. Voice calls can be made or received while a data call is in progress. Model 7407D has the same data feature functionality as the Model 7404D, but does not have a built-in data module. An optional base containing a data module for use with the 7407D is available.

The AT&T Personal Terminal 510D, which operates in alphanumeric and graphics character set mode, provides the equivalent of a Model 7405D voice terminal equipped with a Digital Terminal Data Module, a 513 Business Communications Terminal (BCT), and a Digital Display Module.

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

The 510D terminal or 515 BCT provides an all-digital interface with the system. Through its built-in RS-232C interface, the 510D terminal or 515 BCT can connect to other data equipment.

The DLC, which provides eight ports to connect user's asynchronous Electronic Industries Association (EIA) RS-232C interface to Data Terminal Equipment (DTE), can be used as an alternative to DTDM or PDM.

All data modules except the MPDM and 3270 provide a standard RS-232C interface. The MPDM provides either RS-232C, V.35 or RS-449 interface. The MPDM can also emulate an automatic calling unit (ACU) and supports the RS-366 interface. The ACU emulation and RS-366 interface are required for Keyboard Dialing, discussed later in this section. The 3270 Data Module provides a Category A coaxial Digital Communications Equipment (DCE) interface for connection to 3270-type data terminals or a cluster controller. It also provides a DCP interface for connection to the digital switch.

The 3270 Data Module is available in the following three models:

- 3270T (Terminal)—Connects to a Category A 3270-type terminal, such as the IBM* 3278 Information Delivery System. The 3270T Data Module must connect through the switch to a 3270C (Controller) Data Module.

- 3270A (Asynchronous)—Provides the same function as the 3270T Data Module. It also allows the 3270-type terminal to emulate a Digital Equipment Corporation VT100 or an AT&T Information Systems asynchronous terminal.

- 3270C (Controller)—Connects an IBM 3274 or 3276 cluster controller to the switch. A 3270C Data Module can contain as many as eight ports.

System 75 supports digital-to-digital, digital-to-analog, analog-to-digital, and analog-to-analog data calls. Digital data endpoints are data modules and associated data equipment, 510D terminals or 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 data equipment connected to the system through analog lines or trunks. For data calls, the user can access the system through these digital or analog data endpoints.

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 510D terminal or 515 BCT, a 7403D or 7405D voice terminal with a DTDM, a 7404D with its built-in data module, and a 7407D with an optional data module base. Calls to or from other equipment are either voice-only or data-only.

Data Management Features

The following features are associated with Data Management:

Data Call Setup
Data Hot Line (V2)
Data-Only Off-Premises Extensions
Data Privacy
Data Restriction
Digital Multiplexed Interface (V2)
DS1 Tie Trunk Service (V2)
EIA Interface (V2)
Modem Pooling
Permanent Switched Calls (V2)
Uniform Call Distribution (UCD)

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* Trademark of International Business Machines

Data Networking

Data networking connects two or more data endpoints. System 75 is a highly reliable, centralized switch that provides switched access between the endpoints. Typical data communications configurations for the system are shown in Figure 2-1.

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 and their implementation within System 75 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 switches, 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 2-2.

  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:

- DATAPHONE* Digital Service
- Distributed Communications System (DCS)
- Electronic Tandem Network (ETN)
- Enhanced Private Switched Communications Service (EPSCS)
- Private line (PL)
- Tandem tie trunk

* Service Mark of AT&T

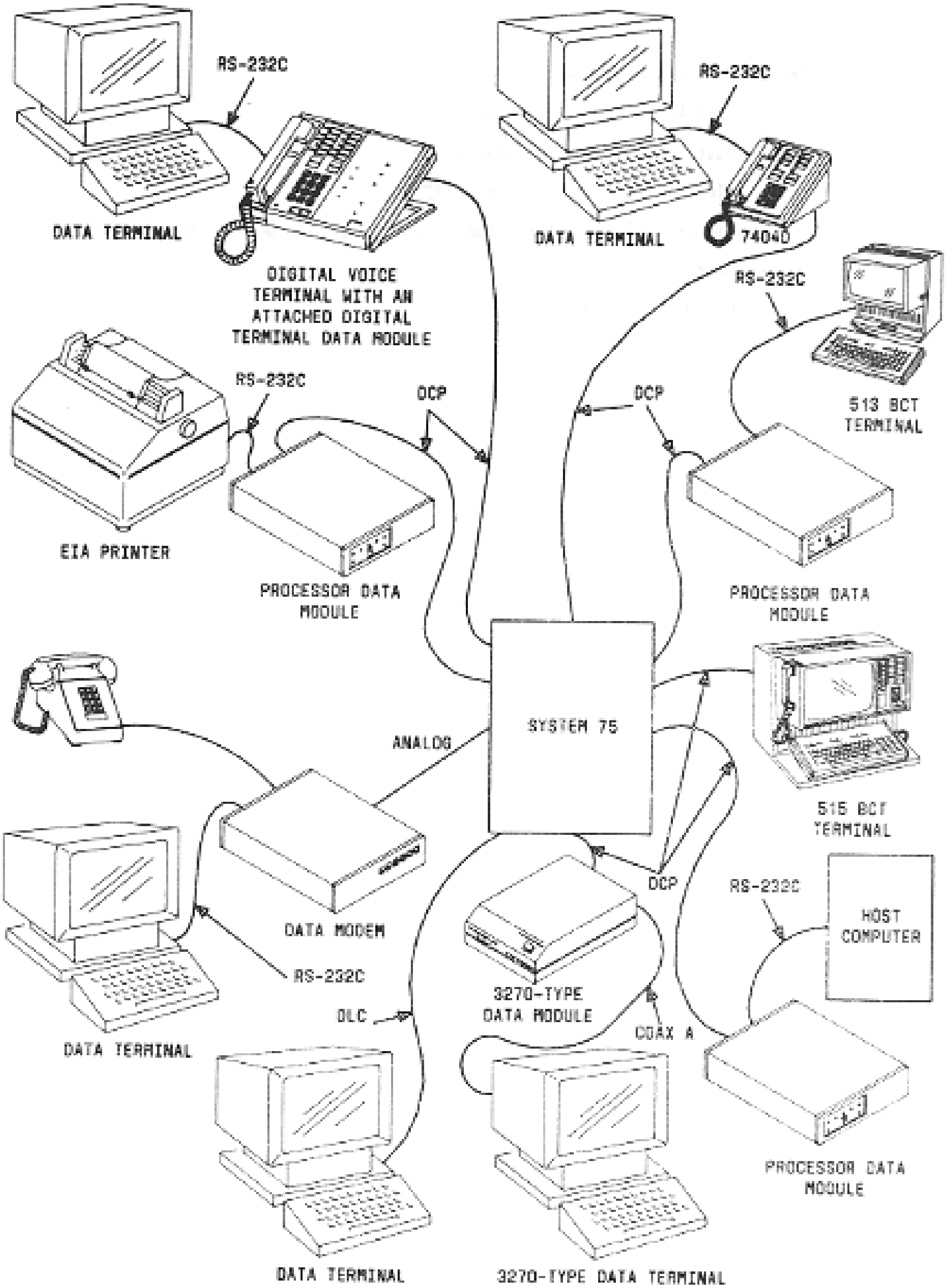


Figure 2-1. System 75 Data Communications Configuration

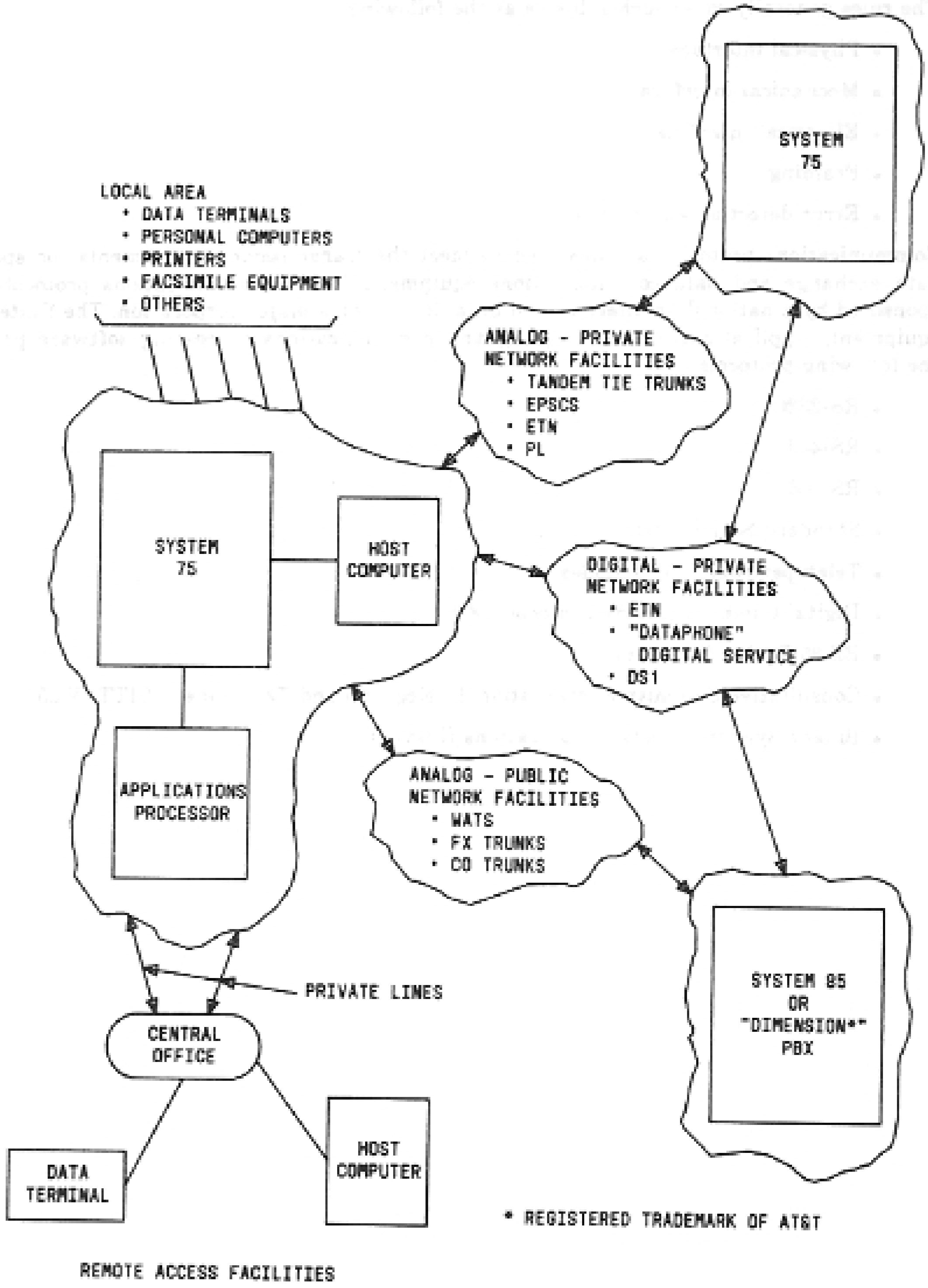


Figure 2-2. System 75 Networking Configurations

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
- Digital Communications Protocol (DCP)
- BX.25 Packet Switching
- Consultative Committee International Telegraph and Telephone (CCITT) V.35
- 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 19.2 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. Modular Data Modules (MDMs) can be configured as either DCE or DTE.

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 20 kbps data rate.

RS-366

The RS-366 communications protocol specifies the standards for interfacing computers to Automatic Call Units (ACUs). This permits 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.

SSI

The SSI communications protocol is used with the 500-series 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 American Standard Code for Information Exchange (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.

Digital Communications Protocol (DCP)

The DCP is used by the system's digital switch, digital voice terminals, data modules, and the 510D terminal or 515 BCT. This protocol permits 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 510D terminal or 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 2-3). 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.

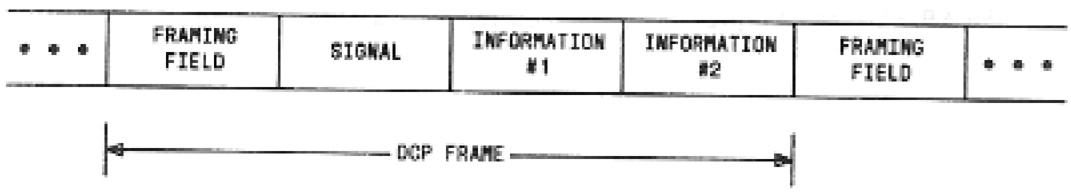


Figure 2-3. Digital Communications Protocol Frame Structure

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.

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 BX.25 protocol has three layers which are not specified for the X.25 protocol. These layers are the Application, Presentation, and Session layers. The Application and Presentation layers (see Table 2-A) are defined in the Transaction Oriented Protocol (TOP) of the BX.25.

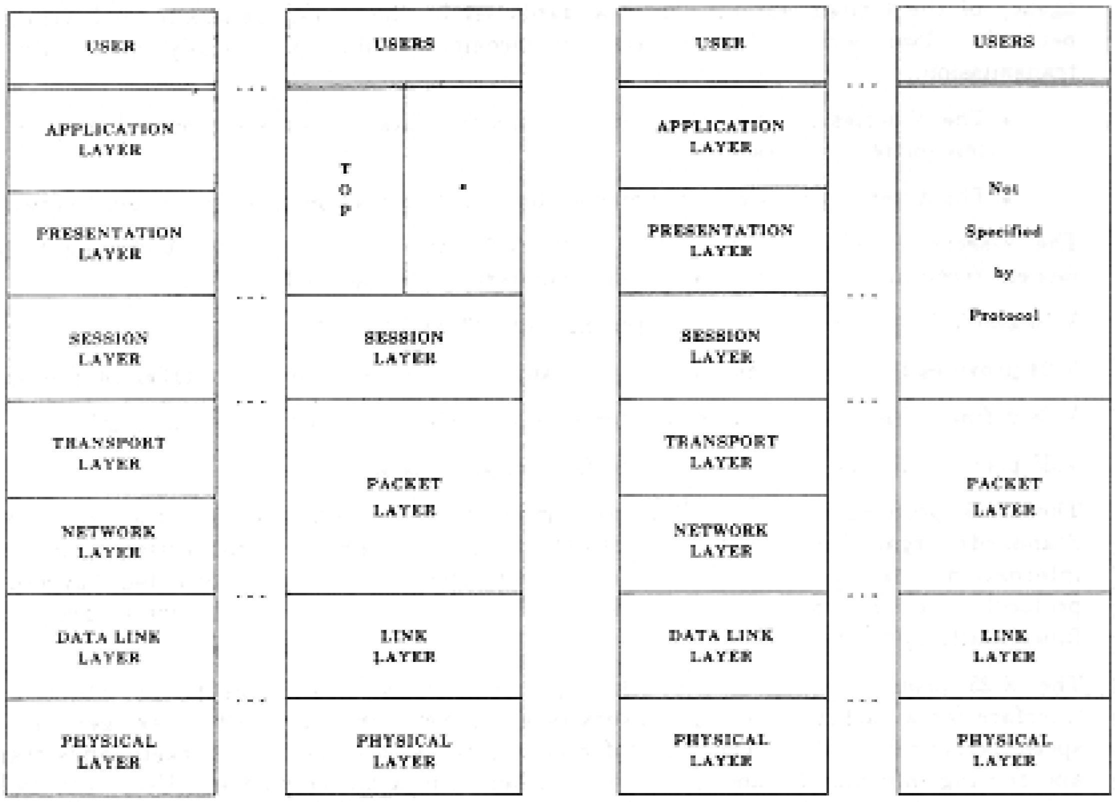
The TOP is a high level protocol, intended to standardize communications between transaction oriented systems. Transaction oriented communications involve communication of small messages or requests describing a single unit of work, which may result in a reply being sent back to the originating system. The Session layer is intended to establish, manage, and terminate sessions for use by higher level protocols, or in some cases, by user applications directly. Other 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 2-A 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 2-A. Packet Switching Protocols



BX.25 Protocol Layers **X.25 Protocol Layers**

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 V-series includes the V.10, V.11, V.24, V.28, and V.35. Also V.26, V.27, and V.28 are modem recommendations for 2400, 4800, and 9600 bps, respectively.

V.10 and V.11 are the equivalent to the EIA RS-423 and RS-422.

V.24 provides the definitions for all interchange circuits which cross the DTE/DCE interface.

V.28 defines a set of electrical characteristics which are compatible with RS-232C.

V.35 provides the constant current interface for 48 kbps operation.

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.

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.

NETWORK SERVICES OVERVIEW

Network Services provides the means to configure multiple System 75s, or one or more System 75s with one or more other systems, in an arrangement that best fits the needs of a medium- to large-sized corporation. Possible arrangements include an Electronic Tandem Network, Distributed Communications System, and Main/Satellite/Tributary. Each is briefly described in this section.

Do not assume System 75 has any capabilities other than those explicitly stated herein. Refer to the *Reference Manual—Network and Data Services*, 555-025-200, for differences between System 75 and other AT&T systems. (Check the *AT&T System 75 Documentation Guide*, 555-200-010, for the availability of this document.)

Network Services Features

The following features are associated with Network Services:

Automatic Alternate Routing (V2)
Automatic Circuit Assurance (V2)
Automatic Route Selection (V1)
Automatic Route Selection (V2)
DCS Alphanumeric Display For Terminals (V2)
DCS Attendant Control of Trunk Group Access (V2)
DCS Attendant Direct Trunk Group Selection (V2)
DCS Attendant Display (V2)
DCS Automatic Callback (V2)
DCS Automatic Circuit Assurance (V2)
DCS Busy Verification of Terminals and Trunks (V2)
DCS Call Forwarding All Calls (V2)
DCS Call Waiting (V2)
DCS Distinctive Ringing (V2)
DCS Leave Word Calling (V2)
DCS Multi-Appearance Conference/Transfer (V2)
DCS Trunk Group Busy/Warning Indication (V2)
Facility Restriction Levels and Traveling Class Marks (V2)
Network Access—Private
Network Access—Public
Off-Premises Station
Subnet Trunking (V2)
Uniform Dial Plan (V2)

Private Network Configurations

A private network is a configuration of trunk and switching facilities dedicated to the use of a business or organization. It may have as few as two switches or it may have hundreds of switches located throughout the country (DCS, however, is limited to 20 switches). The following configurations make it possible for organizations of all sizes to realize the benefits of a private network.

- Electronic Tandem Network—Serves the needs of clients with many locations in a large geographic area. This configuration normally serves moderate to heavy calling between locations without accessing toll facilities.
- Distributed Communications System—Serves the needs of clients with several locations in a small or large geographic area. This configuration normally serves

moderate to heavy calling between locations. A Distributed Communications System appears as a single switch with respect to certain attendant and voice terminal features.

- Main/Satellite/Tributary—Serves the needs of clients with a few locations in a small geographic area. This configuration normally serves moderate to heavy calling between locations.

System 75 also can be used within a Tandem Tie Trunk Network (TTTN). A TTTN is a nonhierarchial network of tie trunks interconnecting three or more switches. User dialing into each switch in the call's path is required. That is, the user at one switch dials the trunk access code for a tie trunk group to another switch, receives dial tone from that switch, and then dials another trunk access code to reach another switch. When dial tone is received from the final (desired) switch, the user dials the desired extension number.

Electronic Tandem Network (ETN)

An ETN is a hierarchical network of privately owned trunk and switching facilities that can provide a cost-effective alternative to toll calling between locations. An ETN consists of tandem switches, the intertandem tie trunks that interconnect them, the access or bypass tie trunks from a tandem switch to a main switch, and the capability to control call routing over these facilities. Figure 2-4 shows a typical ETN configuration. As shown in the figure, a Main/Satellite/Tributary configuration can be served by an ETN. Although not shown in the figure, a DCS can also be part of an ETN.

System 75 has limited capabilities to serve as an ETN tandem switch.

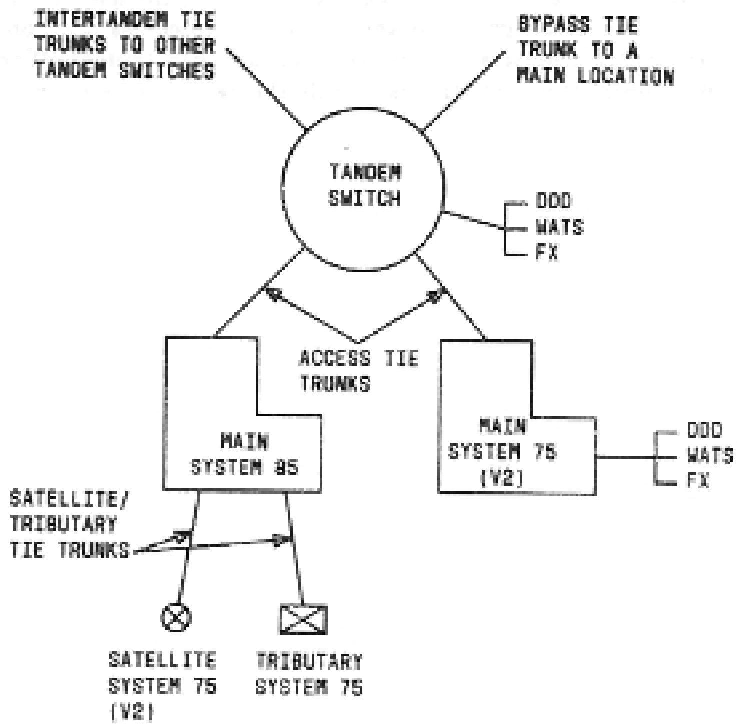


Figure 2-4. Typical ETN Configuration

Within an ETN, each location is identified by a unique private network office code, called an RNX. An RNX never matches an Area Code, so there are 640 possible RNXs available for each ETN. After accessing the ETN, the user simply dials the RNX plus the desired extension number. At most, seven digits are required.

Public network office codes (NXXs) are unique within an Area Code, whereas RNXs are unique within an ETN. RNXs are assigned when the ETN is established and, for convenience, may match NXXs (although this is not always possible). When Direct Inward Dialing (DID) is provided by the local central office, the extension numbers (last four digits of the number) will match. Network Inward Dialing (NID) is the ETN equivalent of DID and can be provided without DID.

The software program that uses the RNXs to control call routing over an ETN is called Automatic Alternate Routing (AAR). AAR not only determines the route for a call, but, through the Facilities Restriction Level (FRL) function, defines up to eight levels of calling privileges for users of the ETN. Another function of AAR, Subnet Trunking, can convert an on-network number to a public network or international number. This function is useful when all on-network routes are busy or are not provided. Details of System 75 Automatic

Alternate Routing, Facilities Restriction Level, and Subnet Trunking are given in Section 3 of this manual.

Distributed Communications System (DCS) (V2)

DCS is a cluster of two or more switches (nodes) that provides certain attendant and voice terminal features as if the cluster were a single large switch. This provides simplified dialing procedures between locations, as well the convenience of using some of the system's features between locations. DCS is particularly attractive if there is frequent interlocation calling.

Each DCS node is connected with every other DCS node via tie trunks for voice communications. DCS nodes are also connected via a data link(s) for sending and receiving control and feature information. The data links and voice channels may be directly between nodes or may pass through an intermediate node. Nodes that cannot serve as an intermediate node—that is, those that cannot receive information from one node and pass it on to another node—are called endpoints (or endpoint nodes). Nodes that can pass information are simply referred to as nodes. Within a DCS, System 75 can only serve as an endpoint. Figure 2-5 shows a typical DCS in simplified form.

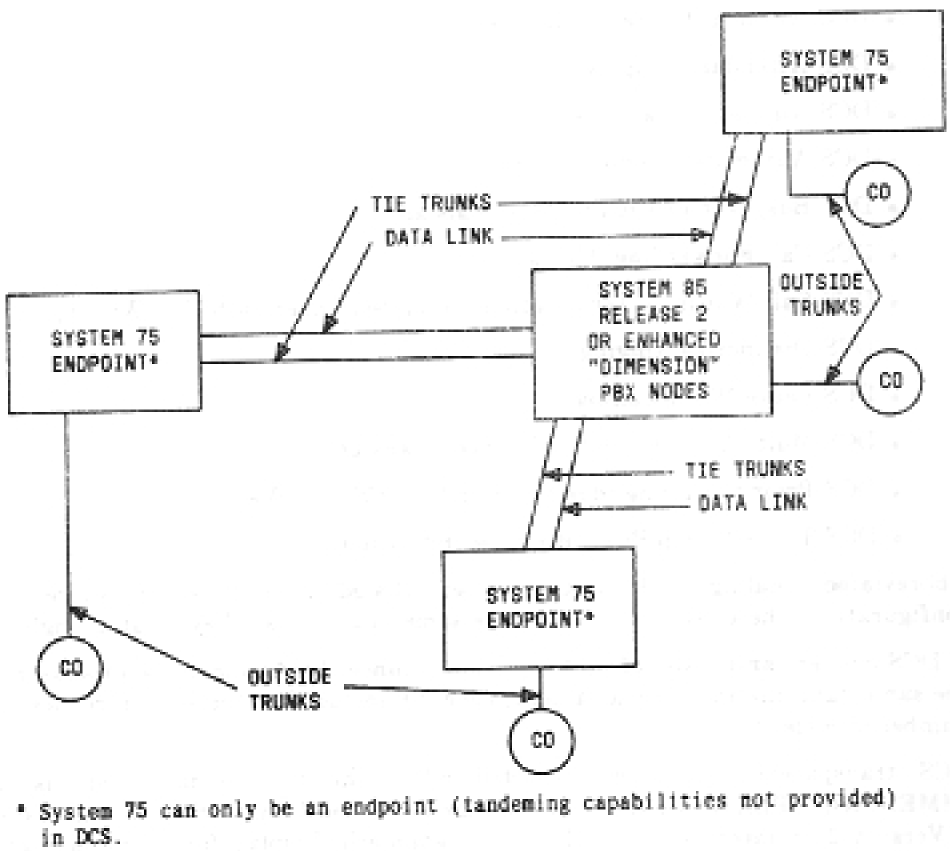


Figure 2-5. Typical Distributed Communications System

A DCS can consist of all endpoints. That is, each node in the DCS may be directly connected via data links and voice channels with every other node in the DCS. In this case, System 75 can serve as all nodes.

Some of the applications of the DCS configuration are as follows:

- In a "campus environment" that has two or more separate buildings and the nodes are connected by local cable
- In a larger area such as a city, several states, or even the entire country, where the nodes 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. Transparency is the ability of the system, from the user's standpoint, to operate across several nodes in the same way it does at the local node. This allows users to dial from any terminal to any other terminal within the DCS without regard for which nodes are involved. Likewise, transparency allows certain voice features to be used across nodes.

Some System 75 voice features have transparency in a DCS configuration. The following System 75 voice features have unique aspects in a DCS environment and are described in detail in Section 3 of this manual.

- DCS Alphanumeric Display for Terminals
- DCS Attendant Call Waiting (described under DCS Call Waiting)
- DCS Attendant Control of Trunk Group Access
- DCS Attendant Direct Trunk Group Selection
- DCS Attendant Display
- DCS Automatic Callback
- DCS Automatic Circuit Assurance
- DCS Busy Verification of Terminals and Trunks
- DCS Call Forwarding All Calls
- DCS Call Waiting—Termination (described under DCS Call Waiting)
- DCS Distinctive Ringing
- DCS Leave Word Calling
- DCS Multi-Appearance Conference/Transfer
- DCS Priority Calling (described under DCS Call Waiting)
- DCS Trunk Group Busy/Warning Indication

Abbreviated Dialing and Last Number Dialed also have transparency in a DCS configuration. These features operate the same in a DCS as they do at a single switch.

A DCS cluster can consist of up to 20 nodes. Since the Applications Processor (AP) requires the same data link facilities as a node, each AP included in the system reduces the maximum number of nodes by one.

DCS transparency is more restricted when the intermediate node is an Enhanced DIMENSION* PBX or a System 85 Release 2 Version 1 than when it is a System 85 Release 2 Version 2 or later. (See the DCS Alphanumeric Display for Terminals and DCS Leave Word Calling features.)

* Registered Trademark of AT&T

Certain feature capabilities are unique to a particular type of node; for example, a System 75 endpoint node. Therefore, a detailed feature description should be consulted for each type of node.

Centralized Attendant Service (CAS) can be used to advantage in DCS networks where all attendants are at one node. CAS reduces traffic volume on interconnecting tie trunks caused by incoming attendant-seeking calls at the endpoint nodes. System 75 Centralized Attendant Service capabilities are given in Section 3 of this manual.

DCS calls can redirect via the Call Coverage feature. The Coverage tone, which indicates that the call has redirected to Coverage, is heard by the calling party at the distant node. However, the call cannot redirect to a distant node. The principal and the covering user must be located at the same node.

Main/Satellite/Tributary

Figure 2-6 shows a Main/Satellite/Tributary configuration. It can function independently or serve as an Electronic Tandem Network (ETN) access arrangement. For a Main/Satellite configuration, attendant positions and public network trunk facilities are concentrated at the Main, and calls to or from satellite locations pass through the Main. To a caller outside the Main/Satellite complex, the system appears to be a single switch with one Listed Directory Number.

A Tributary location is similar to a Satellite location with the following exceptions: (1) a Tributary has one or more attendant positions, and (2) a Tributary has its own Listed Directory Number.

System 75, Version 1, can serve as a Satellite or Tributary. Version 2 can serve as a Main, Sateilite, or Tributary.

A small business can start with a single Main/Satellite or Main/Tributary complex and add trunk and switching facilities as the business grows. In this situation, tie trunks connect the main locations within an urban area and intercity traffic is routed via the public network. This arrangement favors a medium-size organization or one that has small isolated locations where the intercity traffic is too small to justify the cost of tie trunks.

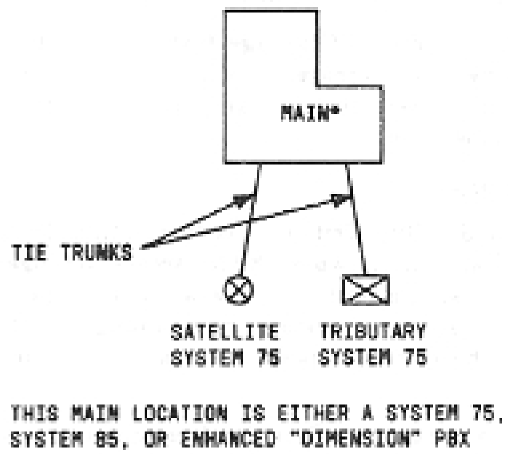


Figure 2-6. Main/Satellite/Tributary Configuration

Trunking

Trunking is the use of communications links to interconnect two switching systems, such as the System 75 to a local central office or the System 75 to another switch. These links, called trunks, can be grouped together in Trunk Groups when all the trunks in the group perform the same function. This grouping simplifies administration since the required service characteristics (parameters) are assigned to the group rather than to each individual trunk. Grouping also simplifies call processing. Calls requiring a trunk are routed to the appropriate trunk group. An idle trunk, if available, is selected from the group.

Several different types of trunk groups can be used in the System 75. They are the following:

- Auxiliary—Provides trunk applications such as Loudspeaker Paging and Music-on-Hold.
- Central office (CO)—Provides a link with the local central office for calls except Direct Inward Dialing (DID) calls.
- Direct Inward Dialing (DID)—Provides a link with the local central office.
- DS1 Tie Trunk—Provides for two types of digital tie trunk interfaces: Voice-Grade DS1 and Alternate Voice/Data (AVD) DS1 Tie Trunks. The Voice-Grade DS1 tie trunks are an alternative to 4-wire analog E&M tie trunks and may be used to interface with other properly-equipped switching systems. AVD DS1 tie trunks permit alternate voice and data calling between System 75s and System 85.
- Foreign exchange (FX)—Type of central office trunk that provides a link with a central office other than the local central office.

- Tie and Release Link (V2 only)—Provide a link with another private switching system for calls between the systems. Release Link trunks are used only with Centralized Attendant Service. Tie trunks are used on calls to or from the following:
  - A Private Branch Exchange (PBX)
  - An Electronic Tandem Network (ETN) switch
  - An Enhanced Private Switched Communications Service (EPSCS) or Common Control Switching Arrangement (CCSA) office
- Wide Area Telecommunications Service (WATS)—Type of central office trunk that provides a link with an Outward WATS office or an 800 Service office.

Incoming calls to the System 75 via a tie trunk may be treated as internal or external calls. Tie trunks designated as internal cause one-burst ringing at a called voice terminal and calls routing via the Call Coverage feature route according to the internal coverage criteria. (See Call Coverage in the Voice Management section for details.) Tie trunks designated as external cause two-burst ringing and route according to the external coverage criteria.

Trunk groups can be one-way incoming, one-way outgoing, or two-way. Whether the trunk group is available for incoming, outgoing, or two-way traffic is called direction. A two-way loop start trunk is subject to glare. This means that the distant switch is trying to use a given trunk for a call to the System 75 at the same time the System 75 is trying to use the same trunk for a call to the distant switch. No incoming calls are aborted because of glare. An outgoing call retains its place in the queue and is completed later. Queuing at both ends of a two-way trunk group compounds glare and is, therefore, not recommended.

Selection of the trunk group to be used for a given call is determined by digit translation on the trunk access code. Assuming an idle trunk in the selected group is found, a seizure signal (service request) is sent to the distant switch. If the distant switch requires the called number, a start dial signal is normally returned to the calling switch, indicating readiness to accept digit transmission.

With the System 75, the start dial signal(s) used is dictated by the serving FX office, WATS office, or local CO. For interconnection with other private switching systems, the System Manager may select the start dial signal(s) to be used.

"Trunk type" refers to the physical design of a trunk circuit. Trunk type and the start dial signal are often used interchangeably, although trunk type is a more accurate term. The trunk types available and a brief description of each are given below.

- Ground Start—A ground signal is sent over the trunk ring lead and is received over the trunk tip lead.
- Loop Start—A closure signal is sent through the loop formed by the trunk leads.
- Immediate Start—No start dial signals are used. On outgoing calls, the system waits at least 80 ms after sending the seizure signal before sending the digits required at the distant switch. This allows the distant switch enough time to attach a digit receiver to the call.
- Wink Start—A momentary signal (wink) is sent to the distant switch.
- Delay Dial—A steady signal is sent to the distant switch and is removed when the trunk is ready to receive digits.
- Automatic—No start dial signals are used. The seizure signal sent or received is sufficient to route the call. The call destination is specified when the trunk group is administered. The destination can be the attendant group or any extension number assigned in the system.

Trunk groups connecting with a WATS office, FX office, or local CO can be ground or loop start. DID trunk groups can be immediate or wink start. Tie trunk groups can be delay dial, wink start, immediate start, or automatic.

Each outgoing and two-way trunk group can have a queue. If all trunks in the group are busy, the call waits in the queue until a trunk becomes idle. The queue length, which is the number of calls waiting, may be from 0 (zero) (no queue) to 100. This information is entered on the trunk group form when the trunk group is administered.

Dual tone multifrequency (DTMF) signaling or rotary dial (dial pulse) signaling can be used between switches. (DTMF is also referred to as touch-tone signaling.) System 75 can send or receive either type of signaling required by the distant switch. The type to be used is specified on the associated trunk group form during administration.

An incoming trunk call to the System 75 can be connected to another trunk, a voice terminal, an attendant console, or an announcement. When the call is answered "an answer supervision" signal is sent to the distant EPSCS, local CO, FX, WATS, or 800 Service office. This is the signal to initiate the recording of the call details normally used for charging. Any call routed outward is deemed "answered" when the called number is sent to the distant switch. If there is also a trunk incoming from one of the previously listed offices on a call of this type, then answer supervision is sent to that office. Also, a Direct Department Calling or Uniform Call Distribution recorded delay announcement is deemed "answered" when the calling party is connected to the announcement. Other types of announcements, such as unassigned number announcements, are treated as an unanswered call.

SYSTEM MANAGEMENT OVERVIEW

System Management provides the capabilities to control and maintain the System 75, and also provides system usage reports to help determine if the system is being used as intended. In short, System Management allows the System Manager to establish the system, monitor its use, and make additions and/or changes as necessary.

System Management comprises the features and functions described in Section 3 of this document. Functions are more fully described in the following documents.

- *AT&T System 75 Implementation Manual, Release 1 Version 1*, 555-200-650
- *AT&T System 75 Implementation Manual, Release 1 Version 2*, 555-200-651
- *AT&T System 75 Administration Manual, System Management*, 555-200-500
- *AT&T System 75 Service Manual, Switch Maintenance*, 555-200-105

Changes made to system translations are effected only at the single system for which the changes were made. If a system is part of a network, changes may have to be made at more than one system to effect the desired changes to the network. Similarly, changes intended for only a single system could affect the network. Therefore, the System Manager must consider the effect on the network before making any changes.

System Management Features

The following features are associated with System Management:

Facility Test Calls
Station Message Detail Recording
System Measurements
System Status Report (V2)

System Administration

Allows the user to implement (initialize) and administer all the System 75 terminal and system features and system parameters. System Administration allows the following:

- Initializing the system
- Managing system, voice terminal, and data terminal features on a day-to-day basis
- Performing system backup procedures
- Monitoring, detecting, and determining system performance
- Maintaining system security

System Administration is performed at the System Access Terminal (SAT), a Remote Administration terminal, or Initialization and Administration System (INADS) location. System maintenance can be performed from the SAT or INADS.

The SAT is a 513 Business Communications Terminal (BCT) located within 50 feet of the system cabinet and connected directly to the maintenance board. The SAT consists of a video display and keyboard which allows a System Manager to input System 75 commands and translations. The SAT is first used by the System Manager to initialize the system. After initialization, the SAT is used to reconfigure translations and to monitor system performance.

A SAT located more than 50 feet from the system cabinet is considered a remote administration terminal. A remote terminal may require either a Processor Data Module or Digital Terminal Data Module as an interface to the system, depending on the type of terminal used.

INADS is a service available from an AT&T Service Center and has the same administrative capabilities as the SAT.

Remote Administration

Allows System 75 to be administered from a remote terminal located either on or off the client's premises. A local System Administration Terminal (SAT) is located on-premises within 50 feet of the system cabinet. The 513 BCT with a 12-foot cable is supplied for this application. A terminal located more than 50 feet from the system cabinet is considered remote. A remote administration terminal can be on the same premises as the local SAT, or can be off-premises. The remote terminal performs the same functions as the local SAT.

The 510D terminal, 513 BCT, or 515 BCT may be utilized for the on-premises remote terminal. Normally, only the 513 BCT is used as the off-premises remote terminal.

If the remote terminal is a 513 BCT, it must be connected to the system through a Processor Data Module (PDM) or Digital Terminal Data module (DTDM). If a 510D terminal or 515 BCT is used as a remote terminal, a PDM or DTDM is not required. The cabling distance from the System 75 to the remote terminal is determined by the type of module associated with the terminal. Distance limitations are as follows:

- Remote terminal to PDM—500 feet using 24-gauge wire or 4000 feet using 26-gauge wire
- Remote terminal to DTDM—3400 feet using 24-gauge wire or 2200 feet using 26-gauge wire

For a detailed description of the data modules and BCTs, refer to the *AT&T System 75 Reference Manual, System Description*, 555-200-200.

Only two users can be logged into the administration functions at one time. This includes a user of the SAT.

Initialization and Administration System (INADS)

Allows System 75 administration and maintenance from a remote location.

INADS allows its user to access the System 75 and perform administrative tasks assigned to the System Manager. The administrative commands used by the System Manager are also available to the INADS users. INADS can also be used to perform maintenance routines.

During system access, INADS automatically receives major and minor alarm notifications from the System 75. When an alarm is received, INADS users can access the System 75 and perform the following tasks:

- Display alarms
- Display errors
- Clear errors
- Test and busyout circuit packs, voice terminals, and trunks
- Set time and date

- Receive backup translations for the System 75
- Download a copy of the System 75 tape

FEATURE DESCRIPTIONS

CONTENTS

LIST OF FIGURES

LIST OF TABLES

OVERVIEW

This section defines the system features associated with Voice Management, Data Management, Network Services, and System Management. The features are arranged in alphabetical order, regardless of the functional area to which they apply. The information for each feature is presented under five headings: Description, Considerations, Interactions, Administration, and Hardware and Software Requirements.

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

- Interactions

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

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

- Administration

  States whether or not administration is required, how the feature is administered, who administers the feature, and lists items requiring administration.

- Hardware and Software Requirements

  Lists any additional hardware and/or software requirements needed for the 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. With Version 2, certain stored numbers can also be accessed by attendants.

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, and 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. With Version 1, a user can have one Personal Number List with 5 or 10 list entries. With Version 2, a user can have up to 3 Personal Number Lists with 5 or 10 entries per list. As many as 400 (V1) or 600 (V2) 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 up to 15 (V1) or 90 (V2) list entries (in multiples of 5). An individual user can access up to 3 specific Group Number Lists, as set by the System Manager.

- System Number List

  Can have up to 30 (V1) or 90 (V2) 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 (V1) or 11 through 99 and 00 (V2). 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. With V1, 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. With V2, the three lists may be any combination of the above as long as there is no more than one System List. 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 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 and List 2 are 101 and 102, respectively. One voice terminal may have List 2 administered as "group 42." Another voice terminal may have List 1 administered as "group 42." In this case, the access code for "group 42" is 102 for the first voice terminal and 101 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 (V1) and Automatic Alternate Routing (V2) features.] 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 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. In Version 2 systems, which have 748B tone detectors, outpulsing will resume as soon as precise dial tone is received, if it is received before 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 a dial pulse trunk was used to set up the call. As a typical application, a data call can be made over a dial pulse trunk.

- 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 lower-case letter s instead of the stored 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.

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 and attendants 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 on a multi-appearance voice terminal. An AD button on an attendant console can be programmed to access a Group List or the System List and associated list entry number.

  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 (V1) or 24 digits (V2)], 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 (V1) or 24 digits (V2)], 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

Abbreviated Dialing provides easy access to selected numbers by decreasing the number of dialed digits required to place the call. Instead of dialing the entire number, the user merely dials a short code to access the desired number. The system then dials the stored number automatically. For frequently called numbers, an abbreviated dialing button can be assigned, allowing the call to be placed by merely pressing the button. By assigning a Privileged list of numbers, a user is allowed to place calls to selected numbers which might otherwise be restricted.

Users can be assigned access to three AD lists. With Version 1, the three lists can be made up of any combination of one Personal Number List, up to three Group Number Lists, and the System List. With Version 2, the three lists can be made up of any combination of up to three Personal Lists, up to three Group Lists, and the System List. Abbreviated Dialing is not the same as Repertory Dialing.

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. However, if any special characters (Mark, Wait, Pause, and/or Suppress) are included in the previous call, they are not used on the Last Number Dialed call.

  With Version 2, if the previously called number was in an Abbreviated Dialing Privileged List, and if the user is not normally allowed to dial the number because of his or her Class of Restriction, Intercept Treatment is given when using Last Number Dialed. To redial the number, the user must again use the Abbreviated Dialing Privileged List.

- Bridged Call Appearance

  A user, accessing Abbreviated Dialing while on a bridged call appearance, accesses the Abbreviated Dialing lists of the primary extension associated with the bridged call appearance.

Administration

Abbreviated Dialing is administered by the System Manager. However, an Abbreviated Dialing Personal List can be programmed by either the System Manager or the voice terminal user.

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.

The following items, if required for a given system, are set by the System Manager:

- Feature Access Codes for List 1, List 2, and List 3
- Voice Terminal Assignments
  - AD buttons, if desired
  - Wait, Mark, Pause, Suppress, and Function Entry buttons, if desired
  - Access to as many as three lists
- Data Module Assignments (Access to as many as three lists)
- Abbreviated Dialing Lists
  - Personal Number Lists
  - Group Number Lists
  - System Number List
  - 7103A Group Number List

Incoming tie trunks cannot access a Personal Number List.

An Abbreviated Dialing Personal List cannot be administered to an attendant console.

A maximum of 502 (V1) or 704 (V2) lists and a maximum of 2500 (V1) or 3010 (V2) entries are allowed for the system. The 502 (V1) or 704 (V2) lists include a maximum of 400 (V1) or 600 (V2) Personal Number Lists, 100 (V1) or 102 (V2) Group Number Lists, a 7103A Group Number List, and a System Number List.

A number stored in any list in the switch can contain up to 16 (V1) or 24 (V2) digits. A special character used for Pause, Wait, Mark, or Suppress counts as two digits.

Hardware and Software Requirements

No additional hardware or software is required for Version 1 systems. Version 2 systems require no additional software, but additional 748B tone detectors (up to five per system) may be required if the special "wait" character is used frequently.

AP DEMAND PRINT

Description

Allows the voice terminal user to print his or her own undelivered messages without calling the AP-based 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.

To illustrate how the AP Demand Print feature functions, assume the following:

- The Message lamp lights at extension 321.
- Extension 432 is to activate the AP Demand Print feature to print the message(s) for extension 321.
- Extension 432 is associated with an overriding printer.

In the given situation, extension 321 is the requesting extension and extension 432 is the originating extension. Since extension 432 is associated with an overriding printer, the message(s) for extension 321 will be printed on that printer. If there was not an overriding printer, the messages would be printed on the printer assigned to extension 321. Under the given conditions, the following sequence of events will take place:

1. To print extension 321's message(s), extension 432 activates the AP Demand Print feature either by dial access or button.

2. Extension 432 then enters the extension number and authorization password for extension 321.

3. The messages are then printed at the overriding printer associated with extension 432 and the Message lamp at extension 321 goes dark.

Considerations

AP Demand Print reduces the Message Center work load by allowing voice terminal users to print their own undelivered messages. The Message Center benefits from this feature by gaining more time for other operations. The voice terminal users also benefit from this feature, because they have control of retrieving their own messages. Up to 10 overriding printers can be assigned.

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.

Administration

AP Demand Print is administered by the System Manager. The following items require administration:

- Requesting Extensions
  - Authorization passwords
  - Printers
- Originating Extensions
  - Print Msgs buttons, if desired
- Overriding Printers
  - Up to 10 overriding printers can be assigned
- Feature Access Code
  - Access code for activating AP Demand Print

Hardware and Software Requirements

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

ATTENDANT AUTO-MANUAL SPLITTING

Description

Allows the attendant to announce a call or consult privately with the called 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

Attendant Auto-Manual Splitting automatically provides a splitting of the called party. Splitting allows the attendant to privately determine if the called party can and will accept the call.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

ATTENDANT CALL WAITING

Description

Allows an attendant originated or extended call to a busy single-line voice terminal to wait at the called terminal. The attendant is free to handle other calls.

Attendant Call Waiting is activated automatically whenever an attendant originates or extends a call to a busy single-line voice terminal. The attendant hears a Call Waiting 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.

When the Attendant Call Waiting is activated the attendant may choose to cancel the call, release the call, or hold the call on the console. However, releasing an attendant-originated call results in the call being dropped completely. 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. With V1 systems, the call in progress at the voice terminal cannot be placed on hold. It must be terminated. With V2 systems, the call in progress at the voice terminal can be placed on hold in order to answer the waiting call. After answering the waiting call, the voice terminal user can return to the held call or toggle back and forth between the two calls. The single-line voice terminal user can only be connected to one call at a time.

As an example of how Attendant Call Waiting is used, assume extension 123, a single-line voice terminal, is busy. An attendant extends a call to extension 123 and hears the Call Waiting Ringback Tone which indicates that Attendant Call Waiting is activated. The attendant may choose to announce the call waiting condition to the calling party. However, after doing this, the attendant cannot cancel the call. The attendant could cancel the call and ask the calling party to call again later, or the attendant could release the call or place the call on hold at the console. This allows the attendant to handle other calls. The voice terminal user at extension 123 hears a 2-burst tone and knows a call is waiting. The voice terminal user at extension 123 can then terminate the call in progress (V1), or place the call in progress on hold (V2), and answer the waiting call. If the waiting call is not answered before a preassigned time interval expires, the call returns to the attendant.

Considerations

Attendant Call Waiting allows an attendant to originate or extend calls to a busy single-line voice terminal, while allowing the attendant to handle other calls. Since the attendant is able to handle other calls while a call is waiting, fewer calls are lost.

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.

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 ringing 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.

Administration

Attendant Call Waiting is a standard system feature. No administration is required for the feature itself. However, the call waiting interval is administered through the Timed Reminder feature by the System Manager.

Hardware and Software Requirements

No additional hardware or software is required.

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.

Each attendant console has 12 Trunk Group Select buttons to be used with the Attendant Direct Trunk Group selection feature. Each button allows the attendant direct access to an outgoing trunk group by merely pressing the button assigned to that trunk group. Each of the 12 buttons has a Busy lamp which lights when all trunks in the associated trunk group are busy. Six of these buttons have two additional lamps and are used for Attendant Control of Trunk Group Access. The two additional lamps are as follows:

- Warn (warning) lamp

  Lights when a preset number of trunks are busy in the associated trunk group

- Cont (control) lamp

  Lights when the attendant activates Attendant Control of Trunk Group Access for the associated trunk group

Attendant Control of Trunk Group Access is activated by pressing a Cont Act button followed by the desired Trunk Group Select button. If a user attempts to access a controlled trunk group directly, the call automatically redirects to the attendant. If the attendant decides to allow the call to go through, the attendant can connect the user to the desired trunk group by pressing the associated Trunk Group Select button. The attendant can then release the call or hold the call on the console.

Calls that are already in queue for a trunk are not affected by the activation of Attendant Control of Trunk Group Access for that trunk group. For example, if an attendant activates Attendant Control of Trunk Group Access for a specific trunk group while a user is waiting in queue for an outside trunk in that trunk group, the call is not affected. The call will remain in queue until an idle trunk becomes available, at which time the call is connected to that idle trunk.

Attendant Control of Trunk Group Access is deactivated by pressing the Cont Deact button followed by the desired Trunk Group Select button. Attendant Control of Trunk Group Access is activated and deactivated separately for each of the six trunk groups associated with the feature.

After an attendant presses a Cont Act or Cont Deact button, the attendant can perform other operations before pressing the desired Trunk Group Select button. This has no effect on the activation or deactivation of the feature. For example, if the attendant presses the Cont Act button and then has to answer another call, the desired Trunk Group Select button can be pressed after answering the call. Attendant Control of Trunk Group Access is then activated for the associated trunk group.

Considerations

By activating Attendant Control of Trunk Group Access, the attendant obtains control of access to specific trunk groups. This allows the attendant to monitor the use of these trunk groups. By watching the lamps associated with each of the six trunk groups, the attendant can determine if the number of busy trunks in a specific trunk group has reached a preset warning level and if all trunks in a specific trunk group are busy. The attendant can then handle other calls to these trunk groups accordingly.

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.

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 shows that the call has attempted to access a controlled trunk group.

- 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.

Administration

Attendant Control of Trunk Group Access is assigned on a per-attendant console basis by the System Manager. The following items require administration:

- Attendant Console
  - Trunk groups which are to be controlled
  - Cont Act and Cont Deact buttons

Hardware and Software Requirements

No additional hardware or software is required.

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.

The Group Select and DXS buttons are located on the selector console. There are 8 Group Select buttons and 100 DXS buttons. The eight Group Select buttons are labeled with up to eight different hundreds group numbers used in the system. For example, if a system uses 4-digit extension numbers, the Group Select buttons could be labeled 2400, 2500, 2800, etc. Likewise, a 3-digit system could have these buttons labeled as 100, 200, 300, etc. A 2-digit system would have a 0 Group Select number. A 5-digit V2 system, for example, could have group select buttons labeled 28400, 28500, 28600, etc.

The 100 DXS buttons are labeled 00 to 99, and each button represents the last 2 digits of an extension number. Each DXS button, when combined with a Group Select button, represents a unique extension number. To place a call to an extension number, the attendant merely presses the appropriate Group Select button followed by the appropriate DXS button. For example, to call extension 4321, the attendant would press Group Select button 4300 followed by DXS button 21.

A lamp associated with each Group Select button indicates the selected hundreds group. A selected hundreds group remains selected until another Group Select button is pressed. The associated lamp lights and remains lighted until another Group Select button is pressed. Each DXS button also has an adjacent lamp, which is used to determine the idle/busy active status of the facility associated with the button. When a facility is busy active, the lamp at the associated DXS button is lighted. When the associated facility is idle the lamp is dark. The 100 lamps adjacent to the DXS buttons are referred to as a busy lamp field.

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.

When an extension number represented by a DXS button is assigned to a group, such as a Terminating Extension Group, trunk group, etc., the lamp next to that DXS button lights only when all members of the group are busy and the queue, if provided, is full.

Considerations

With the Attendant Direct Extension Selection With Busy Lamp Field feature, the attendant can place calls to system users without having to dial the extension number. The attendant simply presses a Group Select button and a DXS button. If the desired Group Select button is already pressed, the attendant needs only to press the desired DXS button. This feature also provides the attendant with a visual indication of the idle/active status of the extension numbers assigned to the selected hundreds group.

A maximum of 100 extension numbers can be monitored for idle/active status at any one time, using the selector console busy lamp field.

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.

Administration

The only administration required is the hundreds group assignment for each of the up to eight Group Select buttons. Assignments are made by the System Manager.

Hardware and Software Requirements

Requires a selector console. No additional software is required.

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.

Each attendant console has 12 Trunk Group Select buttons associated with the Attendant Direct Trunk Group Selection feature. Each button allows the attendant direct access to an outgoing trunk group by simply pressing the button assigned to that trunk group. Each Trunk Group Select button has a Busy Lamp which lights when all trunks in the associated trunk group are busy.

Six of the Trunk Group Select buttons also have a Cont (control) lamp and a Warn (warning) lamp. The Warn lamp lights when a preset number of trunks in the associated trunk group are busy. The Cont lamp lights when the attendant has activated the Attendant Control of Trunk Group Access feature for the associated trunk group.

In addition to trunk groups, Loudspeaker Paging zones can also be assigned to Trunk Group Select buttons. In this case, the Busy lamp indicates the idle/busy status of the associated Loudspeaker Paging zone.

Considerations

Attendant Direct Trunk Group Selection eliminates the need for the attendant to memorize, or look up, and dial the trunk access codes associated with frequently used trunk groups. A label associated with each Trunk Group Select button identifies its destination or use, for example, Chicago, FX, or WATS. Pressing the button selects an idle trunk in the desired group.

Each attendant console has 12 Trunk Group Select buttons.

Interactions

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

Administration

Attendant Direct Trunk Group Selection is assigned on a per-attendant basis by the System Manager. Administration consists of assigning trunk groups or Loudspeaker Paging zones to the Trunk Group Select button.

Hardware and Software Requirements

No additional hardware or software is required.

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 the eight 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. The display must be in the normal mode to answer incoming calls.

- 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 handset 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

  - Version 1

    When the call is from a system user, the display shows the caller's extension number, the caller's name, or a unique identification administered for the voice terminal being used. When the call is from outside the system, the display shows the trunk identification, such as CHICAGO, assigned to the trunk group used for the call.

  - Version 2

    When the call is from a system user, the display shows the caller's name or a unique identification administered for the voice terminal being used, along with the calling party's extension number. When the call is from outside the system, the display shows the trunk identification, such as CHICAGO, and the trunk access code assigned to the trunk group used for the call. If an attendant is active on a call, and receives a subsequent call, the display automatically shows the identification of the subsequent caller.

- Called Party Identification

  - Version 1

    On calls to a system user, the display shows the digits as they are dialed. After the dialing is complete, the display shows the called party's name. If no name is assigned, the called party's extension number is displayed.

    On outgoing calls, the display shows the digits as they are dialed, followed by the name assigned to the trunk group being used. The System Manager can suppress the name of any trunk group. If such a trunk group is accessed, the called party portion of the display is blank.

– Version 2

On calls to a system user, the display shows the digits as they are dialed. After the dialing is complete, the display shows the called party's name and extension number. If no name is assigned, only the called party's extension number is displayed.

On outgoing calls, the display shows the digits as they are dialed, followed by the name and trunk access code assigned to the trunk group being used. The System Manager can suppress the name of any trunk group. If such a trunk group is accessed, the name portion of the display is blank.

- Internal Caller's Class of Restriction (COR)

  All system users have a COR to define their calling privileges. The COR is a 2-digit number followed immediately by a hyphen and a 4-character identifier. With V1, the display shows a user's COR whenever the attendant makes or answers an internal call. With V2, a COR button must be pressed to display a user's COR. The COR information can be obtained from the System Manager. 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.

  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.

  d—Don't Answer or Cover—Indicates that the called voice terminal was not answered or that the calling system user has sent the call to coverage, or the called voice terminal user is not available.

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

Some typical displays are as follows:

- Internal call originated by the attendant (V1):

a=3602

then

a=	TOM BROWN	04-NONE

or

a=	EXT 3602	04-OTWD

- Internal call originated by the attendant (V2):

a=3602

then

a=	TOM BROWN	3062

or

a=	EXT 3602	3602

- Outgoing trunk call originated by the attendant (V1):

```
b=87843541
```

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

then

```
b=   OUTSIDE CALL
```

or

```
b=          WATS
```

- Outgoing trunk call originated by the attendant (V2):

```
b=87843541
```

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

then

```
b=   OUTSIDE CALL               8
```

or

```
b=          WATS            101
```

Where 101 is the trunk access code of the outgoing trunk group.

- Incoming trunk call to the attendant (V1):

```
a=   OUTSIDE CALL
```

- Incoming trunk call to the attendant (V2):

```
a= . OUTSIDE CALL            102
```

Where 102 is the trunk access code of the incoming trunk group.

- 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=    EXT 3174   to   EXT 3077    d
```

or

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

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

- Incoming trunk call redirected to coverage:

```
b=   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 R. Sims the morning of October 16. The second message was stored at 11:40 a.m. 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

The Attendant Display feature gives the attendant considerable call handling capabilities by displaying call-related information. With this feature, the attendant receives detailed information on incoming and outgoing calls. The display provides such information as the called number on call originations, identification of trunk groups and internal users on a call, and calling party restrictions of internal callers requesting assistance.

Attendant Display also provides the attendant with information associated with certain features such as Leave Word Calling and Integrated Directory. For these features, the display provides such information as names, extension numbers, and messages.

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.

Interactions

With the Bridged Call Appearance feature, a call from the primary extension number or a bridged call appearance of the primary extension number is displayed as a call from the primary extension number.

Administration

The Attendant Display feature is administered on a per-attendant basis by the System Manager. Administration consists of assigning feature related buttons to each attendant console. The following buttons can be assigned:

- 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.

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

Attendant Recall provides a convenient means for a voice terminal user, on a call held on the console, to recall the attendant if further assistance is required.

The call must be held on the console.

Interactions

- Individual Attendant Access

  If a hunt group call to an individual attendant is being held on the console, a system user, active on the call, cannot recall the attendant. However, he or she can transfer calls or make conference calls.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

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.

When an incoming call arrives on a call appearance at an attendant console and is answered, extended, and released by the attendant, the call is released from that call appearance. The console is then available to receive the next call.

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

Attendant Release Loop Operation improves efficiency in handling calls by allowing the attendant to release from a call without having to wait for an answer. The attendant is immediately available to handle other calls.

Interactions

- Timed Reminder

  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.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

AUTOMATIC ALTERNATE ROUTING (V2)

Description

Provides alternate routing choices for private on-network calls. Also provides digit modification to allow on-network calls to route via the public network when on-network routes are not available.

Automatic Alternate Routing (AAR) provides up to 6 routes for each of the 640 possible private network office codes (RNXs). To use AAR, the user dials the AAR access code and the called number. Feature operation is completely transparent to the user. The AAR access code is normally the digit 8. The called number may be a 7-digit on-network number, a 10-digit public network number, a service code, an International Direct Distance Dialing (IDDD) number, an operator code (0), or a customer-dialed and operator-serviced (CDOS) number (0+ or 01+ the number).

On-network numbers are handled by the AAR feature. All other numbers are directed to the Automatic Route Selection (ARS) feature for processing. An on-network number can be changed into a 7- or 10-digit public network direct distance dialing number, a CDOS number, or an IDDD number depending on the route selected.

The 640 private network RNXs may match public network office codes (NXXs). Therefore, the only way to determine the intended network for 7-digit calls is by the dialed access code. Ten-digit public network calls are recognizable because an RNX does not match any Area Code. When an Area Code is detected, a 10-digit public network call is assumed.

The principal use of AAR is to provide routing of private network calls, that is, calls that originate and terminate at a client location without accessing the public network. The normal scenario is: the calling party accesses the switch serving his or her location and dials a 7-digit on-network number, AAR then routes the call to the on-network switch serving the calling party.

AAR and Subnet Trunking provide a convenient means to place IDDD calls to a frequently called foreign city. Such calls route as far as possible over the private network before exiting the network. The RNX is, of course, reserved to represent a particular country and city. At the final on-network switch, the RNX is deleted. The international prefix code (011), the country code, and the city code are inserted. The inserted digits plus the last four digits of the originally dialed number constitute the IDDD number. Subnet Trunking, which also has ARS applications, is discussed separately in this section.

Similar to the IDDD case, certain domestic calls may reach a point on the network where they can route no further, because tie trunks to the next switch are busy or none are provided. In this case, the RNX can be deleted and the appropriate public network code inserted. Calls of this type route off-network via a central office. The central office may be connected to either an Electronic Tandem Network (ETN) tandem or main switch. Toll charges, if any, are from the final ETN switch to the destination.

Attendant-seeking ETN calls are of the form RNX-0111. Calls of this type route on the network to an attendant at the location specified by the RNX. Attendant-seeking calls cannot access the public network. Reorder tone is returned when a Wide Area Telecommunications Service (WATS), central office (CO), or foreign exchange (FX) trunk group is selected for such a call.

Each RNX can point to any one of 254 Routing Patterns, numbered 1 through 254. More than one RNX can point to the same pattern. A blank pattern provides intercept treatment and pattern 254 is the default for all RNXs. Routing Patterns are shared with ARS. Access to a route within the pattern is controlled by Facility Restriction Level (FRL) assignments. FRLs are fully described elsewhere in this section.

Although System 75 has limited capabilities to serve as an ETN tandem switch, it may serve such a function in some networks. In this case the system can access or be accessed by Intertandem Tie Trunks to/from other tandem switches and/or Access Tie Trunks to/from ETN main switches. The system can also access Bypass Tie Trunks to an ETN main switch. This distinction as a tandem switch is important with respect to the routing of certain calls.

Considerations

AAR provides efficient use of private network facilities.

AAR provides up to 254 Routing Patterns, each containing up to six routing preferences. Patterns are shared with ARS.

Up to 640 RNXs can be provided. An RNX can represent an actual location on the network, or can be a dummy code to be converted into a public network or IDDD number.

If a client changes ARS routing assignments, it is the client's responsibility to notify the Regional Support Center (RSC) network designer and the System Control Office (SCO) technician of the changes in order to receive their continued support.

Interactions

- Uniform Dial Plan (UDP)

  The leading two digits of the 4- or 5-digit called DCS extension are converted into an RNX. AAR is used to route the UDP call.

- Automatic Route Selection (ARS)

  ARS and AAR can access the same trunk groups and share the same Routing Patterns.

- Abbreviated Dialing

  FRL checking is bypassed on an AAR call made via a privileged Abbreviated Dialing Group List.

- Ringback Queuing

  Ringback Queuing can be used on AAR calls originated at the switch that provides the queuing. Incoming tie trunk calls will not queue on an outgoing trunk group.

- Code/Toll Restriction

  Code/Toll Restriction is not checked on AAR calls.

- Controlled Restriction, Origination Restriction, and Outward Restriction

  These features prohibit access to AAR.

- Miscellaneous Trunk Restrictions

  Miscellaneous Restrictions are not checked on AAR calls.

- Attendant Control of Trunk Group Access

  Attendant control of a trunk group, in effect, removes a trunk group from the Routing Pattern. A controlled trunk group is never accessed by AAR.

- Station Message Detail Recording (SMDR)

  An AAR call using a trunk group marked for SMDR is indicated by the dialed access code and by a Condition Code. The dialed number is recorded as the called number.

Subnet Trunking does not affect SMDR.

The originating FRL associated with the the call is recorded. However, if 15-digit SMDR account codes are used, the FRL value is overwritten.

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 with an ARS call, it must be dialed before the ARS access code is dialed.

Administration

AAR is initially assigned on a per-system basis by an AT&T service technician. After the feature is activated, the following items are administered by either the System Manager or the service technician:

- AAR Access Code (1 to 3 digits)
- RNX Translation Table—Points to the appropriate Routing Patterns. Pattern Number 254 is initially assigned to all RNXs.
- Routing Patterns—In addition to normal trunking data, provides subnetwork trunking information which extends a call through a chain of subtending switches. (See Subnet Trunking for details.)
- FRLs—Must be assigned via a Class of Restriction to each originating facility. The minimum FRLs required to access a route are assigned as part of the Routing Pattern. Assignment of these values determines the calling privileges of each individual user of the ETN.

Hardware and Software Requirements

AAR may require additional tie trunks. These additions are, however, cost effective when compared to the other alternatives for call routing.

AAR is provided as a part of the optional Private Networking software.

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.

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.

When Automatic Callback is activated, the system monitors the called voice terminal. When the called voice terminal becomes available, the system then originates the Automatic Callback call. A busy voice terminal becomes available when the user hangs up after completing the current call. An unanswered voice terminal becomes available after it is used for another call and is then hung up.

When the called voice terminal becomes available, the system originates the Automatic Callback call and the calling party receives 3-burst ringing. The calling party then lifts the handset and the called party receives the same ringing provided on the original call. The ringing at the called voice terminal occurs immediately after the calling voice terminal user lifts the handset.

Considerations

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

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 ringing 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).

Automatic Callback eliminates the need for voice terminal users to continually re-dial busy or unanswered calls to internal voice terminals. Instead, the user simply activates Automatic Callback. The system then calls the user back when the called voice terminal becomes available.

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.

Interactions

- Bridged Call Appearance

Automatic Callback calls cannot originate from a bridged call appearance. When a call is originated from a primary extension number, the return call notification rings at all bridged call appearances.

- 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.

- Ringback Queuing

  An Automatic Callback button is used to activate the Ringback Queuing feature.

Voice terminals with the following features cannot activate Automatic Callback:

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

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

Administration

The System Manager assigns Automatic Callback to individual voice terminals by their Class of Service. The following items also require administration:

- No Answer Time-out Interval (number of times the callback call rings before it is canceled). This interval is assigned on a per-system basis.
- Feature Access Codes—For activating and deactivating Automatic Callback
- Automatic Callback Buttons—For multi-appearance voice terminals

Hardware and Software Requirements

No additional hardware or software is required.

AUTOMATIC CIRCUIT ASSURANCE (V2)

Description

Assists users in identifying possible trunk malfunctions. The system maintains a record of the performance of individual trunks relative to short and long holding time calls. The system automatically initiates a referral call to an attendant or display-equipped voice terminal user when a possible failure is detected.

Holding time is the elapsed time from the time a trunk is accessed to the time a trunk is released. When the Automatic Circuit Assurance (ACA) feature is enabled by the System Manager, the system measures the holding time of each call.

A short holding time limit and a long holding time limit are preset by the System Manager for each trunk group. The short holding time limit can be from 0 to 160 seconds. The long holding time limit can be from 0 to 10 hours. The measured holding time for each call is compared to the preset limits for the trunk group being used.

A short holding time counter and a long holding time counter associated with each trunk group member are kept by the system. When the measured holding time of a call is compared to the preset limits, these counters are incremented or decremented as follows:

- Measured holding time less than short holding time limit—Short holding time counter is incremented.
- Measured holding time greater than short holding time limit and less than long holding time limit—Short holding time counter is decremented.
- Measured holding time greater than long holding time limit—Long holding time counter is incremented.

The short holding time counter is constantly compared to a preset threshold. This threshold can be from 0 to 30 and is set by the System Manager. The threshold for the long holding time counter is always 1. Each time a counter reaches a preset threshold, two things occur:

- If ACA referral has been activated by an attendant or voice terminal user, a referral call is sent by the system to a designated attendant or display-equipped voice terminal.
- An entry is made in an audit trail which stores information on the occurrence.

When ACA is enabled by the System Manager, the ACA measurements are made and the audit trail is updated each time a preset counter threshold is reached. However, in order for a referral call to be sent, ACA referral must be activated. ACA referral is activated whenever an attendant or user presses an ACA button. When this is done the system can send referral calls to the destination specified by the System Manager.

The referral call destination can be the attendant group, a specific attendant, or a display-equipped voice terminal. The information appearing on the display identifies the call as an ACA call, identifies the trunk group access code and the trunk group member number, and shows the reason for referral (short or long holding time). When the call is answered, this information is displayed and remains displayed until the call is released.

Each time a counter threshold is reached, a record of the information is stored in the audit trail. The audit trail records are available to the System Manager. Each record contains the following information:

- Time and Date of occurrence
- Trunk group access code and trunk group member
- Type of referral (short or long holding time)
- Whether or not the referral call was successful

If the referral call destination does not answer the call within 3 minutes, the call times out and this information is entered in the audit trail. The audit trail is examined once each hour. If any entries indicate a referral call was not completed, the call is tried again.

ACA can be enabled or disabled for the entire system by the System Manager. The System Manager can also enable or disable ACA for each individual trunk group. When ACA is disabled, ACA measurements are not made.

Two extensions must be assigned for the purpose of letting the referral call destination identify the type of ACA call (short or long holding time). The two extensions are assigned as a short holding time origination extension and a long holding time origination extension. These extension numbers do not require hardware circuit packs.

As an illustration of how ACA functions, assume the following:

- The ACA is enabled for the entire system.
- The ACA referral destination is extension 389.
- The ACA long holding time origination extension is 423.
- The long holding time limit for trunk group 3 (trunk access code is 9) is 1 hour.
- The ACA referral is activated.

With the above information, assume a call is made on a trunk in trunk group 3 and the call lasts more than 1 hour. Then, the threshold for the long holding time counter is reached, a referral call is made to extension 389, the display reflects a long holding time call, and the information is entered in the audit trail. The referral destination can then have the operation of the trunk checked and taken out of service if defective.

Considerations

The ACA feature provides better service through early detection of faulty trunks and consequently reduces out-of-service time. Some types of trunk failures cause people to shorten their calls. For example, an excessive number of short calls may indicate a noisy trunk. Similarly, a trunk that remains busy for an abnormally long time may be permanently busy due to a trunk fault. The ACA feature takes advantage of these characteristics to identify possibly defective trunks. Once the trunk has been identified as possibly being defective, the Busy Verification of Terminals and Trunks (V2) feature can be used to check the trunk.

The audit trail contains a maximum of 64 records at any one time. The oldest information is overwritten by the newest information.

Measurements are not made on personal central office lines, out-of-service trunks, or trunks undergoing maintenance testing.

Interactions

- Centralized Attendant Service (CAS)

  When CAS is activated, the referral call destination must be on the local System 75. A referral destination of "0" is interpreted as the local attendant, if one exists.

  The CAS attendant cannot activate or deactivate ACA referral calls at a branch location.

- Night Service—Night Station Service

  Referral calls will not be placed if the system is in the Night Service mode.

Administration

ACA is administered by the System Manager. The following items require administration:

- Whether ACA is enabled or disabled (per system).
- Short holding time origination extension (per system). Assigned name must reflect short holding time nomenclature.
- Long holding time origination extension (per system). Assigned name must reflect long holding time nomenclature.
- Referral destination (per system).
- Whether ACA is assigned (per trunk group).
- Short holding time limit (per trunk group).
- Long holding time limit (per trunk group).
- Threshold for short holding time counter (per trunk group).
- ACA activate/deactivate button on attendant console or voice terminal (one per system).

Hardware and Software Requirements

No additional hardware or software is required.

AUTOMATIC INCOMING CALL DISPLAY (V2)

Description

Provides multi-appearance voice terminal users with the identity of a second or subsequent caller while busy on an existing call. The identity is displayed on a display-equipped terminal.

The 40-character alphanumeric display can be the digital display module associated with a 7405D voice terminal, a 515 BCT, or line 1 of the display on a 7407D voice terminal.

This feature applies when an incoming call terminates at a user's voice terminal while the user is active on another call appearance. The information displayed on the current call is replaced by the identity of the incoming call. The identity of the incoming call normally remains displayed for 30 seconds unless there is another incoming call, the user hangs up, or the calling party hangs up. After 30 seconds, the display returns to the current call information.

A third or subsequent incoming call overwrites the information displayed on the previous call and restarts the 30-second interval. In any case, the most recent call to terminate at the user's voice terminal is the call identified by the display.

If the party whose identity is currently being displayed hangs up, the display returns to the current call information. If the user hangs up on the current call before the 30-second interval expires on the incoming call, the display is cleared.

Considerations

The Automatic Incoming Call Display feature lets certain users, while active on one call, know the identity of another incoming caller. This is done without the use of an Inspect button. By knowing who is calling, the user can handle the calls accordingly.

The incoming call must terminate at the user's voice terminal in order to be displayed. Calls forwarded to another extension are not displayed.

The digital display module, or 515 BCT must be in the normal mode to display the identity of the incoming call. This is not required of the 7407D.

Interactions

This feature enhances the Voice Terminal Display feature by providing automatic identification of incoming calls. The same incoming call information can be provided by putting the display in the inspect mode; however, this is not automatic and must be done manually for each call.

Administration

None required.

Hardware and Software Requirements

Requires a 7405D voice terminal with a Digital Display Module, a 515 BCT, or a 7407D and one port on a TN754 Digital Line circuit pack. No additional software is required.

AUTOMATIC ROUTE SELECTION (V1)

Description

Routes calls over the public network based on the preferred (normally the least expensive) route available at the time the call is placed.

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.

An Automatic Route Selection (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)
- Tie trunks

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 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 most privileges. A trunk group with an FRL of 0 is least restricted; an FRL of 7 is most restricted.

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 or local Area Code.
- Calls made to Foreign Numbering Plan Area (FNPA) Codes. FNPA Codes are simply Area Codes other than the local Area Code.
- Calls made to COs located within a particular FNPA. Calls to these COs are referred to as Remote Home Numbering Plan Area (RHNPA) calls. Pattern selection is based on a CO code rather than on the Area Code alone.
- Special service calls such as 911.
- International calls.

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 type of call determines which table a call enters.

A call within the Home Area Code enters the HNPA table. The HNPA table contains an ARS pattern number for each individual office code within the Home Area Code. When a particular office code within the Home Area Code is called, the corresponding Routing Pattern is selected.

Calls to area codes outside the Home Area Codes and special service codes such as 911 enter the FNPA table. The FNPA table contains either an ARS pattern number or an RHNPA table number for each of the 160 possible 3-digit area codes and service codes that can be accessed by the system.

When an area code outside the Home Area Code is called and the FNPA table contains an ARS pattern number for that area code, the call follows the assigned Routing Pattern.

When an area code outside the Home Area Code is called and the FNPA table contains an RHNPA table number for that area code, the call enters the assigned RHNPA table. Each RHNPA table contains an ARS pattern number for each individual office code within the associated area code. When a call enters an RHNPA table, a Routing Pattern is selected according to the called office code within that RHNPA table.

If a call begins with 0, it does not enter any of the six tables stored in memory but, instead, enters the prefix table. All calls beginning with the following 0-type codes use the same assigned Routing Pattern:

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

All calls beginning with 10 use the same Routing Pattern as assigned in the prefix table. 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 digit dialed (0-type call), first three digits dialed (office or area code), or the first six digits dialed (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 compatibility requirements, the system automatically checks for an available trunk within the selected trunk group. If there is an idle trunk, 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.

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.
- Intercept—Indicates that the originating FRL is not sufficient to allow the call.

Considerations

With ARS, voice terminal users do not have to worry about accessing a particular trunk group to make a long-distance call. The user simply activates ARS and dials the desired number. The system then routes the call to the outgoing trunk group best suited for that call.

A system can have up to 16 ARS Routing Patterns. Each pattern can contain up to six trunk groups.

There is one HNPA table, one FNPA table, and four RHNPA tables per system. The HNPA table and the four RHNPA tables can each contain up to 800 office codes. The FNPA table can contain up to 160 Area Codes.

If a client chooses a single primary long-distance carrier for all long distance (1+) calls, then any International Direct Distance Dialing (IDDD) (011), operator (0), Customer-Dialed and Operator-Serviced (CDOS) (0+ or 01+), 700, and 900 calls also go to that carrier. In order to place a call to an area not served by the primary long-distance carrier, the appropriate 10xxx code must be dialed to access a different carrier that has access to the desired area.

If a client changes ARS routing assignments, it is the client's responsibility to notify the Regional Support Center (RSC) network designer and the System Control Office (SCO) technician of the changes in order to receive their continued support.

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)

  A 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.

- 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 with an ARS call, it must be dialed before the ARS access code is dialed.

  An ARS call is identified by the access code dialed and by a Condition Code. The dialed number is recorded as the called number.

Administration

ARS is administered on a per-system basis by the System Manager. The following items must be established, defined, and administered before ARS can be activated.

- Facility 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, RHNPA, or prefix tables. Patterns are administered on the FNPA, HNPA, or RHNPA, or Prefix forms.

  Patterns 2, 3, 4, and 5 are default patterns for the RHNPA Tables 1, 2, 3, and 4, respectively. Pattern 1 is the 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 most-preferred 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 Numbering 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 assigned as Advanced Private Line Termination (APLT) can be assigned as a trunk group in an ARS pattern. For calls using an APLT trunk group, ARS inserts the HNPA, if not dialed, since private networks require ten digits on all calls destined for the public network.

- Toll Tables

  Up to four Toll Tables can be established. Toll tables 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 are assigned to CO, FX, and WATS trunk groups.

- Prefix Table

  This table specifies the Routing Pattern number used for calls beginning with 0 or 10. This includes operator-assisted calls, directly dialed international calls, and long-distance carrier calls. The ARS pattern used for these calls is assigned on the ARS Prefix Codes form.

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" (that is, RHNPA) 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

No additional hardware is required. ARS software is required.

AUTOMATIC ROUTE SELECTION (V2)

Description

Routes calls over the public network based on the preferred (normally the least expensive) route available at the time the call is placed.

Automatic Route Selection (ARS) provides a choice of up to six routes for any given public network call. The following types of trunk groups can be accessed by ARS:

- Local central office—Used for local calls and to provide access to a long-distance carrier. Access to the long-distance carrier can be provided either automatically by the central office or by a carrier access code.
- Foreign exchange—Used to emulate local calling in an area not served by the local central office. Like the local central office, the foreign exchange office provides a choice of long-distance carriers.
- Wide Area Telecommunications Service (WATS)—Used to provide calling to predefined geographic areas at a rate based on expected usage.
- Tie trunks—Used to provide access to an Electronic Tandem Network (ETN), or to an Enhanced Private Switched Communications Service (EPSCS) or Common Control Switching Arrangement (CCSA) office. (In some cases, it is preferable to allow a private network to handle the routing of calls destined for the public network.)

ARS is particularly useful when one or more long-distance carriers and WATS are provided. The system selects the most-preferred (normally, least-expensive) route for the call. Long-distance carrier code dialing is not required on routes selected by the system. Long-distance carrier codes are set in translations to best benefit the client on any given call. These codes are inserted as needed to guarantee automatic carrier selection.

Dial access to a long-distance carrier's operator is also provided. Carrier access codes are of the form 10xxx, where xxx are digits to identify a particular carrier. If 10xxx plus a 0-type (operator or operator-assisted) number is dialed, the call routes to the long-distance carrier's operator.

Although System 75 has limited capabilities to serve as an Electronic Tandem Network (ETN) tandem switch, it may serve such a function in some networks. In this case the system can access or be accessed by Intertandem Tie Trunks to/from other tandem switches and/or Access Tie Trunks to/from ETN main switches. The system can also access Bypass Tie Trunks to an ETN main switch. This distinction as a tandem switch is important with respect to the routing of certain calls.

Dialing

The ARS access code is normally the digit 9. The called number may be a service code, a 7- or 10-digit public network number, an International Direct Distance Dialing (IDDD) number, an operator code (0), a customer-dialed and operator-serviced (CDOS) number ("0+" or "01+"), or a long-distance carrier's operator (10xxx + 0+). Dialing 10xxx plus a 7- or 10-digit number is also possible, although route selection is based on the 7- or 10-digit number, not on the 10xxx.

To use ARS from a System 75, the user simply dials the ARS access code and the called number. Users at subtending switches access the System 75, then follow the same dialing procedures as a user at the System 75.

Domestic Call Routing

The domestic calling area is divided into areas called numbering plan areas, or NPAs. Each NPA is identified by an NPA code, normally just called an Area Code. There are 160 such codes: 1 for the local, or home numbering plan area (HNPA), and 159 for the other (foreign) numbering plan areas (FNPAs). Within a given NPA, all office codes (NXXs) are unique.

With ARS, call routing is determined by the first three or six digits of the called public network number, in other words, by the NPA or the office code or by both the NPA and the office code. Two 3-digit translators are provided: one for the office codes within the home NPA and one for the foreign NPAs. Thirty-two 6-digit translators are provided, allowing call routing based on the office codes within the foreign NPA rather than on the NPA alone. (Six-digit translators are actually 3-digit translators that are accessed from the foreign NPA translator. The foreign NPA translator can yield one of the 6-digit translators based on the NPA. The 6-digit translator only translates the office code, which is the second three digits of the called number. At this time, six digits have been translated. Thus, for clarity, it is common to refer to these translators as 6-digit translators. With System 75, these translators are also known as Remote Home Numbering Plan Area [RHNPA] tables since the call routes on an office code the same as a call within the home NPA does.)

Digit translation yields one of 254 Routing Patterns, numbered 1 through 254. More than one translator can point to the same pattern. A blank entry instead of a Routing Pattern number provides intercept treatment. However, with ARS, digit translation should always point to a Routing Pattern. This way, calls to unassigned office codes will be intercepted by the central office, not by the system. By allowing the unassigned codes to be intercepted by the central office, the System Manager does not have to keep track of which office codes are in service. If calls to some codes are to be denied, this should be handled by Facility Restriction Level (FRL) assignment, not by intercept on the codes. FRLs are discussed elsewhere in this section.

The Routing Pattern applicable for a given call contains a list of the trunk groups that can be used for the call. Trunk group access is controlled by FRLs. If access to the public network is through a main switch (an Access trunk group is selected for the call), then the call will route through the main to one of the public network offices serving the main. The digit manipulation necessary to route the call is controlled by the Subnet Trunking function. (Subnet Trunking is discussed elsewhere in this section.) Otherwise, the digit string to be outpulsed is controlled by ARS. ARS digit manipulation is called code conversion. Code conversion is used to determine whether or not to outpulse the digit 1 on toll calls and whether to insert, keep, or delete the NPA on toll calls. Whether or not the digit 1 should be dialed on an ARS call is a completely separate subject and has nothing to do with outpulsing a 1. Each of these items is discussed separately in the following paragraphs.

Digit 1 Dialing

Normally, the prefix digit 1 is not dialed on a 7- or 10-digit call routed by ARS.

However, there are two cases where the digit 1 must be dialed. Some metropolitan areas are so densely populated that there simply are not enough traditional office codes, that is, those that do not conflict with NPAs. In areas where NPA codes also serve as office codes, the digit 1 must be dialed if a toll (NPA) call is intended. The digit tells the system whether to route the call as a 7-digit call via the home NPA translator (1 not dialed) or as a 10-digit call via the foreign NPA translator (1 dialed). Digit 1 dialing may also be required in areas near an NPA boundary. In these areas, certain calls to the adjacent NPA may be local calls rather than toll calls. However, office codes are duplicated in the home and adjacent NPAs. Thus, if the digit 1 is not required on certain adjacent NPA calls, then it must be dialed on the home NPA calls so the system can differentiate between the intended destinations.

Digit 1 Outpulsing

The digit 1 may or may not be required at the public network office to which the call will be routing. (If "1" is dialed on 7-digit calls at a System 75 stand-alone (non-ETN) switch, the "1" is outpulsed by the system.) In the other cases, the "1" requirements are indicated in the system. Since any given call may have a choice of up to six routes, some of which may require a 1 and some of which may not, this indication is associated with each individual route. Four choices are available and are identified in translations by a Prefix Mark. Values and their meaning are as follows:

- Prefix Mark 0—Never send a 1.
- Prefix Mark 1—Send a 1 on 10-digit calls, but not on 7-digit calls.
- Prefix Mark 2—Send a 1 on all toll calls.
- Prefix Mark 3—Send a 1 on all toll calls and keep or insert the NPA to insure that all toll calls are 10-digit calls.

Which of the four possible treatments of the 1 prefix digit is applicable on a given route is based on the characteristics of the distant office. Prefix Marks 2 and 3 are associated with step-by-step switches. Prefix Marks 0 and 1 are associated with the other switches. Prefix Mark 0 is straightforward. The system never sends a 1 prefix digit. Prefix Mark 1 causes the system to send a 1 prefix on all 10-digit calls.

With Prefix Marks 2 and 3, the decision is based on whether the call is a toll call. Toll Lists are provided in the system to furnish this information. A Toll List simply indicates if the office code associated with the call constitutes a toll call from the interconnecting office (not from the local System 75). Up to 32 Toll Lists are provided. The applicable list number, if any, for the call is given in the Routing Pattern.

Prefix Marks are only applicable on public network routes. No Prefix Mark is given on tie trunks. Also, the digit 1 is never transmitted over an intertandem tie trunk, even if dialed by the calling party. Requirements for a 1 are specified via Prefix Marks when the call accesses the public network. Thus, a 1 is never needed on an intertandem tie trunk.

NPA Deletion and Insertion

Each public network route in the ARS Routing Pattern contains an indication of the NPA of the distant end of the trunk group. If this NPA is the same as the NPA associated with the call, the NPA is deleted prior to outpulsing unless the Prefix Mark is 3 and the call is a toll call in the associated Toll List.

The NPA is inserted on 7-digit calls if the distant NPA is different from the home NPA or if the Prefix Mark is 3 and the call is a toll call in the associated Toll List.

The preceding paragraphs describe NPA deletion or insertion when the call is accessing the public network (and exiting an ETN, if an ETN provided partial routing of the call). When an ARS call accesses an ETN is one case of NPA insertion. If the call is to a destination within the home NPA and if the calling party did not dial the NPA, the system inserts the home NPA before sending the call to the ETN switch. Therefore, all ARS calls (to a domestic destination) accessing an ETN are 10-digit calls. This enables the system to distinguish between ARS calls and the 7-digit on-network calls.

IDDD and Service Code Dialing

IDDD calls other than those generated by Subnet Trunking need not be modified before outpulsing. Since international numbers can be of variable length, the system awaits a dialing time-out before processing the call. (Dialing time-out is 3 seconds for the 0 and 1 prefix digits, but is 10 seconds for the called number.) The calling party can speed up call processing by dialing the end-of-dialing digit (#) after the called number. Receipt of this digit cancels the remaining time-out interval. The system always outpulses the # digit for use by the distant switch, whether dialed by the calling party or not.

No Subnet Trunking is required for service codes. If the prefix digit 1 is dialed before the code, it is outpulsed.

ARS can provide individual Routing Patterns for each 0-type call. A 0-type call can be processed via the foreign NPA translator, meaning that 6-digit translation can be used. This is particularly useful on international calls, since the 6-digit translation can be used on the country code. Thus, call routing can be determined according to the called country, rather than handling all international calls alike.

Operator and Operator-Assisted Calls

Calls to an operator (0 by itself) require a 3-second time-out or dialing of the # digit before the call is processed. Operator-assisted calls (0 plus a 7- or 10-digit number) require 10-digit dialing if the call is within a home NPA and there are office codes within the HNPA which look like NPAs. (On directly-dialed calls, this distinction was made by prefix digit 1 dialing.) All other dialing is the same as direct dialing.

Operator-assisted calls, like IDDD calls, can be routed on the first three digits of the called number. Through the use of Subnet Trunking, this means that different long-distance carriers can be selected for different calls.

Directory Assistance Calls

Local Directory Assistance Calls always route to a telephone company operator. Long-distance Directory Assistance Calls may be routed to the long-distance carrier operator via an associated 6-digit RHNPA table.

Tones

The following tones are associated with ARS:

- Busy—Indicates that the called number is busy.
- Confirmation—Indicates that the call has queued.
- Intercept—Indicates that the originating FRL is not sufficient to allow the call.
- Reorder (fast busy)—Indicates that the call cannot be completed at this time because at least one required facility is not available. (Multi-appearance voice terminal users may be able to queue the call.)

Special Call Routing

The System recognizes certain types of dialing patterns on outgoing calls and routes these calls via special entries in the HNPA or FNPA table. Table 3-A lists the special dialing patterns along with the associated HNPA or FNPA table entry through which that type of call is routed.

TABLE 3-A. ARS Routing Table

DIALED DIGITS	ROUTED ON	IN TABLE
0	000	FNPA
011X...X	011	FNPA
01X...X	010	FNPA
0X...X	001	FNPA
(1)N11	N11	FNPA
(1)NXX-XXXX	NXX	HNPA
(1)800-NXX-XXXX	800	FNPA
(1)NIX-555-XXXX	005	FNPA
(1)HNPA-NXX-XXXX	NXX	HNPA
(1)NIX-NXX-XXXX	NIX	FNPA
10XXX	100	FNPA
10XXX-0	100	FNPA
10XXX-011X...X	111	FNPA
10XXX-01X...X	110	FNPA
10XXX-0X...X	101	FNPA
10XXX (1)555-XXXX	555	HNPA
10XXX (1)NXX-XXXX	NXX	HNPA
10XXX (1)800-NXX-XXXX	800	FNPA
10XXX (1)HNPA-NXX-XXXX	NXX	HNPA
10XXX (1)NIX-555-XXXX	005	FNPA
10XXX (1)NIX-NXX-XXXX	NIX	FNPA

Legend: N — any digit 2-9
I — any digit 0-1
X — any digit 0-9
( ) — an optional digit

Note: ARS ignores the IXC access code unless it is followed by a "0."

Considerations

ARS provides the most-preferred usage of public network facilities available at a System 75.

Up to 254 Routing Patterns, shared with AAR, can be provided.

Two 3-digit translators are provided.

Up to 32 Toll Lists can be provided.

Up to 32 RHNPA tables can be provided.

If a client chooses a single primary long-distance carrier for all long distance (1+) calls, then any IDDD (011), operator (0), CDOS (0+ or 01+), 700, and 900 calls also go to that carrier. In order to place a call to an area not served by the primary long-distance carrier, the appropriate 10xxx code must be dialed to access a different carrier who has access to the desired area.

If a client changes ARS routing assignments, it is the client's responsibility to notify the Regional Support Center (RSC) network designer and the System Control Office (SCO) technician of the changes in order to receive their continued support.

Interactions

- Automatic Alternate Routing (AAR)

  ARS and AAR can access the same trunk groups and share the same Routing Patterns.

- Abbreviated Dialing

  FRL checking is bypassed on an ARS call made via a privileged Abbreviated Dialing Group List.

- Ringback Queuing

  Ringback Queuing can be used on ARS calls originated at the switch that provides the queuing. Incoming tie trunk calls will not queue on an outgoing trunk group.

- Code/Toll Restriction

  Code/Toll Restriction is not checked on ARS calls.

- Controlled Restriction, Origination Restriction, and Outward Restriction

  These features prohibit access to ARS.

- Miscellaneous Trunk Restrictions

  Miscellaneous Restrictions are not checked on ARS calls.

- Attendant Control of Trunk Group Access

  Attendant control of a trunk group, in effect, removes the trunk group from the Routing Pattern. The trunk group is never accessed by the ARS feature. ARS calls do not route to the attendant.

- Station Message Detail Recording (SMDR)

  An ARS call using a trunk group marked for SMDR is indicated by the dialed access code and by a Condition Code. The dialed number is recorded as the called number. Subnet Trunking does not affect 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 with an ARS call, it must be dialed before the ARS access code is dialed.

Administration

ARS is initially assigned on a per-system basis by an AT&T service technician. After the feature is activated, the following items are administered by either the System Manager or the service technician:

- ARS Access Code (1 to 3 digits)
- Three-digit Home NPA Table—Points to the appropriate Routing Pattern for each office code within the home NPA.
- Three-digit Foreign NPA Table—Points to the appropriate Routing Pattern for each nonlocal NPA or points to a 6-digit translator so the call will be routed on both the NPA and the office code.
- Up to 32 Remote Home NPA Tables—Provides 6-digit translation on selected foreign NPAs. Since calls accessing one of these tables route on an office code, similar to the way home NPA calls route, the term Remote Home NPA is used.

- Toll Lists—Provides an indication of whether each office code (with respect to the distant end of the trunk group) is a local or toll call.
- FRLs—Must be assigned via a Class of Restriction to each originating facility. Minimum FRLs required to access a route are assigned as part of the Routing Pattern. Assignment of these values determines the calling privileges of each individual user of the ETN.
- Routing Patterns—Provide an indication of the NPA at the distant end of the trunk group selected for the call and the applicable Toll List number, if any. The Routing Pattern also provides FRL and Subnet Trunking data. (Refer to the FRL and Subnet Trunking descriptions for details.)

Hardware and Software Requirements

ARS may be used on a stand-alone system or may be an integral part of a private network. No additional hardware is required for a stand-alone system. A private network may require additional tie trunks and TN748B Tone Detector circuit packs. These additions are, however, cost effective when compared to the alternatives for call routing.

ARS is provided via the optional ARS software.

BRIDGED CALL APPEARANCE

Description

Allows multi-appearance voice terminal users to have an appearance of another user's primary extension number. The bridged call appearance can be used to originate, answer, and bridge onto calls to or from the other user's primary extension number.

The appearance of a voice terminal's primary extension number at another voice terminal is called a bridged call appearance. The Bridged Call Appearance feature is used by lifting the handset and pressing the Bridged Appearance button. The user is then bridged onto the other voice terminal's primary extension number and can handle calls on that extension number. The bridged appearance can be used to originate calls from, and answer calls to, the other voice terminal's primary extension number. The user can also bridge onto an existing call to or from the other voice terminal.

An incoming call rings the primary extension number's voice terminal and all voice terminals that have a bridged call appearance of the voice terminal's primary extension number. Each voice terminal has the option of audible or silent ringing for all bridged call appearances on the voice terminal.

A bridged call appearance can be assigned to any 2-lamp button. It does not require the use of a regular call appearance. A bridged call appearance can be used just like a regular call appearance for most features. For example, the Hold, Transfer, and Priority Calling features can be used from a bridged appearance, just as they would be used from a regular call appearance.

Considerations

The Bridged Call Appearance feature allows calls to be handled from more than one voice terminal. Some practical uses of this capability are as follows:

- A secretary making or answering calls on an executive's primary extension

  These calls can be placed on hold for later retrieval by the executive, or the executive can simply bridge onto the call. In all cases, the executive handles the call as if he or she had placed or answered the call. It is never necessary to transfer the call to the executive.

- A secretary taking care of details for an executive who is already active on a call

  A secretary can bridge onto an active call and take down information such as an address or telephone number. This frees the executive for more important matters.

- Visitor telephones

  An executive may have another voice terminal in his or her office which is to be used by visitors. It may be desirable that the visitor be able to bridge onto a call which is active on the executive's primary extension number. A bridged call appearance makes this possible.

- Service environments

  It may be necessary that several people be able to handle calls to a particular extension number. For example, several users may be required to answer calls to a hotline number in addition to their normal functions. Each user may also be required to bridge onto existing hotline calls. A bridged call appearance provides this capability.

- A user frequently using voice terminals in different locations

  A user may not spend all of his or her time in the same place. For this type of user it is convenient to have his or her extension number bridged at several different voice terminals.

A voice terminal's primary extension number can have an appearance on up to seven other voice terminals. The number of bridged call appearances allowed at each voice terminal is limited only by the number of 2-lamp buttons available on the voice terminal.

A maximum of 400 (V1) or 500 (V2) bridged call appearances are allowed per system.

It is recommended that a bridging voice terminal have a bridged call appearance corresponding to each call appearance of the primary extension number at the bridged voice terminal. For example, if a bridged voice terminal has three call appearances of its primary extension, a bridging voice terminal should have three bridged call appearances of that extension. This allows users to refer to the individual call appearances when talking about a specific call.

A single-line voice terminal cannot have any bridged call appearances and cannot bridge onto other extensions.

If a user's primary extension is assigned to a call appearance on a Call Coverage module, that call appearance cannot be bridged.

Bridged call appearances may result in the reduction of available feature buttons, thereby reducing a user's capabilities. A Call Coverage module can be used to provide up to 20 bridged call appearances. This leaves the other call appearances available for use with other features.

If a call terminates at a voice terminal on an extension number other than the primary extension number (for example, Terminating Extension Group extension number), a bridged call appearance is not maintained.

The Bridged Call Appearance feature should not be considered as a replacement for Call Coverage.

Interactions

- Abbreviated Dialing

  A user, accessing Abbreviated Dialing while on a bridged call appearance, accesses the Abbreviated Dialing lists of the primary extension associated with the bridged call appearance.

- Attendant Display and Voice Terminal Display

  A call from the primary extension number or a bridged call appearance of the primary extension number is displayed as a call from the primary extension number.

- Automatic Callback

  Automatic Callback calls cannot originate from a bridged call appearance. When a call is originated from a primary extension number, the return call notification rings at all bridged call appearances.

- Call Coverage

  Coverage criteria for bridged call appearances is based entirely on the criteria of the primary extension associated with bridged appearance.

- Call Forwarding All Calls

  Call Forwarding All Calls can be activated or canceled for the primary extension number from a bridged call appearance of that number. When activated, calls to the primary extension number do not terminate at the bridged call appearances. Bridged call appearances do not receive redirection notification when a call to the primary extension is forwarded.

- Call Park

  When a call is parked from a bridged call appearance, it is parked on the primary extension assigned to the bridged call appearance.

- Call Pickup

  If a voice terminal receives ringback tone on a bridged call appearance, the incoming call can be picked up by members of that voice terminal's Call Pickup group. This causes all bridged call appearances to be dropped. The call is parked on the primary extension number of the answering voice terminal.

- Class of Restriction

  The Class of Restriction assigned to a voice terminal's primary extension also applies to calls originated from a bridged call appearance.

- Conference—Attendant and Conference—Terminal

  A Bridged Appearance button can be used to make conference calls.

- Consult

  Bridged call appearances of the primary extension do not ring on a consult call to the primary extension.

- Hold

  Any user, active on a bridged call, can place the call on hold. If a call on a bridged call appearance is placed on hold and no other users with a bridged call appearance of the same extension number are connected to the call, the status lamp at the Bridged Appearance button indicates that the call is on hold. If the primary extension or another bridged appearance is connected to the call, the status lamp at all bridged appearances indicates an active status for the call.

- Last Number Dialed

  Activation of the Last Number Dialed feature causes the last number dialed from the voice terminal to be redialed, regardless of the extension number used (primary or bridged call appearance).

- Leave Word Calling

  A Leave Word Calling message left by a user on a bridged call appearance leaves a message for the called party to call the primary extension number assigned to the bridged call appearance. When a user calls a primary extension, and activates Leave Word Calling, the message is left for the primary extension, even if the call was answered at a bridged call appearance.

- Personal Central Office Line

  If a user is active on his or her primary extension number on a Personal Central Office Line (PCOL) call, bridged call appearances of that extension number cannot be used to bridge onto the call. The call can only be bridged onto if another voice terminal is a member of the same PCOL group and has a PCOL button.

- Privacy—Manual Exclusion

  When Privacy—Manual Exclusion is activated, all other users are prevented from bridging onto the active call.

- Ringback Queuing

  Ringback Queuing is not provided on calls originated from a bridged call appearance.

- Send All Calls

  If a voice terminal user has activated Send All Calls on its primary extension, incoming calls will still ring bridged call appearances of that extension as long as a Temporary Bridged Appearance of the call is maintained at the primary extension.

Administration

The Bridged Call Appearance feature is administered on a per-voice terminal basis by the System Manager. The following items require administration:

- Bridged Appearance buttons (per voice terminal)
- Audible or silent ringing for bridged appearances (per voice terminal)

Hardware and Software Requirements

No additional hardware or software is required. A Call Coverage module can be used to provide up to 20 bridged call appearances.

BUSY VERIFICATION OF TERMINALS AND TRUNKS (V2)

Description

Allows attendants and specified multi-appearance voice terminals to make test calls to trunks, voice terminals, and hunt groups [Direct Department Calling (DDC) and Uniform Call Distribution (UCD) groups]. These test calls are used to determine the status of the called voice terminal, hunt group, or trunk.

Busy verification of voice terminal extensions, hunt group extensions, and trunks can be done by either multi-appearance voice terminal users or attendants. Feature activation is via a Busy Verify button.

An attendant or multi-appearance voice terminal user can activate Busy Verification of Terminals and Trunks by pressing the Busy Verify button. The attendant then dials an extension number if a voice terminal or hunt group is to be verified. If a trunk is to be verified, the attendant dials a trunk access code, followed by a 2-digit number (leading 0s may be required) to specify which member of the trunk group is to be verified.

After an attendant or multi-appearance voice terminal user has activated the Busy Verification of Terminals and Trunks, the system checks the validity of the entered extension number or trunk access code and member number. If the entered number is not a voice terminal extension number, a DDC/UCD group extension number, or a trunk access code with a valid member number, the verification attempt is denied.

If an attendant activates Busy Verification of Terminals and Trunks for a valid voice terminal extension number, the system initiates a priority call to that extension. One of the following then occurs:

- Voice terminal is idle.

  Priority ringing is heard at the voice terminal and the voice terminal is successfully verified. The call proceeds as a normal attendant-originated call.

- Voice terminal is active on a call.

  The system first searches for an idle call appearance on the voice terminal. If one is found, that call appearance is rung. If an idle call appearance cannot be found, or if the voice terminal is a single-line voice terminal, the attendant will bridge onto the active call. All parties on the active call receive a warning tone (2 second burst of 440 Hz tone) to let them know that the attendant is bridging onto the call. A 1/2 second burst of warning tone is repeated every 15 seconds, as long as the attendant is bridged onto the call.

- Voice terminal is out of service.

  Busy verification is denied and the attendant receives reorder tone.

If an attendant activates Busy Verification of Terminals and Trunks for a valid UCD or DDC group, the system initiates a priority call to that group. One of the following then occurs:

- At least one group member is available for incoming calls.

  The call rings the available group member and is treated as a priority call from an attendant to the group.

- All group members have activated the Make Busy function.

  Busy verification is denied and the attendant receives reorder tone.

- Not all group members have activated Make Busy, but no group members are available for incoming calls.

  The call will not queue if a queue is available. Busy verification is denied.

If an attendant or a multi-appearance voice terminal user activates Busy Verification of Terminals and Trunks for a valid trunk, the system checks the status of that trunk. One of the following then occurs:

- The trunk is idle.

  If the trunk is an outgoing trunk, the originator of the busy verification receives dial tone and can make a call on that trunk to verify that it is in working order. If the trunk is an incoming trunk, the originator of the busy verification receives confirmation tone as an indication that the trunk is available for use.

- The trunk is busy with an active call.

  The originator of the busy verification is bridged onto the active call. All parties on the active call receive a warning tone (2 second burst of 440 Hz tone) to let them know that the originator of the busy verification is bridging onto the call. A 1/2 second burst of warning tone repeats every 15 seconds, as long as the busy verification originator remains on the call.

- The trunk is out of service.

  The busy verification is denied. The attendant receives intercept tone.

If busy verification is denied for any other reason, intercept tone or reorder tone is returned to the user.

Considerations

Busy Verification of Terminals and Trunks provides attendants with an easy method of checking the condition of certain extensions and trunks. An attendant or multifunction voice terminal can distinguish between a voice terminal that is truly busy and one that only appears busy because of some trouble condition. Attendants or multifunction voice terminal users can also use the feature to quickly identify faulty trunks. As a result, better communications service is provided and faulty trunks can be corrected more quickly.

A busy verification can be performed on the following:

- Voice terminal extensions
- UCD and DDC hunt group extensions
- Members of the following types of trunk groups:
  - Direct Inward Dialing (DID)
  - Central Office (CO)
  - Foreign Exchange (FX)
  - Wide Area Telecommunications Service (WATS)
  - Advanced Private Line Termination (APLT)
  - Tie
  - Remote Access
  - Release Link Trunk (RLT)

The bridging capability associated with Busy Verification of Terminals and Trunks is not provided on verification attempts to UCD and DDC groups or RLTs.

Outgoing test calls cannot be made on DID trunks.

Interactions

- Automatic Callback

  Once the called party in an Automatic Callback call hangs up, neither extension number can be busy verified until both the calling and called parties are connected or the callback attempt is canceled (by the activating party or by time-out of the callback interval).

- Call Coverage

  Since the busy verification call to an extension number is originated as a priority call, the call does not go to coverage.

- Call Forwarding

  A busy verification made to an extension with call forwarding activated, does not busy verify the forwarded-to extension. Only the called extension is busy verified.

- Call Waiting Termination

  A busy verification cannot be made to an extension which is waiting to be answered at another extension.

- Conference—Attendant and Terminal

  If a conference call involves six parties, busy verification on any extension number in the conference is denied. If the number of parties in the conference is five or less, a busy verification can be performed on any of the associated extension numbers.

- Data Privacy

  Busy verification is denied if it results in a bridging attempt on a voice terminal which has activated Data Privacy.

- Data Restriction

  If Data Restriction is active on a call, and a busy verification bridging attempt is made on that call, the busy verification is denied.

- Hold

  A busy verification of a multi-appearance voice terminal is denied if all call appearances have calls on hold.

- Individual Attendant Access

  An attendant cannot make a busy verification of another individual attendant console or of the attendant group.

- Loudspeaker Paging Access

  If the voice terminal or trunk to be verified is connected to paging equipment, the verification attempt is denied.

- Voice Terminal Origination Restriction

  A voice terminal that is origination restricted can be assigned a Busy Verify button. However, the button cannot be used.

- Voice Terminal Termination Restriction

  Voice terminals that are termination restricted cannot be busy verified.

- Transfer

  Once the originator of the busy verification has bridged onto a call, any attempt to transfer the call is denied until the originator drops from the call.

Administration

Busy Verification of Terminals and Trunks is administered on a per-voice terminal basis by the System Manager. The only administration required is the assignment of a Busy Verify button to the desired attendant consoles and multi-appearance voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

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. If a coverage path is not assigned to a particular facility, calls will not be redirected from that facility, unless another feature such as Call Forwarding All Calls is assigned. 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 rung 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 (1 to 9 ringing 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

  Allows users, when making a call to another internal extension, to send the call directly to coverage. This is optionally assigned to a button on a voice terminal and is activated by the internal calling party. Use of Go to Cover 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 rung on 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.

TEG calls are not affected by the activation of Send All Calls.

- 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.

Attendant-extended calls that redirect to Call Coverage use the type of redirection criteria (external or internal) applicable for the calling party. In other words, the call is treated as if no attendant is involved.

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 rings 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 when 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 ringing) 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. Busy means no call appearances are 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 until the calling party activates Go to Cover. These calls take precedence over the redirection criteria and seize the call appearance normally reserved for outgoing calls, if no other call appearances are available.

An internal calling party is informed that a call is redirecting to coverage by a single, short burst of ringing, 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 seizes any idle call appearance. If there is no idle call appearance, 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 ring simultaneously. Anyone in the group can answer the call. A Coverage Answer Group member already handling a group call is rung when another call is redirected to that Coverage Answer Group. If a Coverage Answer Group member is also a member of another Coverage Answer Group, he or she can also receive calls for the other group. A second call directed to a Coverage Answer Group lights a Coverage Incoming Call Identification lamp.

- 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. A second coverage call takes control of the Coverage ICI lamp and does not return control to the previous call when the second call is released.

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 rings the individual voice terminal or member of a group specified for that point or queues on the group. Once a call is ringing 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 (1 to 9 ringing 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 ringing 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. Establish two groups, if required. Note 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 ring 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 system allows for as many as 100 (V1) or 125 (V2) Coverage Answer groups with up to 8 voice terminals in each group.

Up to 200 (V1) or 250 (V2) coverage paths can be established. Each coverage path can have one, two, or three coverage points.

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 ringing, 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.

- Bridged Call Appearance

  Coverage criteria for bridged call appearances is based entirely on the criteria of the primary extension associated with the bridged call appearance.

  If a voice terminal user has activated Send All Calls on its primary extension, incoming calls will still ring bridged call appearances of that extension as long as a Temporary Bridged Appearance of the call is maintained at the primary extension.

- 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 has 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 and released.

  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.

- Night Service—Night Station Service (V1)

  A call routed to the DID LDN night extension via Night Station Service does not go to coverage, even if the coverage criteria of the DID LDN night extension is met.

- 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)

  If a user has a Make Busy button, and activates or deactivates Send All Calls, the Make Busy function associated with DDC or UCD is activated or deactivated simultaneously.

  If a user has no Make Busy button, activating or deactivating Send All Calls still makes the user available or unavailable for DDC or UCD calls, but Make Busy is not activated or deactivated. The Make Busy activate or deactivate code and the DDC or UCD extension must be dialed to activate the Make Busy function.

  Activating or deactivating the Make Busy function does not activate or deactivate Send All Calls.

Administration

Call Coverage is administered by the System Manager. The following items require administration:

- Coverage Paths

  The same coverage path can be used for as many voice terminal users as desired.

- Cover Answer Groups

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

  The Don't Answer Interval specifies the number of ringing cycles heard at the principal's terminal before the call is redirected to the first coverage point. This interval is recommended to be two rings, but can be administered from one to nine rings. 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 rings at a covering terminal before the call attempts to redirect to the next coverage point. This interval is recommended to be two rings, but can be administered from one to nine rings. This interval is administered as a system parameter.

- Caller Response Interval

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

- 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.

- Feature Access Codes for Activation and Deactivation of Send All Calls

- Buttons on Multi-Appearance Voice Terminals, as desired:

  — Consult

  — Coverage Callback

  — Go To Cover

  — Coverage ICI

  — Send All Calls

Hardware and Software Requirements

No additional hardware or software is required.

CALL FORWARDING ALL CALLS (V1)

Description

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

Call Forwarding All Calls can be activated or deactivated by voice terminal users.

Voice terminal users activate Call Forwarding All Calls by dialing a feature access code followed by the designated (forwarded-to) extension number. The feature is deactivated by dialing a different feature access code.

Calls can be forwarded only once. Calls forwarded to a designated (forwarded-to) extension number do not forward again. These calls ring 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 (if administered to do so) receive a redirection notification signal that a call is being forwarded.

Call Forwarding All Calls can also be activated or deactivated at data terminals.

Considerations

With Call Forwarding All Calls, voice terminal users can have their incoming calls forwarded to another extension number. This allows voice terminal users to have their calls follow them when they know they will be temporarily near another extension. There is no limit to the number of calls that can be forwarded simultaneously.

Calls to attendants cannot be forwarded. However, calls can be forwarded to the attendant group.

Interactions

- Automatic Callback and Ringback Queuing

  Callback calls do not forward.

- Bridged Call Appearance

  Call Forwarding All Calls can be activated or canceled for the primary extension number from a bridged call appearance of that number. When activated, calls to the primary extension number do not terminate at the bridged call appearances. Bridged call appearances do not receive redirection notification when a call to the primary extension is forwarded.

- Call Coverage

  If a forwarding extension number's Call Coverage criteria are met at the designated (forwarded-to) extension number, the forwarded call goes to coverage based on the forwarding extension's coverage criteria. 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.

- Night Service—Night Station Service

  A call routed to the Direct Inward Dialing Listed Directory Number (DID LDN) night extension via Night Station Service does not forward to another extension, even if Call Forwarding All Calls has been activated at the DID LDN night extension.

- Personal Central Office Line (PCOL)

  PCOL calls are not forwarded.

Administration

Call Forwarding All Calls is assigned on a per-extension number basis by the Class of Service. The following items require administration by the System Manager:

- Voice Terminals
  - Class of Service
  - Redirection Notification
- Feature Access Codes for Activation and Deactivation of Call Forwarding All Calls

Hardware and Software Requirements

No additional hardware or software is required.

CALL FORWARDING ALL CALLS (V2)

Description

Allows all calls to an extension number to be forwarded to a selected internal extension number, external (off-premises) number, the attendant group, or a specific attendant. This feature is activated or deactivated by dial access code or by a Call Forwarding button.

Call Forwarding All Calls can be activated or deactivated by voice terminal users and data terminal users. Also, an attendant can activate or deactivate the feature for a particular extension number, Terminating Extension Group, Direct Department Calling group, or Uniform Call Distribution group.

Voice terminal users activate Call Forwarding All Calls by dialing a feature access code or pressing a Call Forwarding button, and then dialing the designated (forwarded-to) extension number. The feature is deactivated by dialing a different feature access code or pressing the Call Forwarding button again.

An attendant activates Call Forwarding All Calls by dialing a feature access code, followed by the forwarding extension number plus the forwarded-to extension number. The attendant deactivates the feature by dialing a different access code, followed by the extension number for which the feature is to be canceled. The attendant cannot have a Call Forwarding button assigned to the console.

When a Call Forwarding button is used to activate the feature, the status lamp associated with the button remains lighted until the feature is deactivated.

Calls can be forwarded only once. Calls forwarded to a designated (forwarded-to) extension number do not forward again. These calls ring 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 (if administered to do so) receive a redirection notification signal that a call is being forwarded.

Considerations

With Call Forwarding All Calls, users can have their incoming calls forwarded to another extension number. This allows users to have their calls follow them when they know they will be temporarily near another extension. A user can also forward calls to an outside number when temporarily at an off-premises location. There is no maximum number of calls that can be forwarded simultaneously.

For Terminating Extension Groups, Uniform Call Distribution groups, and Direct Department Calling groups, Call Forwarding All Calls can only be activated by the attendant.

Calls to attendants cannot be forwarded. However, calls can be forwarded to the attendant group.

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 has 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.

- Personal Central Office Line (PCOL)

  PCOL calls cannot be forwarded.

- Station Message Detail Recording (SMDR)

  When a call is forwarded to an off-premises number, the call is recorded in SMDR records as a call from the forwarding station.

- SMDR Account Code Dialing

  If forced entry of account codes is required, calls cannot be forwarded to off-premises destinations.

Administration

Call Forwarding All Calls is assigned on a per-extension number basis by the Class of Service. The following items require administration by the System Manager:

- Voice Terminals
  - Class of Service
  - Call Forwarding Buttons
  - Redirection Notification
- Feature Access Codes for Activation and Deactivation of Call Forwarding All Calls

Hardware and Software Requirements

No additional hardware or software is required.

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. The call is parked on the user's extension number.
- A multi-appearance voice terminal user—Press the Transfer or Conference button, dial the Call Park access code or press the Call Park button (if assigned), 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.

A systemwide expiration interval can be set for parked calls. When the interval expires, the parked call will redirect to an attendant console and will no longer be parked on the extension number. However, if the parked call has already been retrieved when this interval expires, the call will not redirect to an attendant console. If two parties are connected on a parked call, a third party can also answer the call before the interval expires, creating a 3-way conference.

The attendant console group can have up to 10 common shared extension numbers used exclusively for Call Park. These extension numbers are not assigned to a voice terminal, but are stored in system translations and used to park a call. These extension numbers are particularly useful when one party is paged at the request of another party. The calling party is parked and the extension number is announced. Common shared 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.

Considerations

Call Park can be used whenever a voice terminal user who is on a call needs to go elsewhere and obtain information, and wishes to complete the call at another extension. Call Park also allows users to answer a call from any station after being paged by a voice terminal user or an attendant.

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 can be parked. The sixth conferee will be the retrieving party.

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

Interactions

- Bridged Call Appearance

  If a user, active on a bridged call appearance, activates Call Park, the call is parked on the primary extension associated with the bridged call appearance.

- 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.

Administration

Call Park is administered on a per-system basis by the System Manager. The following items require administration:

- Call Park access code
- Answer Back access code
- Call Park expiration interval (from 1 to 90 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)

Hardware and Software Requirements

No additional hardware or software is required.

CALL PICKUP

Description

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

Call Pickup groups are established so that when one member of a group is away other members of the group can answer that member's calls. A Call Pickup group usually consists of users who are located in the same area or have similar functions.

When a member of a Call Pickup group is away and receives an incoming call, any member of the Call Pickup group can answer the call. A member simply goes off-hook and dials the Call Pickup access code or presses a Call Pickup button. That group member is then connected to the calling party.

A Temporary Bridged Appearance is maintained at the called voice terminal. This allows the called party to bridge onto the call after it has been picked up by another member of the Call Pickup group.

Considerations

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

Up to 200 (V1) or 300 (V2) Call Pickup groups can be established. Each group can have up to 25 members. However, a voice terminal can be a member of only one Call Pickup group.

Interactions

- Automatic Callback and Ringback Queuing

  Callback calls cannot be answered by Call Pickup group members.

- Bridged Call Appearance

  Activating Call Pickup while on a bridged call appearance will pick up a call in the Call Pickup group of the bridged extension.

  If a voice terminal receives ringback tone on a bridged call appearance, the incoming call can be picked up by members of that voice terminal's Call Pickup group. This causes all bridged call appearances to be dropped. The call is parked on the primary extension number of the answering voice terminal.

- Hold

  A call, picked up and placed on hold at an extension, remains on that extension, even if the called party answers the call.

- 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.

Administration

Call Pickup is administered by the System Manager. The following items require administration:

- Call Pickup group number
- Members (extension number) of each Call Pickup group
- Call Pickup access code
- Call Pickup buttons

Hardware and Software Requirements

No additional hardware or software is required.

CALL WAITING TERMINATION

Description

Provides for calls to busy single-line voice terminals to wait and sends a distinctive call waiting tone to the called party.

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.

With V1 systems, the call in progress at the voice terminal cannot be placed on hold. It must be terminated. With V2 systems, the call in progress at the voice terminal can be placed on hold in order to answer the waiting call. After answering the waiting call, the voice terminal user can return to the held call or toggle back and forth between the two calls. The single-line voice terminal user can only be connected to one call at a time.

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.

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

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.

Considerations

With Call Waiting Termination, the party who calls a busy single-line voice terminal does not have to hang up and try the call again later. Instead, the call will wait at the called voice terminal until the called party hangs up on the current call.

Call Waiting Termination applies only to busy single-line voice terminals. Calls to multi-appearance voice terminals are routed to an idle call appearance and do not wait.

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

Administration

Call Waiting Termination is assigned on a per-voice terminal basis by the System Manager. The only administration required is the assignment of Call Waiting Termination to the desired voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

CENTRALIZED ATTENDANT SERVICE (V2)

Description

Allows services performed by attendants in a private network of switching systems to be concentrated at a central, or main, location. Each branch in a Centralized Attendant Service (CAS) has its own listed directory number (LDN). Incoming trunk calls to the branch, as well as attendant-seeking voice 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.

When all RLTs are busy, CAS attendant-seeking calls are placed in a CAS queue.

Backup service provides for all CAS calls to be sent to a backup extension in the local branch if all RLTs are maintenance busy or out of service. The backup extension can be assigned a button and associated status lamp to provide notification that backup service is in effect. The status lamp remains lighted as long as backup service is in effect.

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 CAS 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 handset.

- Incoming trunk call: 480 Hz (100 ms), 440 Hz (100 ms), 480 Hz (100 ms) in sequence; heard immediately after attendant lifts handset
- 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 handset
- Call extended to idle station or recall on don't answer: ringback tone for 300 ms followed by connection to normal ringing 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)

Considerations

CAS reduces the number of attendants required at each branch location. More efficient call handling is provided by letting one group of centralized attendants handle calls for the individual branches. For example, a chain of department stores can have a centralized attendant location at the main store. The centralized attendant can then handle calls for the individual stores.

Up to 16 RLTs can be assigned to a System 75 serving as a branch location.

A System 75 can be a branch to only one main location.

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.

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.

Interactions

- Call Coverage

  Calls are not redirected to a centralized attendant by Call Coverage. Calls to a CAS backup extension for backup service are not redirected to the backup extension's coverage path. Other calls to the backup extension, however, can still redirect to coverage.

- Call Forwarding

  Calls to a CAS backup extension are not forwarded and do not terminate at the backup extension.

- Night Service—Night Console Service

  Calls do not go to CAS attendants when Night Service has been activated at the branch.

Administration

CAS is administered by the System Manager. The following items require administration:

- 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

Requires a TN760B Tie Trunk circuit pack. The TN760B will serve all other tie trunk applications in addition to CAS. As an alternative, the DS1 Tie Trunk circuit pack (TN722) can be used for the release link trunks of the CAS network.

CAS software is required.

CLASS OF RESTRICTION

Description

Defines up to 64 different classes of call origination and termination privileges. Systems may have only a single COR, one with no restrictions, or 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)
- Individual Attendant Consoles
- 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
- Forced entry of account codes (V2)
- 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) or Automatic Alternate Routing (AAR) (V2) 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 3-1. The values shown on the form are the default values. However, these values can be changed to implement the desired restrictions. The Forced Entry of Account Codes field applies only to Version 2. This field does not appear on Version 1 forms.

```
                                                                Page 1 of 1
                              CLASS OF RESTRICTION

  COR Number: _                                                    FRL: 7

     APLT?: y                              Calling Party Restriction: none

                                            Called Party Restriction: none

                               (V2) Forced Entry of Account Codes? __

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 3-1. 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 certain calls with a 0 or 1 as the first or second digit, unless the called office code, area code, or service code is on an allowed calls list. This list can contain up to ten entries. (In areas where area codes can also serve as office codes, the system requires the prefix 1 on area code calls to differentiate them from local calls. In this case, local calls with a 0 or 1 as the second digit are not subject to toll restriction.)

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 3-1), 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, data modules, the attendant group, and individual attendant extensions (V2).

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 is 12, the storeroom voice terminal(s) is assigned COR 12.

Forced Entry of Account Codes (V2)

Account Codes are used to associate calling information with specific projects or account numbers. This is accomplished by dialing a specific account code before making an outgoing call. With V2, account code dialing can be optional or mandatory (forced) on a per-COR basis.

Looking at the screen form used to administer CORs (Figure 3-1), the field labeled Forced Entry of Account Codes is preset as "n." This means that account code dialing is optional. A "y" in the field would indicate that account code dialing is mandatory.

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 "n" (no) can use off-network calling. Users assigned a COR that has APLT set to "y" (yes) 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 3-1), the field labeled APLT is preset as "y." This means that a facility with this COR is not allowed to access CCSA or EPSCS off-network capabilities for public network calling. An "n" in this field would indicate that the facility can access CCSA or EPSCS off-network capabilities.

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

If the system does not use AAR or ARS to determine the most-preferred routing of calls, then the FRL field can be ignored. If AAR or 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 AAR/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 AAR/ARS Routing Pattern and the FRL in the COR of the call originator (typically, a voice terminal user).

The FRL field (see Figure 3-1) 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 (V1) or 254 (V2) 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 AAR and/or 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.

For a detailed description of AAR, ARS, and FRLs, refer to the individual feature descriptions given 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 in the following paragraphs.

The simplest way to understand miscellaneous restrictions is to look at the screen form used during implementation (see Figure 3-1). 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 3-1, no restrictions apply because all 64 CORs are specified as "y."

The V1 screen form in Figure 3-2 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.

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 3-2, 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.

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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)

    Ø? 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   1Ø? n   18? y   26? y   34? y   42? y   5Ø? y   58? y

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

    4? y   12? y   2Ø? y   28? y   36? y   44? y   52? y   6Ø? y

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

    6? y   14? y   22? y   3Ø? 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 3-2. Screen Form Used to Explain Miscellaneous Restriction Groups

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.

Class of Restriction Examples

The examples given here 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 central office calls only (no FX or WATS calls)
- COR 37—Voice terminals that can place in-house, local central office, 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 3-1) 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 nor a screening COR, it does 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 are 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 this 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 20—Manual Terminating Line Restriction.
- COR 21—Inward Restriction.
- COR 22—Toll Restriction.

- COR 23—Outward Restriction.

  Note: A new Manual Terminating Line Restriction for voice terminals was not established. COR 20, above, can be assigned.

- COR 24—Unrestricted.
- COR 25—COR for WATS.
- COR 26—COR for FX.
- COR 27—Provides Miscellaneous Restrictions for WATS and FX. Enter an "n" beside COR 25 and COR 26 on the form for COR 27.
- COR 28—Provides Miscellaneous Restriction for WATS. Enter an "n" beside COR 25 on the form for COR 28.

Now assign the appropriate COR to each physical or screening facility:

- Central office trunks—COR 24 (unrestricted)
- WATS—COR 25 (WATS COR)
- FX—COR 26 (FX COR)
- Attendant group—COR 24 (unrestricted)
- Voice terminals—COR 22 (toll), COR 23 (outward), COR 20 (manual), COR 24 (unrestricted), COR 27 (WATS and FX miscellaneous), or COR 28 (WATS miscellaneous), as required
- Data Modules—COR 24 (unrestricted)
- Terminating Extension Group—COR 21 (inward)
- Loudspeaker Paging trunks—COR 20 (manual)

This latter method is probably more difficult to use, but it 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.

Interactions

- ARS/AAR

  Originating FRLs are assigned via a COR.

- Bridged Call Appearance

  The Class of Restriction assigned to a voice terminal's primary extension also applies to calls originated from a bridged call appearance.

- 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.

- Forced Entry of Account Codes

  This feature can be assigned 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.)

Administration

Class of Restriction (COR) is administered by the System Manager. For each COR which is assigned, the following items must be administered:

- COR Number
- Facility Restriction Level (FRL)
- Permission to access EPSCS or CCSA off-net facilities
- Calling Party Restriction
- Called Party Restriction
- Permission to call other CORs
- Forced Entry of account codes for SMDR (yes or no)

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 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 3-2, 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 central office or FX 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. If the originating facility is specified as something other than "code" or "toll," and the trunk group is specified as "code" or "toll," the call will be neither toll nor code restricted. This paragraph is summarized below:

Originating Facility Restriction	Trunk Group Restriction	Restriction Applied On Call
toll	toll	toll
toll	code	toll
code	toll	toll
code	code	code
other	toll or code	none

Attendant Consoles (as a group) and Individual Attendant Extensions

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. Through Miscellaneous Restriction groups certain users are allowed access to certain facilities, while other users are denied access. For example (looking at Figure 2-2), 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).

Hardware and Software Requirements

No additional hardware or software is required.

CLASS OF SERVICE

Description

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 or individual attendant can or cannot activate the feature. Four features with two choices each yields 16 possible combinations. A Class of Service (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, Release 1 Version 1*, 555-200-650, and the *AT&T System 75 Implementation Manual, Release 1 Version 2*, 555-200-651. 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 and individual attendant 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.

Interactions

None.

Administration

COS is administered by the System Manager. The parameters for each COS can be changed. However, since the preassigned values include all possible combinations, it is not necessary to change these parameters. The only administration required is the assignment of a COS to each individual attendant (V2) and voice terminal.

Hardware and Software Requirements

No additional hardware or software is required.

CODE CALLING ACCESS

Description

Allows attendants, voice terminal users, and tie trunk users to page with coded chime signals.

As many as nine individual paging zones can be provided. (A zone is the location of the loudspeakers, for example, conference rooms, warehouses, etc.) In addition, one zone can be provided to activate all zones simultaneously. Each paging zone requires a separate Code Calling Access code.

A paging party dials the Code Calling Access code and the extension number assigned to the person to be paged. The paging party is automatically parked (through the Call Park feature) on the paged party's extension number. The system translates the number to a chime code and then plays the code over loudspeakers. The paged party, recognizing the chime code, can answer the call from any voice terminal within the system by dialing the Call Park Answer Back access code and his or her own extension number.

Considerations

With Code Calling Access, users do not have to be at their own voice terminal in order to answer calls. Users who are frequently away from their voice terminal or at a location where a ringing voice terminal might be disturbing can be assigned a chime code. When a user's chime code is heard, that user can answer the parked call from a nearby voice terminal.

The system can have up to nine individual zones plus one zone to activate all zones simultaneously.

As many as 125 3-digit chime codes can be provided. Only one extension number can be assigned to each chime code.

Interactions

- Call Park

  This feature is automatically provided with Code Calling Access.

- Conference—Attendant

  A call cannot be conferenced while accessing paging equipment. The attendant can, however, release the call after paging the called party.

- Conference—Terminal

  A call cannot be conferenced while accessing paging equipment.

- Controlled Restriction

  Controlled Total restriction prohibits use of Code Calling Access.

- Loudspeaker Paging Access

  It is not possible to use a PagePac* paging system for Code Calling Access when

* Trademark of Harris Corporation Dracon Division

multi-zone paging is desired. The PagePac systems expect a 2-digit code to access a particular zone. System 75, however, immediately plays the chime code once a connection is established.

- 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.

- Transfer

  A call cannot be transferred while accessing paging equipment.

Administration

Code Calling Access is administered by the System Manager. The following items can be administered:

- Trunk access code and Class of Restriction for each of the nine individual paging zones and for the zone used to activate all zones simultaneously.
- Number of times (1 to 3) the chime code will play.
- Loudspeaker locations (name of zone).
- 3-Digit chime codes for extensions. The codes are combinations of the digits 1 through 5.

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.

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

Whenever an attendant needs to talk with more than one party at the same time, the Attendant Conference feature can be used. An attendant can also establish a conference call for other voice terminal users or parties outside the system.

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.

The attendant cannot handle any other calls while setting up a conference call.

Interactions

- Bridged Call Appearance

  A Bridged Appearance button can be used to make conference calls.

- Trunk-to-Trunk Transfer

  If Trunk-to-Trunk Transfer is disabled and the attendant releases from a conference call involving only trunk conferees, the trunks are also disconnected.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

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

With the Conference—Terminal feature, voice terminal users can set up their own conference calls without assistance from an attendant.

With assistance from other users, a single-line voice terminal can have more than three parties on a conference call. For example, one user can add a party, who can add another party, and so on.

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.

Interactions

A Bridged Appearance button can be used to make conference calls.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

CONSULT

Description

Allows a covering user, after answering a coverage call, to call the principal (called party) for private consultation.

Consult is activated by first pressing the Conference or Transfer button followed by the Consult button to call the principal. This places the calling party on hold and establishes a connection between the principal and the covering user. The covering user can then add the calling party to the conversation, transfer the call to the principal, or return to the calling party.

Details of how Consult is used in conjunction with Call Coverage are given in the Call Coverage feature description, elsewhere in this section.

Considerations

Consult can be used to let a covering user consult with the principal, to determine whether he or she wishes to speak with the called party (for example, an executive's secretary may wish to consult the executive on an established call).

Interactions

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.

Bridged Call Appearances of the principal's extension are not alerted on a Consult call to the principal extension.

Consult is only used in conjunction with the Call Coverage feature.

Administration

Consult is administered by the System Manager on a per-voice terminal basis. The only administration required is the assignment of a Consult button.

Hardware and Software Requirements

No additional hardware or software is required.

COVERAGE CALLBACK

Description

Allows a covering user to leave a message for the principal (called party) to call the calling party.

Coverage Callback is activated by pressing the Cover Callback button after answering a coverage call.

Details of how Coverage Callback is used in conjunction with Call Coverage are given in the Call Coverage feature description, elsewhere in this section.

Considerations

Coverage Callback is useful whenever it is necessary to let the principal know that he or she has received a call from a certain party.

Interactions

Coverage Callback is only used in conjunction with the Call Coverage feature.

Administration

Coverage Callback is administered on a per-voice terminal basis by the System Manager. The only administration required is to assign a Cover Callback button.

Hardware and Software Requirements

No additional hardware or software is required.

COVERAGE INCOMING CALL IDENTIFICATION

Description

Allows multi-appearance voice terminal users without a display in a Coverage Answer Group to identify an incoming call to that group.

When an incoming call is directed to a Coverage Answer Group, the status lamp associated with the Coverage Answer Group button lights at group member's voice terminal.

Details of how Coverage Incoming Call Identification (ICI) is used in conjunction with Call Coverage are given in the Call Coverage feature description, elsewhere in this section.

Considerations

With Coverage ICI, members of Coverage Answer Groups do not have to have a display in order to identify incoming calls to the group.

A second coverage call takes control of the Coverage ICI lamp and does not return control to the previous call when the second call is released.

Interactions

Coverage ICI is used only in conjunction with the Call Coverage feature.

Administration

Coverage ICI is administered on a per-voice terminal basis by the System Manager. The only administration required is to assign a Coverage Answer Group button.

Hardware and Software Requirements

No additional hardware or software is required.

DATA CALL SETUP

Description

Provides three methods to set up a data call: Data Terminal (keyboard) Dialing, Voice Terminal Dialing, or dedicating a voice terminal for data calls. Typically, when a data terminal is available, keyboard dialing is more convenient and requires less steps; therefore, it should be used whenever possible. Data calls can be made at any of the industry standard data rates 19.2, 56, or 64 kbps.

In addition to data terminal dialing and voice terminal dialing, the system accepts calls from other devices, such as a Modular Processor Data Module (MPDM) equipped with an Automatic Calling Unit (ACU) interface module. An analog modem interfaced with an ACU can also be used to provide dialing capability for a host computer.

Voice Terminal Dialing

Voice terminal dialing must be used under the following conditions:

- The Data Terminal is connected to an analog modem.
- The Data Terminal is not accessible for 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 transferring 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 administration.

The voice terminal Data Extension buttons control the One-Button Transfer to Data, Return-to-Voice, and Data Call Preindication operations for the associated data module. These operations are discussed below. 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 module can be controlled from Data Extension buttons on up to four voice terminals.

Voice terminal dialing has the advantage that the user may hear the different types of network tones (particularly for V1).

For off-premises dialing, particularly for toll calls, the user may opt for voice terminal dialing, instead of keyboard dialing (in V1 it is the only way to detect tones, in V2 the user may prefer to hear than to see text).

The following options, either alone or combined, 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 established. Return of a data call to the voice terminal implies that the same (data) call will be continued in the voice mode, or transferred to another data endpoint.

- Data Call Preindication

  Allows the user, before dialing the distant data endpoint, to reserve the associated data module by pressing the Data Extension button. This insures that a conversion resource, if needed, and the data module are reserved for the call. Use of Data Call Preindication before one button transfer to data is recommended when establishing data calls that use toll network facilities. Needed conversion resources are reserved before any toll charges are incurred.

Data Terminal (Keyboard) Dialing

Allows a user to set up and disconnect data calls directly from a data terminal. A voice terminal is not needed. The voice terminal functions of switchhook and the audible call progress tones are replaced with keyboard dialing and text known as call progress messages. The message "DIAL:" prompts the user to enter the called data number manually from the keyboard, and "RINGING" informs the user the called data number is being rung. With Version 2 systems, if the data call is placed in queue, the message "WAIT, xx IN QUEUE" is received (xx represents queue position). This queue number is updated by the system as the call moves up in the queue. Table 3-B lists the call progress messages.

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 handset [call origination] or hanging up [call disconnect].) If the terminal being used does not generate a 2-second continuous break signal, the user can press the ORIGINATE/DISCONNECT button on the data module. Then, the data terminal allows the user to enter digits from the data terminal keyboard, after the message "DIAL:" (which is the equivalent of dial tone on a voice terminal).

In addition to the numeral, #, and * characters found on a touch-tone pad, the dialing information may contain the following special characters:

- SPACE, —, (, and #) may be used to improve legibility. These characters are ignored by the system during dialing.
- + (V2) character (wait) may be used to "interrupt" or "suspend" dialing when dial tone is expected from the distant switch. The + is ignored in V1.
- %, (V2) (pause) character may be used to place a 1.5-second pause in dialing. (multiple % can be used). The % should not be used in V1.
- $, (V2) (mark) character may be used to indicate the following digits are for end-to-end signaling (touch-tone). This is required when the trunk is rotary. It is not required when the trunk is "Touch-Tone." The $ should not be used in V1.
- UNDERLINE or BACKSPACE characters may be used to correct previously typed characters on the same line.
- @ character may be used to delete the entire line and start over with a new DIAL: prompt.

Each line of dialing information may contain up to 36 characters.

Examples of dialing are as follows:

- DIAL: 3478
- DIAL: 9+(201) 555-1212
- DIAL: 8% 555-2368
- DIAL: 9+555-2368+$9999+123 (remote access)

Single-Line Dialing

All of the dialing information, including pauses and ignored characters, are typed on a single line. The line with the DIAL: prompt must be complete; that is, the dialing information must specify a complete call before the carriage return or line feed.

Single-line dialing is recommended if all dialing information can be entered on one line.

Multiple-Line Dialing (Version 2)

Automatically invoked when a single line of dialing information is incomplete. Multiple-line dialing is used only with off-premises calling.

In multiple line dialing, the DIAL: prompt follows on the next line when all of the dialing information of the previous line has been sent and dial tone has occurred; additional dialing information is requested.

This is a typical off-premises dialing sequence:

- DIAL: 9
- DIAL: (201) 555-2368
- RINGING
- ANSWERED

TABLE 3-B. 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.
RINGING	Placing a call	Equivalent to ringing tone. Called terminal (far-end) is ringing.
BUSY	Placing a call	Equivalent to busy tone. Called number is in use, or out of service.
ANSWERED	Placing or receiving a call	Notifies calling and called users that call has been answered.
OUTGOING TRKS	Placing a call	Notifies calling user that outpulsing is complete.
ANSWERED - NOT DATA(V2)	Placing a call	Notifies calling and called users that call has been answered and a modem answer tone has not been detected.
TRY AGAIN	Placing a call	Equivalent to reorder tone. System facilities are currently not available.
DENIED	Placing a call	Equivalent to intercept tone. Call cannot be placed as dialed.
ABANDONED(V2)	Receiving a call	Notifies called user that the calling user abandoned the call.
NO TONE(V2)	Placing a call	Notifies user that tone was not detected.
CHECK OPTIONS	Placing a call	Notifies calling terminal that data module options are incompatible.
XX IN QUEUE(V2)	Call in queue	Current position of the user in queue. XX-indicates position.
PROCESSING*(V2)	Call in queue	Notifies user when out of queue. Facility is available.
TIMEOUT*(V2)	Call in queue	Notifies user when time has been exceeded. Call will be terminated.
FORWARDED*	Receiving a call	Equivalent to redirection notification signal. Called terminal has activated Call Forwarding and received a call, and call has been forwarded.
INCOMING CALL-*	Receiving a call	Equivalent to ringing.
PLEASE ANS-	Receiving a call	Originating voice terminal user has transferred call to data module using One-Button Transfer to Data.
-TRANSFER	Call is transferred to voice	Notifies calling terminal when Data Call Return-to-Voice occurs.

* Bell sounds when message is displayed.

TABLE 3-B. Call Progress Messages for Keyboard Dialing (Contd)

Displayed Message	Application	Meaning
CONFIRMED	Activating or deactivating a feature	Equivalent to confirmation tone. Feature request is accepted, or call has gone to a local coverage point.
-OTHER END(V2)	During a call	Notifies user that the other end terminated the call.
DISCONNECTED*	Call is terminated	Call or call attempt is disconnected from system
WAIT	Placing a call	Notifies user that normal processing is continuing.
WAIT, XX IN QUEUE(V2)	Placing a call	Notifies user that call entered a local hunt group queue. XX indicates position.

* Bell sounds when message is displayed.

Considerations

All systems have Data Call Setup capability. This facilitates data calling by eliminating the need to dedicate a voice terminal for data calls. Version 2 offers the enhancement of off-premises Multiple Line Dialing and additional call progress messages.

When a voice terminal user places a data call to a digital data endpoint, and does not transfer the call to another digital data endpoint but uses a modem or acoustically coupled modem, the user must dial the Data Origination access code assigned in the system before dialing the distant endpoint.

Data Call Preindication is activated by pressing a Data Extension button before dialing the distant data endpoint. Preindication is in effect until the associated Data Extension button is pressed again for a one-button transfer; there is no time-out.

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 using Data Call Return-to-Voice.

When placing outgoing or off-premises calls via keyboard dialing, the call progress message "WAIT" indicates recognition of the nature of the call and acceptance of the call. The "ANSWERED" text indicates completion of outpulsing over the selected trunk. Since no tone detection or analysis is done for these calls, no further messages are given to the user. The best success ratio (V1 only) for placing outgoing calls is achieved by placing the call from a voice terminal and using One-Button Transfer to Data.

Interactions

- Abbreviated Dialing

  This feature can be used by voice terminal or Data Terminal (Keyboard) Dialing users on calls to data endpoints.

- Data Call Hot Line

  Upon going off-hook for origination, the system automatically places a call to a predesignated local or off-premises destination.

- Call Forwarding All Calls

  Calls incoming to a data module can be forwarded. That is, calls can be redirected to another endpoint. This feature is activated using Data Terminal (Keyboard) Dialing. If the forwarded-to endpoint is an analog data endpoint, and the calling user is a digital endpoint, modem pooling is activated automatically.

- Modem Pooling

  This feature is automatically available on data calls when the system ascertains the need for a conversion resource. The system automatically inserts the conversion resource. Data Call Preindication (optionally) or Data Origination is used to indicate that conversion resource is needed.

- Uniform Call Distribution (UCD)

  UCD can provide a group of data modules or analog modems for answering calls to facilities, such as computer ports, connected to the data modules or modems.

- Information Systems Network (ISN) Interface

  ISN consists of packet data switches which support data calls between data endpoints and System 75. The physical connection to System 75 is via the Data Line Circuit (DLC) board. The DLC provides eight ports for connection with asynchronous Electronics Industries Association (EIA) RS-232C compatible Data Terminal Equipment.

  Data Terminal (Keyboard) Dialing is used to access ISN endpoints.

Administration

Data Call Setup does not require assignment as such; however, the following related items require administration by the System Manager:

- Data Origination Access Code—Allow users to indicate a need for a conversion resource on an analog to digital data call origination.
- Port Assignments—Assign the data modules, BCTs, DLCs, 7404D, analog modems.
- Modem Pooling—Assign Circuit Packs or ports.
- Data Extension buttons—Assign Data Extension buttons to multi-appearance voice terminals.

Hardware and Software Requirements

Data Call Setup is a means of using data equipment to establish data calls. Requirements for data modules, 510D terminals or 515 BCTs, and modems are given below.

- ***Data Modules:*** Each data module requires one port on a TN754 Digital Line circuit pack. [A Digital Terminal Data Module (DTDM) shares the port with the associated voice terminal.]
- ***510D Terminals or 515 BCTs:*** Each 510D terminal or 515 BCT requires one port on a TN754 Digital Line circuit pack for shared use of voice and data.
- ***7404D or 7407D:*** Each Voice Terminal requires one port on a TN754 Digital Line circuit pack for shared use of voice and data.
- ***Modems:*** Each modem requires one port on a TN742 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.)
- ***Modem Pooling:*** Version 1 requires a TN758 Modem Pool circuit board (two conversion resources per board).

  Version 2 requires either a TN758 Modem Pool circuit board or one digital port with a Trunk Data Module (either TDM/2 or MTDM), and one analog port with analog modem for each conversion resource.
- Keyboard Dialing to off-premises data endpoints requires the use of a TN748B Tone Detector circuit pack. Extensive use of features and services using tone detection may necessitate adding additional TN748B circuit packs (several other features also use a TN748B).

No additional software is required.

DATA HOT LINE (V2)

Description

Provides for automatic nondial placement of a data call to a digital data endpoint, upon receiver pickup.

Data Hot Line calls are automatically placed, by the system, from specified voice or data terminals to preassigned extension numbers or off-premises numbers. Hot Line calling endpoints are destinations that are associated with Data Communications Equipment (DCE), Data Terminal Equipment (DTE), or other devices such as, DTDM, PDM, 51X BCT, or a port of a Data Line Circuit (DLC). The destination number is stored in the Abbreviated Dialing List.

Considerations

Data Hot Line offers fast and accurate call placement to commonly called data endpoints. Voice and data terminal users that constantly call the same destination number can use Data Hot Line to automatically place the call by simply lifting the handset or going off hook.

The number of terminals that can be assigned Data Hot Line is not limited, and the number of terminals that can be assigned the same destination number is not limited. The only limit, if any, would be on the number of entries stored in the Abbreviated Dialing List.

Interactions

- Call Forwarding—All Calls

  A Hot Line originator cannot activate Call Forwarding, since an off-hoc ...ended to dial the Call Forwarding Feature Access Code will cause activation of the Data Call Hot Line feature instead.

- Data Terminal (Keyboard) Dialing (V2)

  Any Terminal Dialing text may occur when a Data Hot Line Call is being established, with the exception of the inhibition of the initial dial text message prompt normally given on off-hook for origination.

- System 75/ISN Access

  A data call to an ISN data endpoint from a System 75 digital data endpoint requires a 2-stage dialing. A Hot Line Destination may be an extension number for an outgoing ISN group only, the Hot Line originator must then interact with ISN and manually enter the second address (data endpoint).

Administration

Data Hot Line is administered on per-voice or data terminal basis by the System Manager. The following item requires administration:

- Hot Line Destination Number—Assign the destination number, for that station, in the Abbreviated Dialing List.

Hardware and Software Requirements

No additional hardware or software is required.

DATA-ONLY OFF-PREMISES EXTENSIONS

Description

Allows users to establish data calls involving data communications equipment (DCE) or Data Terminal Equipment (DTE) 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 (Modular) Trunk Data Module located on-premises. The communication with the remote data equipment is accomplished through the private line facility linking the on-premises (Modular) Trunk Data Module and the remote data equipment.

The Trunk Data Module and DCE or DTE constitute a digital data endpoint. Data calls to this type of data endpoint can be placed using Voice Terminal Dialing or Data Terminal (Keyboard) Dialing. Since there is no voice terminal at the remote site, 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.

Considerations

All systems have the capability for Data-Only Off-Premises Extensions to allow for data calls to remote DCE using DATAPHONE digital service or other private line data facilities.

Data-Only Off-Premises Extensions provides 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.

Interactions

- Voice Terminal Dialing

  An on-premises 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 voice terminal user and the remote data equipment user 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 on-premises voice terminal user can transfer a call to the Data-Only Off-Premises Extension. The Data Extension button on the voice terminal lights and 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, the voice terminal user may press the associated busy Data Extension button to transfer the call to the voice

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* Service mark of AT&T

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 user receives the BUSY message when attempting to originate a call.

Administration

Data-Only Off-Premises Extensions is assigned on a per-line basis by the System Manager. The following item requires administration:

- Digital Line Circuit Pack—Assign the associated data module to a vacant port.

Hardware and Software Requirements

Requires a Trunk Data Module and one port on a TN754 Digital Line circuit pack. No additional software is required.

DATA PRIVACY

Description

Protects analog data 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

All systems have the capability for Data Privacy to provide interruption protection by denying the system the ability to gain access to the protected, analog data, call.

Connections involving one or more digital data endpoints (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 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.

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.

- Busy Verification cannot be done when data privacy is active.

Administration

Data Privacy is activated by a system code, administered by the System Manager. The feature is assigned on a per class-of-service basis.

Hardware and Software Requirements

No additional hardware or software is required.

DATA RESTRICTION

Description

Protects analog data 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

All systems have the capability for Data Restriction to prevent overriding or ringing features from interrupting the voice or data call.

Connections involving one or more digital data endpoints (data modules) are automatically protected from receiving system-generated tones. In this case, the Data Restriction feature is not needed.

Data Restriction applies to both voice and data calls. Also, Data Restriction cannot be assigned to attendant consoles.

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.

- Busy Verification cannot be done when Data Restriction is assigned.

Administration

Data Restriction is assigned on a per-line or trunk basis by the System Manager.

Hardware and Software Requirements

No additional hardware or software is required.

DCS ALPHANUMERIC DISPLAY FOR TERMINALS (V2)

Description

Allows calls to or from terminals equipped with alphanumeric displays to have transparency with respect to the display of call-related information.

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 node to a terminal on another node. Transparency in this area is limited by the type of systems at the endpoint nodes and at the intermediate node, if any.

Considerations

DCS Alphanumeric Display for Terminals gives the user considerable call handling capabilities by displaying call related information on calls to and from other DCS nodes.

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 tandem node is a System 85 Release 2 Version 2 or later.
- The other party is at a System 85 Release 2 Version 2 or later.
- The call is not routed through a tandem System 85 Release 2 Version 1 or Enhanced DIMENSION PBX node. (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 distant node. The called name display only works between System 75s.

Within the same System 75 node in a DCS, complete transparency of Calling and Called Name Display exists.

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 or later as an intermediate node
- A System 75 connected to a System 85 Release 2 Version 2 or later

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 nodes 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 *AT&T System 75 User's Guide, Voice Terminal Operations*, 555-200-701, 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 node 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 node 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 node 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 node 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.

- Conference

  When a DCS call is conferenced either at a remote node 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 node to a user on any node, all DCS Calling and Called Name Display transparency is lost to users on the local System 75.

Administration

DCS tie trunk groups between nodes must be administered by the System Manager with the Outgoing Display disabled. This enables the called party's name to be displayed at the calling terminal.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required.

DCS ATTENDANT CONTROL OF TRUNK GROUP ACCESS (V2)

Description

Allows an attendant at any node in the DCS to exercise control over an outgoing trunk group at a different node in the cluster.

Each attendant console has 12 Trunk Group Select buttons to be used with the Attendant Direct Trunk Group selection feature. Each button allows the attendant direct access to an outgoing trunk group by merely pressing the button assigned to that trunk group. Each of the 12 buttons has a Busy lamp which lights when all trunks in the associated trunk group are busy. Six of these buttons have two additional lamps and are used for Attendant Control of Trunk Group Access. The two additional lamps are as follows:

- Warn (warning) lamp

  Lights when a preset number of trunks are busy in the associated trunk group.

- Cont (control) lamp

  Lights when the attendant activates Attendant Control of Trunk Group Access for the associated trunk group.

Attendant control of a remote trunk group in the DCS network is activated by pressing the Cont Act button followed by the desired Remote Trunk Group Select button. Then the initiating node sends a message to the remote node where the trunk group to be controlled resides. The message indicates that control of that trunk group has been initiated.

When the remote node receives the control activation message from the initiating node, it has 4 seconds to send a reply message back to the initiating node if control of the remote trunk group can be activated. A confirmation message will be sent to the initiating node and the Cont lamp at the corresponding Trunk Group Select button is lighted at the remote node if control of the remote trunk group can be activated. An error message is sent to the attendant at the initiating node if the trunk access code is invalid, if the trunk group is already controlled, or if the remote node is a System 85 or Enhanced DIMENSION PBX and the attendant does not have a Trunk Group Select button with Cont lamp for that trunk group.

When a trunk group is controlled in a DCS environment, calls to the trunk group by anyone other than an attendant are routed to the local attendant at the node where the trunk group resides. If that node does not have an attendant, the call is routed to a Centralized Attendant Service (CAS) main attendant or an attendant at a location arranged for Inter-PBX Attendant Calls. However, if CAS or the Inter-PBX Attendant Calls feature is not provided, the party attempting to call on the controlled trunk receives intercept tone.

A detailed description of the CAS and Inter-PBX Attendant Calls are given elsewhere in this section.

Considerations

DCS Attendant Control of Trunk Group Access allows attendants to obtain control of access to specific trunk groups at any node in the DCS network. This allows the attendant to monitor the use of the controlled trunk group.

There must be direct DCS tie trunk connections between the initiating node and the remote node where the trunk group to be controlled originates. Otherwise, control of remote trunk groups is denied.

If the remote node (where the trunk group to be controlled resides) is a System 75, it is not necessary for that node to have an attendant console with corresponding 3-lamp Trunk Group Select button. However, if the remote node is a System 85 or Enhanced DIMENSION PBX, control of the trunk group is not allowed unless an attendant at that node has a corresponding 3-lamp Trunk Group Select button.

The attendant must use the Remote Trunk Group Select button to directly access the controlled remote trunk group. If an attendant controls a remote trunk group, and that attendant dials the trunk access codes of the DCS tie trunk and the controlled remote trunk group, the call is routed to the attendant at the node where the trunk group resides.

Interactions

- DCS Attendant Display

  When a user attempts to access a controlled trunk group and is routed to the local attendant, the display shows the reason the call was redirected. If the call is routed via CAS or the Inter-PBX Attendant Calls feature, the display does not show the reason the call was redirected.

- Uniform Dial Plan

  DCS tie trunks should not be attendant controlled. This would result in all Uniform Dial Plan calls on the controlled tie trunk being routed to the controlling attendant instead of to the desired destination.

Administration

The ability of an attendant to control access to a remote trunk group is dependent on the administration by the System Manager of Trunk Group Select buttons for remote trunk groups in the DCS.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required.

DCS ATTENDANT DIRECT TRUNK GROUP SELECTION (V2)

Description

Allows attendants at one node to have direct access to an idle outgoing trunk at a different node in the DCS.

A Trunk Group Select button can be assigned to access a trunk group at the local node or a trunk group at a remote node. A Trunk Group Select button assigned to access a remote node is referred to as a remote Trunk Group Select button. Pressing a remote Trunk Group Select button has the same affect as dialing the tie trunk group access code for the remote node and the trunk access code of the selected trunk.

DCS Attendant Direct Trunk Group Selection functions the same as the regular Direct Trunk Group Selection feature (fully described elsewhere in this section). The only difference is an attendant can access a trunk group at a remote node.

Considerations

With DCS Attendant Direct Trunk Group Selection, an attendant can have faster access to trunk groups at remote nodes. There is no need to look up trunk access codes, because the press of a button connects the attendant to the desired trunk group.

There must be a direct DCS tie trunk connection between the initializing node and the remote node where the trunk group to be accessed originates. Otherwise, access to the remote trunk group is denied.

Interactions

None.

Administration

A remote Trunk Group Select button must be assigned both the tie trunk access code to the remote node and the trunk access code of the remote trunk group to be selected. These assignments are made by the System Manager.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required.

DCS ATTENDANT DISPLAY (V2)

Description

Provides some transparency with respect to the display of call-related information.

Calls to and from a System 75 in a DCS environment have Calling/Called Party Identification transparency under the following conditions:

- The other party is at another System 75 and the intermediate node is a System 85 Release 2 Version 2 or later.
- The other party is at a System 85 Release 2 Version 2 or later.
- The call is not routed through an intermediate System 85 Release 2 Version 1 or Enhanced DIMENSION PBX node. (Such calls will display only the extension number of the calling or called party.)

A detailed description of the Attendant Display feature is given elsewhere in this section.

Considerations

DCS Attendant Display gives the attendant considerable call handling capabilities by displaying call related information on calls to and from both local and remote nodes. This detailed information can be very useful in processing calls.

System 75 Classes or Restriction (CORs) may not correspond to those used by an Enhanced DIMENSION PBX or System 85. Therefore, if the DCS network contains nodes other than System 75s, the display CORs may be misinterpreted. If it is important that certain CORs between various systems correspond with each other, those CORs should be administered accordingly.

On outgoing calls, the display of called party information may be delayed a few seconds until the required information arrives from the remote node. The called party information is displayed only if both nodes are System 75s.

DCS tie trunks between nodes must be administered with the Outgoing Display disabled. This enables the called party's name to be displayed at the calling attendant's display.

Interactions

None.

Administration

The administration required for DCS Attendant Display is the same as that required for the Attendant Display feature. This information is given elsewhere in this section.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required.

DCS AUTOMATIC CALLBACK (V2)

Description

Allows a user at one node to make an automatic callback call to a user at another node in the DCS.

A callback call can be activated from a voice terminal at one node to a voice terminal at another node the same way as if at a local node, with the following exceptions:

- If the called party is at a System 85 or Enhanced DIMENSION PBX node, the callback call can only be activated if the called node is returning busy tone or special audible ringback.
- If the called party is at a System 75 node, the callback call can be activated if the called node is returning busy tone, Call Waiting ringback tone, or ringback tone.
- The calling party must disconnect within 6 seconds after hearing the confirmation tone for Automatic Callback activation.

The callback of the calling or called parties is as follows when a callback call has been made to a user at another node:

- When the calling party answers the callback call, and no tie trunk to the called party's node is available, Automatic Callback is reactivated toward the called party. The calling party hears confirmation tone instead of ringback when this occurs.
- If the calling party is on a System 85 or Enhanced DIMENSION PBX node and is unable to receive the callback call (for example, a busy single-line voice terminal without Call Waiting), Automatic Callback is reactivated by the calling party's node. If the calling party is on a System 75 node and is unable to receive the callback call, the callback call is canceled.
- If the calling party is on a System 85 or Enhanced DIMENSION PBX node, the callback call will be canceled if the calling party calls the called party, or vice versa. If the calling party is on a System 75 node, the callback call is not canceled when one party calls the other.

A detailed description of the Automatic Callback feature is given elsewhere in this section.

Considerations

DCS Automatic Callback eliminates the need for voice terminal users to continuously redial busy or unanswered calls to voice terminals within the DCS network.

An Automatic Callback request is canceled automatically if the called party does not become available within 40 minutes, or if the calling party does not hang up within 6 seconds after activating Automatic Callback.

DCS Automatic Callback does not work on the last trunk between nodes. Thus, if "*n*" trunks are provided, there can be up to "*n - 1*" Automatic Callback calls.

Interactions

- Attendant Control of Trunk Group Access and DCS Attendant Control of Trunk Group Access

  Automatic Callback cannot be activated if the call uses a controlled trunk group.

- Call Forwarding All Calls and DCS Call Forwarding All Calls

  Automatic Callback call cannot be activated toward a voice terminal at a System 75 node that has Call Forwarding activated.

Administration

The administration required for DCS Automatic Callback is the same as that required for the Automatic Callback feature. This information is given elsewhere in this section.

Hardware and Software Requirements

AP/DCS interface and DCS software are required.

DCS AUTOMATIC CIRCUIT ASSURANCE (V2)

Description

Allows a voice terminal user or attendant at a System 75 endpoint node to activate or deactivate Automatic Circuit Assurance (ACA) referral calls for the entire DCS network. This transparency also allows the referral calls to be generated at a node other than the node that detects the trunk problem.

If referral calls are generated at a System 75 node for one or more remote nodes, the remote nodes are notified when ACA referral is activated or deactivated at the System 75 node.

If referral calls are generated at a remote node for a System 75 node, the System 75 node is notified when ACA referral is activated or deactivated at the remote node. This notification is accomplished via the ACA button located on the attendant console or voice terminal at the System 75 node. The lamp associated with the ACA button lights when ACA referral is activated and goes dark when ACA referral is deactivated. The ACA button serves no other purpose when a remote node generates the System 75 referral calls.

A detailed description of the ACA feature is given elsewhere in this section.

Considerations

The DCS Automatic Circuit Assurance feature provides better service through early detection of faulty trunks and consequently reduces out-of-service time.

Interactions

None.

Administration

DCS Automatic Circuit Assurance requires that the System Manager administer whether ACA referral calls are to be "local," "remote," or "primary":

- If administered as local, referral calls are generated at the System 75 node for the System 75 node.
- If administered as remote, referral calls are generated at a remote node for the System 75 node. In this case, the remote node identification must also be entered.
- If administered as primary, referral calls are made at the System 75 node for a remote node.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required.

DCS BUSY VERIFICATION OF TERMINALS AND TRUNKS (V2)

Description

Allows attendants and multi-appearance voice terminal users to make test calls to voice terminals and trunk groups that are located at other nodes within the DCS.

Attendants and multi-appearance voice terminal users can busy verify voice terminals at a remote location. This is done by first pressing the Verify button and then entering the desired Uniform Dial Plan extension number. The verification then continues the same as if the voice terminal being verified is on the same node.

Multi-appearance voice terminal users can busy verify a trunk at a remote location. This is done by first pressing the Verify button, then dialing the trunk access code of the tie trunk group to the remote node, pressing the Verify button a second time, and then entering the desired trunk access code and the trunk group member number to be verified. The verification of the trunk then continues as if the trunk being verified is on the same node.

Attendant operation is the same except a Trunk Group Select button can be used to access the tie trunk to the remote node. A detailed description of the Busy Verification of Terminals and Trunks feature is given elsewhere in this section.

Considerations

DCS Busy Verification of Terminals and Trunks provides an easy method of checking the working condition of extensions and trunks at remote nodes.

Interactions

If the Trunk Identification by Attendant feature is used during busy verification of a trunk (Trunk ID button is pressed), the trunk access code and trunk group member number of the DCS tie trunk being used is displayed.

Administration

The administration for DCS Busy Verification of Terminals and Trunks is the same as that for the Busy Verification of Terminals and Trunks feature, which is fully described elsewhere in this section.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required.

DCS CALL FORWARDING ALL CALLS (V2)

Description

Allows all calls to an extension number to be forwarded to a selected extension number within the DCS network, or to an external (off-premises) number. This feature is activated or deactivated by dial access code or by a Call Forwarding button. The feature can be activated or deactivated only by voice terminal users within the DCS.

Activation and deactivation of the feature is the same as described for the Call Forwarding All Calls feature, described in detail elsewhere in this section.

Considerations

With DCS Call Forwarding All Calls, voice terminal users can have their calls follow them to any location within the DCS network or outside the DCS network.

Calls to an attendant cannot be forwarded. However, an attendant can activate or deactivate the feature for other extension numbers within the DCS.

Interactions

If the forwarding extension and the designated extension are at different nodes, and the designated extension's coverage criteria are met on a forwarded call, the call is redirected to a point in the designated extension's coverage path.

If the forwarding extension and the designated extension are at different nodes, Leave Word Calling and Coverage Callback cannot be activated at the designated extension for a forwarded call.

Administration

The administration for DCS Call Forwarding All Calls is the same as that for the Call Forwarding All Calls feature, which is fully described elsewhere in this section.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required.

DCS CALL WAITING (V2)

Description

Allows calls from one node to busy single-line voice terminals at another node to wait until the called party is available to accept the call.

DCS Call Waiting includes the following features:

- Attendant Call Waiting
- Call Waiting—Origination (not a System 75 feature)
- Call Waiting—Termination
- Priority Calling

Attendant Call Waiting, Call Waiting Termination, and Priority Calling function the same between System 75 nodes in a DCS as they do from a single System 75. These features are fully described elsewhere in this section.

Call Waiting—Origination is not a feature of System 75, but is supported in a DCS for calls into a System 75 from nodes that do provide it. When activated before a call, this feature rings an idle single-line terminal with 3-burst priority ringing. If the called party is busy, the call waits and the busy party hears 3-burst call waiting tone through the handset. The waiting party hears Call Waiting ringback tone.

Call Waiting—Origination can also be activated *after* the caller receives busy tone. After activation, the call waits, the busy party hears 3-burst call waiting tone through the handset, and busy tone changes to Call Waiting ringback tone.

Considerations

With DCS Call Waiting, a System 75 single-line voice terminal user, by knowing a call is waiting, can quickly process calls from locations within the DCS.

Call Waiting—Origination can only be received in System 75, not activated.

Interactions

DCS 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

Administration

None required.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required.

DCS DISTINCTIVE RINGING (V2)

Description

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.

A detailed description of the Distinctive Ringing feature is given elsewhere in this section.

Considerations

DCS Distinctive Ringing allows the user to identify the type of incoming calls. By knowing the type of incoming call, the user can answer each call properly.

When DCS transparency is lost for any reason, terminal-to-terminal calls made between nodes produce 2-burst ringing instead of the usual 1-burst ringing. Loss of transparency may occur when the data link between nodes is down or when data transmission delay exceeds the trunk signaling time.

Interactions

- Intercom—Automatic

  This feature and its distinctive ringing are not provided between nodes in a DCS.

- Intercom—Dial

  This feature and its distinctive ringing are not provided between nodes in a DCS.

- Manual Signaling

  This feature and its distinctive ringing are not provided between nodes 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 ringing. Calls from external tie trunk groups are treated as externally originated calls and receive 2-burst ringing.

Administration

None required.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required.

DCS LEAVE WORD CALLING (V2)

Description

Enables System 75 terminal users to leave preprogrammed "call me" messages at other terminals within the DCS network. Messages can be left by calling, called, or covering users.

Leave Word Calling (LWC) transparency in a DCS configuration allows messages from a System 75 to another node, depending on the storage capability of the remote node.

Considerations

DCS LWC lets users within the DCS leave short, simple messages for other users.

Both System 75 and System 85 Release 2 Version 2 can store LWC messages internally. An Applications Processor (AP) is not required. However, both System 85 Release 2 Version 1 and Enhanced DIMENSION PBX must be connected to an AP in order to store LWC messages.

LWC cannot be successfully activated toward any system that is not capable of storing the messages, either internally or in an associated AP.

Messages from one node, through an intermediate node, to a remote node do not require storage capability at the intermediate node.

The following configurations have LWC transparency in a DCS:

- From System 75 to System 85 Release 2 Version 2
- From System 75 through any intermediate node to another System 75 or a System 85 Release 2 Version 2
- To System 75 from any other node

Retrieval of LWC messages is permitted only from a terminal at the node where the messages are stored.

Interactions

- DCS Multi-appearance Conference/Transfer (V2)

  Activation of LWC is denied after a DCS call has been conferenced or transferred.

- DCS Call Forwarding All Calls (V2)

  If the forwarding extension and the designated extension are at different nodes, Leave Word Calling cannot be activated at the designated extension for a forwarded call.

Administration

The administration for DCS LWC is the same as that for LWC, which is fully described elsewhere in this section.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required.

DCS MULTI-APPEARANCE CONFERENCE/TRANSFER (V2)

Description

Provides transparency of conference calls and the transfer of calls within a DCS network. A user in the DCS can make conference calls or transfer calls originated from any extension in the DCS network to another extension within the DCS.

In a DCS, if a party in a conference hangs up or completes a transfer leaving only outgoing trunks on the call, an attempt is made to preserve the connection if any of the remaining parties on the call is a DCS tie trunk. This can be accomplished if the DCS tie trunk can signal the remote node when the party hangs up. The remote node sends a reply to the originating node, and disconnect supervision is provided for that trunk.

Conference Calls can be placed and calls can be transferred to users within the DCS by dialing the Uniform Dial Plan extension number.

A detailed description of the Conference—Attendant, Conference—Terminal, and Transfer features is given elsewhere in this section.

Considerations

DCS Multi-Appearance Conference/Transfer is useful when it is necessary to talk to more than one party at one time within a DCS. Multi-appearance voice terminals must have an idle appearance in order to transfer a call.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required.

DCS TRUNK GROUP BUSY/WARNING INDICATION (V2)

Description

Provides attendants with a visual warning that the number of busy trunks in a remote group has reached an administered level. A visual indication is also provided when all trunks in a trunk group are busy.

If an attendant has a Trunk Group Select button assigned to a remote trunk group, the button's Busy lamp lights when all trunks in the trunk group are busy. The lamp goes dark when one of the busy trunks becomes available.

If an attendant has a 3-lamp Trunk Group Select button assigned to a remote trunk group, the button's Warn lamp lights when the number of busy trunks in the trunk group reaches the Busy Warning Threshold. The lamp goes dark when the number of busy trunks in the trunk group falls below the Busy Warning Threshold.

To insure that the busy, warning, and control status of all Trunk Group Select buttons in the DCS remain consistent with the status of the corresponding trunk groups, some nodes in the DCS broadcast the status of a different local trunk group, every 50 seconds, to all directly connected nodes. For example, a node with 30 trunk groups would take 1500 (50 x 30) seconds to broadcast the status of all 30 trunk groups. This is called a lamp audit. When a node receives a lamp audit message, its lamps are updated accordingly. As a traffic consideration, a System 75 node only receives lamp audit messages. It does not broadcast lamp audit messages.

Considerations

Trunk Group Busy and Trunk Group Warning Indication is particularly useful with the Attendant Control of Trunk Group Access feature. The indicators alert the attendant when control of access to local and remote trunk groups is necessary.

Interactions

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 at the remote location as well as at the local node.

Administration

Administration for DCS Trunk Group Busy/Warning Indication is the same as that for the Trunk Group Busy/Warning Indicator feature, which is fully described elsewhere in this section.

Hardware and Software Requirements

AP/DCS interface hardware and DCS software are required.

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

With Dial Access to Attendant, voice terminal users can dial 0 to access an attendant whenever attendant aid is needed.

A voice terminal user calling the attendant by dial access cannot be added to an existing conference by the attendant.

Interactions

Restriction—Origination (administered to a voice terminal by the Class of Restriction) prohibits placing any calls, including Dial Access to Attendant calls.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

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 with Version 1 systems. Version 2 systems can have 2-, 3-, 4-, or 5-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 can range from 400 to 499. Also, if a 4-digit number with a 6 as the first digit is administered, the extension numbers can range from 6000 to 6999.

- Attendant

  Dial access to the attendant group is always by the single digit "0." Version 2 provides for Individual Attendant Access (V2) by assigning each attendant an individual extension number.

- 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 Wide Area Telecommunications Service (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.

With Version 2 systems, a Uniform Dial Plan may also be established during administration as part of the Dial Plan. This plan provides a common extension number plan that can be shared among a group of switches. If a Uniform Dial Plan is to be established, all extension numbers must be the same length (4 to 5 digits). A Uniform Dial Plan also requires the following information, so that calls will route to the desired switch.

- A PBX Code, which represents the first 2 digits of a 4- or 5-digit extension number and can range from 10 to 99 for a maximum of 90 PBX Codes.
- Whether or not the PBX Code is local to this System 75—this information is required for each PBX Code.
- An RNX, which is associated with the PBX code and is used to select an Automatic Alternate Routing (AAR) pattern for the call—this information is required for each PBX code.
- A PBX ID (1 to 63), which represents a specific switch—this information is required for each PBX Code when the switch is located within a Distributed Communications System (DCS).

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 #.

Interactions

All dial access features and services provided by the system require the Dial Plan.

Administration

The Dial Plan is administered on a per-system basis by the System Manager. The following items require administration:

- Area code where system 75 is located
- Whether or not the serving central office requires the digit 1 to indicate a long-distance call
- Whether or not a Uniform Dial Plan is to be established (V2)
- The type of code and the number of digits in the code for each first digit

If a Uniform Dial Plan is to be established, the following items must also be administered:

- Number of digits in plan (4 or 5)
- PBX Codes
- Whether or not each PBX Code is local to the PBX being administered
- RNX (Per PBX Code)
- PBX ID (Per PBX Code)

Hardware and Software Requirements

No additional hardware or software is required.

DIGITAL MULTIPLEXED INTERFACE (V2)

Description

Supports two signaling techniques: Bit Oriented Signaling and Message Oriented Signaling for direct connection to host computers.

System 75 supports only the Bit Oriented Signaling version of the Digital Multiplexed Interface (DMI) for multiplexed data communication over Data Services Level 1 (DS1) Tie Trunk Service digital facilities between a host computer and System 75.

In the United States and Japan, the DMI provides twenty-three 64-kbps data channels, plus one 64-kbps channel for Common Channel Signaling. In Europe, the DMI provides thirty 64-kbps data channels, plus one 64-kbps channel for Common Channel Signaling. Within the data channel, DMI provides control information exchange and data formats supporting data transport at all standard data rates; each data channel can be used in one of the following transfer modes:

- Mode 0 — 64 kbps Clear Channel
- Mode 1 — 56 kbps DDS Compatible
- Mode 2 — 0-19.2 kbps Synchronous/Asynchronous
- Mode 3 — Multiple Virtual Channels

The signaling mode of the TN722 DS1 circuit pack must be optioned, via the System Access Terminal (SAT), for Common Channel Signaling. In the United States and Japan, the format used is a 1.544-Mbps digital signal that consists of a 1.536-Mbps signal multiplexed with an 8-kbps framing channel. The 1.536-Mbps signal is divided into 24 channels of 64 kbps each. In Europe, the DMI operates at the rate of 2.048 Mbps.

DMI trunks are accessed the same as tie trunks. The only difference is that DMI trunks are connected to host computers while tie trunks are connected to another switch. Each trunk functions like a Processor Data Module (PDM) since the DMI protocol is identical to the Digital Communications Protocol (DCP) format used by the data modules.

Considerations

System 75 DMI support offers high volume (high speed, high capacity) data transmission, via DS1 digital facilities, between host computers and analog or digital data endpoints.

DMI is widely supported. To date, more than 40 data processing suppliers, communications equipment suppliers, and device manufacturers have licensed DMI specifications and have obtained the rights to implement DMI in their products.

DMI trunks can only be connected to host computers. Also, queuing for DMI trunks is not provided.

Interactions

- Data Restriction

  DMI trunks cannot be data restricted.

- Modem Pooling

  Data calls dialed from a local analog data endpoint to a DMI trunk must contain the Data Origination Access Code to obtain a conversion resource. Data calls on DMI trunks to local analog data endpoints will automatically obtain conversion resources, if available.

Administration

DMI support is assigned on a per-system basis by the System Manager. The following items require administration.

- DS1 Circuit Pack—Assign the circuit pack to the system before administration of the associated trunks.
- DMI Trunk Group—Associate the trunks to the groups.

Hardware and Software Requirements

One TN722 DS1 circuit pack is required for every 23 DMI trunks required.

No additional software is required.

DIRECT DEPARTMENT CALLING AND 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.

A Direct Department Calling (DDC) or Uniform Call Distribution (UCD) answering group can consist of voice terminals and individual attendants. In addition, a UCD group can consist of data modules, data line circuit ports, or modems.

One extension number is assigned to all voice terminals, individual attendants (V2), data modules, data line circuit ports, or modems in a group or department, that is, to a set that serves the same function and requires call distribution among the members of the group. Incoming calls to a Direct Department Calling (DDC) group or Uniform Call Distribution (UCD) group can be internal or external.

The hunting algorithm used by the system to select an idle terminal or console is the only difference between DDC and UCD.

With DDC, an incoming call rings the first available voice terminal or individual attendant (V2) in the administered sequence. If the first group member in the sequence is active on a call (busy), or has had his or her calls temporarily redirected (via Send All Calls, Call Forwarding All Calls, or the Make Busy Function discussed later), the call routes to the next group member, and so on. In other words, incoming calls always try to complete at the first group member in the administered sequence. Therefore, the calls are not evenly distributed among the DDC group members.

With UCD, an incoming call will ring the member of the group that has not received a UCD group call for the longest period of time (the most-idle member). In other words, incoming calls to a UCD group extension number will be distributed evenly among the group members.

When DDC or UCD 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 or UCD 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.

Any voice terminal or individual attendant (V2) can be a member of one or more DDC and/or UCD groups. Data modules, data line circuit ports, and modems are limited to UCD groups and can be a member of one or more groups. Each member of a group also has its own unique extension number and can be called individually. Multi-appearance voice terminals and attendant consoles (V2) can have an assigned status lamp that identifies an incoming DDC or UCD call. However, the voice terminal or individual attendant (V2) must be idle (not active on any call appearance) before a group call will be directed to the terminal or console (V2). Therefore, a voice terminal can receive only one DDC or UCD call at a time.

A queue can be established for a DDC or UCD group. When all members of 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 (1 to 6 ringing cycles) may also begin when the call enters the DDC or 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. Once a call is connected to a delay announcement, it remains in queue until a group member becomes 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. When the call begins ringing a member of the hunt group, the calling party hears audible ringing.

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 (voice terminals or individual attendants only) 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 or 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 there is an incoming call to a DDC or 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 DDC or UCD group members are busy.
- The call is the fifth call in the queue.

Since all members in the DDC or 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 DDC or 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 DDC or UCD group member becomes idle, the longest queued call is directed to that group member. 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 and UCD are 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.

If DDC and UCD 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.

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.

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 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 or UCD call at a time. A voice terminal is idle for a DDC or UCD call only if all call appearances are idle.

A Make Busy option can be administered for the system. When a voice terminal user or individual attendant (V2) dials the Make Busy activation code followed by the DDC or UCD group extension number, or presses the Make Busy button, the terminal or console (V2) appears busy to the DDC or UCD group. This effectively removes the member 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 or UCD 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 or UCD 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 or console (V2), 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 or UCD group and can be retrieved by a member of the DDC or 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 DDC or UCD group. The status lamp associated with this button informs the user that at least one message has been left for the group.

Members of a UCD group used for data communications must be of the same type and serve the same function. Either data modules or analog 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 data UCD group can be used on a given call, option settings must be the same for all group members. This minimizes call setup failures because of incompatible options between the origination 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. Individual data modules or modems can originate and receive calls.

Each UCD group and each individual UCD member is assigned a Class of Restriction (COR). Miscellaneous Restrictions, described in this section, 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.

Interactions

- Attendant Call Waiting

  An attendant can originate or extend a call to a DDC or UCD group. Attendant Call Waiting cannot be used on such calls. However, such calls can enter the group queue, if provided.

- Automatic Callback

  Automatic Callback calls cannot be activated toward a DDC or UCD group.

- Call Coverage

  Calls can redirect to or from a DDC or UCD group.

  If a user has a Make Busy button, and activates or deactivates Send All Calls, the Make Busy function associated with DDC or UCD is activated or deactivated simultaneously.

  If a user has no Make Busy button, activating or deactivating Send All Calls still makes the user available or unavailable for DDC or UCD calls, but Make Busy is not activated or deactivated. The Make Busy activate or deactivate code and the DDC or UCD extension must be dialed to activate the Make Busy function.

  Activating or deactivating the Make Busy function does not activate or deactivate Send All Calls.

  For a call to a DDC or 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 DDC or UCD group.

- Data Call Setup (to or from a member of a UCD group)

  Voice Terminal Dialing of Data Terminal (Keyboard) Dialing can be used on calls to a UCD group.

- Direct Inward Dialing

  If DID is provided and the DDC or 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.

- Individual Attendant Access (V2)

  Individual Attendant Extensions can be assigned to DDC and UCD groups. Unlike voice terminal users, individual attendants can answer DDC and UCD calls as long as there is an idle call appearance and no other DDC or UCD call is on the console.

- Multi-Appearance Preselection and Preference

  All assigned call appearances must be idle before a DDC or UCD group call is directed to a voice terminal.

- Music-on-Hold Access

  A call placed in a DDC or UCD group queue can receive a delay announcement followed by music.

- Priority Calling

  A priority call directed to a DDC or UCD group is treated the same as a nonpriority call, except that the distinctive 3-burst ringing is heard.

- Terminating Extension Group

  A Terminating Extension Group cannot be a member of a DDC or UCD group.

- Voice Terminal Display

  On calls dialed directly to a DDC or UCD group extension number, the DDC or UCD group's identity is displayed at the calling extension.

Administration

DDC and UCD are administered by the System Manager. The following items can be administered for each DDC or UCD group:

- Delay announcement
- Delay announcement interval
- Group extension number, name, and type (DDC and UCD)
- Group Path
- Class of Restriction
- 4-Digit security code (for use with AP Demand Print feature)
- Whether or not the group is used as the AP Message Center
- Whether or not the group is served by a queue

- Queue length (0 to 35 calls)
- Queue Warning Threshold (0 to 35 calls)
- Port Number assigned to queue warning level lamp
- Group Members (extension numbers)

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.

DIRECT INWARD DIALING

Description

Connects calls from the public network directly to the dialed extension number without attendant assistance.

Direct Inward Dialing reduces the attendant's workload and provides the calling party immediate contact with the called party.

Considerations

Direct Inward Dialing (DID) trunk group(s) from the local telephone company central office 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.

When a DID trunk is accessed via a Listed Directory Number (LDN), the call is routed to the attendant. The attendant display indicates that the call is an LDN call.

Administration

Direct Inward Dialing is administered by the System Manager. The only administration required is the administration of a DID trunk.

Hardware and Software Requirements

Requires one port on a TN753 DID Trunk circuit pack for each DID trunk. No additional software is required.

DIRECT OUTWARD DIALING

Description

Allows voice terminal users to access the public network without attendant assistance.

The user simply lifts the handset and dials the trunk access code for the desired trunk group. The user is then connected to the public network.

Considerations

Direct Outward Dialing reduces the attendant workload and saves the user time by allowing direct access to the public network. The user does not have to call and make the request to an attendant.

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.

Interactions

Calling party restrictions (assigned by the Class of Restriction) prevent placing Direct Outward Dialing calls from the restricted voice terminal.

Administration

Direct Outward Dialing is administered on a per-system basis by the System Manager. Administration consists of assigning the various trunk groups and their associated trunk access codes.

Hardware and Software Requirements

Requires one port on a TN747 CO Trunk circuit pack for each assigned trunk. No additional software is required.

DISTINCTIVE RINGING

Description

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

The ringing cycle, which begins when a voice terminal or attendant console receives an incoming call, is heard by the voice terminal user or attendant. Since the ringing cycle is different for different types of calls, the voice terminal user or attendant can tell what type of call is being received and can handle the call accordingly.

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

Associated Call Type	User	Ringing Cycle (In Seconds)
Internal voice terminal, internal tie trunk, and Remote Access	All voice terminals	1-burst ringing (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 ringing (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 ringing (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 ringing (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 ringing cycles are received at attendant consoles:

Call Type	Ringing 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

Ringing allows the user to identify the type of incoming call. By knowing the type of incoming call, the user is able to answer each call in a suitable manner for that type of call.

The 2- and 3-burst ringing is optional only on single-line voice terminals. If Ringing 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.

Interactions

The Distinctive Ringing cycles are altered when the Personalized Ringing (V2) feature is used.

Administration

Ringing is a standard system feature. No administration is required except for the following items which are set by the System Manager:

- Redirection Notification can be assigned for any voice terminal.
- Distinctive 2- or 3-burst ringing can be disabled for single-line voice terminals.

Hardware and Software Requirements

Requires a 500-type, 2500-type, or 7100-series voice terminal to be installed and connected to a TN742 Analog Line circuit pack. No additional software is required.

DS1 TIE TRUNK SERVICE (V2)

Description

Provides for two types of digital tie trunk interfaces: Voice-Grade DS1 and Alternate Voice/Data (AVD) DS1 tie trunks. The Voice-Grade DS1 tie trunks are an alternative to 4-wire analog E&M tie trunks and may be used to interface with other properly-equipped switching systems. AVD DS1 tie trunks permit alternate voice and data calling between System 75s and System 85.

Voice-Grade DS1 tie trunks can also be used as the following:

- Electronic Tandem Network (ETN) or Tandem Tie Trunk Network (TTTN) tie trunks
- Main/Satellite tie trunks
- Tie trunks used to interface with Enhanced Private Switched Communications Service (EPSCS) and Common Control Switching Arrangement (CCSA) networks
- Release Link trunks for Centralized Attendant Service (CAS)
- Access Trunks

AVD DS1 tie trunks can be used to connect the System 75 with other digital switches.

Both types of tie trunks use DS1 transmission facilities in the "Domestic DS1" format, which is a 1.544-Mbps digital signal that consists of a 1.536-Mbps signal multiplexed with an 8-kbps framing signal.

The TN722 DS1 circuit pack is used to support Voice-Grade DS1 tie trunks in the Robbed Bit Signaling mode, and AVD DS1 tie trunks in the Common Channel Signaling mode. The Robbed Bit Signaling mode supports 24 trunks for transmission on the circuit pack because the least significant bit ("robbed") in every sixth frame of data transmission is replaced by a signaling bit. The Common Channel Signaling mode supports 23 trunks for data transmission and 1 trunk for signaling purposes.

Considerations

DS1 tie trunks offer voice and data transmission, via DS1 digital facilities, at lower cost and faster speed than conventional analog trunks. Since AVD DS1 tie trunks can be used for Acunet T1.5 Service facilities, data transmission costs are lower than if large analog trunk groups are used. In the future, digital transmission is expected to cost less, thus adding to savings over analog facilities.

Each DS1 circuit pack can support up to 24 trunks: 24 trunks for transmission in the Robbed Bit Signaling mode or 23 trunks for clear data transmission, and one trunk to transmit Common Channel Signaling for the other 23 trunks in the Common Channel Signaling mode.

Interactions

- Centralized Attendant Service (CAS) (V2)

  Voice-Grade DS1 tie trunks can be used as Release Links for CAS, but not AVD DS1 tie trunks.

- Distributed Communications System (DCS)

  AVD DS1 tie trunks can only be used in a DCS between System 75 and System 85.

- Electronic Tandem Network (ETN)

  AVD DS1 tie trunks can be used in an ETN only between System 75 and System 85.

- Modem Pooling

  When AVD DS1 tie trunks are used, a conversion resource is not automatically inserted into the connection because the system cannot determine whether the transmission is voice or data. A conversion resource is connected between Voice-Grade tie trunks and digital endpoints.

- Private Network Access

  AVD DS1 tie trunks cannot be used as EPSCS or CCSA access trunks.

Administration

Voice-Grade DS1 and AVD DS1 tie trunks are assigned on a per-line basis by the System Manager. The following items require administration:

- DS1 Circuit Pack—Assign the circuit pack to the system before administration of the associated trunks.
- Trunk Group—Associate the trunks to groups, if desired.

Hardware and Software Requirements

One TN722 DS1 circuit pack is required for every 24 Voice-Grade DS1 tie trunks required or for every 23 AVD DS1 tie trunks required. A Version 2 TN741 Tone Generator/Clock circuit pack is required to provide synchronization for the DS1 tie trunks.

No additional software is required.

EIA INTERFACE (V2)

Description

Provides a lower cost alternative to Digital Terminal Data Modules (DTDMs) and Modular Processor Data Modules (MPDMs), within the system hardware, for interconnection between RS-232 compatible Digital Terminal Equipment (DTE) and the system. The EIA Interface consists of a Data Line circuit pack port and an Asynchronous Data Unit (ADU).

The EIA Interface supports speeds of LOW, 300, 1200, 2400, 9600, and 19200 bps.

A data line port differs from a data module in that the functions (options) are set in the system rather than at the physical hardware. The user does not have physical access to the data line port, but has access to all data module related functions; that is, the user can examine and change such items as speed, parity, etc., via a menu-driven selection mode at the DTE. Also, the System Access Terminal (SAT) can be used to examine and change the functions.

A data line port in conjunction with an ADU can be used to connect the system to the Information Systems Network (ISN). The ISN consists of packet data switches that support data calls between data endpoints. Data line ports provide the most economical access to the ISN.

Digital Communications Equipment (DCE) may also be connected to a data line port by use of a null modem.

Considerations

System 75 EIA Interface support offers a convenient and lower cost alternative to data modules. DTEs can connect directly to a Data Line circuit pack which functions as a data module connected to a Digital Line port. Since the user does not have physical access to the data module, all related data module options are settable from the DTE. With a density of eight data line ports per circuit pack, each port provides connections of user's asynchronous EIA RS-232 compatible DTE.

There is no limit to the number of Data Line circuit packs the system can support, subject to slot availability and the system limit of digital data endpoints.

Interactions

- Data Hot Line

  Data Terminal (Keyboard) Dialing permission must be granted before Keyboard Dialing can be accessed.

- Data Terminal (Keyboard) Dialing

  Access to ISN endpoints requires 2-stage dialing, the first stage consisting of dialing a hunt group extension number to access ISN, then the second stage consisting of an ISN address.

Administration

EIA Interface support is assigned on a per-data terminal basis by the System Manager. The following items require administration:

- Data Line circuit pack—Assign a vacant port, port options, and permissions on the circuit pack to the associated DTE.
- Data Extension buttons—Assign Data Extension buttons to multi-appearance voice terminals.

The following permissions can be administered on a data line port to allow DTEs to be used.

- Keyboard Dialing (KYBD)—Must be set to allow data endpoints to receive and send text during data call origination or termination. Text prompts are provided.
- Configuration—Must be set to allow DTEs to change their data module options; that is, examine and change options, such as speed, from the DTE. Keyboard Dialing permission must be granted first.
- Busy Out—Should be set for DTEs that are members of a hunt group, and to allow "busy out" (when DTE turns power off) so that calls will not terminate on that DTE.

The following options can be examined and changed from the DTE if the "configuration" permission has been granted:

- Speed—All speeds (up to 19.2 kbps) at which the DTE can operate are selectable, including Autoadjust. Autoadjust is the capability of the data line port to determine what speed and parity the associated DTE is transmitting at and match it for terminal dialing and/or text feedback purposes.
- Parity—All choices of parity (even, odd, mark, or space) can be selected.
- Permit Mismatch—The EIA Interface may be operated at a higher transmission speed rate than the rate between the Data Line circuit and the far end data module. This allows for calls between digital endpoints with different speeds without changing the speed of the DTE.
- Dial Echoing—Can be set to echo typed characters back to the DTE during dialing.
- Disconnect—Set the signal to indicate "disconnect." Choices are one break greater than 2 seconds or two breaks within 1 second.
- Answer Text—Can be selected when DTE is an intelligent device, to allow text messages to be delivered to the DTE when a call is being answered; also, applies to text generated by the data line circuit and received from the system. The following call progress messages may be answered:
  - INCOMING CALL
  - PLEASE ANSWER
  - TRANSFER
  - FORWARDED
  - ANSWERED
  - ABANDONED
  - DISCONNECTED
  - OTHER END
- Connected Indication—Can be set to allow the text "CONNECTED SPEED= XXXX" to be sent to the DTE when the data call has been established.

- Other Characteristics—The Data Line circuit always operates in automatic answer (provided the DTE is on), asynchronous, and full duplex modes. The "Loss of Carrier Disconnect" is set to off; that is, the Data Line circuit, unlike other data modules, does not disconnect upon loss of EIA updates in the previous 4 seconds.

Hardware and Software Requirements

One TN726 Data Line circuit pack is required for each eight EIA interfaces provided.

No additional software is required.

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, terminating extension group, a hunt group (Direct Department Calling or Uniform Call Distribution 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 automatically selects an idle call appearance and places a call to the resource.

Considerations

With Facility Busy Indication, a user can monitor the busy or idle status of a frequently called extension number. By knowing when the monitored facility is busy or idle, the user can wait until the facility is idle to make a call. This reduces the time spent trying to call busy facilities.

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.

Extension numbers, trunk group access codes, and Loudspeaker Paging Access codes can be stored in a Facility Busy Indication button. However, an access code followed by other numbers cannot be stored.

It is possible that an incoming call which causes the lamp to flash may go unanswered. If the lamp reresents the status of a trunk group and all trunks in the trunk group become busy before the flashing call is answered, the system software lights the lamp steadily to indicate that all trunks are busy. When a trunk in that trunk group becomes idle, the system software turns off the busy indication and the lamp goes dark. Therefore, the lamp flashes, lights steadily, and goes out while the call has neither been answered nor dropped.

Interactions

None.

Administration

Facility Busy Indication is administered on a per-voice terminal basis by the System Manager. The only administration required is to assign the Facility Busy Indication button to a voice terminal or attendant console.

Hardware and Software Requirements

No additional hardware or software is required.

FACILITY RESTRICTION LEVELS AND TRAVELING CLASS MARKS (V2)

Description

Provides up to eight levels of restriction for users of the Automatic Alternate Routing (AAR) and/or Automatic Route Selection (ARS) features.

Facility Restriction Levels (FRLs) and Traveling Class Marks (TCMs) provide a method of allowing certain calls to specific users, while denying the same calls to other users. For example, certain users may be allowed toll calling only to other corporate locations. Similarly, certain users may be allowed toll calling into more areas than other users. International calling may be denied to all except a few users.

FRLs and TCMs are transparent to the user. Appropriate values are predetermined and programmed into the system. Dialing procedures are unaffected.

Call routing for each call is determined by the dialed Area Code and/or office code (either public or private network). Translation on the first three or six digits of the called number yields one of 254 Routing Patterns, numbered 1 through 254. More than one translation can point to the same pattern. Pattern 0 provides intercept treatment and is used for unassigned private network office codes. Each Routing Pattern contains up to six routing preferences. Each preference includes the following information:

- Trunk Group Number
- Minimum FRL required to access the trunk group

Routing preferences are listed in ascending FRL order.

Each facility, such as a trunk or voice terminal, capable of originating a call also has an associated FRL. Whether a given call is allowed or not depends on two things: compatibility between FRLs and availability of an idle trunk.

Compatibility is determined by a comparison of the minimum FRL associated with the trunk group and the originating-side FRL. Either can have a value of 0 through 7. Access to the associated trunk group is permitted if the originating-side FRL is greater than or equal to the minimum FRL. Note that lower originating-side FRLs can access fewer routing preferences, whereas lower minimum FRLs permit greater access. Stated another way, a 0 originating-side FRL is the most restricted and a 7 is the least restricted. A 0 minimum FRL is the least restrictive, and a 7 is the most restrictive. Compatibility checking begins with the first-choice route (the first one in the pattern). Assuming access is permitted, availability is checked; that is, is there an idle trunk in the group? If so, the call continues. If not, compatibility is checked on the next routing preference.

If the compatibility check fails on the first-choice route, intercept tone is returned to the user. This call will always fail and need not be retried. If the compatibility check fails on the second or subsequent routing preference, or if all accessible trunk groups are busy, the call may queue on the first choice trunk group. (See Ringback Queuing, elsewhere in this section, for details.)

If the trunk group selected for a call is an intertandem tie trunk group, then a TCM is outpulsed as the last digit. A TCM is equivalent to the originating-side FRL. At the next tandem switch, compatibility and availability checking are done, as before. In this case, the FRL assigned to the incoming intertandem tie trunk group is used as the originating-side FRL. However, if this fails to yield a route and if the TCM is higher than the tie trunk FRL, then the TCM is used in another attempt to complete the call.

Call Originating Facilities

At a System 75 serving as the call origination point, any of the following can be the originator of an ARS or AAR call:

- Voice terminal
- Remote Access user
- Attendant
- Incoming tie trunk group from a subtending location
- Data terminal capable of Keyboard Dialing

At a tandem switch, either of the following can be the originator of an ARS or AAR call:

- Incoming Intertandem tie trunk group
- Incoming Access tie trunk group—links a remote main switch to a tandem switch

Each of these facilities is assigned an FRL via an associated Class of Restriction (COR), either directly or indirectly.

Voice terminals and all incoming tie trunk groups use the FRL contained within the assigned COR. Attendants use the FRL contained within the COR assigned to the attendant group. Data terminals use the FRL contained within the COR assigned to the associated data module.

The Remote Access feature can be accessed via a Direct Inward Dialing (DID) trunk group, tie trunk group, dedicated central office trunk group, 800 Service trunk group, and/or dedicated foreign exchange trunk group. In the absence of a Remote Access Barrier Code, the applicable FRL is contained in the COR assigned to the trunk group. If a Barrier Code is required on Remote Access calls, the applicable FRL is contained in the COR assigned to the Barrier Code.

Call Terminating Facilities

Any of the following trunk types can serve as the termination point for an ARS or AAR call:

- Tie trunk—excluding release link trunks (RLTs), but including Common Control Switching Arrangement (CCSA) and Enhanced Private Switched Communications Service (EPSCS) Access trunks
- Wide Area Telecommunications Service (WATS)
- Local central office (CO)
- Foreign exchange (FX)

Each of these outgoing trunk groups has an assigned COR that contains an FRL. However, this FRL is never used. Terminating-side FRLs are assigned in the Routing Pattern, not to the outgoing trunk group.

Considerations

FRLs provide the means to restrict certain users from placing selected calls while allowing other users to place the same calls.

Originating-side FRLs are assigned via the COR of the originating-side facility, such as an incoming tie trunk group or voice terminal. If an FRL is not assigned, the system assumes an FRL of 0 for all originating facilities except the attendant group. An FRL of 7 is assumed for the attendant group.

A COR is also assigned to each trunk group. If the COR specifies an FRL, the FRL is ignored. The minimum FRL specified in the Routing Pattern is the only FRL used on the terminating side of the call.

On attendant-extended calls, the attendant group FRL is used rather than the FRL of the calling party.

Interactions

FRLs apply only on ARS and AAR calls.

If Station Message Detail Recording (SMDR) 15-digit account codes are used, the FRL field in the SMDR record is overwritten.

Administration

FRLs are assigned by the System Manager as a part of ARS and/or AAR administration. Originating FRLs are assigned on a per-COR basis. Terminating FRLs are assigned on a per-Routing Pattern basis. TCMs do not require assignment.

Assignment Guidelines

The FRL assigned to the facility answering a call is not checked. Terminating-side FRLs apply to trunk groups only. This simplifies assignments. At each switch, the trunk groups available to handle a given call must be listed in the preferred order within the Routing Pattern. The most-preferred choice must be at the top of the list. Up to six choices can be specified. Now the relative value of access to each of the listed trunk groups must be determined. This, of course, is specified via an FRL. On a scale of 0 through 7, the relative value is determined and assigned. Decisions are normally based on the cost of using the facility, although other criteria can be used. The same FRL value can be assigned to more than one trunk group if there is no reason to prefer one trunk group over the other.

If there will be users within the system who are not allowed to make outside calls, use some value other than 0 as the value for the first-choice trunk group. By assigning these users an FRL of 0, none of the trunk groups can be accessed (since all trunk group FRLs are greater than 0). Such calls are denied.

Each Routing Pattern must be individually constructed. The same trunk group can be used in more than one pattern. The associated FRL is assigned within the pattern and is not associated with the trunk group itself. The same trunk group can have a different FRL in a different pattern.

Be consistent in FRL assignments. Do not use a range of 0 through 5 in one pattern and a range of 2 through 7 in another pattern if all users can access the first-choice route. Admittedly, the trunk group with an FRL of 2 may be more expensive than the trunk group with an FRL of 0, but there is no real reason to assign a 2 to a trunk group that everyone can access. For ease of assignments, always use a 0 for such a trunk group.

There should be a COR established for each FRL used in a Routing Pattern. The appropriate COR is then assigned to the users who can access the routes restricted by the FRL value. For example, a middle executive might be able to access all routes with an FRL of 5 or lower, whereas the president can access all routes. In this case, the executive is assigned a COR with an FRL of 5 and the president is assigned a COR with an FRL of 7.

Remote Access users can access the system's features and services the same as an on-premises user. If Barrier Codes are not used, then the FRL associated with the COR assigned to the originating facility is used as the originating FRL. The choice of the FRL value should be consistent with the calling privileges of comparable on-premises users. Use of the single FRL, however, is not completely satisfactory. Invariably, some users will have less calling privileges when they use Remote Access rather than normal on-premises access. Likewise, some users will gain more calling privileges by using Remote Access. Use of Barrier Codes alleviates this problem. Up to 10 Barrier Codes, each with its own COR (and FRL), can be assigned. Although the COR defines other restrictions, 10 Barrier Codes are enough to also provide a range of FRL assignments. Assignment of Barrier Code FRLs is the same as if the user were on-premises. The simplest way to assign these FRLs is to duplicate the on-premises FRLs, then merely relate the appropriate Barrier Code to those that will be using Remote Access.

Hardware and Software Requirements

No additional hardware is required. The optional ARS software is required with Version 1 systems. The optional Private Network Access or ARS software is required with Version 2 systems.

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 central office (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

If a user has a problem with a specific system facility, Facility Test Calls can be used to test that facility for proper operation.

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.

Interactions

None.

Administration

Facility Test Calls is administered on a per-system basis by the System Manager. The only administration required is to assign the Facility Test Calls access code.

Hardware and Software Requirements

No additional hardware or software is required.

GO TO COVER

Description

Allows users, when making a call to another internal extension, to send the call directly to coverage.

Go To Cover is activated by pressing a Go To Cover button. If activated prior to ringing, the call does not attempt to direct to the called extension, but goes directly to coverage. Go To Cover can also be used later in the call.

Details of how Go To Cover is used in conjunction with Call Coverage are given in the Call Coverage feature description, elsewhere in this section.

Consideration

Go To Cover gives the calling party the option to send calls directly to coverage.

Interactions

The following features do not redirect to coverage unless the caller presses the Go To Cover button:

- Intercom—Automatic
- Intercom—Dial
- Priority Calling

Go To Cover can only be used if the called party is assigned a call coverage path; that is, the called party must have alternate answering positions assigned.

Administration

Go To Cover is administered on a per-voice terminal basis by the System Manager. The only administration required is to assign a Go To Cover button.

Hardware and Software Requirements

No additional hardware or software is required.

HOLD

Description

Allows 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

With the Hold feature, voice terminal users can temporarily disconnect from one call and handle another call. For example, a busy voice terminal user who receives another call can place the first call on hold and answer the second call. This results in fewer missed calls. The Hold feature can also be used when a user receives a call and needs to make another call to obtain information for the calling party.

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 then return to the original call by pressing Recall or flashing the switchhook again.

Interactions

A held multi-appearance voice terminal user can activate Leave Word Calling toward the holding user.

With V2, a single-line voice terminal user can place a call on hold to answer a waiting call. The waiting call is connected automatically. At the conclusion of the call, the user hangs up and is re-rung by the held call.

Any user, active on a bridged call, can place the call on hold. If a call on a bridged call appearance is placed on hold and no other users with a bridged call appearance of the same extension number are connected to the call, the status lamp at the Bridged Appearance button indicates that the call is on hold. If the primary extension or another bridged appearance is connected to the call, the status lamp at all bridged appearances indicates an active status for the call.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

HOT LINE SERVICE

Description

Allows single-line voice terminal users, by simply lifting the handset, to automatically place a call to a preassigned extension number, public or private network telephone number, or feature access code.

The Hot Line Service destination number is stored in an Abbreviated Dialing List. When the Hot Line Service user lifts the handset, 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 Alternate Routing (AAR), 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), 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.

Considerations

The Hot Line Service feature is useful in any application where very fast service is required. Also, if a voice terminal is used only for accessing a certain facility, it can be assigned to Hot Line Service. The Hot Line Service voice terminal user simply lifts the handset and is connected to that facility.

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.

Interactions

A Hot Line Service user cannot activate any feature unless the access code is, or is part of, the destination number.

- Loudspeaker Paging Access

  Loudspeaker Paging Access can be used with Hot Line Service to provide automatic access to paging equipment.

- Ringback Queuing

  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.

Administration

Hot Line Service is administered on a per-voice terminal basis by the System Manager. The following items require administration:

- Abbreviated Dialing Lists
- Hot Line Destination Number

Hardware and Software Requirements

No additional hardware or software is required.

HUNTING

Description

Checks for the active or idle status of extension numbers in one or more ordered groups. If all members of a group are active, the call can route to another group via Call Coverage or can wait in a queue for an available group member, if provided.

Hunting is accomplished through the Call Coverage, Direct Department Calling, and Uniform Call Distribution features. The order of hunting is defined under each individual feature.

Considerations

Hunting is useful whenever a group of voice terminal users receives a high volume of calls. It minimizes call completion time and attendant assistance is not required.

Interactions

With V2, individual attendant extensions can be in hunt groups.

Administration

Hunting is administered through the Call Coverage, Direct Department Calling, and Uniform Call Distribution features. Administration of each of these features is discussed under that feature, elsewhere in this section.

Hardware and Software Requirements

No additional hardware or software is required.

INDIVIDUAL ATTENDANT ACCESS (V2)

Description

Allows users to access a specific attendant console. Each attendant console can be assigned an individual extension number, to provide access to each individual attendant.

A user can access an individual attendant by simply lifting the handset and dialing the extension number assigned to the desired attendant. An individual attendant extension number can also be assigned to users' abbreviated dialing button for fast access to the specific attendant.

Individual attendants can be accessed by voice terminal users, incoming trunks, Remote Access, and other attendants. A specific attendant, when called, can extend the call to another trunk or extension.

Each individual attendant has a queue that allows one or two incoming calls to wait. This individual attendant queue has priority over all other attendant seeking calls.

Whenever a call is in an individual attendant's queue, the top lamp of the Forced Release button lights to indicate this condition.

An individual attendant can be a part of a hunt group. The hunt group can be a Direct Department Calling (DDC) group or a Uniform Call Distribution (UCD) group. Calls to individual attendants and calls to the attendant group have priority over hunt group calls to an individual attendant.

Any call made from an attendant console which is assigned an individual extension is considered to be made from the individual attendant, not the attendant group.

Considerations

With Individual Attendant Access, attendant consoles can become more flexible by assigning each one an individual extension number. An individual attendant extension allows an attendant to use features that an attendant group cannot use; for example, individual attendant extensions can be a member of a DDC or UCD group. An individual attendant can also be accessed when the Centralized Attendant Service feature is in effect. Another advantage is that each individual attendant extension can have its own Class of Restriction and Class of Service.

The Position Available lamp on the attendant console only indicates whether or not attendant group calls can be accepted. It does not indicate whether or not individual attendant calls can be accepted.

Each attendant console has one position busy button. When the lamp associated with this button is lighted, the attendant will not receive attendant group calls but can still receive individual attendant calls.

Interactions

- Abbreviated Dialing

  Individual attendant extensions can be in Abbreviated Dialing lists. Individual attendants, however, cannot have their own Abbreviated Dialing lists.

- Attendant Display

  For calls to or from individual attendants, individual attendant names (when specified) will be displayed instead of the individual attendant extensions.

- Bridged Extension

  Individual attendant extensions cannot be assigned to a bridged call appearance.

- Busy Verification of Terminals and Trunks

  An individual attendant extension cannot be busy verified.

- Call Coverage

  Individual attendant extensions can be points in a coverage path but cannot be a member of a coverage answer group.

- Call Park

  Individual attendants can park calls on their own extension or another individual attendant extension.

- Call Pickup

  Individual attendant extensions cannot be in Call Pickup groups.

- Centralized Attendant Service (CAS)

  Individual attendants can be accessed when CAS is in effect.

- Class of Restriction and Class of Service

  Each individual attendant extension has its own Class of Restriction and Class of Service.

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

  Individual attendant extensions can be assigned to DDC and UCD groups. Unlike voice terminal users, individual attendants can answer DDC and UCD calls as long as there is an idle call appearance and no other DDC or UCD call is on the console.

- Facility Busy Indication

  An individual attendant extension can be stored in a Facility Busy Indication button.

- Integrated Directory

  The names and extensions of the individual attendants are stored in the directory associated with this feature.

- Leave Word Calling

  A message from an attendant will indicate whether it is from the attendant group or whether it is from an attendant which has an individual extension.

- Night Service—Night Console Service

  Activation and deactivation of this feature affects only calls to the attendant group. Calls to individual attendant extensions are allowed when night service is active. A night-only attendant console with an individual extension can receive individual attendant calls when night service is not active.

- Privacy—Attendant Lockout

  This feature applies only to attendant group calls. Individual attendant calls are not affected.

- Voice Terminal Display

  For calls from individual attendants, individual attendant names (when specified) will be displayed instead of the individual attendant extensions.

Administration

Individual Attendant Access is administered on a per-attendant console basis by the System Manager. The following items require administration for each attendant console:

- Extension number
- Name
- Class of Restriction
- Class of Service

Hardware and Software Requirements

No additional hardware or software is required.

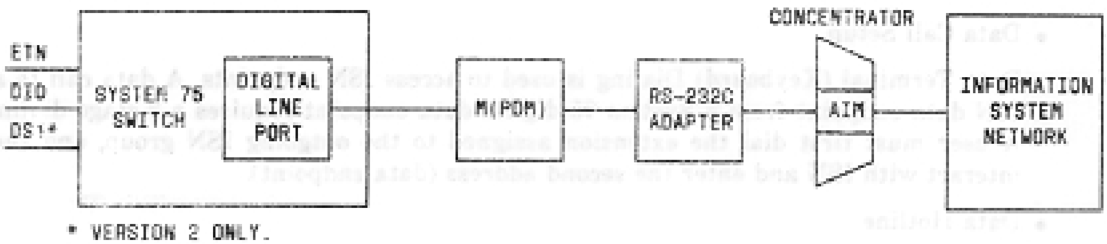
INFORMATION SYSTEM NETWORK (ISN) INTERFACE

Description

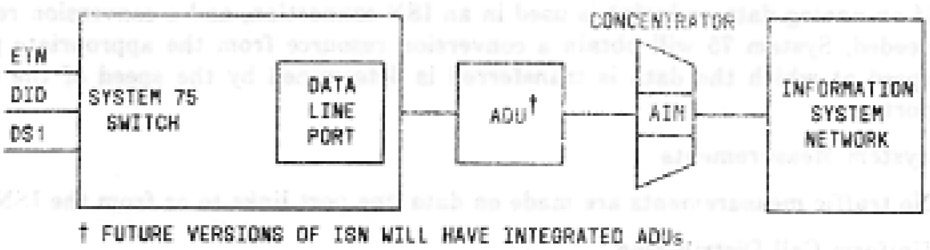
Provides links between mainframe computers, minicomputers, word processors, storage devices, personal computers, printers, terminals, and communications processors so that they function as a single system.

The ISN is a packet-switched local area network. This local area network is made up of one or more modular data-only digital communications switches.

The interface between System 75 and the ISN is via a Data Line Port in conjunction with an Asynchronous Data Unit (ADU). A (Modular) Processor Data Module [(M)PDM] may be used instead of the ADU, but the ADU is more economical. Also, future versions of the ISN will have integrated ADUs. The (M)PDM or ADU connects to an Asynchronous Interface Module (AIM) on the Packet Controller or Terminal Concentrator (see Figure 3-3). This interface allows System 75 and the ISN to share data capabilities.



VERSION 1 OR VERSION 2



VERSION 2 ONLY

Figure 3-3. System 75 to ISN Connectivity

Data is transferred between System 75 and the ISN on one-way trunks (either incoming or outgoing). Each data line port is administered for a specific data rate, which can be any of the common asynchronous data rates ranging from LOW to 19,200 bps.

Considerations

Connectivity between ISN and System 75 provides the following major benefits:

- Users on ISN may (in addition to having access to other endpoints directly connected to ISN) have access to any endpoint connected to System 75 or addressable from System 75.
- Users who either connect to or have access to System 75 may also access endpoints connected to ISN.

Since the ISN switches are modular, the local area data communications network can be designed so that it is both versatile and cost-effective. A single packet controller can be configured to support from 40 to 1920 data ports.

Interactions

- Abbreviated Dialing

  Outgoing lines cannot use Abbreviated Dialing.
- Automatic Circuit Assurance

  Automatic Circuit Assurance is not provided for data line port links to or from the ISN.
- Data Call Setup

  Data Terminal (Keyboard) Dialing is used to access ISN endpoints. A data call to an ISN data endpoint from a System 75 digital data endpoint requires a 2-stage dialing. A user must first dial the extension assigned to the outgoing ISN group, and then interact with ISN and enter the second address (data endpoint).
- Data Hotline

  Outgoing lines cannot use hot line calling.
- Modem Pooling

  If an analog data endpoint is used in an ISN connection, and a conversion resource is needed, System 75 will obtain a conversion resource from the appropriate pool. The speed at which the data is transferred is determined by the speed of the data line port.
- System Measurements

  No traffic measurements are made on data line port links to or from the ISN.
- Uniform Call Distribution

  Outgoing lines should be members of a Uniform Call Distribution (UCD) group. This way, the system automatically selects an idle port when a user tries to access the ISN.

Administration

Data module extensions used to access ISN must be administered by the System Manager as data lines connected to the ISN. The System Manager can then administer the other options required for each data line. These options include:

- Keyboard Dialing—If the line is incoming (to System 75), Keyboard Dialing should be enabled so that System 75 can be accessed by the ISN. If the line is outgoing (to the ISN), Keyboard Dialing should be disabled.
- Configuration—This option should be disabled on both incoming and outgoing lines to prevent the ISN from changing the data line configuration.
- Busy Out—This option should be enabled for outgoing lines so that a member of the outgoing ISN group can be "busied out" and let calls go through another member of the group.
- Speeds—Data speeds should be selected according to individual needs, and should be the same as those at connecting ISN ports. Only one speed should be assigned to each data line port.
- Autoadjust—This option is not needed with the ISN, and should be disabled on incoming lines. This option can only be set if Keyboard Dialing is enabled.
- Permit Mismatch—This option should be disabled on both incoming and outgoing data lines.
- Disconnect—The disconnect sequence should be administered as one long break. This option can only be set if Keyboard Dialing is enabled.
- Parity—This option should be administered as even. This option can only be set if Keyboard Dialing is enabled.
- Dial Echoing—This option should be disabled so that characters are not echoed back to the ISN. This option can only be set if Keyboard Dialing is enabled.
- Answer Text—This option should be disabled, and can only be set if Keyboard Dialing is enabled.
- Connected Indication—This option should be disabled, and can only be set if Keyboard Dialing is enabled.
- Class of Restriction (COR)—Outgoing lines should be origination restricted. Incoming lines should be termination restricted.

Hardware and Software Requirements

One TN726 Data Line circuit pack is required for each eight ISN interfaces. No additional software is required.

INTEGRATED DIRECTORY

Description

Allows internal system users with display-equipped terminals 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 display-equipped voice terminal users or Attendants with 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 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

With Integrated Directory, users spend less time looking up names and extension numbers. Instead of searching through lists or directories, a user simply keys in the desired name and the display shows the name and extension number. Less dialing time is also required if a Return Call button is provided. When the desired extension is displayed, the user just presses the Return Call button to automatically place the call.

The maximum size of the directory is 400 entries (V1) or 500 entries (V2). The maximum length of the name is 15 characters (including spaces and commas). The extension number cannot exceed four digits (V1) or five digits (V2).

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 cannot be used to make calls or access features by dial code. It can, however, 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.

Interactions

- Touch-Tone Dialing

  Call origination and feature access by dial code is not allowed when the Integrated Directory feature is active.

Administration

Integrated Directory is administered on a per-voice terminal basis by the System Manager. The following items require administration:

- Display Module
- Integrated Directory Button
- Return Call Button
- Messaging Cartridge (for 7404D)
- Next button

Hardware and Software Requirements

No additional hardware or software is required.

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 handset 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

The Intercept Tone lets a user know when a call cannot be completed as dialed. The user can then hang up or try the call again. When Direct Inward Dialing and Private Network Access calls cannot be completed as dialed, a recorded announcement can be provided or, for more personal service, the calls can be routed to the attendant.

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.

Interactions

Attendant Intercept and Recorded Announcement Intercept (both optional) cannot be used together. Direct Inward Dialing calls cannot be assigned Intercept Treatment—Tone.

Administration

The Intercept Tone is standard and requires no administration. However, administration is required to determine whether Direct Inward Dialing and Private Network Access calls are routed to the attendant or to an announcement. If an announcement is to be used, the announcement must be administered.

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.

INTERCOM — AUTOMATIC

Description

Provides a talking path between two voice terminal users. Calling users press the Automatic Intercom button and lift the handset, or vice versa. The called user receives a unique intercom alerting signal, and the status lamp associated with the Dial or Automatic Intercom button, if provided, flashes.

Considerations

With Automatic Intercom, users who frequently call each other can do so by pressing one button instead of dialing an extension number.

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.

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.

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.

Administration

Automatic Intercom is assigned on a per-voice terminal basis by the System Manager. Before Automatic Intercom can be assigned, the associated Intercom group must be established. Each Intercom group requires the following administration:

- Intercom group number
- Length of dial code
- Extension number within the group
- Dial codes to access Intercom group members

Once the Intercom group is established, Automatic Intercom buttons can be assigned to members of the group. The following items must be administered for each button.

- Intercom group number to be accessed
- Dial code assigned to group member to be accessed

Hardware and Software Requirements

No additional hardware or software is required.

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 handset, 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

With Dial Intercom, a group of users who frequently call each other can do so by pressing a Dial Intercom button and dialing a 1- or 2-digit code instead of dialing an extension number.

Up to 32 Intercom groups can be established. Each group can have up to 32 members, with a maximum of 128 members per system.

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.

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.

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.

Administration

Dial Intercom is administered by the System Manager. The following items require administration:

- Intercom groups
  - Group number
  - Length of dial code
  - Extension numbers within the group
  - Dial codes to access Intercom group members
- Dial Intercom buttons

Hardware and Software Requirements

No additional hardware or software is required.

INTER-PBX ATTENDANT CALLS (V2)

Description

Allows attendant positions for more than one branch location to be concentrated at one central, or main, location. Each branch location has its own Listed Directory Number (LDN). Incoming trunk calls to the branch location, as well as attendant-seeking voice terminal calls, are routed over tie trunks to the attendants at the main location.

Inter-PBX Attendant Calls are incoming tie trunk calls from the branch location to the main location with the attendant group as the destination. If no attendant in the attendant group is available, these calls are queued. When an attendant becomes available, the call is routed to that attendant for handling. Calls can be extended the same as if the call were an incoming LDN to the main location. When the attendant releases the call, the tie trunk associated with the call is tied up with the call until the call is dropped.

A System 75 can be a branch or a main location for this feature. A branch location can have local attendants. These local attendants can be accessed by the Individual Attendant Access (V2) feature. The attendants at the main location are the local attendants for the main location and also the attendant for the Inter-PBX Attendant Calls.

Considerations

With Inter-PBX Attendant Calls, the number of attendants required at each branch location is reduced. Also, users at each branch location can use the main location to access each of the other branch locations.

Interactions

- Attendant Control of Trunk Group Access

  If Inter-PBX Attendant Calls is enabled, and a call at a branch location attempts to access a controlled trunk group, the call is routed to the local attendant at the branch location, if there is one. If there is no local attendant, the call is routed to the attendant group at the main location.

- Attendant Display and Distributed Communications System (DCS) Attendant Display

  In a DCS environment, an incoming Inter-PBX Attendant Call from a branch location is displayed at the attendant console the same as a local call.

  In a non-DCS environment, an incoming Inter-PBX Attendant Call is displayed at the attendant console as an incoming tie trunk call.

- Attendant Recall

  If an attendant at the main location holds an Inter-PBX Attendant Call, the calling parties at the branch location cannot recall the attendant.

- Call Coverage

  At a branch location with Inter-PBX Attendant Calls enabled, a call redirected to a coverage path with the attendant group as a coverage point will skip that coverage point.

- Centralized Attendant Service (CAS) (V2)

  CAS and Inter-PBX Attendant Calling cannot be enabled at the same time.

- Dial Access to Attendant

  When a user at a branch location dials a single digit "0" and the Inter-PBX Attendant Calls feature is enabled, the call is routed to an attendant at the main location.

- Night Service

  The Inter-PBX Attendant Calls feature is deactivated when the branch location is put into night service, and reactivated when the branch location is taken out of night service.

Administration

Inter-PBX Attendant Calls is administered by the System Manager. The following items require administration:

- Branch location access to Inter-PBX Attendant Calls
- Inter-PBX Attendant Calls Trunk Group
- Inter-PBX Attendant Access Code

Hardware and Software Requirements

Requires a tie trunk group between the branch and main locations. No additional software is required.

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 (V1) or 24 digits(V2) of the last number dialed whether the call attempt was manually dialed or an Abbreviated Dialing button was pressed.

Considerations

Last Number Dialed prevents the user from having to redial a busy number. If a user has dialed a busy number and that was the last number dialed, the user simply activates Last Number Dialed by button or dial access code. The system automatically dials the same number again.

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.

Interactions

- Abbreviated Dialing

  With Version 2, if the previously called number was in an Abbreviated Dialing privileged list, and if the user is not normally allowed to dial the number because of his or her Class of Restriction, Intercept Treatment is given when using Last Number Dialed. To redial the number, the user must again use the Abbreviated Dialing privileged list.

- Automatic Callback

  Automatic Callback can be used after the Last Number Dialed feature is used on a call to an internal voice terminal.

- Bridged Call Appearance

  Activation of the Last Number Dialed feature causes the last number dialed from the voice terminal to be redialed, regardless of which extension number is used (primary or bridged call appearance).

Administration

Last number dialed is administered by the System Manager. The following items require administration:

- Feature Access Code for Last Number Dialed
- Last Number Dialed button

Hardware and Software Requirements

No additional hardware or software is required.

LEAVE WORD CALLING

Description

Allows internal system users to leave a short preprogrammed message 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.

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.

A calling user who left an LWC message can cancel that message if it has not already been accessed. The calling user lifts the handset, 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

Leave Word Calling lets users automatically leave short, simple messages for other users. When a voice terminal's message lamp is lighted, the user simply has the message retrieved by an authorized user. This reduces the time spent making handwritten notes.

Ten terminals, or nine terminals and the attendant console group, can be administered as systemwide message retrievers.

A system maximum of 1000 (V1) or 2000 (V2) 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 capacity. Authorized retrievers can selectively delete messages to gain storage space. Old messages are not automatically purged by the system.

Interactions

- Bridged Call Appearance

  A Leave Word Calling message left by a user on a bridged call appearance leaves a message for the called party to call the primary extension number assigned to the bridged call appearance. When a user calls a primary extension, and activates Leave Word Calling, the message is left for the primary extension, even if the call was answered at a bridged call appearance.

- 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.

Administration

Leave Word Calling is administered by the System Manager. The following items require administration:

- 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 (per voice terminal)
- 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)

All buttons associated with the display modes are administered through the Attendant Display and Voice Terminal Display features.

Hardware and Software Requirements

No additional hardware or software is required.

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 handset is still lifted, the voice terminal is taken out of service.

The out-of-service condition remains in effect until the voice terminal user hangs up.

Considerations

The out-of-service condition provided by Line Lockout does not tie up switching facilities or call processing time. The facilities are then available for other users.

This feature does not apply to multi-appearance voice terminals.

Interactions

Call intercept is provided by the Intercept Treatment feature.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

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 by the system. (A zone is the location of the loudspeakers, for example, conference rooms, warehouses, or storerooms.) In addition, one zone can be provided by the system to activate all zones simultaneously.

Each of the ten zones provided by the system is assigned an individual trunk access code. The trunk access codes are used to activate Loudspeaker Paging Access. A user can activate Loudspeaker Paging Access by dialing the trunk access code of the desired paging zone. In addition, the trunk access codes can be stored in Abbreviated Dialing lists. This allows multi-appearance voice terminals to activate the feature via Abbreviated Dialing buttons. Attendants can use a Direct Trunk Group select button to activate Loudspeaker Paging Access, if the desired paging zone's trunk access code is assigned to one of the buttons.

Once a user has activated Loudspeaker Paging Access for the desired zone, the user can speak into the handset and make the announcement.

In addition to, or instead of the System 75 loudspeaker paging equipment, a PagePac* paging system can be used. A PagePac paging system has a distinct advantage over the System 75 paging system in that a PagePac system requires only one port on one circuit pack to provide as many as 39 paging zones. The System 75 paging system requires a separate port for each paging zone with a maximum of nine zones. Three different PagePac paging systems are available for use with System 75:

- PagePac 20

  This is the smallest PagePac system. The basic system provides a single paging zone with an input source for music over the paging system. The music can also serve as the music for the Music-on-Hold Access feature. Additional add-on hardware is available to provide multi-zone paging for 3, 9, or 39 paging zones.

- PagePac VS

  This system provides one to three paging zones. It also permits the paging of all zones simultaneously. Additional hardware is available to provide music and/or talkback over the paging system.

- PagePac 50/100/200

  This system provides up to 24 paging zones. Additional hardware is available to provide music and/or talkback over the paging system.

PagePac equipment is also easy to use. A user simply dials the extension number (PagePac 50/100/200 only) or trunk access code assigned to the PagePac system. This connects the user to the PagePac equipment. If there is only one paging zone, the user then uses the handset of the voice terminal to page someone. If there are multiple zones, the user, after hearing a steady tone, dials a 1- or 2-digit code to access the desired zone(s) before paging.

* Trademark of Harris Corporation Dracon Division

Considerations

With Loudspeaker Paging Access, a user can be paged at any location with loudspeaker paging equipment. This feature is particularly useful when used in conjunction with the Call Park feature. When a user is away from his or her location and receives a call, an incoming call can be answered and parked by another user. The called party can then be paged and told what extension number the call is parked on. The called party can then answer the parked call from a nearby voice terminal.

The system can have up to nine individual zones plus one zone to activate all zones simultaneously. A PagePac paging system can be used to provide up to 39 paging zones.

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.

Interactions

The following features cannot be used with Loudspeaker Paging:

- Attendant Conference
- Terminal Conference
- Data Call Setup
- Hold
- Ringback Queuing
- Transfer

Normally, a call to a busy single-line voice terminal results in a call waiting tone being heard by the called voice terminal user. If that user is in the process of paging, the call waiting tone is not heard.

It is not possible to use a PagePac paging system for Code Calling Access when multi-zone paging is desired. The PagePac systems expect a 2-digit code to access a particular zone. System 75, however, immediately plays the chime code once a connection is established.

Administration

Loudspeaker Paging Access is administered by the System Manager. The following items require administration:

- Up to ten (one 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 features.
- Trunk access codes and Class of Restriction (per zone provided).
- Paging expiration interval (from 10 seconds to 10 minutes).
- Station Message Detail Recording activation.

If a PagePac paging system is to be used, it must be assigned a trunk access code or extension number (PagePac 50/100/200 only).

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, 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.)

If PagePac equipment is used, one port on a TN747 CO Trunk circuit pack, TN742 Analog Line circuit pack, or TN763 Auxiliary Trunk circuit is required (depending on which PagePac system is used).

No additional software is required.

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.)

Interactions

None.

Administration

Manual Message Waiting is administered on a per-voice terminal basis by the System Manager. The only administration required is the assignment of the Manual Message Waiting buttons to the voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

MANUAL ORIGINATING LINE SERVICE

Description

Connects users to attendant automatically when the user lifts the handset.

The attendant code is stored in an Abbreviated Dialing list. When the Manual Originating Line Service voice terminal user lifts the handset, the system automatically routes the call to the attendant using the Hot Line Service feature.

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.

Considerations

Manual Originating Line Service is useful in any application where all call originations are screened by the attendant. The user simply lifts the handset and is connected to the attendant.

The number of voice terminals that can be assigned Manual Originating Line Service is not limited.

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.

When a Night Service feature is activated, the Manual Originating Line Service call redirects.

Administration

Manual Originating Line Service is administered on a per-voice terminal basis by the System Manager. The following items require administration:

- Abbreviated Dialing Lists (the attendant code must be a list entry)
- Hot Line Destination

Hardware and Software Requirements

No additional hardware or software is required.

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.

The signal is sent each time the button is pressed. If the receiving voice terminal is already being rung with 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.

Considerations

With Manual Signaling, one voice terminal user can signal another voice terminal user. The meaning of the signal is prearranged between the sender and the handset.

When a voice terminal user presses the Manual Signaling button, the associated status lamp lights for 2 seconds.

Interactions

None.

Administration

Manual Signaling is assigned on a per-voice terminal basis by the System Manager. The only administration required is the assignment of the Manual Signaling button to the originating voice terminal.

Hardware and Software Requirements

No additional hardware or software is required.

MODEM POOLING

Description

Allows switched connections between digital data endpoints (data modules) and analog data endpoints, and acoustic coupled modems. The analog data endpoint can be either a trunk or line circuit.

Data transmission between a digital data endpoint and an analog endpoint requires a conversion resource since the Digital Communications Protocol (DCP) format used by the data module is not compatible with the modulated signals of an analog modem. The conversion resource translates the DCP format into modulated signals and vice versa.

The Modem Pooling feature provides pools of conversion resources.

Integrated conversion resources (Versions 1 and 2) and combined conversion resources (Version 2) are available with System 75. The integrated type has functionality integrated on the TN758 Pooled Modem circuit pack, which provides two conversion resources and each one emulates a Trunk Data Module (TDM) cabled to a 212 modem. The combined type is a TDM/2 cabled to any TDM-compatible modem to provide a conversion resource.

When a conversion resource is required, the system queries the digital data module associated with the call to determine if its options are compatible with those supported by the pools. If the data module options are not compatible, the originating user receives intercept treatment. If the options are compatible, the system obtains a conversion resource from the appropriate pool. If a conversion resource is not available, the user receives reorder treatment. If all data calls, including analog, are not successfully established, the call will be disconnected within 15 seconds (handshake time-out).

In almost all cases, the system can detect the need for a conversion resource. Data calls originated from an analog data endpoint to a digital data endpoint require that the user indicate the need for a conversion resource, since the system considers an analog call origination as a voice call. This need is indicated by dialing the Data Origination Access Code before dialing the digital data endpoint. Use of Data Call Preindication before One-Button Transfer To Data is recommended when establishing data calls that use toll network facilities. Needed conversion resources are reserved before any toll charges are incurred.

Version 2 provides a "HOLD Time" parameter to specify the maximum time any conversion resource may be held but not used (while a data call is in queue).

Version 1 Modem Pooling supports asynchronous transmissions at 0 to 300 (LOW), 300 and 1200 bps, and synchronous transmission at 1200 bps.

Version 2 additionally supports the following configurations:

- IBM* bisynchronous protocols typically used in 3270 and 2780/3780 applications. Both require 2400 bps or 4800 bps, half duplex, synchronous transmission.
- Interactive IBM-TSO applications using 1200 bps, half duplex, asynchronous transmissions.
- DATAPHONE† II switched network modems supporting asynchronous and synchronous communications, and autobaud at 300, 1200, or 2400 bps.

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* Trademark of International Business Machines

† Service Mark of AT&T

- Version 2 can operate up to 19.2 kbps.
- Different pools (V2) can have different data transmission characteristics.

The following modem options are supported by the integrated (only) pool:

- Receiver Responds to Remote Loop
- Loss of Carrier Disconnect
- Send Space Disconnect
- Receive Space Disconnect
- CF-CB Common
- Options TDM and Modem, for Combined Conversion resources (on devices)
- Speed, Duplex, and Synch (administered)

Considerations

Modem Pooling offers a pool of conversion resources which increase data call flexibility. Conversion resources allow analog data endpoints, using modems, to communicate with digital data endpoints (using data modules). Also, pooling of conversion resources allows maximum use of such facilities.

Data Call Preindication is recommended for off-premises data calls involving toll charges.

On data calls between a data module and an analog data endpoint, Return-to-Voice releases the conversion resource and returns it to the pool. The voice terminal user is then connected to the analog data endpoint.

For traffic purposes, Version 2 accumulates data on modem pooling calls separate from voice calls. Measurements on the pools are also accumulated.

Version 1 can support one pool of 32 integrated conversion resources. Version 2 can support up to five pools; all five combined, integrated, or any mix. Each pool has a capacity of up to 32 conversion resources. Version 2 has a limit of 160 conversion resources for all five pools.

Use of Modem Pooling cannot be restricted. Also, queuing for conversion resources is not provided, although calls queued on a hunt group retain reserved conversion resources while queued.

Mixing of modems from different vendors within a combined pool should be avoided since a difference in transmission characteristics may exist. Mixing is possible, but satisfactory results are not guaranteed.

Data transmission characteristics (speed, duplex, and synchronization mode), as administered, must be identical to the TDM and modem optioning by the client.

Interactions

- Data Call Setup

  Data calls to or from a TDM cannot use Modem Pooling.

- Data-Only Off-Premises Extensions

  Modem Pooling is not possible on calls to or from a Data-Only Off-Premises Extension, when this type of digital data endpoint uses a TDM.

- Digital Multiplexed Interface (DMI)

  Data calls originated from a local analog data endpoint to a DMI trunk must dial the Data Origination Access Code to obtain a conversion resource. Data calls on DMI trunks to local analog data endpoints automatically obtain conversion resources.

Administration

Modem Pooling is assigned on a per-pool basis by the System Manager. The following items require administration.

- Conversion Resources—For integrated conversion resources, assign Pooled Modem circuit packs. For combined conversion resources, assign TDM and associated modems ports, speed (up to three speeds), and duplex and synchronization characteristics.
- HOLD Time (per pool basis)—Specify the maximum time (1 to 99 minutes) any conversion resource may be held and not used (while a data call waits in a queue). Default value is 5 minutes.
- Data Origination Access Code—Allow users to indicate a need for a conversion resource on an analog data call origination.

Hardware and Software Requirements

One TN758 Pooled Modem circuit pack is required for each two integrated conversion resources provided. With Version 2, combined conversion resource requires one port on the Digital Line circuit pack and one port on an Analog Line circuit pack.

MULTI-APPEARANCE PRESELECTION AND PREFERENCE

Description

Provides multi-appearance voice terminal users with options for placing or answering calls on selected appearances.

- Ringing Appearance Preference

  When a user lifts the handset to answer an incoming call, the system automatically connects the user to the ringing call appearance. If more than one call is incoming, the user is automatically connected to the eldest (first-in) ringing call appearance. The in-use (red) lamp tracks the ringing appearance and the answered appearance.

- Idle Appearance Preference

  When a user lifts the handset to place a call, the system automatically connects the user to an idle appearance even if an incoming call is ringing at another appearance. The in-use (red) lamp tracks an idle appearance when the handset is lifted.

- Preselection

  Before lifting the handset 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.

  The Preselection option overrides both Preference options. If the user does not lift the handset 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 handset within 5 seconds, the call is automatically placed. Preference only applies if there is a ringing call and if the user lifts the handset. Preference dictates whether the user is connected to the ringing call appearance or to an idle call appearance. If there is no incoming call, the user is automatically connected to an idle call appearance upon lifting the handset. This is true, regardless of the Preference option assigned.

Considerations

Multi-Appearance Preselection and Preference is used to select the call appearances to which users will be connected when they lift the handset.

Multi-appearance voice terminals can have from two to ten 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 nonpriority 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 ten 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 rings 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.

Interactions

- If Cover All Calls (part of the Call Coverage feature) 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 handset without being accidentally connected to a call which should be screened.

Administration

The Idle Appearance Preference option is administered on a per-terminal basis by the System Manager. If Idle Appearance Preference is not administered, the voice terminal will have Ringing 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 Ringing Appearance Preference. No administration is required for preselection.

Hardware and Software Requirements

No additional hardware or software is required.

MULTIPLE LISTED DIRECTORY NUMBERS

Description

Allows a publicly published number for each incoming and two-way (incoming side) foreign exchange (FX) and local central office (CO) trunk group assigned to the system. Also allows up to eight Direct Inward Dialing (DID) numbers to be treated as Listed Directory Numbers (LDNs).

When a CO or FX LDN is called, a trunk group is accessed. The trunk group then routes the call to the incoming destination designated for that trunk group. The incoming destination for an FX or CO trunk group can be one of the following:

- Attendant group
- Direct Department Calling (DDC) group
- Uniform Call Distribution (UCD) group
- Remote Access

All DID LDN calls route directly to the attendant group.

Considerations

Multiple Listed Directory Numbers provide publicly published numbers for a business. These numbers allow public access to an attendant. LDNs are also useful when it is necessary that the public be able to contact a particular DDC or UCD group. The feature can also be used for Remote Access.

A unique display for incoming call identification can be provided for each LDN, including the DID numbers.

Interactions

If Night Service has been activated and a night console is not assigned or is not operational, incoming LDN calls route as follows:

- DID LDN calls route to a designated DID LDN night extension. If no DID LDN night extension is designated, DID LDN calls route to the attendant.
- Other incoming calls on trunk groups route to the night destination specified for the trunk group. If the night destination is the attendant, calls route to the DID LDN night extension, if specified. If no DID LDN night extension is specified, calls route to the attendant. If no night destination is specified for the trunk group, the calls route to the normal incoming destination for that trunk group. If that destination is an attendant, calls route to the DID LDN night extension.
- Internal calls and coverage calls to the attendant route to the DID LDN night extension.

Administration

Multiple Listed Directory Numbers is administered by the System Manager. The following items require administration:

- Incoming destination for each CO trunk group and each FX trunk group used for LDNs

- Up to eight DID LDNs
- DID LDN night extension
- A unique name for each LDN (optional, for display purposes)

Hardware and Software Requirements

No additional hardware or software is required.

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.

The system provides automatic access to the music source.

Considerations

The music provided by Music-on-Hold Access lets the waiting party know that he or she is still connected. Waiting parties are less likely to hang up. This results in a greater number of completed calls.

If a multiple-party connection is on hold, waiting in queue, or parked, music is not provided.

The number of parties that can be connected to Music-on-Hold Access simultaneously is not limited.

Interactions

When any one of the following features is activated, music is provided when one party is waiting or held:

- Hold
- Conference—Terminal
- 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.

Administration

Music-On-Hold Access is administered on a per-system basis by the System Manager. The only administration required is to assign the port number used to provide the feature.

Hardware and Software Requirements

Requires the music source and one port on a TN763 Auxiliary Trunk circuit pack. Also, a 36A voice coupler may be required to provide an interface and system protection for the music source. No additional software is required.

NETWORK ACCESS—PRIVATE

Description

Allows calls to be connected to the following types of networks:

- Common Control Switching Arrangement (CCSA)
- Electronic Tandem Network (ETN)
- Enhanced Private Switched Communications Service (EPSCS)
- Tandem Tie Trunk Network (TTTN)

A private network provides call routing over facilities dedicated to the client.

Considerations

With Network Access—Private, calls can be made to other switching systems without having to use the public network.

A total of 50 (V1) trunk groups or 60 (V2) 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 calling 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.

Interactions

None.

Administration

Network Access—Private is administered by the System Manager. The following items require administration:

- Tie trunk groups used with private networks.
- Whether or not access to CCSA and/or EPSCS off-network calling is provided. (This assignment is made on a per-COR basis.)

Hardware and Software Requirements

Requires one port on a TN760B or DS1 Tie Trunk circuit pack for each trunk assigned. No additional software is required.

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

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 dialed for manual route selection. Long-distance carrier access codes can be dialed to select particular carriers.

Interactions

None.

Administration

Network Access—Public is administered by the System Manager. All trunk groups used for Network Access—Public must be administered.

Hardware and Software Requirements

Requires one port on a TN747B CO Trunk circuit pack for each trunk assigned. No additional software is required.

NIGHT SERVICE—NIGHT CONSOLE SERVICE

Description

Directs all calls for the primary and daytime attendant consoles to a night console.

Night Service—Night Console Service is activated when an attendant presses the Night button on the primary attendant console. Night Service is deactivated by pressing the Night button again. When Night Service is activated, all attendant-seeking calls and calls waiting in queue are directed to the night console.

Considerations

Night Service—Night Console Service calls to the attendant group are still handled by an attendant, even though the primary and daytime attendant consoles are out of service.

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.

Interactions

None.

Administration

Night Service—Night Console Service is administered by the System Manager. The only administration required is the assignment of a night console. The night console must be administered identically to the primary console.

Hardware and Software Requirements

Requires an attendant console. No additional software is required.

NIGHT SERVICE—NIGHT STATION SERVICE

Description

Redirects incoming attendant-seeking trunk calls to designated extension numbers whenever the system is placed in Night Service.

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.

When the above conditions have been met, incoming calls to the attendant route as follows:

- Direct Inward Dialing (DID) Listed Directory Number (LDN) calls route to a designated DID LDN night extension.
- Internal calls to the attendant route to the DID LDN night extension.
- Incoming calls on trunk groups (other than DID trunk groups) which have the attendant as their destination route to the night destination specified for the trunk group. If no night destination is specified, the calls route to the DID LDN night extension.

When Night Station Service is activated, all trunk and internal calls to the attendant (other than calls redirected via Call Coverage or Call Forwarding All Calls) route to either the DID LDN night extension or the trunk group's specified night destination as discussed above. A different extension number can be assigned as the night destination for each incoming central office, foreign exchange, or 800 Service trunk group. Both the DID LDN night extension and the extension number assigned as a trunk group's night destination can be a voice terminal or an answering group, that is, Direct Department Calling group, Uniform Call Distribution group, or Terminating Extension Group.

Calls redirected to the attendant via Call Coverage or Call Forwarding All Calls do not route to the DID LDN night extension. These calls enter the attendant queue, and can be answered via the Trunk Answer From Any Station feature, if administered.

Considerations

Night Station Service provides for the answering of attendant-seeking calls when all attendant consoles are out of service due to Night Service activation.

When the Night Station Service feature is active but night station extension numbers have not been established, the Trunk Answer From Any Station feature can be activated.

Interactions

- Call Coverage (V1)

  A call routed to the DID LDN night extension via Night Station Service does not go to coverage, even if the coverage criteria of the DID LDN night extension is met.

- Call Forwarding All Calls

  A call routed to the DID LDN night extension via Night Station Service does not forward to another extension, even if Call Forwarding All Calls has been activated at the DID LDN night extension.

- 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.

- Timed Reminder

  Timed Reminder Calls returning to a console which has been placed in Night Service and has an assigned DID LDN night extension will not be redirected to the DID LDN night extension, but will be dropped.

Administration

Night Station Service is assigned by the System Manager. The following items require administration:

- DID LDN night extension
- Trunk group night destination (per trunk group)

Hardware and Software Requirements

No additional hardware or software is required.

NIGHT SERVICE—TRUNK ANSWER FROM ANY STATION

Description

Allows voice terminal users to answer all incoming attendant-seeking 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.

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.

Considerations

When Trunk Answer From Any Station is activated, any user can answer the attendant-seeking trunk call. Even though an attendant is not available, the call is still answered. This reduces the number of lost calls.

Interactions

Inward-restricted voice terminals can activate TAAS for incoming trunk calls. Night Service features override Inward Restriction.

Calls which are redirected to the attendant via the Call Coverage and Call Forwarding All Calls features while the Night Station Service feature is activated can be answered via TAAS.

Administration

Trunk Answer From Any Station (TAAS) is administered on a per-system basis by the System Manager. The following items require administration:

- Dial access code for TAAS (to answer a call)
- Port for the ringing device

Hardware and Software Requirements

Requires a ringing device and one port on a TN742 Analog Line circuit pack. No additional software is required.

OFF-PREMISES STATION

Description

Allows a remotely located FCC-registered analog voice terminal to be connected to the system.

Considerations

Off-Premises Stations are useful whenever it is necessary to have a voice terminal located away from the main location.

The maximum loop distance for Off-Premises Stations is 14,000 feet, without repeaters.

Interactions

The Distinctive Ringing feature might function improperly at an Off-Premises Station due to the distance. However, the Distinctive Ringing feature can be disabled when the Off-Premises Station is administered. If the Distinctive Ringing feature is not used with an Off-Premises Station, the terminal will receive 1-burst ringing for all calls.

Administration

Off-Premises Stations are administered by the System Manager.

Off-Premises Stations are administered the same as on-premises voice terminals.

Hardware and Software Requirements

Requires cross-connecting capabilities and one port on a TN742 Analog Line circuit pack. No additional software is required.

PERMANENT SWITCHED CALLS (V2)

Description

Provides for calls that are intended to be present while the system is active. The specified calls are automatically placed when the system is started or restarted, and remain active until the system becomes inactive.

If a call is inadvertently dropped, the system attempts to reestablish the call. The system attempts to reestablish all nonactive Permanent Switched Calls (PSCs) at 2-minute intervals. These attempts continue until all calls are completed.

Considerations

PSCs make the system responsible for placing and maintaining calls that should be present while the system is active. Only data calls can be placed in this manner.

The system can support up to 18 PSCs, indicated in a PSC list. Each PSC listed can contain up to 36 characters of dialing information (see "DATA CALL SETUP" for the dialing format).

Interactions

- Call Forwarding

  The called endpoint should not have Call Forwarding activated since the endpoint should be a final destination.

- Data Restriction

  All PSCs should be administered with Data Restriction set to prevent imposing system tones on the call. Such tones interfere with data transmission.

Administration

PSCs are assigned on a per-system basis by the System Manager. The following items require administration:

- Call List—Establish or change the list of PSCs.
- Classes of Restriction—Determine Classes of Restriction so that only PSC endpoints are allowed to call other PSC endpoints. Other users should be denied permission to call a PSC endpoint.

A PSC between the Interface 3 circuit pack and a data module does not appear in the call list, but is administered as a link assignment.

A PSC can be dropped temporarily, for maintenance purposes, by disabling the call in administration at the System Access Terminal (SAT). The call remains in the list, but is dropped until it is enabled.

Hardware and Software Requirements

PSCs do not require additional hardware or software.

PERSONAL CENTRAL OFFICE LINE

Description

Provides a dedicated trunk for direct access to or from the public network for multi-appearance voice terminal users.

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 rings 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.

Central office (CO), foreign exchange (FX), and Wide Area Telecommunications Service (WATS) trunks can be assigned to this feature.

PCOLs are not assigned a Class of Restriction.

Considerations

PCOLs are useful to users such as 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).

The system will support 25 PCOLs. These lines (trunks) are not included in the trunk groups supported by the system. They are, however, included in the 200-trunk system limit.

Interactions

- Abbreviated Dialing

  Abbreviated Dialing can be used with the PCOL feature. However, the accessed lists are associated with the individual voice terminals.

- Bridged Call Appearance

  If a user is active on his or her primary extension number on a PCOL call, bridged call appearances of that extension number cannot be used to bridge onto the call. The call can only be bridged onto if another voice terminal is a member of the same PCOL group and has a PCOL button.

- Leave Word Calling

  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.

- Station Message Detail Recording

  The Station Message Detail Recording (SMDR) feature can be activated for PCOL calls, but the SMDR record will not specifically identify the call as PCOL. A PCOL call can however be identified by the trunk access code used on the call. The call will be recorded to the extension number assigned to the voice terminal where the call was originated or answered.

- Temporary Bridged Appearance

  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 following features cannot be used with the PCOL feature:

- Automatic Route Selection
- Call Forwarding All Calls
- Ringback Queuing

Administration

PCOLs are administered by the System Manager. The following items require administration:

- 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.

PERSONALIZED RINGING (V2)

Description

Allows users of certain voice terminals to uniquely identify their own calls. Each user can choose one of a number of possible ringing patterns.

The eight ringing patterns are tone sequences consisting of different combinations of three tones. The eight different combinations are listed below. The tones are heard in the sequence given for each combination.

- 750 Hz, 750 Hz, 750 Hz (normal ringing)
- 1060 Hz, 1060 Hz, 1060 Hz
- 530 Hz, 530 Hz, 530 Hz
- 530 Hz, 1060 Hz, 1060 Hz
- 1060 Hz, 1060 Hz, 530 Hz
- 1060 Hz, 530 Hz, 530 Hz
- 1060 Hz, 530 Hz, 1060 Hz
- 530 Hz, 1060 Hz, 530 Hz

Each ringing pattern requires 0.6 second (0.2 second for each tone) in the 5.2 seconds ringing cycle. This 0.6 second of personalized ringing occurs at the given time during the ringing cycles of the following types of calls (times indicated are in seconds):

- Internal voice terminal, internal tie trunk, and remote access calls

  0.6 on, 0.6 personalized ringing, 4.0 off

- Attendant extended, attendant originated, and incoming trunk calls, including external tie trunk calls

  0.2 on, 0.4 off, 0.6 personalized ringing, 4.0 off

- Automatic Callback, Priority Calling, and Ringback Queuing Callback calls

  0.1 on, 0.1 off, 0.1 on, 0.3 off, 0.6 personalized ringing, 4.0 off

- Intercom Calls (7404D and 7407D voice terminals only)

  0.6 personalized ringing, 4.6 off

One of the eight ringing patterns can be specified for each eligible voice terminal (7303S and 7305S) by the System Manager. In addition, the 7404D, 7407D, and 7103A programmable voice terminal users have the capability of setting their own ringing pattern. The 7404D user can select the desired ringing pattern via the given menu options. The 7407D user can select the desired ringing pattern by using the Select Ring and PR (Personalized Ringing #) buttons. The 7103A programmable voice terminal user can select one of four ringing patterns via a slide switch on the voice terminal.

Considerations

With Personalized Ringing, users working closely in the same area can each specify a different ringing pattern. This enables the users to distinguish their own ringing voice terminal from other voice terminals in the same area.

Up to eight different ringing patterns are available.

Interactions

The normal ringing cycles are altered as described in the Description of this feature.

Administration

Personalized Ringing is administered for the 7303S and 7305S voice terminals on a per-voice terminal basis by the System Manager. Administration consists of assigning one of the eight ringing patterns to each eligible voice terminal. Also, a 7404D, 7407D, or 7103A programmable voice terminal user can specify his or her own ringing pattern. The user specified ringing pattern for a 7404D or 7407D, however, is lost in the event of a power failure.

Hardware and Software Requirement

No additional hardware or software is required.

POWER FAILURE TRANSFER

Description

Provides service to and from the local telephone company central office (CO) if power fails.

Considerations

Power Failure Transfer provides for the use of certain voice terminals during a power failure to access the local CO. These voice terminals can be used to make important or emergency calls.

From 5 to 35 (maximum) voice terminals can be connected to from 5 to 35 CO trunks for the Power Failure Transfer feature. The Power Failure Transfer feature is available in multiples of five.

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.

Interactions

During the Power Failure Transfer mode, no other system features can be activated.

Administration

None required.

Hardware and Software Requirements

Requires one emergency transfer panel for every five or six trunks assigned to Power Failure Transfer, depending on the transfer panel used. Two emergency transfer panels are available:

- Z1A Panel—Each unit serves up to six power failure transfer terminals. A ground-start key is required at each preselected voice terminal when ground-start trunks are used.
- Porta-Systems Model 573-5 Panel—Each unit serves up to five failure transfer terminals. The unit provides automatic ground start.

No additional software is required.

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, including the call appearance normally reserved for call origination, are active, the caller receives busy tone. If the call appearance normally reserved for call origination is the only idle call appearance, an incoming priority call will ring at that call appearance.

A user activates priority calling by dialing a Priority Calling access code, followed by the desired extension number.

Whether or not a user can activate Priority Calling is determined by the user's Class of Service.

Considerations

With Priority Calling, a voice terminal user can ring another voice terminal with a distinctive signal that tells the called party the incoming call requires immediate attention. The called party can then handle the call accordingly.

Call Coverage Consult calls and callback calls from Automatic Callback and Ringback Queuing are Priority Calling calls.

Interactions

- Ringing

  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.

- Bridged Call Appearance

  A Bridged Call Appearance recieves ringing on a priority call the same as the called primary extension.

- 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 ringing 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 Call Waiting Termination feature is not assigned to the voice terminal. The active single-line voice terminal user receiving the call hears a distinctive 3-burst priority Call Waiting tone.

- Dial Access to Attendant

  A Priority Calling call cannot be originated to the attendant. However, the attendant can originate Priority Calling calls.

Administration

Priority Calling is administered by the System Manager. The following items require administration:

- Priority Calling access code
- Permission to activate Priority Calling (per Class of Service)

Hardware and Software Requirements

No additional hardware or software is required.

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

Privacy—Attendant Lockout provides privacy for parties on a multi-party call held on the console. The held parties can hold a private conversation without being interrupted by the attendant.

Interactions

- Trunk-to-Trunk Transfer

  Privacy—Attendant Lockout does not function when a call using the Trunk-to-Trunk Transfer feature is held on the console.

- Individual Attendant Access (V2)

  Privacy—Attendant Lockout applies only to attendant group calls. Individual attendant calls are not affected.

Administration

Privacy—Attendant Lockout is administered on a per-system basis by the System Manager. The only administration required is to administer whether or not attendant lockout is active.

Hardware and Software Requirements

No additional hardware or software is required.

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.

Exclusion is activated by pressing the Exclusion button on a per-call basis. If the Exclusion button is pressed while other users are bridged onto the call, the other users are dropped from the call. The Privacy—Manual Exclusion feature is automatically deactivated when the Exclusion button is pressed a second time or when the party who activated Privacy—Manual Exclusion is dropped from the call.

Privacy—Manual Exclusion is used with the Personal Central Office Line, Terminating Extension Group, and Bridged Call Appearance features.

Considerations

Privacy—Manual Exclusion prevents users who have an appearance of another terminal's extension from bridging onto that extension.

Interactions

- Bridged Call Appearance

  When Privacy—Manual Exclusion is activated, all other users are prevented from bridging onto the active call.

Administration

Privacy—Manual Exclusion is administered on a per-voice terminal basis by the System Manager. The only administration required is the assignment of the Exclusion button to the desired voice terminals.

Hardware and Software Requirements

No additional hardware or software is required.

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 twice.

Considerations

Recall Signaling provides a single-line voice terminal user with the ability to place a call on hold and use the voice terminal for other operations. The user can then return to the held call.

Recall Signaling cannot be used to answer a waiting call (Version 1 only).

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

RECORDED TELEPHONE DICTATION ACCESS

Description

Permits voice terminal users, including Remote Access and incoming tie trunk users, to access dictation equipment.

The dictation equipment is accessed by dialing an access code or extension number (depending on how the feature is administered). After the dictation equipment is accessed, the start/stop function can be voice- or dial-controlled. Other functions such as initial activation and playback are controlled by additional dial codes. The specific dial codes depend on the dictation equipment selected.

Considerations

This feature provides dictation equipment which users can access at their own convenience. Dictation can be recorded, corrected, and played back by the user.

Interactions

The Recorded Telephone Dictation Access feature cannot be used with the following features:

- Automatic Route Selection
- Conference—Attendant
- Conference—Terminal

Administration

Recorded Telephone Dictation Access is administered on a per-system basis by the System Manager. The following items require administration:

- 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.

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, as 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, is 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.

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, 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

Remote Access provides a caller with access to the system and its features from the public network. An executive can make business calls from home or use the Recorded Telephone Dictation Access feature to dictate a letter.

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.

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.

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.

Administration

Remote Access is administered by the System Manager. The following items require administration:

- Extension number
- Barrier code length (from four to seven digits)
- Barrier codes
- COR (per Barrier code)
- Trunk groups

Hardware and Software Requirements

If Remote Access is not available via DID, dedicated trunks must be provided. No additional software is required.

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

Controlled Restriction gives the attendant control of outward and total restriction for voice terminals or groups of voice terminals.

All voice terminals with the same COR are affected by a group restriction.

Interactions

Controlled Restriction overrides restrictions assigned by a COR.

Administration

Controlled Restriction is administered on a per-system basis by the System Manager. The only administration required is the assignment of Controlled Restriction Activation and Deactivation access codes. Separate access codes are needed for outward and total restriction.

Hardware and Software Requirements

No additional hardware or software is required.

RESTRICTION—MISCELLANEOUS TERMINAL

Description

Restricts callers at specified voice terminals from accessing certain other voice terminals.

Miscellaneous Terminal Restrictions can be used whenever it is undesirable for users at certain voice terminals to access other specific 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.

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.

Administration

Miscellaneous Terminal Restriction is administered via the Class of Restriction feature by the System Manager. The only administration required is the permission for each COR to access other CORs.

Hardware and Software Requirements

No additional hardware or software is required.

RESTRICTION—MISCELLANEOUS TRUNK

Description

Restricts users at specified voice terminals from accessing certain trunk groups, such as Wide Area Telecommunications Service (WATS).

For a detailed description of Miscellaneous Trunk Restrictions, see the Class of Restriction (COR) description.

Considerations

Miscellaneous Trunk Restriction can be used whenever it is necessary to restrict users at certain voice terminals from accessing specific trunk groups.

The Miscellaneous Trunk Restriction is controlled by the 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.

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.

Administration

Miscellaneous Trunk Restriction is administered via the Class of Restriction feature by the System Manager. The only administration required is the permission for each COR to access other CORs.

Hardware and Software Requirements

No additional hardware or software is required.

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).

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.

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. (In areas where area codes can also serve as office codes, the system requires the prefix 1 on area code calls to differentiate them from local calls. In this case, local calls with a 0 or 1 as the second digit are not subject to toll restriction.)
- 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.

If the System 75 is connected to a central office that uses a step-by-step switch, all seven of the digits which are normally dialed for a local call may not be required by the central office to route the call. For example the central office may only require the last five of the normally dialed seven digits. If all seven digits are dialed, the step-by-step switch uses digit absorption to absorb the unneeded digits. Digit absorption can be provided within the System 75 to emulate the absorption at the central office. This prevents users from bypassing code and toll restriction by dialing uneeded digits. For example, assume that the central office absorbs leading 7s before processing a number and that a toll-restricted user wants to call someone in area code 201. The user could dial 77-1-201 plus seven more digits. The Toll Restriction feature would not recognize the call as a toll call and the central office would route the call. With digit absorption, the 77 is absorbed by the system before Toll Restriction is used. Thus, the call would be denied, as intended. Up to five digit absorption lists can be assigned.

Considerations

Toll and/or Code Restriction is used whenever it is necessary to restrict users at certain voice and data terminals from making calls to certain central offices, area codes, and/or service codes.

The Allowed Calls List can include up to ten 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.

Two code-restriction tables are established. One table lists certain 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.

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.

Administration

Toll or Code Restriction (but not both) is administered by the System Manager to each foreign exchange and central office trunk on a trunk group basis.

Toll or Code Restriction (but not both) is administered by the System Manager 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
- Data Modules on a per-module basis

Other items that can be administered are as follows:

- Allowed Calls List containing up to ten 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
- Digit absorption lists containing absorption treatment of each digit 0-9

Hardware and Software Requirements

No additional hardware or software is required.

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.

Calls can redirect to an inward-restricted voice terminal. The Class of Restriction (COR) of the originally called extension number is the only one checked.

Considerations

Inward Restriction is used whenever it is necessary that users at certain voice terminals receive only internal calls from other voice terminals.

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.

Administration

Inward Restriction is administered by the System Manager to voice terminals by the Class of Restriction feature.

Hardware and Software Requirements

No additional hardware or software is required.

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.

Calls can redirect to a voice terminal assigned this feature. The Class of Restriction (COR) of the originally called extension number is the only one checked.

Considerations

Manual Terminating Line Restriction is used whenever it is necessary to have users at certain voice terminals receive only calls from an attendant.

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.

Administration

The Manual Terminating Line Restriction feature is administered by the System Manager to voice terminals by the Class of Restriction feature.

Hardware and Software Requirements

No additional hardware or software is required.

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. A voice terminal can, however, activate certain features by dialing the assigned feature access codes.

Considerations

Origination Restriction is used whenever a voice terminal is to be used only for answering incoming calls.

Interactions

The Controlled Restriction overrides restrictions assigned by the Class of Restriction.

Administration

The Origination Restriction feature is administered by the System Manager to voice terminals by the Class of Restriction feature.

Hardware and Software Requirements

No additional hardware or software is required.

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

Outward Restriction is used whenever it is desired that a voice terminal make only internal calls.

The attendant or an unrestricted voice terminal user can extend a call to an outside number for the outward-restricted voice terminal user.

Interactions

The Controlled Restriction overrides restrictions assigned by the Class of Restriction.

Administration

The Outward Restriction feature is administered by the System Manager to voice terminals by the Class of Restriction feature.

Hardware and Software Requirements

No additional hardware or software is required.

RESTRICTION—VOICE TERMINAL—TERMINATION

Description

Restricts voice terminal users on specified extension numbers from receiving any calls. Voice terminal users can originate calls.

Considerations

Termination Restriction is used whenever a voice terminal is to be used only for making calls.

Interactions

The Controlled Restriction overrides restrictions assigned by the Class of Restriction (COR).

Administration

The Termination Restriction feature is administered by the System Manager to voice terminals by the COR.

Hardware and Software Requirements

No additional hardware or software is required.

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 waits for 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.

Queuing can be specified for any outgoing only trunk group, or for the outgoing direction of a 2-way trunk group.

Considerations

With Ringback Queuing, users do not have to keep trying to access a trunk group when all trunks in the group are busy. This feature provides for the caller of a busy trunk group to automatically be called back when a trunk becomes available.

Queuing can reduce the number of trunks required.

The system allows a maximum of 100 (V1) or 120 (V2) calls in queue for all the trunk groups in the system.

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 ringing 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.

Incoming tie trunk calls cannot queue on an outgoing trunk group. The system does not know the calling number and cannot originate the callback call.

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:

- Bridged Call Appearance

  Ringback Queuing is not provided on calls originated from a bridged call appearance.

- 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.

Administration

Ringback Queuing is administered by the System Manager. The following items require administration:

- Callback call no-answer time-out (from 2 to 9 ringing cycles)
- Automatic Callback button (per multi-appearance voice terminal)
- Ringback Queuing cancellation code
- Queue length (per outgoing trunk group)

Hardware and Software Requirements

No additional hardware or software is required.

ROTARY DIALING (V2)

Description

Allows use of rotary dialing voice terminal at a System 75 location.

When a number is dialed at a rotary dialing voice terminal, the voice terminal outpulses at a rate of 10 pulses per second. Each digit dialed sends out the corresponding number of pulses. For example, dialing a 7 results in 7 pulses being sent from the voice terminal. Version 2 software recognizes that the voice terminal is rotary when the user lifts the handset, and expects to receive dial pulses instead of tones.

Considerations

With Rotary Dialing, existing rotary dialing voice terminals can be used in situations where very simple call processing functions are required.

Any functions requiring the * and # symbols cannot be performed on a rotary dialing voice terminal.

Interactions

None.

Administration

Rotary Dialing voice terminals must be administered as a 500 set.

Hardware and Software Requirements

No additional hardware is required. V2 software is required.

SEND ALL CALLS

Description

Allows users to temporarily direct all incoming calls to coverage regardless of the assigned Call Coverage redirection criteria. Send All Calls also 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. It is deactivated by pressing the button a second time or by dialing the deactivate access code.

Details of how Send All Calls is used in conjunction with Call Coverage are given in the Call Coverage feature description, elsewhere in this section.

Considerations

Send All Calls gives a user the option to have all incoming calls sent directly to coverage. This is useful when a user needs to be away from his or her desk temporarily.

Interactions

Send All Calls is used only in conjunction with the Call Coverage feature.

Administration

Send All Calls is administered by the System Manager. The following items require administration:

- Send All Calls button (per voice terminal)
- Activate and Deactivate access codes for Send All Calls (per system)

Hardware and Software Requirements

No additional hardware or software is required.

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

This feature provides quicker service to remote touch-tone receiving facilities.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

SMDR ACCOUNT CODE DIALING

Description

Allows certain calls to be associated with a particular project or account number. This is accomplished by dialing specified account codes before making outgoing calls. This information is recorded by the Station Message Detail Recording (SMDR) feature and can be used later for accounting and/or billing purposes.

To associate an account code with a particular call, a user first dials an SMDR access code. The user then dials the desired account code, which can contain up to 5 digits (V1) or 15 digits (V2). The user then dials the desired trunk access code, Automatic Alternate Routing (AAR) access code, or Automatic Route Selection (ARS) access code.

SMDR Account Code Dialing is optional in Version 1. The user may or may not dial an account code, and the call is not affected. With Version 2, SMDR Account Code Dialing can be optional or mandatory (forced). With Version 2, forced entry of account codes can be assigned for any of the following:

- All toll calls—those calls which have a 0 or 1 as one of the first two digits of the called number, except service calls (for example, 911 and 411), directory assistance calls, and 800 Service calls.

  This affects all calls made by AAR, ARS, or trunk access codes.

- Individual Classes of Restriction (CORs)

  If forced entry of account codes is assigned to a specific COR, any voice terminal assigned that COR must dial an account code before making outgoing trunk calls. Any trunk group assigned that COR cannot be accessed until an account code is dialed. If a call is being routed via AAR or ARS, account code checking is not done on the COR.

With Version 2, any time an account code is required and the user does not enter an account code, intercept tone is heard. An account code is never required for the following:

- Attendant originated call
- Busy verification of a trunk by an attendant or voice terminal user
- Distributed Communications System (unless required by the trunk group's COR)
- Personal Central Office Line
- Remote Access without Barrier codes
- Trunk-To-Trunk Transfer

Considerations

SMDR Account Code Dialing provides an easy method of allocating the costs of specific calls to the correct project, department, etc. Call information is recorded by the SMDR feature for this purpose.

Account Code length can be up to 5 digits (V1) or 15 digits (V2).

The validity of the entered account codes cannot be checked by the system.

Interactions

- Automatic Alternate Routing (V2) and Automatic Route Selection (V2)

  If a trunk group is accessed via AAR or ARS, the trunk group's COR is not used to determine if an account code needs to be entered.

- Busy Verification of Terminals and Trunks (V2)

  An attendant or voice terminal user is never required to enter an account code when making a busy verification.

- Call Forwarding All Calls (V2)

  If a user is required to enter an account code to call a particular destination, the calls cannot be forwarded to that destination.

- Last Number Dialed

  The SMDR access code and account code dialed are stored as part of the Last Number Dialed. However, some digits may be lost due to the limit on the number of digits stored for this feature.

Administration

SMDR Account Code Dialing is administered by the System Manager. The following items require administration for forced entry of account codes (V2):

- Whether or not all toll calls require account code entry (per system)
- Whether or not each individual COR requires account code entry

Hardware and Software Requirements

No additional hardware is required. Optional SMDR Account Code Dialing software is required.

SPEECH SYNTHESIS (V2)

Description

Allows attendants, voice terminal users, and remote access users to retrieve Leave Word Calling (LWC) and Call Coverage messages in the form of a voice output.

Speech Synthesis is only used for the retrieval of messages. It can be used to retrieve a user's own messages or to retrieve messages for another user. However, a different user's messages can only be retrieved by a user at a voice terminal or attendant console in the associated coverage path, by an administered systemwide message retriever, or by a Remote Access user when the extension number and associated security code are known.

Messages are protected by restricting unauthorized users from retrieving 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. The systemwide access codes and security code used for the Lock and Unlock functions are the same as those used for LWC message retrieval by display. A status lamp can be assigned to show the locked or unlocked status of a terminal.

Speech Synthesis is activated by dial access code. Separate access codes are used for Message Retrieval and Coverage Message Retrieval (someone else's messages). Speech Synthesis message retrieval is activated as follows:

- To retrieve one's own messages:

  Dial the access code for Speech Synthesis retrieval of LWC messages. Then, dial a # sign to indicate that the extension whose messages are to be retrieved is the dialing extension, or enter a specific extension and corresponding password (same as the security code used for the Lock and Unlock functions).

- To retrieve someone else's messages:

  Dial the access code for Speech Synthesis retrieval of Call Coverage messages, followed by the extension of the user (within the same coverage path) whose messages are to be retrieved.

After Speech Synthesis has been activated, message retrieval may not be allowed for the following reasons:

- Speech synthesizer circuit pack fails or all voice channels are busy—Reorder tone is provided.
- If the system has an Applications Processor (AP) and the AP link is down—"Messages are not available, please try later" is heard and the user is disconnected.
- An attempt is made to retrieve a message for a user not in the same coverage path or an invalid password has been entered—"Message Retrieval Denied" is heard and the user is disconnected.
- Message retrieval for the terminal whose messages are to be retrieved is locked—"Message Retrieval is locked" is heard and the user is disconnected.
- No message is heard within 10 seconds—"Press 4 for help" is heard and if no messages are heard within another 10 seconds, the user is disconnected.

When Speech Synthesis has successfully been activated, voice messages are heard as follows:

- If no messages are left, one of the following is heard:
  - "No messages for [abc]"—where [abc] are the initials associated with the user whose messages are to be retrieved.
  - "No messages for extension [xxxxx]"—where [xxxxx] is the extension number when no name is associated with the user whose messages are to be retrieved.
- If one message, or more, is left, one of the following is heard:
  - "[n] messages(s) for [abc]"—where [n] is the number of messages left for the user whose messages are being retrieved.
  - "[n] messages(s) for [xxxxx]"
  - "Messages for [abc]"—when an AP is provided
  - "Messages for [xxxxx]"—when an AP is provided

When a user's initials are given in a voice message, as described above, the initials are computed in the "first name(s) followed by last name" order. If a single name (Brown, for example) is administered, the entire name is spelled out.

When a user has activated Speech Synthesis and has heard the number of messages that has been left, one of the following functions, as described below, can be performed:

- NEXT—dial #
- REPEAT—dial 5 or *5
- DELETE—dial 3 or *3
- CALL—dial 6 or *6
- HELP—dial 4 or *4

If the NEXT function is selected, the next message, if there is one, is played. The following messages may be heard:

- "No more messages for [abc]" or "No more messages for extension [xxxxx]"—when there are no more messages.
- "No more messages, press pound to repeat messages"—when already at the last message and no AP-based messages are left.
- "Please call Message Center for more messages"—when already at the last message and there are still AP-based messages left.-
- "[abc] called [n] times, last message at [time] [date] extension [xxxxx]"—when the user has received more than one message from the same caller; [abc] are the caller's initials; [n] is the number of times called; [time] is expressed an hour followed by minute (for example, "Nine Thirty-Five PM"); [date] is expressed as month followed by day (for example, "July third") or "today," if applicable; [xxxxx] is the calling party's extension number.
- "extension [xxxxx] called [n] times, last message at [time] [date]"—when the user has received more than one message from the same caller and no name is associated with the extension.
- "[abc] called at [time] [date] extension [xxxxx]" or "extension [xxxxx] called at [time] [date]"—when only one message has been left by a particular extension number.

If the REPEAT function is selected, the synthesized voice repeats the previously retrieved message with the calling party's name spelled out (instead of initials). The last name is spelled out first followed by a pause, then the first name(s) spelled out (with pauses between first names if there is more than one). If no name is associated with the extension, the current message is repeated. If a message has not been retrieved, "Press pound for the next message" is heard.

If the DELETE function is selected, the previously retrieved message is deleted and the user hears "Message is deleted." If no message was previously retrieved, "Press pound for next message" is heard. After a message is deleted, the user can still place a call to calling party of the deleted message, via the CALL function, as long as no other function has been entered between DELETE and CALL.

If the CALL function is selected, the extension of the calling party from the previously retrieved message is called. If no message was previously retrieved, "Press pound for next message" is heard. Otherwise, the call is initiated and the user leaves the message retrieval mode.

If the HELP function is selected, the following speech synthesized message is heard: "Press pound for the next message, press 3 to delete the message, press 4 for help, press 5 to repeat the message, press 6 to place the call."

The system expects the user to enter a function after each voice message. If a function is not entered before a specified time or if an invalid digit (digit other than #, *, 3, 4, 5, or 6) is dialed, the voice message "Press 4 for help" is heard. If no other input is entered within 10 seconds after this message, the user is automatically disconnected.

Speech Synthesis can be deactivated to get out of the voice message retrieval mode by doing any of the following:

- Hang up.
- Press the Drop of Disconnect button.
- Activate CALL function.

Considerations

With Speech Synthesis, a display-equipped voice terminal is not required to retrieve messages. Authorized users on any touch-tone terminal can retrieve messages. This results in significantly reduced traffic to the Message Centers and systemwide message retrievers.

The number of simultaneous Speech Synthesis users possible is dep ndent on the number of speech synthesizer circuit packs used in the system.

Speech Synthesis cannot be accessed by rotary dialing voice terminals.

Certain voice terminals and attendants can be designated for systemwide message retrieval. These systemwide retrievers are the same as those used for display message retrieval and have the same privileges.

When a terminal is in the Speech Synthesis mode, it cannot be used to make calls or access other features.

Interactions

- Bridged Call Appearance

  Activation of Speech Synthesis on a Bridged Call Appearance functions the same as if it was activated by the primary extension associated with the Bridged Call Appearance.

- Call Forwarding

  A forwarded-to user cannot retrieve messages for a forwarding user unless both users are in the same coverage path.

- Call Pickup

  A user cannot retrieve messages for a member of his or her Call Pickup group, unless both users are in the same coverage path.

- Leave Word Calling

  Speech Synthesis enhances Leave Word Calling (LWC) by allowing any authorized touch-tone terminal user to retrieve messages.

Administration

Speech Synthesis is administered by the System Manager. The following items require administration:

- Speech Synthesis Access Code for LWC message retrieval (per system)
- Speech Synthesis Access Code for Call Coverage message retrieval (per system)
- Lock and Unlock Access Codes (per system)
- Unlock Security Code (per voice terminal)
- Identities of authorized systemwide message retrievers

Hardware and Software Requirements

Requires a TN725 Speech Synthesizer circuit pack. Each circuit pack has four ports to provide Speech Synthesis. No additional software is required.

STATION MESSAGE DETAIL RECORDING

Description

Records detailed call information on all incoming and outgoing calls on specified trunk groups and sends this information to a Station Message Detail Recording (SMDR) output device. Internal 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.

* 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 15 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 Automatic Alternate Routing access code, or Automatic Route Selection code. Information in this field is left adjusted. These account codes allow the System Manager associated calling information with projects or account numbers. The access code is not recorded because it is not part of the SMDR account code.

  Account code dialing is optional in Version 1. The user may or may not dial an account code and the call is not affected. With Version 2, account code dialing can be optional or mandatory (forced). Forced account code entry is set on a per-Class of Restriction (COR) basis. If the trunk group used for a call requires an account code and one is not dialed, the call is denied. Forced account code entry can also be assigned for all toll calls in which the first or second digit is a 0 or 1. Service calls, directory assistance calls, and Wide Area Telecommunications Service (WATS) calls are excluded.

- Calling Number (up to 4 digits in V1 and 5 digits in V2)

  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 3-C. 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 3-D defines the overrides. To illustrate how to use this matrix, assume that condition codes 7 and A apply to the same call. The matrix contains nine horizontal rows (1, 4, 7, 8, 9, A, C, E, and F) and nine vertical columns (1, 4, 7, 8, 9, A, C, E, and F). To find the condition code that overrides, look at the point of intersection between row 7 and column A. In this case, condition code 7 overrides. This can also be found by looking at the point where row A and column 7 intersect.

TABLE 3-C. 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 Alternate Routing or 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 either 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 Class of Restriction).

Note: When more than one condition applies to a call, the overriding code is shown in Table 3-D.

TABLE 3-D. Condition Code Override Matrix

	CONDITION CODE								
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).

- Facility Restriction Level (FRL) (1 digit)

  FRLs, numbered 0 through 7, are associated with the Automatic Alternate Routing and Automatic Route Selection features 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.

- Interexchange Carrier (IXC) Code (1 digit)

  IXC codes, numbered 1 through 15, are associated with the Version 2 Automatic Alternate Routing and Automatic Route Selection features and depict the carrier used on the call.

  An IXC access number is used to access a specific common carrier for a call. This number is of the form 10XXX, 950—1XXX, or NXX—XXXX, where X is any digit 0 through 9. The IXC access numbers applicable at a given location are associated with an IXC code in translations. When an IXC access number is used on a call, the associated IXC code is recorded. If no IXC access number is used, a 0 is recorded. In this case, either an interexchange carrier is not used on the call or the carrier is selected at the central office.

- 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 3-4 (V1) and Figure 3-5 (V2).

The call detail information sent to the printer is shown in Figure 3-6 (V1) and Figure 3-7 (V2).

The call detail information sent to the 94A Local Storage Unit (LSU) is shown in Figure 3-8 (V1) and Figure 3-9 (V2).

The call detail information sent to the COMM-STOR II unit is shown in Figure 3-10 (V1) and Figure 3-11 (V2).

Alternatively, the client may elect to send SMDR data to a central collection point via private line or connect to their own equipment.

ASCII Character Position	Data Field Description
01-03	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	Space
50-56	Space or Account Code† §
57	Space
58-59	Space or Account Code† §
60	Space
61	FRL or Account Code† §
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 3-C.
† Data is right justified and padded with blanks.
‡ Data is right justified and padded with 0s.
§ If a 15-digit account code is used, the FRL associated with the call, if any, is overwritten.

Figure 3-4. SMDR Data Format—TELESEER Unit (V1)

ASCII Character Position	Data Field Description
01-03	Space
04	Time Hour- (tens)
05	Time Hour- (units)
06	Time Minute (tens)
07	Time Minute (units)
08	Duration Hour
09	Duration Minute (tens)
10	Duration Minute (units)
11	Duration Minute (tenths)
12	Condition Code*
13-15	Access Code #1†
16-18	Access Code #2†
19-33	Dialed Number†
34-38	Calling Number†
39-53	Account Code†
54	FRL
55	IXC
56-57	Incoming Circuit ID‡
58	Space
59-60	Outgoing Circuit ID‡
61-76	Space
77	Carriage Return
78	Line Feed
79-81	null

* Refer to Table 3-C.

† Data is right justified and padded with blanks.

‡ Data is right justified and padded with 0s.

Figure 3-5. SMDR Data Format—TELESEER Unit (V2)

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	Space
50-56	Space or Account Code† §
57	Space
58-59	Space or Account Code† §
60	Space
61	FRL or Account Code† §
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 3-C.
† Data is right justified and padded with blanks.
‡ Data is right justified and padded with 0s.
§ If a 15-digit account code is used, the FRL associated with the call, if any, is overwritten.

Figure 3-6. SMDR Direct Output Format From System to Printer (V1)

ASCII Character Position	Data Field Description
01	Time Hour- (tens)
02	Time Hour- (units)
03	Time Minute (tens)
04	Time Minute (units)
05	Space
06	Duration Hour
07	Duration Minute (tens)
08	Duration Minute (units)
09	Duration Minute (tenths)
10	Space
11	Condition Code*
12	Space
13-15	Access Code #1†
16	Space
17-19	Access Code #1†
20	Space
21-35	Dialed Number†
36	Space
37-41	Calling Number†
42	Space
43-57	Account Code†
58-69	Space
70	FRL
71	Space
72	IXC
73	Space
74-75	Incoming Circuit ID‡
76-77	Space
78-79	Outgoing Circuit ID‡
80-82	Space
83	Carriage Return
84	Line Feed

* Refer to Table 3-C.
† Data is right justified and padded with blanks.
‡ Data is right justified and padded with 0s.

Figure 3-7. SMDR Direct Output Format From System to Printer (V2)

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 or Account Code† §
45	FRL or Account Code† §
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 3-C.

† Data is right justified and padded with blanks.

§ If a 15-digit account code is used, the FRL associated with the call, if any, is overwritten.

Figure 3-8. SMDR Direct Output Format From System to 94A Local Storage Unit System (V1)

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 or Account Code† §
45	FRL or Account Code† §
46	Calling Number (fifth digit)
47-48	Incoming Circuit ID
49	Space
50-51	Outgoing Circuit ID
52-53	Space
54	IXC
55	Carriage Return
56	Line Feed
57-59	null

* Refer to Table 3-C.

† Data is right justified and padded with blanks.

§ If a 15-digit account code is used, the FRL associated with the call, if any, is overwritten.

Figure 3-9. SMDR Direct Output Format From System to 94A Local Storage Unit System (V2)

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 or Account Code† §
57	FRL or Account code† §
58	Carriage Return
59	Line Feed
	null
	null
	null

* Refer to Table 3-C.

† Data is right justified and padded with blanks.

§ If a 15-digit account code is used, the FRL associated with the call, if any, is overwritten.

Figure 3-10. SMDR Direct Output Format From System to COMM-STOR II Unit (V1)

ASCII Character Position	Data Field Description
00	Time Hour- (tens)
01	Time Hour- (units)
02	Time Minute (tens)
03	Time Minute (units)
04	Duration Hour
05	Duration Minute (tens)
06	Duration Minute (units)
07	Duration Minute (tenths)
08	Condition Code*
09-11	Access Code #1†
12-14	Access Code #2†
15-29	Dialed Number†
30-34	Calling Number†
35-49	Account Code†
50	FRL
51	IXC
52-53	Incoming Circuit ID
54	Space
55-56	Outgoing Circuit ID
57	Space
58	Carriage Return
59	Line Feed
	null
	null
	null

* Refer to Table 3-C.
† Data is right justified and padded with blanks.

Figure 3-11. SMDR Direct Output Format From System to COMM-STOR II Unit (V2)

Date Record Format

Three formats are available for date records, one for the 94A LSU (Figure 3-12), one for the COMM-STOR II unit, printer, and AP (Figure 3-13), and one for the TELESEER unit (Figure 3-14). 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 3-12. 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 3-13. 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 3-14. 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 3-15.

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 3-16.

An Activity Report lists call record details for each extension number assigned to the System 75. The Activity 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 3-17.

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 3-18.

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 bits long.

94A Local Storage Unit (LSU)

The 94A LSU collects and stores Message Detail Records (MDRs) data from System 75. The 94A LSU stores MDRs 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 bits 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 bits 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 station message detail records are generated to enable the client to assess individual voice terminals in the System 75 for trunk usage. A station message detail record is created for the following types of calls:

- Outgoing call—a call originating at a voice terminal in System 75 and going out on a trunk
- Incoming call—a call incoming on a trunk and terminating at a voice terminal in System 75

The client has the option of turning off station message detail record generation for incoming calls, specific trunk group(s), and ineffective call attempts via an administration procedure.

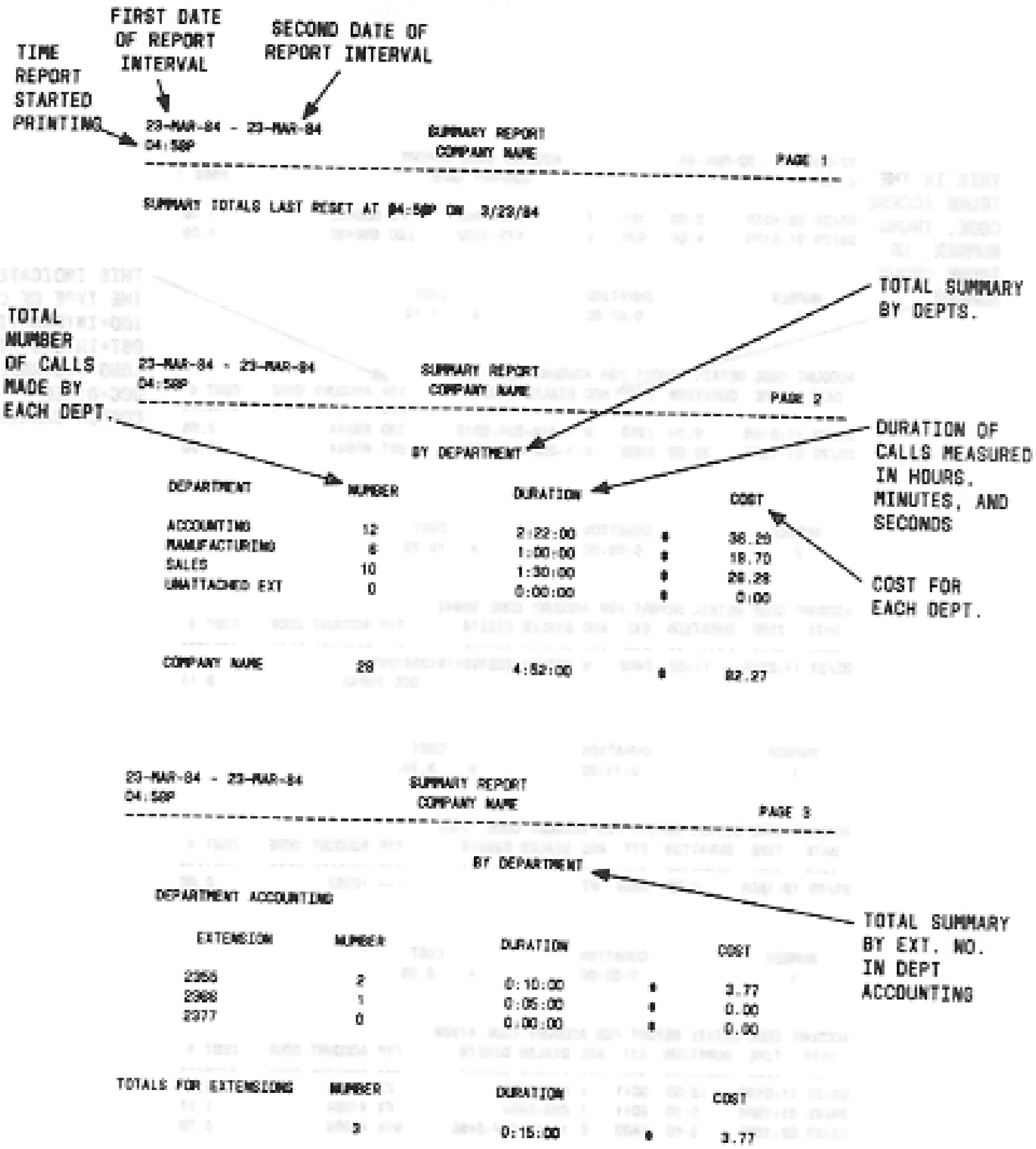

Figure 3-15. Example of a TELESEER Unit Summary Report

THIS IS THE TRUNK ACCESS CODE, TRUNK NUMBER, OR TRUNK GROUP NUMBER

THIS INDICATES THE TYPE OF CALL
IOD=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    9        473-1502    LOC 556432             0.08

     NUMBER                DURATION                   COST
       2                   0:07:00            #       1.13

ACCOUNT CODE DETAIL REPORT FOR ACCOUNT CODE 88844
 DATE    TIME  DURATION  EXT  ACC DIALED DIGITS      TYP ACCOUNT CODE    COST #
 ----    ----  --------  ---  --- ------ ------      --- ------- ----   -------
03/23 11:31AM     8:00  1799    9 1-416-324-5012     IOD 88844             2.55
03/23 01:16PM    30:00  2388    9 1-206-888-4567     OST 88844            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 5872011222333441312312456
                                                  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   67                    --- 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    4 663-2929            FX 41364             5.01
03/23 01:15PM     5:00  3011    7 298-5454            FX 41364             1.77
03/23 02:13PM     3:00  2400    8 1-417-987-3498     WTS 41363             0.70

     NUMBER                DURATION                   COST
       3                   0:20:00            #       7.48
```

Figure 3-16. 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
05:07P                                          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   88                          FX                   1.77
03/23 02:53PM       12:00  4155    8 1-206-324-5151          WST                   2.86
03/23 03:12PM        5:00  4155    8 1-312-654-7829          WST                   1.18

TOTALS              22:00
                                                                                   5.81
TOTAL CALLS 3

ACTIVITY REPORT FOR EXTENSION 4388
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 98267             0.46
03/23 01:18PM       30:00  4355    9 1-206-999-5479          OST 54321            13.20

TOTALS              32:00                                                         13.66

TOTAL CALLS 2

ACTIVITY REPORT FOR EXTENSION 4455
NO RECORDS STORED

ACTIVITY REPORT FOR EXTENSION 4500
  DATE    TIME   DURATION   EXT  ACC DIALED DIGITS          TYP ACCOUNT CODE     COST #
  ----    ----   --------   ---  --- ------ ------          --- ------- ----    -------
03/23 06:05AM       11:00  4500    8 1-419-643-7474          OST 34671             3.02

TOTALS              11:00
                                                                                   3.02
TOTAL CALLS 1

ACTIVITY REPORT FOR EXTENSION 4822
  DATE    TIME   DURATION   EXT  ACC DIALED DIGITS          TYP ACCOUNT CODE     COST #
  ----    ----   --------   ---  --- ------ ------          --- ------- ----    -------
03/23 00:05AM       10:00  4822    8 1-419-643-7474          OST 54321             1.49
03/23 02:02PM       1:00/  4822    7 344-7542                 FX 98267             0.33

TOTALS              11:00
                                                                                   1.82
TOTAL CALLS 2

23-MAR-84 - 23-MAR-84                         ACTIVITY REPORT
05:07P                                          COMPANY NAME                        PAGE 3
COST CENTER 520


ACTIVITY REPORT FOR EXTENSION 5300
  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 53643             0.08

TOTALS              14:00
                                                                                   1.87
TOTAL CALLS 2
```

Figure 3-17. Example of a TELESEER Unit 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	6300	64		WST		1.79
03/23	11:01AM	12:00	3011	4	683-2828	FX	41383	5.01
03/23	11:15AM	10:00	1122	9	1-315-681-0848	IST	395878	0.83
03/23	11:29AM	11:00	2400	9	9872011222333413129312459	OCC	95843	3.74
03/23	11:35AM	22:00	1011	8	1-213-324-5012	OST	12345	9.76
03/23	00:05AM	10:00	4622	8	1-419-643-7474	OST	54321	1.49
03/23	06:05PM	11:00	4500	9	1-818-643-7474	OST	34871	3.02
03/23	01:19PM	30:00	4355	9	1-206-999-5478	OST	54321	13.20
03/23	01:19PM	30:00	2388	9	1-206-888-4587	OST	89844	13.20
03/23	02:35PM	15:00	1122	9	9871011278565610156863200	OST		4.80
03/23	02:53PM	12:00	4155	6	1-206-324-5151	WST		2.86
03/23	03:02PM	35:00	2388	8		---		6.30
03/23	03:22PM	25:00	2400	9	98720112634576131536857219	OCC		1.72
TOTALS		3:53:00						67.82

TOTAL CALLS 13

TOTAL TIME SPENT ON 13 CALLS: 3 HOURS AND 15 MINUTES

TOTAL COST FOR 13 CALLS WHICH LASTED FOR 3 HOURS AND 15 MINUTES

Figure 3-18. Example of a TELESEER Unit Selection Report

Considerations

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

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.

The TELESEER unit can store up to 28,000 call records, 500 extension numbers, and 2000 account codes.

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 to a 93B CMDR.

The COMM-STOR II unit can handle up to 2000 voice terminals.

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, Automatic Alternate Routing, 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.

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):

  - Fact an ARS call was made
  - Calling extension number
  - Facility Restriction Level of the calling extension
  - Called number
  - Type of trunk group used for the ARS call
  - Time of call completion
  - Call duration (how long the parties talked)
  - Interexchange carrier code, if any, in Version 2

  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.

- Bridged Calls

  The primary extension on a multi-appearance voice terminal can be bridged onto at least five other voice terminals.

- 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.

- 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 Group (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 Service trunk group will be recorded.

Administration

SMDR is administered by the System Manager. The following items can be administered.

System Parameters

- Type of SMDR output device to be used.
- Extension number assigned to the output device.
- Printer paper width (80 or 132 columns) if a printer is the output device.
- SMDR account code length number (from 1 to 15), the system defaults to 2 digits.
- 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 that are 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.

Class of Restriction

With Version 2, specify if SMDR account code entry is forced.

Feature Access Codes

Assign SMDR account code access code. The system defaults to *6.

IXC Codes

- IXC access numbers
- Name of IXC (optional)

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 numbered 01 to 04 (V1) or netcon channel (V2) must be assigned using a data module form and entering data-channel or netcom 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 PDM form.
- If the SMDR output device is connected to a Trunk Data Module, administer a Trunk Data Module form.
- If the SMDR output device is connected to a 212A-type modem, a 2500 Voice Terminal form and Pooled Modem form must be completed. This allows circuit switched data connections between digital data communications equipment (data modules) 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 3 module. This is used with the TN719 Interface 3 circuit pack.
- Processor Data Module
- Interface Link
- Processor Channel

Hardware and Software Requirements

Hardware requirements depend on the type of output device used for SMDR:

- If the output device is a printer, personal computer, tape unit, or the TELESEER unit (Data Terminal Equipment), the interface equipment consists of either a PDM to a port on a TN754 Digital Line circuit pack or a 212A-type modem to a port on a TN742 Analog Line circuit pack. In the latter case, a standard modem pool facility is required for the data path.
- If the output device is the 94A LSU or COMM-STOR II unit (Digital Communications Equipment), the interface equipment consists of either a Trunk Data Module to a port on a TN754 Digital Line circuit pack or a 212A-type modem to a port on a TN752 Analog Line circuit pack. In the latter case, a modem pool facility is also required.
- If SMDR is connected to a host computer, the computer must be connected over a private line terminated at the System 75 with a Trunk Data Module.
- If SMDR is used with the AP, a TN716 Interface 1, a TN720 (V1) or TN738 (V2) Interface 2, and a TN719 Interface 3 circuit pack must be installed and connected.

For V2, forced entry of account codes software is required.

STRAIGHTFORWARD OUTWARD COMPLETION

Description

Allows an attendant to complete an outgoing trunk call for a voice terminal user.

Considerations

With Straightforward Outward Completion the attendant determines which calls should be allowed and can select the trunk group used for the call.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

SUBNET TRUNKING (V2)

Description

Provides modification of the dialed number so an Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS) call can route over trunk groups that terminate in switches with different dial plans.

Subnet Trunking provides digit insertion, deletion, pauses, and/or wait for dial tone in digit outpulsing, as required, to permit calls to route:

- To or through a remote switch
- Over tie trunks to private network switch
- Over CO trunks to the serving central office.

All AAR and ARS calls ultimately reach a point where they can no longer route on a private network. That is, the call reaches a point where another on-network switch is not available for the call. (In an ARS stand-alone configuration, this is the originating switch.) Assuming the call is not denied at this point, then the call must route to one of the following:

- Directly to a party at the local switch.
- To a party at a remote switch (without accessing the public network).
- Through a remote switch to a party at a subtending location (without accessing the public network).
- Directly to a WATS serving office.
- Directly to a local central office or a foreign exchange (FX) central office, both of which may or may not provide dial access to a long-distance carrier. (The alternative to dial access is for the central office to automatically provide access to a single long-distance carrier of the subscriber's choice.)
- Through a remote switch to the local or FX central office serving the remote switch.
- Through a remote switch to the WATS office serving the remote switch.
- To an Enhanced Private Switched Communications Service (EPSCS), Common Control Switching Arrangement (CCSA) office, or Electronic Tandem Network (ETN) office.

Subnet Trunking is not required on calls terminating directly to a party at the local switch. AAR handles these calls.

Subnet Trunking is required on calls routing to or through a remote switch, regardless of the call's destination.

With direct access to a WATS, EPSCS, CCSA, or ETN office, Subnet Trunking is not normally used. The called number on these types of calls is not normally modified. Subnet Trunking is needed only if the number is modified or if the call passes through some intermediate switch, such as a main.

Calls accessing a local or FX central office directly from the terminating switch normally require Subnet Trunking only if access to a long-distance carrier is other than the carrier automatically provided by the central office. In this case, the appropriate dial access code is inserted into the digit string by the system.

Aside from the normal cases, Subnet Trunking can be used to provide added functionality to the system, for example, to convert an AAR number into an international number. Also, Subnet Trunking can modify a digit string so that a Remote Access trunk group can be used on calls. This capability is called equivalent Direct Inward Dialing (DID) and may be useful when a location has Remote Access but does not have DID or Network Inward Dialing (NID). (NID is the private network equivalent of DID.)

Addition or deletion of an Area Code on an ARS call does not require Subnet Trunking. ARS handles it via code conversion, as required. With AAR, an on-network number can be converted into a public network number. In this case, the conversion may include an Area Code insertion via Subnet Trunking.

Any of three special characters may be used with Subnet Trunking:

- Pause—Delays outpulsing of subsequent digits for 1.5 seconds.
- Wait—Delays outpulsing of subsequent digits for a preprogrammed interval (from 5 to 25 seconds) or, if TN748B Tone Detectors are provided, until dial tone is received from the distant switch or the interval expires, whichever occurs first.
- Convert-to-tone—Causes all remaining digits to be outpulsed using tone signaling.

Use of these special characters is discussed in the following paragraphs.

During outpulsing of a digit string, it may be necessary to pause or wait for the distant switch to act upon the digits already sent. A programmed pause (a "%" symbol) is used when the required action by the distant switch occurs within 1.5 seconds. Multiple pauses can be used. A programmed wait (a "+" symbol) is used to specify a longer interval. Receipt of dial tone automatically cancels the remainder of an interval when TN748B Tone Detectors are provided. If a dial tone detector is not available on a given call, the system uses the wait interval to determine when to resume outpulsing. Multiple waits can be used.

The type of outpulsing, either dial pulse (rotary) or tone, used on a call is specified by the trunk group selected for the call. In some cases, it may be necessary to assure that a portion of the digits are sent using tone signaling. The convert-to-tone character (a "$" symbol) is used to indicate that all digits remaining in the string to be outpulsed will use tone signaling.

Digit deletion always begins with the first digit. Subnet Trunking can delete up to 11 digits and can insert up to 36 digits. The last four digits dialed are normally retained. Thus, the new digit string can be up to 40 digits long. Typical uses of digit insertion are the conversion of an AAR call to an international call and the insertion of a long-distance carrier code, 10xxx, on a domestic call.

The insertion of a long-distance carrier access code in the string of digits to be outpulsed does not usually require a pause or wait symbol. Interconnecting offices, other than crossbar offices, can handle the code and the called number as a single string. However, a crossbar office returns dial tone after receiving the long-distance carrier code. Thus, a pause or wait is required between the long-distance carrier code and the called number.

Considerations

Subnet Trunking allows AAR and ARS calls to access the public network. With AAR, the major advantage is that the call continues although no on-network routes are available to handle the call. With ARS, the major advantage is that calls destined for the public network can route partially over the private network, if there is one. This saves toll charges for a portion of the call.

It is not necessary to include the trunk access code for the trunk group connecting to the distant switch in the string of digits to be outpulsed. In fact, such inclusion must be avoided. Access to the interconnecting trunk group is automatic. Outpulsing the access code, therefore, serves no purpose, and will cause mishandling of the call at the distant end.

The wait interval is a System Parameter. This interval can be from 5 to 25 seconds (in increments of 1 second).

Up to four special characters can be included in a string of digits to be outpulsed. Each special character counts as two digits.

Interactions

Subnet Trunking is a function associated with the AAR and ARS features. Interactions are the same as those given for AAR and ARS.

Administration

Subnet Trunking is set by the System Manager as a part of AAR and/or ARS administration. The following items require administration:

- Wait—Specify the wait interval used with Subnet Trunking.
- Routing Pattern—Specify the number of digits to delete (beginning with the first digit) and the specific string of digits to insert. Special characters, if any, are included in the inserted string.

Hardware and Software Requirements

If Routing Patterns containing wait symbols are heavily used, and if dial tone detection is preferable to waiting for interval time-out, then additional TN748B circuit packs may be required.

Private Network Access or ARS software is required for Subnet Trunking.

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.)

In addition to the trunk group, hunt group, attendant group, and security violation reports already described, Version 2 offers the following reports:

- Automatic Route Selection (ARS) Pattern Measurement

  A maximum of 20 ARS patterns can be measured, as specified by the client.

- Trunk Outage Measurements

  Provides measurements on the four trunk groups which were out of service the most during the measurement period. Separate reports are available for yesterday, today, and the last hour.

- Trunk Lightly Used Measurements

  Provides measurements on the five trunks in each trunk group that have carried the least calls. Separate reports are available for yesterday, today, and the last hour.

- Modem Pool Measurements

  Provides traffic data for Modem Pool groups. Separate reports are available for yesterday's peak usage hour, today's peak usage hour, and the last hour.

- Attendant Group Performance

  Provides measurements on time taken by attendants to answer calls. Separate reports are available for any 8-hour period, yesterday or today.

- Hunt Group Performance

  Provides data on the longest time taken by a group member to answer a call for each hunt group. Separate reports are available for yesterday and today.

- Trunk Group Performance

  Provides data on calls that attempt to access trunk groups but cannot because all trunks are busy or the queue is full. Separate reports are available for yesterday and today.

- Summary Performance Report

  Provides a summary of data found in the ARS Pattern Measurements, Trunk Outage Measurements, Trunk Lightly Used Measurements, and Trunk Group Performance reports. Separate reports are available for yesterday and today.

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*, 555-200-500.

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.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

SYSTEM STATUS REPORT (V2)

Description

Allows the user to view data associated with attendants, major and minor alarms, and traffic measurements. The information is displayed on the System Access Terminal (SAT) and presents a basic picture of the System 75 condition. The report can only be displayed by the System Manager and maintenance personnel.

The Status Report is displayed by entering the "monitor system view1" or "monitor system view2" command. Once the command is entered, the system continually displays the report until it is canceled. Once the report is canceled, the user is automatically logged off the system.

The "view1" report displays the following information:

- Activation status of all attendants
- Maintenance status which includes major and minor alarms for trunk ports, terminal ports, and all maintained objects in the system except terminals and trunks
- Traffic measurements for trunk groups, hunt groups, and attendant groups

The view2 report is a subset of the view1 report and displays the same information listed for the view1 report except the last hour's measurement for the hunt groups.

The attendant status and the maintenance status is updated every minute. The traffic status is updated every hour because the information is taken from existing measurements which is polled every hour by the system.

Considerations

In addition to providing status reports, this feature also provides an indication that the SAT is functioning. Any attempt to stop the reports logs the SAT off the system. Therefore, no unauthorized administration can be performed.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

TEMPORARY BRIDGED APPEARANCE

Description

Allows multi-appearance voice terminal users in a Terminating Extension Group or Personal Central Office Line Group to bridge onto an existing group call. If a call has been answered via the Call Pickup feature, the originally called party can bridge onto the call. Also allows a called party to bridge onto a call that redirects to coverage before the called party can answer it.

A call incoming to a Terminating Extension Group or Personal Central Office Line Group is not a call to an individual, although one particular member of the group may be the most qualified person to handle the given call. If this individual did not answer the call originally, then he or she can simply bridge onto the call. The call does not have to be transferred.

A call to an individual can be answered by a Call Pickup group member. If the called person returns while the call is still connected, he or she simply bridges onto the call and the answering party hangs up.

Call Coverage provides redirection of calls to alternate answering positions (covering users). A Temporary Bridged Appearance is maintained at the called voice terminal.

The called party can answer the call at any time, even if already answered by a covering user. If the called party does not bridge onto the call, the covering user can use the Consult function of Call Coverage to determine if the called party wants to accept the call. The Consult function uses the Temporary Bridged Appearance maintained on the call. When the consult call is finished, the Temporary Bridged Appearance is removed.

Considerations

Temporary Bridged Appearance permits the desired party to bridge onto a call without manually transferring the call. This provides convenience of operation and also saves time.

Temporary Bridged Appearance does not provide any call originating capability or the capability to answer another party's calls. These capabilities are provided by the Bridged Call Appearance feature.

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 be dropped.

The Bridged Call Appearance feature enhances Temporary Bridged Appearance by allowing more than one call to an extension to be bridged and by allowing calls to be originated from bridged appearances.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

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.

A single-line voice terminal administered as a TEG member is rung for a TEG call if it is idle.

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.

Considerations

TEGs are useful when it is desirable to have incoming calls to a specific extension number ring more than one voice terminal simultaneously. For example, the appliance department of a large retailer might have three voice terminals. Anyone in the department can answer the call. The salesperson most qualified to handle the call can bridge onto the call from either of the other two voice terminals.

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 can only handle one TEG call at a time. If any member of a TEG is active on a call to the TEG, a second call to the TEG waits until the first call is terminated before it rings the TEG. The TEG members have no way to know when a TEG call is waiting. If a coverage path is assigned to the TEG, the waiting call routes accordingly.

Interactions

- Automatic Callback

  This feature cannot be activated for a TEG.

- Bridged Call Appearance

  Calls to a TEG cannot be bridged, except via a Temporary Bridged Appearance.

- 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

  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

  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 be dropped.

Administration

Terminating Extension Groups are administered by the System Manager. The following items require administration for each group:

- 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.

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.

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.

Considerations

Through Dialing saves the attendant time by allowing the calling party to dial the called number.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

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

Timed Reminder informs the attendant that a call requires additional attention. After the attendant reconnects to the call, the user can either choose to try another extension number, hang up, or continue to wait. This personal attention can help establish rapport with clients and customers.

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.

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.

Administration

Timed Reminder is administered on a per-system basis by the System Manager. The following items require administration:

- Time a call remains on hold before the attendant is rung
- Time a call remains unanswered before the attendant is rung

Hardware and Software Requirements

No additional hardware or software is required.

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

With Touch-Tone Dialing, user's are more efficient when placing and handling calls.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

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

The Transfer feature provides a convenient way to connect a party with someone better qualified to handle the call. Attendant assistance is not required and the call does not have to be redialed.

Multi-appearance voice terminals must have an idle call appearance in order to transfer a call.

Interactions

None.

Administration

None required.

Hardware and Software Requirements

No additional hardware or software is required.

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.

The two lamps which provide the visual indications are as follows:

- Warn Lamp

  Located at 6 of the 12 Trunk Group Select buttons. The Warn lamp lights when a preset number (warning threshold) of trunks are busy in the associated trunk group.

- Busy Lamp

  Located at each of the 12 Trunk Group select buttons. The Busy lamp lights when all trunks in the associated trunk 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 show the attendant that control of access to trunk groups is necessary.

Each attendant console has 12 Trunk Group Busy Indicators and 6 Trunk Group Warning Indicators.

Interactions

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.

Administration

This feature is administered by the System Manager. The following items require administration:

- Trunk Group Select buttons (per attendant console)
- Warning threshold (per trunk group)

Hardware and Software Requirements

No additional hardware or software is required.

TRUNK IDENTIFICATION BY ATTENDANT (V2)

Description

Allows an attendant or display-equipped voice terminal user to identify a specific trunk being used on a call. This capability is provided by assigning a Trunk ID button to the attendant console or voice terminal.

The Trunk Identification By Attendant feature can be used when a user is on an established call of one of the following types:

- An incoming trunk call
- An outgoing trunk call
- A transferred or conferenced call involving a trunk
- A trunk-to-trunk call

In addition to its use during an established call, the Trunk ID button can be used while a trunk is being seized, while digits are being outpulsed on a trunk, or during intervals between digit outpulsing.

When a user is connected to a trunk, as described above, and presses the Trunk ID button, the identification of the trunk is displayed on the 40-character alphanumeric display. The trunk identification consists of the trunk access code (2-digit) for that trunk group and the trunk group member number (2-digit).

The trunk identification displayed depends on the type of call in process. If the call is incoming, the incoming trunk identification is displayed. If the call is outgoing, the outgoing trunk identification is displayed. If the call is trunk-to-trunk, the identification displayed is of the last trunk added to the call.

Considerations

Trunk Identification By Attendant is useful whenever it is necessary to identify a particular trunk being used. The feature is particularly useful for identification of a faulty trunk. That trunk can then be removed from service and the problem quickly corrected.

A maximum of one Trunk ID button is allowed per each attendant console and voice terminal with a display.

The Trunk Identification By Attendant feature is denied if there are more than two trunks on the call.

Interactions

- Busy Verification

  A trunk being busy-verified can be identified.

- Attendant Display and Voice Terminal Display

  Any action by the user or the system which changes the display removes the trunk identification currently displayed. The lamp associated with the Trunk ID button remains lighted as long as the call on which the button was used remains active. While the lamp is lighted, the user can use the associated button to re-display the trunk identification.

If the Trunk ID button is pressed during a call origination (before all digits have been dialed) the trunk identification appears. On a voice terminal display, any subsequently dialed digits are not displayed. On an attendant display, the subsequently dialed digits overwrite other digits on the dispay.

- Hold

  A trunk held by a user cannot be identified.

Administration

Trunk Identification By Attendant is assigned by the System Manager on a per-voice terminal and per-attendant console basis. The only administration required is the assignment of a Trunk ID button.

Hardware and Software Requirements

No additional hardware or software is required.

Administration

The UDP is administered by the System Manager. The following items require administration:

- Whether UDP has 4- or 5-digit extension numbers
- PBX Codes (expands first 2 digits of dialed extension to an RNX)
- RNX Table (used by AAR to route calls to the correct switch)
- Routing Patterns

Hardware and Software Requirements

AP/DCS interface hardware is required for DCS applications. DCS or UDP software is required.

VOICE TERMINAL DISPLAY

Description

Provides multi-appearance voice terminal users with updated call and message information. This information is displayed on a display-equipped 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. The display must be in the Normal mode to answer incoming calls and to display information associated with the Version 2 Automatic Incoming Call Display feature.

- 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 handset to retrieve messages. Also, messages can be retrieved even if the retriever is active on a call.

TRUNK-TO-TRUNK TRANSFER

Description

Allows the attendant or voice terminal user to connect an incoming trunk call to an outgoing trunk.

Considerations

Trunk-To-Trunk Transfer is particularly useful when a caller outside the system calls a user or attendant and requests a transfer to another outside number. For example, a worker, away on business, can call in and have the call transferred elsewhere.

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 can, however, use the Forced Release button and disconnect all parties associated with the call.

If a voice terminal user has connected two outgoing trunks, the user must remain on the call. Otherwise, the call will be dropped.

Interactions

The Attendant Lockout feature does not function on Trunk-to-Trunk Transfer.

Administration

Trunk-To-Trunk Transfer is administered on a per-system basis by the System Administrator. The only administration required is whether or not Trunk-To-Trunk Transfer is permitted.

Hardware and Software Requirements

No additional hardware or software is required.

UNIFORM DIAL PLAN (V2)

Description

Provides a common 4- or 5-digit dial plan that can be shared among a group of switches (Distributed Communications System [DCS] or Main/Satellite/Tributary arrangement). Interswitch dialing and intraswitch dialing are both via 4- or 5-digit dialing.

In a Uniform Dial Plan (UDP), the first two digits of the 4- or 5-digit extension number determine the particular switch to which a call is directed. When a UDP call is made, the system expands the first two digits of the extension number into a 3-digit private network office code (RNX). Automatic Alternate Routing (AAR) is then used to route the call to the correct switch within the private network, based on the RNX. (A subset of AAR is provided with the UDP software.)

UDP is used with Main/Satellite/Tributary and Distributed Communications System (DCS) configurations. Additionally, UDP can be used alone to provide uniform 4- or 5-digit dialing between two or more private switching systems without Main/Satellite/Tributary or DCS configurations.

Considerations

The UDP feature enables a terminal user at any switch to call any other terminal on any switch in the UDP complex, using only the 4- or 5-digit extension number.

When calling an extension on another switch, there is a slight delay before call progress tones are applied. This delay is due to the trunk signaling necessary to complete the call to the remote switch.

Interactions

- Automatic Alternate Routing (AAR)

  After the system determines the RNX of the switch being called, AAR routes the call to the correct switch. The required subset of AAR is provided with the UDP software.

- Direct Inward Dialing (DID)

  DID calls to 5-digit UDP extension numbers require that the DID trunk group insert enough digits to make a 5-digit extension number.

- Dial Plan

  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.

- Distributed Communications System (DCS)

  UDP is required when DCS is provided. The necessary UDP software is provided with the DCS software.

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

  — Version 1

  When the call is from a system user, the display shows the caller's extension number, the caller's name, or a unique identification administered for the voice terminal being used. When the call is from outside the system, the display shows the trunk identification, such as CHICAGO, assigned to the trunk group used for the call.

  — Version 2

  When the call is from a system user, the display shows the caller's name or a unique identification administered for the voice terminal being used, along with the calling party's extension number. When the call is from outside the system, the display shows the trunk identification, such as CHICAGO, and the trunk access code, assigned to the trunk group used for the call. If a user is active on a call, and receives a subsequent call, the display automatically shows the identification of the subsequent caller.

- Called Party Identification

  — Version 1

  On calls to a system user, the display shows the digits as they are dialed. After the dialing is complete, the display shows the called party's name. If no name is assigned, the called party's extension number is displayed.

  On outgoing calls, the display shows the digits as they are dialed, followed by the name assigned to the trunk group being used. The System Manager can suppress the name of any trunk group. If such a trunk group is accessed, the called party portion of the display is blank.

— Version 2

On calls to a system user, the display shows the digits as they are dialed. After the dialing is complete, the display shows the called party's name and extension number. If no name is assigned, only the called party's extension number is displayed.

On outgoing calls, the display shows the digits as they are dialed, followed by the name and trunk access code assigned to the trunk group being used. The System Manager can suppress the name of any trunk group. If such a trunk group is accessed, the name portion of the display is blank.

- 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.

  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 or Cover—Indicates that the called user has not answered or that the calling system user has sent the call to coverage.

  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 (V1):

a=3602

then

a=	TOM BROWN

or

a=	EXT 3602

- Internal call (V2):

a=3602

then

a=	TOM BROWN	3062

or

a=	EXT 3602	3062

- Outgoing trunk call (V1):

b=87843541

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

then

b= OUTSIDE CALL

or

b= WATS

- Outgoing trunk call (V2):

b=87843541

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

then

b= OUTSIDE CALL	8

or

b= WATS	101

- Incoming trunk call (V1):

a= OUTSIDE CALL

- Incoming trunk call (V2):

```
a=   OUTSIDE CALL                102
```

Where 102 is the trunk access code of the incoming trunk group.

- Conference call originated by the attendant:

```
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=    EXT 3174  to  EXT 3077    d
```

or

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

Where d indicates that Go To Cover was activated by the calling voice terminal user.

- Incoming trunk call redirected to coverage:

```
b=   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 a.m. 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

The Voice Terminal Display feature provides an instant display of information associated with certain system features, functions, and services. Information that allows personalized call answering is available on many calls. Retrieval of stored information, such as messages received and directory information, is easy as well as convenient.

Up to 62 display modules can be provided per system.

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.

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.

Interactions

- Bridged Call Appearance

  A call from the primary extension number or a bridged call appearance of the primary extension number is displayed as a call from the primary extension number.

Administration

Voice Terminal Display is administered on a per-voice terminal basis by the System Manager. The following items require administration.

- Whether or not a display module is provided (per 7405D voice terminal)
- Whether or not to restrict other users from reading or canceling the voice terminal's message (per display module)
- The following buttons (per display module):
  - 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 display-equipped voice terminal and one port on a TN754 Digital Line circuit pack. No additional software is required.

SYSTEM PARAMETERS

CONTENTS

OVERVIEW

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.

FEATURE ADMINISTRATION

Administration Not Required

Administration is not required to activate the following features.

- Attendant Auto-Manual Splitting
- Attendant Call Waiting
- Attendant Recall
- Attendant Release Loop Operation
- Automatic Incoming Call Display (V2)
- Conference—Attendant
- Conference—Terminal
- Hold
- Line Lockout
- Recall Signaling
- Senderized Operation
- Straightforward Outward Completion
- Temporary Bridged Appearance
- Through Dialing
- Touch-Tone Dialing (for Terminals)
- Transfer

Administration Required

Administration is required to activate the following features.

- Abbreviated Dialing
- Attendant Control of Trunk Group Access
- Attendant Direct Extension Selection With Busy Lamp Field
- Attendant Direct Trunk Group Selection
- Attendant Display (Buttons only)
- Automatic Alternate Routing
- Automatic Callback
- Automatic Circuit Assurance (V2)
- Automatic Route Selection
- Bridged Call Appearance
- Busy Verification of Terminals and Trunks (V2)
- Call Coverage
- Call Forwarding All Calls
- Call Park
- Call Pickup
- Call Waiting Termination
- Centralized Attendant Service (V2)
- Code Calling Access
- Data Call Setup
- Data-Only Off-Premises Extension
- Data Privacy
- Data Restriction
- Dial Access to Attendant
- Digital Multiplexed Interface (V2)
- Direct Department Calling
- Direct Inward Dialing

Direct Outward Dialing
Distinctive Ringing
DS1 Tie Trunk Service
EIA Interface (V2)
Facility Busy Indication
Facility Test Calls
Hot Line Service
Hunting
Individual Attendant Access (V2)
Information Systems Network (ISN) Interface
Integrated Directory
Intercept Treatment
Intercom—Dial
Inter-PBX Attendant Calls (V2)
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
Permanent Switched Calls (V2)
Personal Central Office Line
Personalized Ringing (V2)
Power Failure Transfer
Priority Calling
Privacy—Attendant Lockout
Privacy—Manual Exclusion
Recorded Telephone Dictation Access
Remote Access
Restrictions
Ringback Queuing
Rotary Dialing (V2)
SMDR Account Code Dialing
Speech Synthesis (V2)
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 (see Direct Department Calling)
Uniform Dial Plan (V2)
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

- AP Demand Print
- 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)
- SMDR Account Code Dialing

- Speech Synthesis

  Message Retrieval Mode
  Coverage Message Retrieval Mode
  Delete Message
  Repeat Message
  Next Message
  Help
  Call

- 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)

- Automatic Circuit Assurance

- Bridged Call Appearance

- Busy Verification of Terminals and Trunks

- 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
- Trunk Identification by Attendant

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.)

APPLICATIONS PROCESSOR (AP) FEATURES

The following features are available with the AP:

- Automated Building Management
- Call Detail Reporting and Recording
- 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.

Item	R1V1	R1V2
Abbreviated Dial Lists:	502	802
Personal Lists	400	800
Group Lists	100	100
System List	1	1
7103A List	1	1
Applications Processors:		
With Information Exchange Between AP and System 75 switch	1	1
Without Information Exchange Between AP and System 75 switch	8	8
Attendant Consoles:		
Daytime Consoles	6	6
Night-Only Console	1	1
Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS):		
Patterns	16	254
Trunk Groups per Pattern	6	6
Toll Lists	4	32
3-Digit Translation Tables	2	2
6-Digit Translation Tables	4	32
Bridged Call Appearances	400	500
Cabinets (Basic)	1	1
Call Coverage:		
Coverage Paths per System	200	400
Coverage Points per Path	3	3
Coverage Answer Groups	100	200
Members per Coverage Answer Group	8	8
Call Pickup Groups	200	400
Members per Group	25	50
Total Members	400	800
Carriers:		
Control	1	1
Port (5-Carrier Cabinet)	4	4
Port (2-Carrier Cabinet)	1	1
Classes of Restriction	64	64
Classes of Service	16	16
Code Calling Ids	125	125
Communications Interface Links (See Note).	1	4
Digital Data Endpoints	200	400
DS1 Circuit Packs	-	20

Note: Only one link can be used for the Applications Processor (V1 and V2).

Item	R1V1	R1V2
Extension Numbers	600	1200
Facility Busy Indicators	1000	1600
Hunt Groups (DDC and UCD Combined)	32	32
Members per Group	32	32
Members per System	448	448
Queue Slots per Group	35	35
Intercom Groups (Automatic and Dial Combined)	32	32
Members per Group	32	32
Members per System	128	128
Leave Word Calling (Switch-Based, No AP):		
Messages Stored	1000	2000
Messages per User	125	125
Systemwide Message Retrievers	10	10
Remote Message Waiting Indicator:		
Per Extension Number	1	1
Per System	50	50
Loudspeaker Paging Zones	9	9
Personal Central Office Lines	25	40
Pooled Modems:		
Groups	1	5
Members per Group	32	32
Integrated	32	160
Combined	-	160
Port Circuit Packs (excluding Tone Detectors)	85	85
Recorded Announcements	10	10
Calls Connected per Announcement	1	1
Remote Access Barrier Codes	10	10
Ringback Queue Slots	100	120
Speech Synthesizer Circuit Packs	-	6
Terminating Extension Groups	32	32
Members per Group	4	4
Time Slots:		
Total	512	512
Call Switching	473	483
Simultaneous Conversations	236	241
Tone Detectors:		
Call Progress	10	10
Touch-Tone	20	20
Traffic Handling Capability (in Hundred Call Seconds [CCS])	8500	8670
Trunk Access Codes	100	118
Trunks	200	200
Trunk Groups	50	60
Trunks per Group	60	60
Voice Terminals (Includes all voice terminals, 515 or 510D terminals, external alerts, and announcement machines).	400	800

GLOSSARY

Access Code

A 1-, 2-, or 3-digit dial code used to activate or cancel a feature or access an outgoing trunk. The star (*) and pound (#) can be used as the first digit of an access code.

Access Tie Trunks

Tie trunks used to handle normal ETN calls between Main and Tandem switches.

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 transmitting data where each character is preceded by a start bit and followed by a stop bit, thus permitting data elements to occur at irregular intervals. This type transmission is advantageous when transmission is not regular (characters typed at a keyboard).

Asynchronous Data Unit (ADU)

A data communications equipment (DCE) type device that allows direct connection between RS-232C equipment and the system digital switch.

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

A multi-conductor electrical path used to transfer information over a common connection from any of several sources to any of several destinations.

Bus, Time Division Multiplex

See Time Division Multiplex Bus.

Business Communications Terminal

An advanced series of semi-intelligent terminals.

Bypass Tie Trunks

One-way, outgoing tie trunks from a Tandem switch to a Main switch in an ETN. These trunks are used as a "last-choice" route when all trunks to another Tandem switch are busy.

Byte

A sequence of bits, 8 bits long, that is usually shorter than a word. A word is 16 bits long.

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. Two lamps next to the button show the status of the call appearance or status of the call.

Callback Call

A call that is automatically returned to a voice terminal user who activated the Automatic Callback or Ringback Queuing feature.

Call Waiting Ringback Tone

A low-pitched tone identical to the ringback 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.

Central Office

The location housing telephone switching equipment that provides local telephone service and access to toll facilities for long-distance calling.

Central Office Codes

The first three digits of a 7-digit public network telephone number. These codes are numbered from 200 through 999.

Central Office Trunk

A telecommunications channel that provides access from the system to the public network through the local central office.

Channel

A communications path for transmitting voice and data.

Class of 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 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 extension, 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)

The equipment on the network side of a communications link that provides all the functions required to make the binary serial data from the source or transmitter compatible with the communications channel.

Data Terminal Equipment (DTE)

Equipment comprising the source or link of data, or both, that also provides communication control functions (protocol). DTE is any piece of equipment at which a communications path begins or ends.

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.

Distributed Communications System (DCS)

A network of two or more switches, each with its terminals and trunks, configured to function as a single large system.

Digital Data Endpoints

In System 75, digital data endpoints include the following:

- 510D Terminals or 515-Type Business Communications Terminals
- 7404D Terminals
- 7407D Equipped With Optional Data Module Base
- Asynchronous Data Units
- Digital Terminal Data Modules
- (Modular) Processor Data Modules
- (Modular) Trunk Data Modules
- Internal Data Channels

Digital Multiplexed Interface (DMI)

Specifies the remote interface requirements for multiplexed data communications between a host computer and a private switching system.

Digital Terminal Data Module (DTDM)

An adjunct to Model 7403D or 7405D voice terminals. Provides the required interface between the system and a data terminal such as a 513 Business Communications Terminal.

A circuit in a telecommunications channel designed to handle digital voice and 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 special tandem tie trunk network that has automatic call routing capabilities based on the number dialed and most preferred route available at the time the call is placed. Each switch in the network is assigned a unique private network office code (RNX) and each voice terminal is assigned a unique extension number.

Endpoint Node

A node (switch), within a Distributed Communications System (DCS), that cannot recieve information from one node and pass it on to another node.

End-to-End Signaling

The transmission of touch-tone signals generated by 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

One- through five-digit number assigned to each voice terminal, certain groups, data modules, 510 Personal Terminals, and 515 Business Communications Terminals within the system. A 1- or 5-digit number is available for Version 2 only.

External Call

A connection between 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 geographic area.

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

A format used to initiate a data connection between two data module devices.

Home Numbering Plan Area Code

The local area code. The area code does not have to be dialed to call numbers within the local geographic area.

Immediate-Start Tie Trunk

After establishing a connection with the distant switching system for an outgoing call, the system waits a nominal 65 milliseconds before sending the digits of the called number. This allows time for the distant system to prepare to receive the digits. Similarly, on an incoming call, the system has less than 65 milliseconds to prepare to receive the digits.

Information Exchange

The exchange of data between users of two different systems (System 75 and host computer) over a local area network.

In-Use Lamp

A red lamp on a multi-appearance voice terminal that lights to show which call appearance will be selected when the handset is lifted or which call appearance is active when a user is off-hook.

Intercept Tone

An alternating high and low tone; indicates a dialing error or denial of the service requested.

Interface

A common boundary between two systems or pieces of equipment.

Internal Call

A connection between two users within the system.

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.

Main/Satellite/Tributary

A Main switch provides: interconnection, via tie trunks, with one or more subtending switches, called Satellites; all attendant positions for the Main/Satellite configuration; and, access to and from the public network. To a user outside the complex, a Main/Satellite configuration appears as a single switch, with a single Listed Directory Number (LDN). A Tributary is a switch, connected to the Main via tie trunks, but which has its own attendant position(s) and its own LDN.

Message Center

An answering service for calls that might otherwise go unanswered; an agent 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.

Modular Processor Data Module

See Processor Data Module.

Modular Trunk Data Module

See Trunk Data Module.

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.

Private Network Office Code (RNX)

The first three digits of a 7-digit private network number. These codes are numbered 220 through 999, excluding any codes that have a 0 or 1 as the second digit.

Processor Data Module (PDM)

Provides the required interface between the system and an EIA 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 by all customers for local or long-distance calling.

Queue

An ordered sequence of calls waiting to be processed.

Queuing

The process of holding calls in order of their arrival to await connection to an attendant, to an answering group, or to an idle trunk. Calls are automatically connected in first-in, first-out sequence.

Random Access Memory (RAM)

A storage arrangement whereby information can be retrieved at a speed independent of the location of the stored information.

Read Only Memory (ROM)

A storage arrangement primarily for information retrieval applications.

Recall Dial Tone

Three short bursts of tone followed by steady dial tone; indicates the system has completed some action (such as holding a call) and is ready to accept dialing.

Redirection Criteria

The information administered for each voice terminal that determines when an incoming call is redirected to coverage.

Remote Home Numbering Plan Area Code (RHNPA)

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 500, 2500, 2554, 7101A, 7103A, and 7104A).

Software

A set of computer programs that accomplish one or more tasks.

Split

A condition whereby a caller is temporarily separated from a connection with the attendant. This split condition automatically occurs when the attendant, active on a call, presses the Start button.

Standard Serial Interface (SSI)

A communications protocol developed by AT&T Teletype Corporation for use with the 500 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, broken flutter, 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 Switch

A switch within an ETN that provides the logic to determine the best route for a network call, possibly modifies the digits outpulsed, and allows or denies certain calls to certain users.

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

A special bus that is time shared by preallocating short time slots to each transmitter on a regular basis. In a PBX, all port circuits are connected to the time division multiplex bus permitting any port to send a signal to any other port.

Tone Ringer

A device with a speaker, used in electronic voice terminals to alert the user.

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.

Uniform Dial Plan

A feature that allows a unique number assignment (4- or 5-digit) for each terminal in a multi-switch configuration, such as a Distributed Communications System (DCS) or Main/Satellite/Tributary configuration.

Wide Area Telecommunications Service (WATS)

A service that allows calls to a certain area or areas for a flat-rate charge based on expected usage.

Wink-Start Tie Trunk

After establishing a connection with a distant switching system for an outgoing call, the system waits for a momentary signal (wink) before sending the digits of the called number. Similarly, on an incoming call, the system sends the wink signal when ready to receive digits.

Write Operation

The 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

AT&T SYSTEM 75 REFERENCE MANUAL
555-200-201 Issue 1

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