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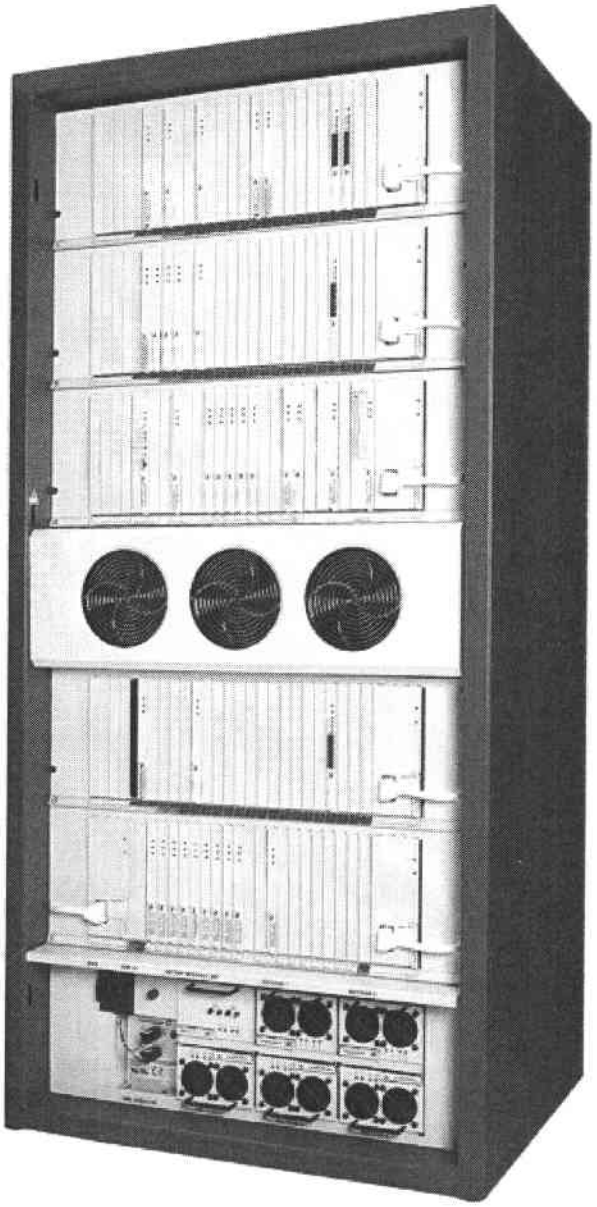
BASIC DEFINITY ADMINISTRATION



Definity Multicarrier Cabinet
PPN

Processor Port Network Cabinet (J58890A)

A Processor Port Network (PPN) cabinet contains the following carriers:

- Port carrier (J58890BB) — 1 to 4
- Control carrier (J58890AH) in R7si and later — 1
- Duplicated control carrier (J58890AJ) R7si and later — 1 in high or critical reliability configurations
- Processor carrier (J58890AP) inR7r and later — 1 in all systems, 2 in high reliability and critical reliability systems
- Switch Node (SN) (J58890SA) in R7r and later with a Center Stage Switch (CSS) — 1 in standard and high reliability systems or 2 in critical reliability systems
- In a R7r and later with ATM, the ATM interface card would be placed in a port carrier.

Carriers in MCCs

The following types of carriers can install in MCCs:

- J58890AH Control Carrier (si model) (J58890AH Control Carrier (si model)) located only in the R7si or later PPN cabinet. Contains SPE circuit packs to perform call processing, maintenance, and administration. These carriers also contain port circuit pack slots.
- Duplicated Control Carrier (si model) (J58890AJ) (optional), in R7si or later PPN only. Contains duplicate SPE circuit packs to perform call processing, maintenance, and administration identical to the Control Carrier. The Duplicated Control Carriers also contain port circuit pack slots. Only R7si/r or later support duplication (uses the R8r Control Carriers for duplication).
- Processor Carrier (r model) (J58890AP), only in the R7r or later PPN cabinets. Contains SPE circuit packs to perform call processing, maintenance, and administration. These carriers do not contain port circuit pack slots. Two J58890AP carriers are in the PPN for high and critical reliability (duplicate processor) systems.
- J58890BB Port Carrier (optional), located in the PPN and EPN cabinets. Contains port, service, and tone/clock circuit packs.

- Expansion Control Carrier (J58890AF), only in the EPN cabinets. Contains extra port circuit packs, tone-clock, maintenance interface, and EI circuit packs.
- Switch Node Carrier (SN) (J58890SA) (optional), in R7r or later, in the PPN cabinet and/or EPN cabinets. Contains SNI circuit packs composing the CSS.

Carrier Circuit-Pack Slots

There are 3 types of circuit pack slots in the carriers: Control, Port, and Service.

NOTE:
The purple-colored and white-colored circuit packs and slots are being replaced by circuit packs and slots labeled with gray and white rectangles, respectively. A label with a solid gray rectangle indicates a port slot/circuit pack. A label with an outlined white rectangle indicates a control slot/circuit pack.

- Port: colored purple or labeled with a gray rectangle and can accept any purple or gray-labeled circuit pack
- Control: colored white or labeled with an outlined white rectangle and can accept only a circuit pack assigned to that slot.
- Service: colored purple or labeled with a gray rectangle; is a special type of circuit pack that does not have an I/O connector

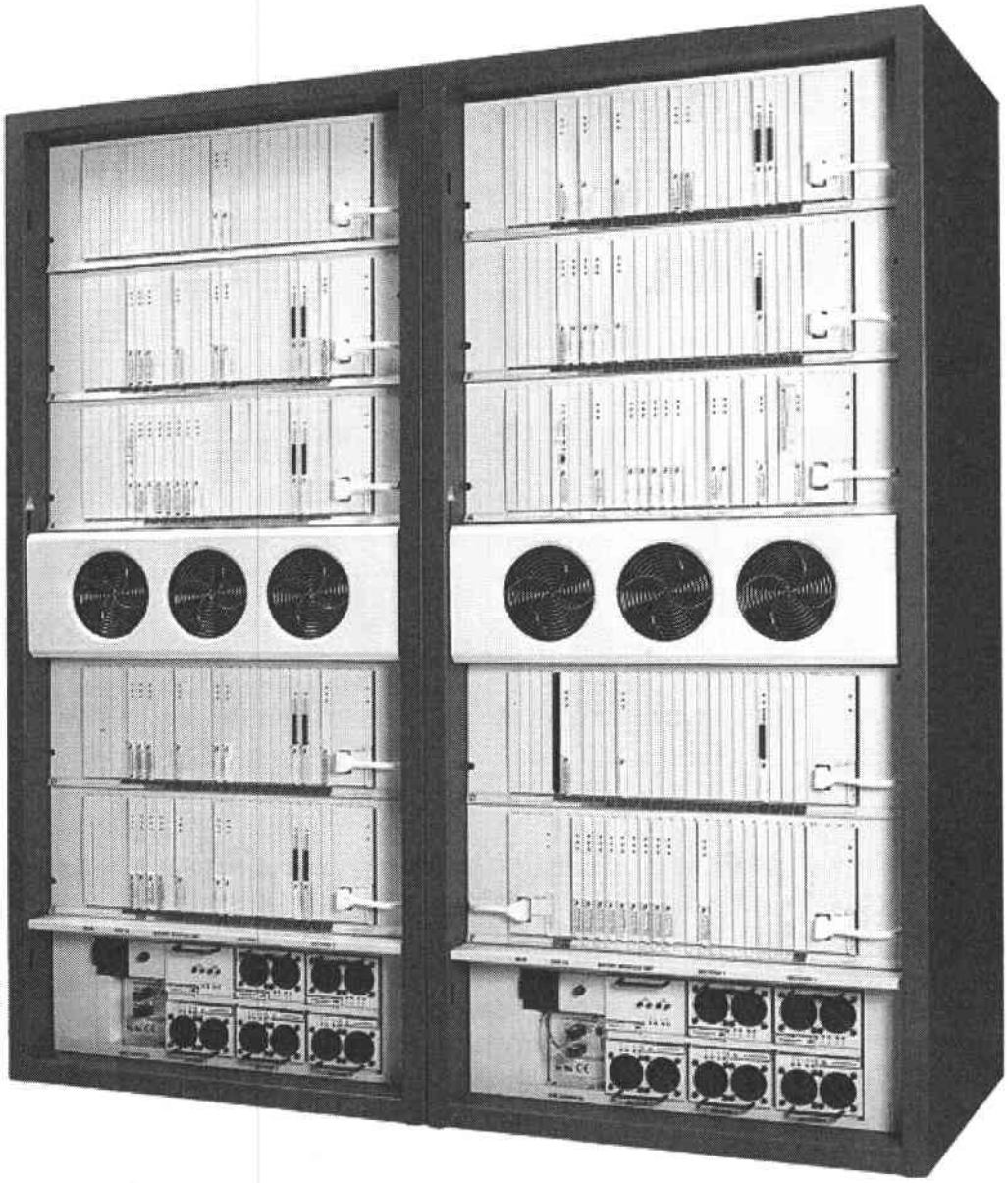
Each port slot attaches to a 50-pin (25-pair) connector on the carrier's rear panel. A cable attaches to each connector and routes to the cross-connect field. Each slot containing a fiber optic interface circuit pack (EI or SNI) uses a fiber optic transceiver on the carrier's rear panel.

A current limiter board (CFY1B) plugs into the backplane of the control carrier located in the *A* position only. The board supplies emergency transfer logic, current-limited power, 5 VDC to trip the main circuit breaker in an over-temperature condition, and the ringing transfer relay. Terminators on the backplane terminate each end of the processor expansion bus.

J58890BB Port Carrier

A Port Carrier contains the following circuit packs:

- Port slot locations 1 to 20 for the port circuit packs. A dedicated slot contains an optional tone-clock circuit pack used for Port Carriers in the B position of an EPN cabinet in critical reliability systems. Slot 2 contains an optional EI or ATM Interface circuit pack
- Power unit service slots in which power unit circuit packs or maintenance circuit packs can install
- AC or DC power units located at each end of the carrier

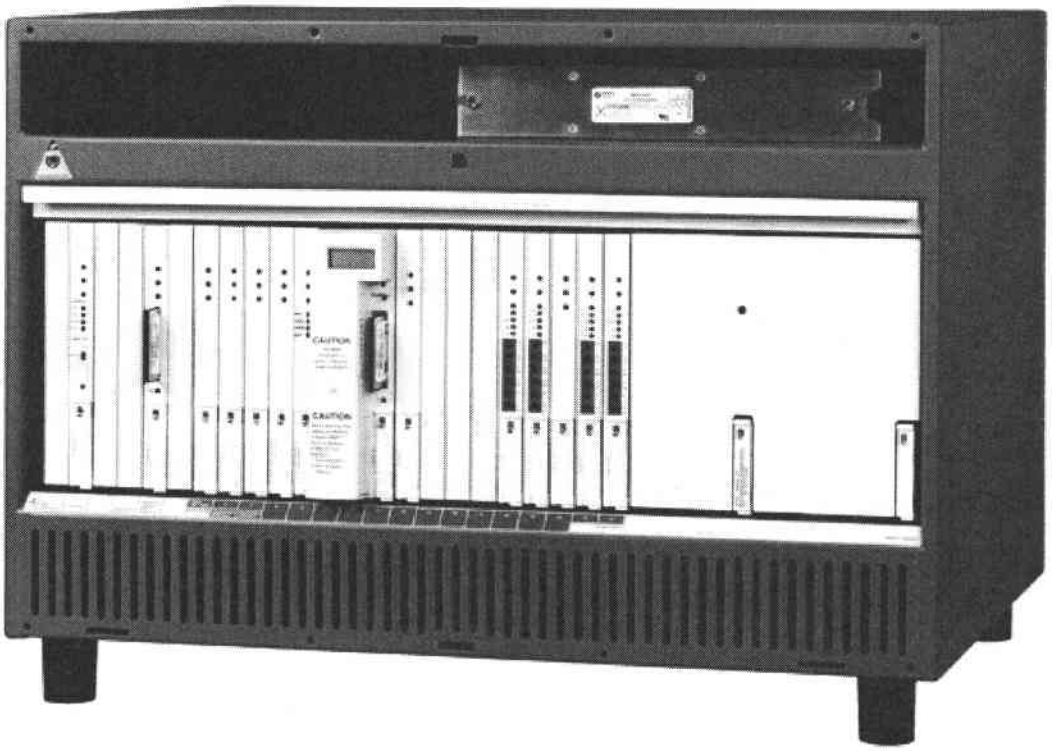


Definity Multicarrier Cabinet
PPN with EPN

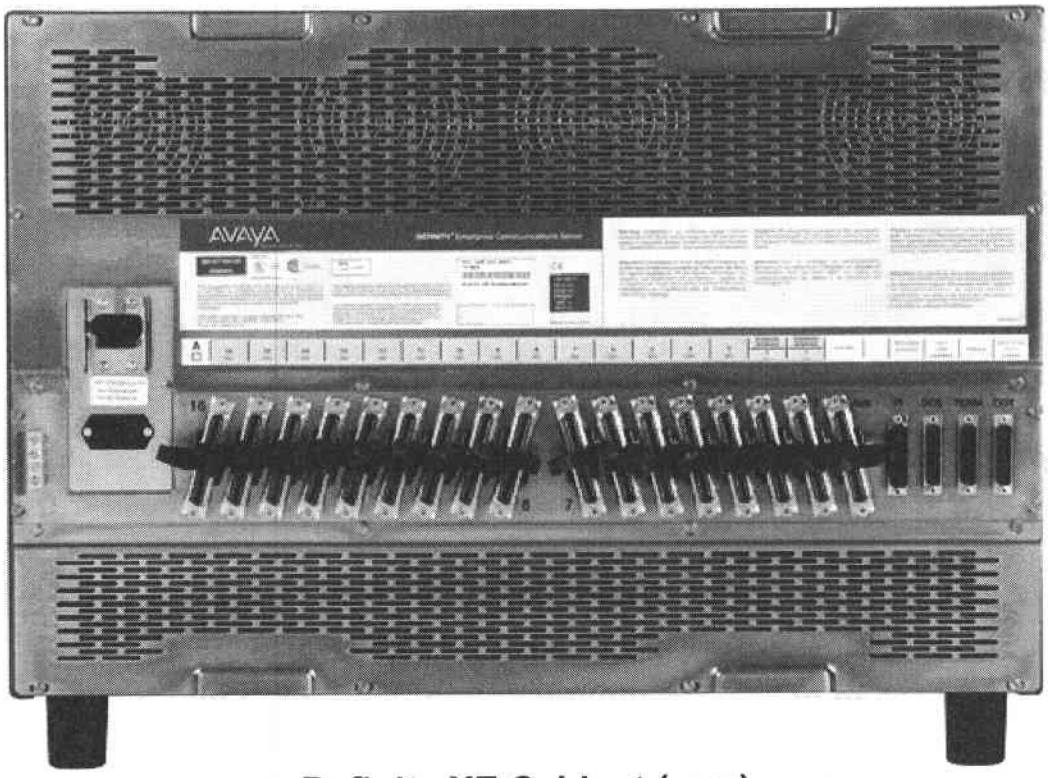
Expansion Port Network Cabinet (J58890A)

An Expansion Port Network (EPN) cabinet contains the following carriers:

- Port carrier (J58890BB) — 1 to 4
- Expansion control carrier (J58890AF) — 1
- SN Carrier (J58890SA) in CSS-connected R7r and later systems — 0, 1, or 2 when required



Definity XE Cabinet (front)



Definity XE Cabinet (rear)

Single-Carrier Cabinets

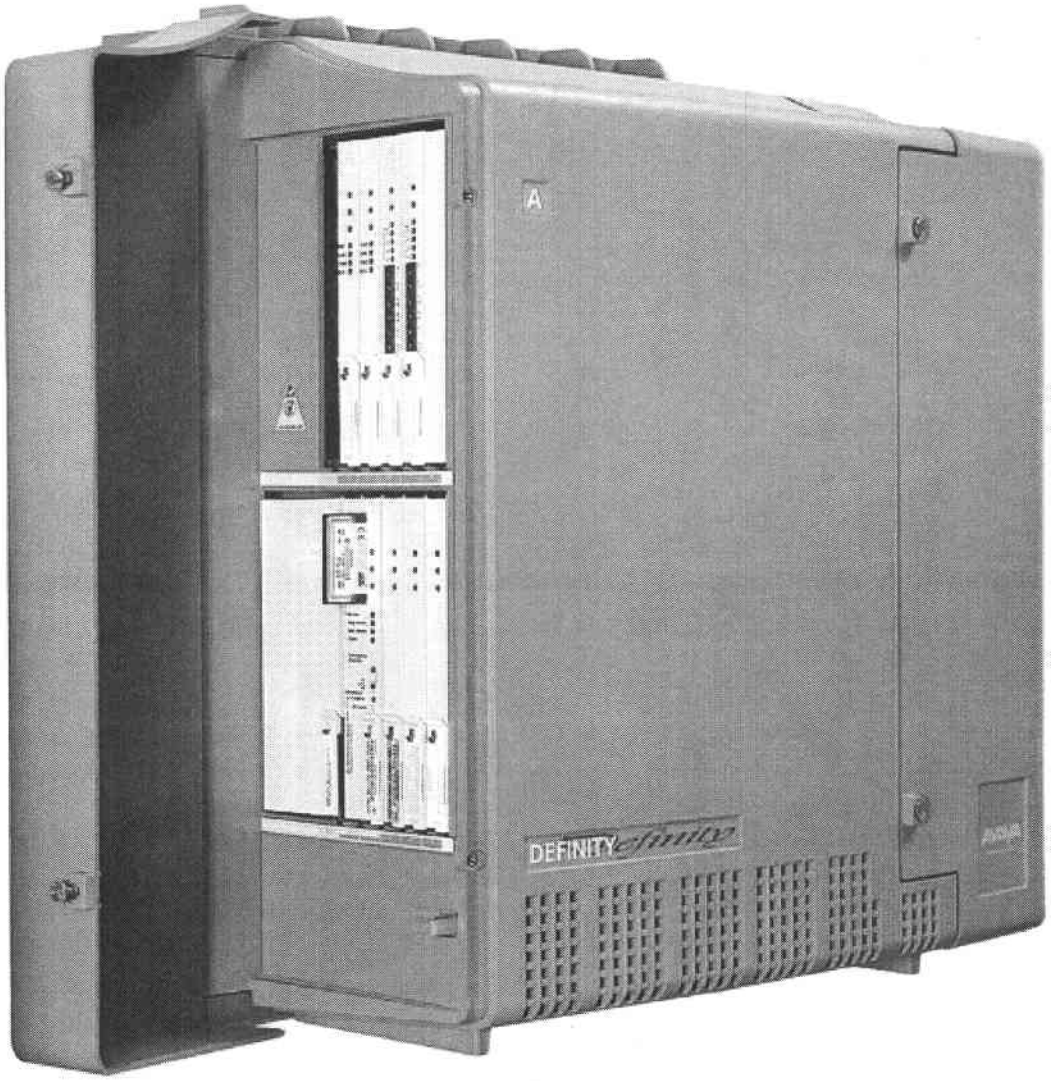
This section describes the following types of Single-Carrier Cabinet (SCC):

- Basic Control Cabinet (si model) (J58890L)
- Duplicated Control Cabinet (J58890M)
- Expansion Control Cabinet (J58890N)
- Port Cabinet (J58890H)
- Compact Modular Cabinet (csi model) (J58890T)
- DC power distribution cabinet

A maximum of 4 SCCs can stack on top of each other. The cabinet positions are labeled A through D. The position of the basic control cabinet or expansion control cabinet is always labeled A. Additional port cabinet positions are labeled B, C, and D, sequentially. The Duplicated Control Cabinet is labeled B.

Each stack of SCCs requires 1 basic- or Expansion-Control Cabinet at the bottom of the stack. Three is the maximum number of port cabinets per stack.

Cabinet clips connect the cabinets together. At the rear of the cabinets, a ground plate connects between cabinets for ground integrity.

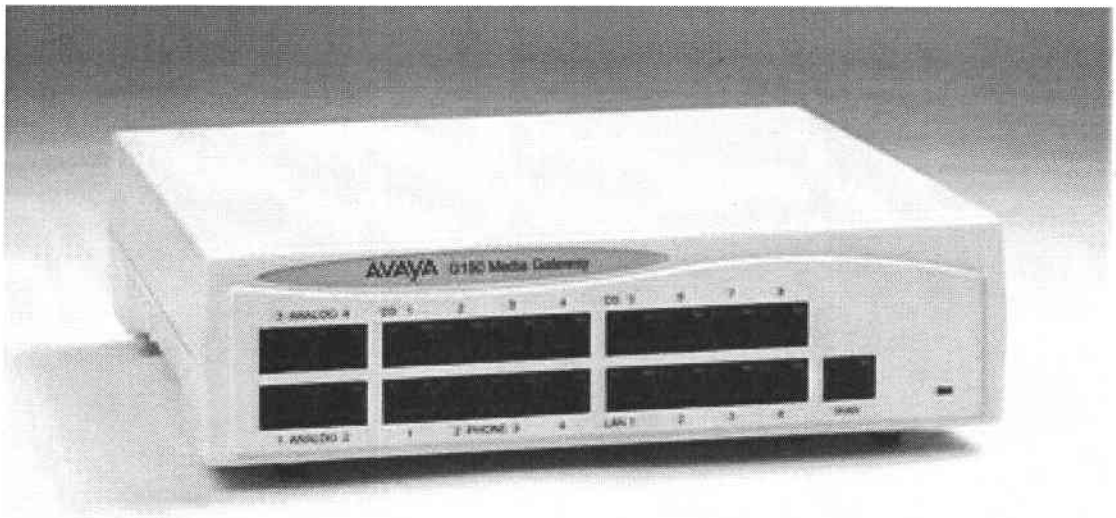


Definity Prologix Cabinet

J58890T Compact Modular Cabinet (csi model)

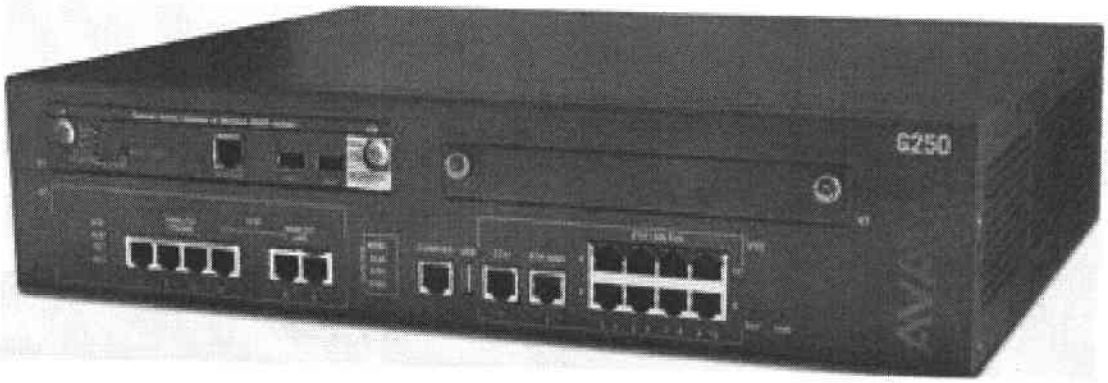
The Compact Modular Cabinet is an economical, small-footprint alternative to a Single-Carrier Cabinet. It can mount on a wall or on the floor, and uses an AC-only power supply. The control carrier contains 2 control slots: the processor has to be in slot 1 and the tone-clock in slot 2. Slots 3 to 10 can contain optional port and service circuit packs.

Up to three Compact Modular Cabinets can be combined in a single installation. Port and service circuit packs fill all ten slots in the second and third cabinets. The first cabinet (A) installs in the middle position, the second cabinet (B) installs on the top, and the third cabinet (C) installs on the bottom.



G150 Media Gateway

The **G150 Media Gateway** is designed for smaller office use. Included in the G150 is a four port switch that can be a T1 or WAN interface. The G150 can provide up to four IP stations.

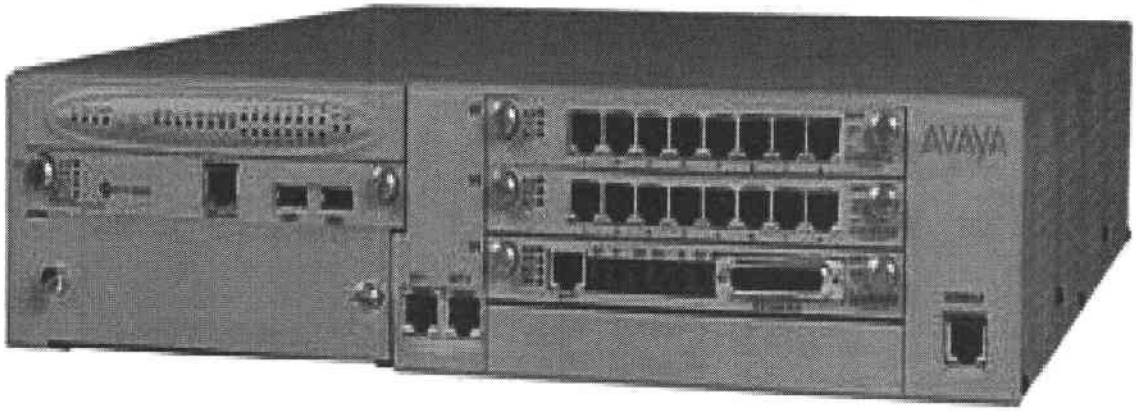


G250 Media Gateway

The **G250 Media Gateway** is designed as an extension gateway for a branch office. It can provide communication applications from a headquarter location and with the addition of an S8300 processor, the G250 can act in a stand-alone or survivable mode. It can provide two to twelve IP and analog stations in a small branch location.

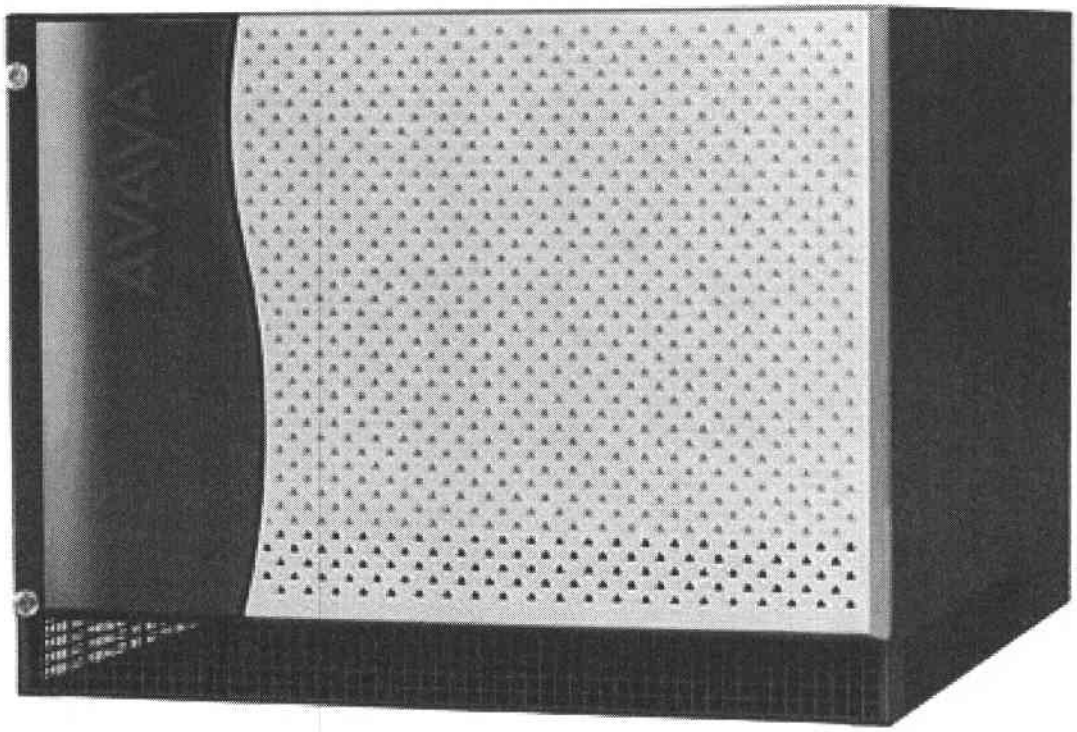
G350 Media Gateway

Similar to the G250 Media Gateway, the **G350 Media Gateway** can provide branch office connectivity to a headquarters office. It can allow between eight and forty extensions to be administered. Once again, an S8300 processor can provide stand-alone or survivable capability.



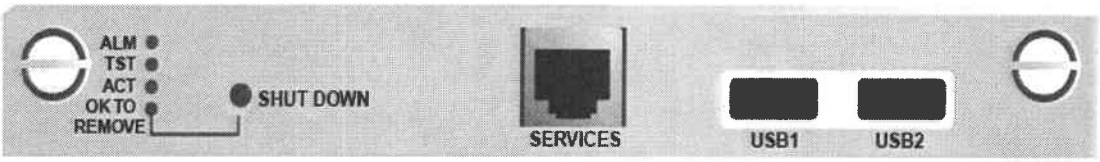
G700 Media Gateway

The **G700 Media Gateway** can be used as a stand-alone or survivable gateway with the addition of an S8300 server. It can accommodate 40 to 450 extensions. As with the G250/G350, the G700 Media Gateway is designed to work as a branch office in a home office configuration.



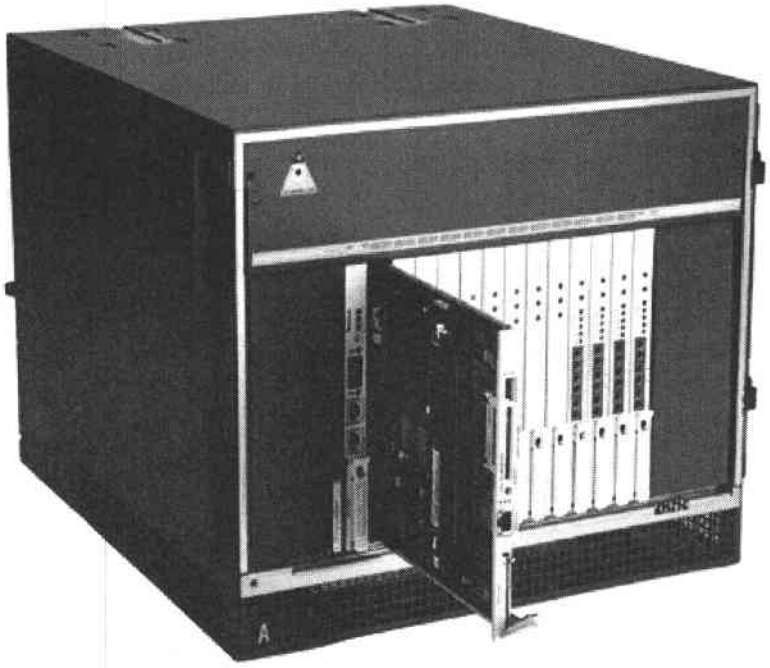
G650 Media Gateway

The **G650 Media Gateway** provides the capability to allow both IP and TDM telephony to exist in one system. It provides a 14-slot cabinet that can accommodate analog, digital, ISDN, T1, and IP circuit packs. Multiple G650 cabinets can be stacked to allow high capacity TDM or IP capability. The G650 Media Gateway connects to a S8400, S8500, or an S8700/8710/8720 server over an IP link using an IPSI or SIPI interface circuit pack.



S8300 Media Server

The **S8300 Media Server** is an Intel Celeron-based Processor that resides in a G250, G350, or G700 Media Gateway. It inserts in a media module slot in the front panel of the gateways and can serve as a stand-alone or local survivable processor. The S8300 can be the primary controller of up to 50 remote G250, G350, and G700 Media Gateways.



S8400 Media Server

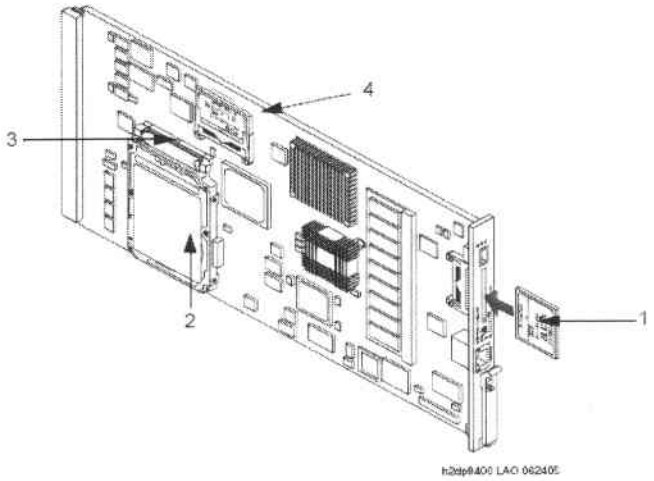


Figure notes:

1. **Compact flash**
2. **Hard disk drive**
3. **Ribbon cable to hard disk drive**
4. **Solid state drive**

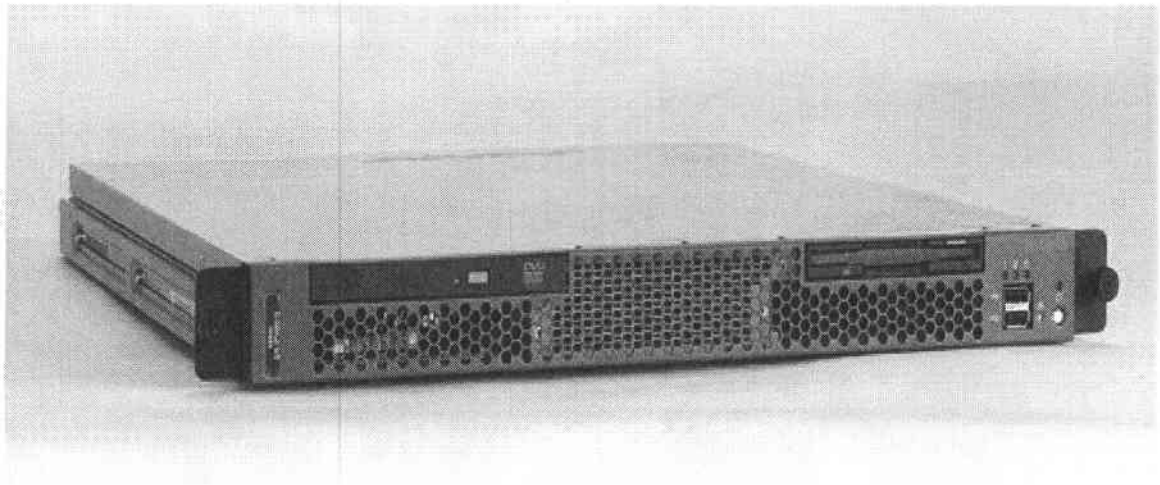
S8400 Media Server Circuit Pack

The **S8400 Media Server** is composed of two circuit packs:

TN8400AP Media Server

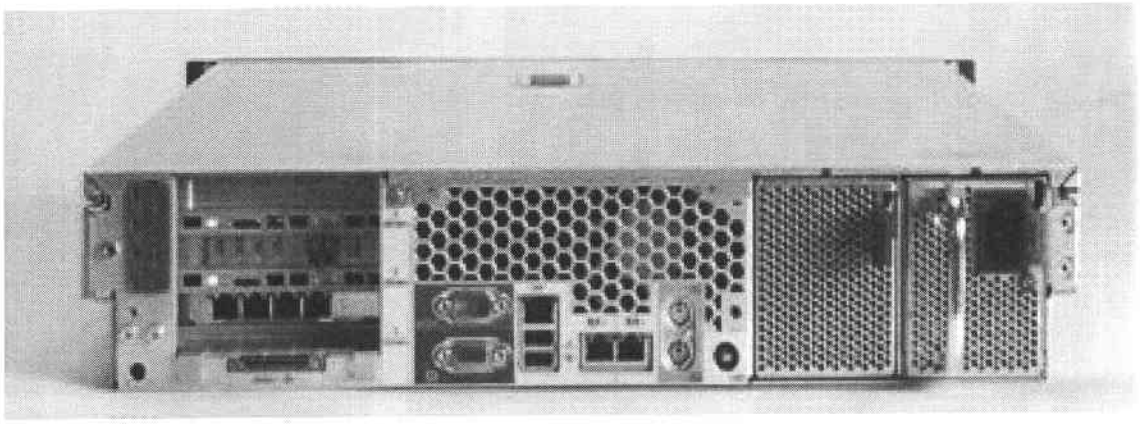
TN8421AP S8400 IP Interface (SIPI)

The S8400 Media Server resides in a G650 Media Gateway and provides processing for both IP and TDM Telephony. The SIPI circuit pack acts as an interface between the S8400 Media Server and the TDM circuit packs in the G650 Gateway. The Server also provides support for up to five gateways including the G700, G350, G250 and the G150.



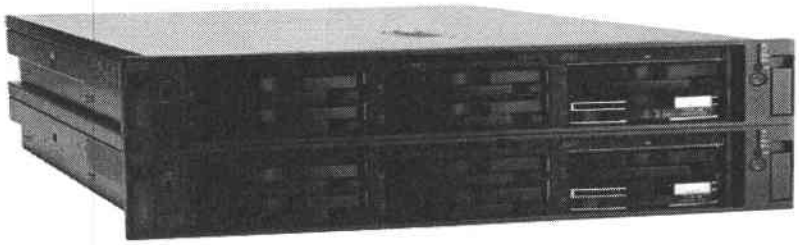
S8500 Media Server

The S8500 Media Server is a Pentium 4 based server serving up to 3200 ports. It is designed for the mid-sized customer.



S8710 Media Server

The S8710 Media Server is the up-graded version of Avaya's S8700 server. The S8710 uses two processors for reliability and they can be separated by over six miles to insure continuity. The S8710 server can support up to 36000 stations of which 12000 can be IP stations.



S8720 Media Server

The S8720 Media Server is Avaya's latest server designed for large multinational corporations. It is powered by an AMD Opteron processor that can support 36000 stations of which 12000 can be IP stations. It can support 8000 trunks and a total of 44000 ports. As with the S8700/S8710, the S8720 Media Server is configured with two processors for reliability.

Logging into the system

You must log in before you can administer your system. If you are performing remote administration, you must establish a remote administration link and possibly assign the remote administration extension to a hunt group before you log in. The members of this hunt group are the extensions of the data modules available to connect to the system administration terminal.

NOTE:
Change your password frequently, at least once a month, to help keep hackers out of your system.

When not using the system, log off for security purposes.

Logging into the system

This procedure provides instructions for logging in from the system terminal not a remote terminal.

To log into the system:
1. Enter your login name and press **RETURN.**
2. Enter your password and press **RETURN.**
For security, your password does not display as you type it.
3. Enter the kind of terminal you have or the type your system emulates and press **RETURN.**
The Command prompt appears.

NOTE:
If you enter the wrong terminal type, it can lock up your system. If the system is not responding to keyboard commands, type **newterm** and press **RETURN.** Enter the correct terminal type on the new screen and press **RETURN.** If this does not work, turn the power off only on the terminal and then turn it back on. The terminal reboots and you can login again.

Accessing the S8700 Series Media Server

To administer the S8700 Series Media Server, you must be able to access it. Personal computers and services laptop computers equipped with a network PCMCIA card, Avaya Site Administration (ASA), and a Web browser are the primary support access for system initialization, aftermarket additions, and continuing maintenance.

You can access the S8700 Series Media Server in one of three ways:

- directly
- remotely over the customer's local area network (LAN)
- over a modem

A direct connection and over the customer's LAN are the preferred methods. Remote access over a modem is for Avaya maintenance access only.

Accessing the S8700 Series Media Server Directly

You can access the S8700 Series Media Server directly by plugging a computer into the services port which defaults to port 2 (Eth1) on the back of the media server. You must use a crossover cable with an RJ45 connector on each end. Plug the other end into the network connector (NIC card) on the your computer. You might need a NIC card adapter. Once connected, you can administer the media server using three tools:

- Media server web interface for server-specific administration.
- ASA for various features of Avaya Communication Manager
- An SSH client, like PuTTY, and an IP address of 192.11.13.6.

Media Server Web Interface - You can access the media server web interface either by connecting directly to the services port on the media server or connecting over the customer's network.

Connected to the services port - To use the media server web interface:

1. Open either the Netscape or MS Internet Explorer browser.
2. In the **Location/Address** field, type **192.11.13.6**. Press **Enter**.
3. When prompted, log in to administer the S8700 Series Media Server and the features of Avaya Communication Manager.

Connected to the customer network - To use the media server web interface:

1. Open either the Netscape or MS Internet Explorer browser.
2. In the **Location/Address** field, type the active server name or IP address. Press **Enter**.
3. When prompted, log in to administer the S8700 Series Media Server and the features of Avaya Communication Manager.

   You can also connect directly to an individual server using its name or IP address.

Accessing the S8700 Series Media Server Remotely over the network

You can access the S8700 Series Media Server from any computer connected through the LAN. To access either server, use the IP address assigned to the server you want to access. You can also use the active server address to connect automatically to the server that is active. Once connected, you can administer the media server using three tools:

- Media server web interface for server-specific administration and call processing features
- Avaya Site Administration for Communication Manager (Only available on the active Communication Manager server)
- An SSH client, like PuTTY, and an IP address of 192.11.13.6.

Using Avaya Site Administration

Avaya Site Administration features a graphical user interface (GUI) that provides access to SAT commands as well as wizard-like screens that provide simplified administration for frequently used features. You can perform most of your day-to-day administration tasks from this interface such as adding or removing users and telephony devices. You can also schedule tasks to run at a non-peak usage time. ASA is available in several languages.

The S8300, S8400, or S8700 Series Media Server can be used to download Avaya Site Administration. A downloadable version of this package can be accessed through the S8300, S8400, or S8700 Series Media Server Web Interface. This software must be installed on a computer running a compatible Microsoft Windows operating system such as Windows 95, 98, NT 4.0, Millennium Edition, Windows 2000, or Windows XP. Once installed, it can be launched from a desktop icon, from the P330 Device Manager, or through a link in the S8300 Media Server Web Interface.

Logging in with Access Security Gateway

Access Security Gateway (ASG) is an authentication interface used to protect the system administration and maintenance ports and logins associated with Avaya Communication Manager. ASG uses a challenge and response protocol to validate the user and reduce unauthorized access.

You can administer ASG authentication on either a port type or login ID. If you set ASG authentication for a specific port, it restricts access to that port for all logins. If you set ASG authentication for a specific login ID, it restricts access to that login, even when the port is not administered to support ASG.

Authentication is successful only when Avaya Communication Manager and the ASG communicate with a compatible key. You must maintain consistency between the Access Security Gateway Key and the secret key assigned to the Communication Manager login. Before you can log into the system with ASG authentication, you need an Access Security Gateway Key, and you need to know your personal identification number (ASG). The Access Security Gateway Key must be pre-programmed with the same secret key (such as, ASG Key, ASG Passkey, or ASG Mobile) assigned to the Avaya Communication Manager login. Verify that the **Access Security Gateway (ASG)** field on the **System-Parameters Customer Options** screen is set to **y**. If not, contact your Avaya representa

Logging in with ASG

To log into the system with ASG:

1. Enter your login ID. Press **Enter**.

   The system displays the challenge number (for example, 555-1234) and system Product ID number (for example, 1000000000). The Product ID provides Avaya Services with the specific identifier of your Avaya MultiVantage communications application.

2. Press **ON** to turn on your Access Security Gateway Key.

3. Type your PIN. Press **ON**.

   The Access Security Gateway Key displays a challenge prompt.

4. At the challenge prompt on the Access Security Gateway Key, type the challenge number without the "-" character (for example, 5551234) from your screen. Press **ON**.

   The Access Security Gateway Key displays a response number (for example, 999-1234).

5. At the response prompt on your terminal, type the ASG response number without the "-" character (for example, 9991234). Press **Enter**.

   The Command prompt displays.

Note:
If you make 3 invalid login attempts, the system terminates the session. For more information, see the appropriate maintenance book for your system.

Logging off the system

For security, log off any time you leave your terminal. If you use terminal emulation software to administer the switch, log off the system and exit the emulation application before switching to another software package.

Saving translations

Instructions

To save translations manually:

1. Type **save translation** and press **RETURN.**

The save process can take up to 10 minutes. You cannot administer your system while the save is in process. The Save Translation screen appears.

SAVE TRANSLATION

Processor	Command Completion Status	Error Code
SPE_A	Success	0
SPE_B	Success	0

2. If there is an error message in the Command Completion Status field and an error code in the Error Code field, clear the error and repeat the save process.

System-wide settings

There are some settings that you enable or disable for the entire system, and these settings affect every user. You may want to look over the various System Parameters screens and decide which settings best meet the needs of your users.

To see a list of the different types of parameters that control your system, type **display system-parameters** and press HELP. You can change some of these parameters yourself. Type **change system-parameters** and press HELP to see which types of parameters you can change. In some cases, an Avaya representative is the only person who can make changes, such as to the System Parameters Customer-Options screen.

System-Parameters Customer-Options

This screen shows you which optional features are enabled for your system, as determined by the installed license file. All fields on this screen are display only. If you have any questions about disabling or enabling one of these features, contact your Avaya representative.

Field descriptions for page 1

```
 display system-parameters customer-options                          Page 1 of x

                                OPTIONAL FEATURES

  G3 Version: V12 123456789012
                                                        RFA System ID (SID):
  Location: 2                                           RFA Module ID (MID):
123456
  Platform: 2
                                                                        USED
                                          Platform Maximum Ports: 300   174
                                                Maximum Stations: 300   174
                                       Maximum XMOBILE Stations: 30     28
                                Maximum Off-PBX Telephones - EC500: 1200  0
                                  Maximum Off-PBX Telephones - OPS: 1200  0
                                Maximum Off-PBX Telephones - SCCAN: 0     0


(NOTE: You must logoff & login to effect the permission changes.)
```

System Parameters Customer-Options screen

G3 Version

Identifies the version of Avaya Communication Manager being used.

Location

Indicates the location of this media server or switch. 1 indicates Canada or the United States. 2 indicates any other location, and allows the use of International Consolidation circuit packs and telephones.

Maximum Off-PBX Telephones - EC500

Valid entries	Usage
0 to license max	A number greater than zero should appear in either this field or the **Maximum Off-PBX Telephones - OPS** field. The "license max" value is defined as follows: • On legacy systems, the upper limit is 1/2 of the maximum number of administrable stations. • On Linux systems, the upper limit is the maximum number of administrable stations. Stations that are administered for any Extension to Cellular/OPS application count against this limit. Default is **0**.

Maximum Off-PBX Telephones – OPS

Valid entries	Usage
0 to license max	A number greater than zero should appear in either this field or the **Maximum Off-PBX Telephones - EC500** field. The "license max" value is defined as follows: • On legacy systems, the upper limit is 1/2 of the maximum number of administrable stations. Note that legacy platforms do not support SIP Enablement Services (SES) trunks. • On Linux systems, the upper limit is the maximum number of administrable stations. Stations that are administered for any Extension to Cellular/OP application count against this limit. Default is **0**.

Maximum Off-PBX Telephones – SCCAN

Valid entries	Usage
0 to license max	The "license max" value is defined as follows: • SCCAN is only available on Linux systems. The upper limit is The maximum number of administrable stations. • Stations that are administered for any Extension to Cellular/OPS application count against this limit. Default is **0**.

Maximum Stations

Displays the maximum number of stations allowed in the system. This feature is set based on the CM license file. Default is **0**.

Maximum XMOBILE Stations

Specifies the maximum number of allowable XMOBILE stations. In general, each XMOBILE station is assigned to a wireless handset. Each XMOBILE station counts as a station and a port in terms of system configuration.

Platform

A display-only field that identifies, via a number mapping, the platform being used for a specific customer.

Platform Maximum Ports

Number of ports active, per contract.

Used

Shows the actual current usage as compared to the system maximum for each field.

Field descriptions for page 2

```
display system-parameters customer-options                          page 2 of x

OPTIONAL FEATURES


IP PORT CAPACITIES                                                       USED

                                  Maximum Administered H.323 Trunks:
                                     Maximum Administered IP Trunks: 100  96
                           Maximum Concurrently Registered IP Stations: 10   10
                             Maximum Administered Remote Office Trunks: 0    0
              Maximum Concurrently Registered Remote Office Stations: 0    0
                          Maximum Concurrently Registered IP eCons: 0    0
                              Maximum Video Capable H.323 Stations: 0    0
                               Maximum Video Capable IP Softphones: 0    0
                                   Maximum Administered SIP Trunks: 500  25
                 Maximum Number of DS1 Boards with Echo Cancellation: 0    0
                                       Maximum TN2501 VAL Boards: 1    0
                                Maximum G250/G350/G700 VAL Sources: 0    0
                      Maximum TN2602 Boards with 80 VoIP Channels: 20  12
                            Maximum TN2602 Boards with 320 VoIP Channels: 4    3
                 Maximum Number of Expanded Meet-me Conference Ports: 0    0


       (NOTE: You must logoff & login to effect the permission changes.)
```

System Parameters Customer-Options screen

IP PORT CAPACITIES

Maximum Administered IP Trunks

Defines limits of the number of IP trunks administered.

Maximum Administered Remote Office Trunks

Defines limits of the number of IP endpoints based on the endpoint. Use the smaller of this number or the number based on the **MAXIMUM IP REGISTRATIONS BY PRODUCT ID** page of this screen.

Maximum Administered SIP Trunks

Defines limits on the number of SIP Enablement Services (SES) trunks administered.

Maximum Concurrently Registered IP eCons

Specifies the maximum number of IP SoftConsoles that can be registered at one time. The maximum number depends on the type of system.

Maximum Concurrently Registered IP Stations

Specifies the maximum number of IP stations that can be registered at one time. This field accepts 6,000 concurrently registered IP stations for the S8700 series servers, and 3,000 for S8500 servers.

Maximum G250/G350/G700 VAL Sources

Specifies the maximum number of VAL announcement sources.

Maximum Number of DS1 Boards with Echo Cancellation

Shows the number of DS1 circuit packs that can have echo cancellation.

Maximum Number of Expanded Meet-me Conference Ports

Displays the license-file based value of the system maximum for the number of Expanded Meet-me Conference ports.

Maximum TN2501 VAL Boards

This display-only field indicates the maximum number of TN2501AP (Voice Announcement over LAN) boards allowed in this system.

Valid entries	Usage
0 to **10** (S8700 Series Media Servers)	• For values greater than 1, the **Val Full 1-Hour Capacity** field on page 4 of the **System-Parameters Customer-Options** screen must be set to **y**.
0 to **5** (DEFINITY CSI, and S8300 Media Server)	• This field updates the **System Limit** field on the **System Capacity report.**

Maximum TN2602 Boards with 80 VoIP Channels

Valid entries	Usage
0 to license truncation limit.	This field defines the total number of TN2602AP boards that can be administered with 80 VoIP channels. The value is based on the value in the Avaya Communication Manager license file.The **USED** value is the total number of TN2602AP boards in the system administered with 80 VoIP channels. Default is **0**.

Maximum TN2602 Boards with 320 VoIP Channels

Valid entries	Usage
0 to license truncation limit.	This field defines the total number of TN2602AP boards that can be administered with 320 VoIP channels. The value is based on the value in the Avaya Communication Manager license file. The **USED** value is the total number of TN2602AP boards in the system administered with 320 VoIP channels. Default is 0

Maximum Video Capable H.323 Stations

Specifies the maximum number of H.323 stations that are video-capable. The maximum number depends on the type of system.

Maximum Video Capable IP Softphones

Specifies the maximum number of IP Softphones that are video-capable. The maximum number depends on the type of system.

Used

For each item with a capacity listed, the **USED** value is the actual number of units currently in
use.

Field descriptions for page 3

```
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                         OPTIONAL FEATURES

      Abbreviated Dialing Enhanced List?                Audible Message
Waiting?
          Access Security Gateway (ASG)?               Authorization Codes?
           Analog Trunk Incoming Call ID?                          CAS
Branch?
       A/D Grp/Sys List Dialing Start at 01?                   CAS Main?
    Answer Supervision by Call Classifier?             Change COR by FAC?
                                  ARS?          Computer Telephony Adjunct Links?
                  ARS/AAR Partitioning?            Cvg Of Calls Redirected Off-net?
             ARS/AAR Dialing without FAC?                   DCS (Basic)?
                ASAI Link Core Capabilities?             DCS Call Coverage?
                ASAI Link Plus Capabilities?            DCS with Rerouting?
           Async. Transfer Mode (ATM) PNC?
     Async. Transfer Mode (ATM) Trunking?          Digital Loss Plan Modification?
                 ATM WAN Spare Processor?                        DS1 MSP?
                                    ATMS?             DS1 Echo Cancellation?
                     Attendant Vectoring?
```

System Parameters Customer-Options screen

Abbreviated Dialing Enhanced List

Provides the capability to store and retrieve dialing lists that simplify or eliminate dialing. You dial an abbreviated code or depress an assigned button. The stored entries are organized in number lists. There are three types of number lists: personal, group, and enhanced.

Access Security Gateway (ASG)

Provides an additional level of security for remote administration.

A/D Grp/Sys List Dialing Start at 01

Allows you to number Abbreviated Dialing group or system lists starting with 01, rather than simply 1. This allows Abbreviated Dialing under Avaya Communication Manager to operate like it did with the DEFINITY G2 system.

Analog Trunk Incoming Call ID

This field allows collection and display the name and number of an incoming call information on analog trunks.

Answer Supervision by Call Classifier

This circuit pack detects tones and voice-frequency signals on the line and determines whether a call has been answered. This field is set to **y** if the system contains a call-classifier circuit pack.

ARS

Provides access to public and private communications networks. Long-distance calls can be routed over the best available and most economical routes. Provides partitioning of ARS routing patterns.

ARS/AAR Partitioning

Provides the ability to partition AAR and ARS into 8 user groups within a single server running Avaya Communication Manager. Can establish individual routing treatment for each group.

ARS/AAR Dialing without FAC

Provides for Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) calls without dialing a feature access code (FAC).

ASAI Link Core Capabilities

Provides linkage between Avaya Communication Manager and adjuncts. CallVisor ASAI improves the call handling efficiency of ACD agents and other system users by allowing an adjunct to monitor, initiate, control, and terminate calls on the server running Communication Manager.

If the **ASAI Link Core Capabilities** field is administered to **y** then it will be associated with the following ASAI capability groups:

- Adjunct Control
- Domain Control
- Event Notification
- Single Step Conference
- Request Feature

- II Digits
- Set Value
- Value Query

ASAI Link Plus Capabilities

Provides linkage between Avaya Communication Manager and adjuncts. If the **ASAI Link Plus Capabilities** field is administered to **y**, then the following ASAI capability groups are enabled:

- Adjunct Routing
- Answering Machine Detection
- Selective Listening
- Switch Classified Outbound Calls
- ISDN Redirecting Number Information - the original dialed number information is provided within the ASAI messages if it arrives in ISDN SETUP messages from the public networks as either Original Dialed Number or Redirecting Party Number.

Asynch. Transfer Mode (ATM) PNC

ATM PNC can be enabled only if:

- all prior fiber-link administration has been removed
- all "switch-node" and "dup-switch-node" carrier types have been removed

Asynch. Transfer Mode (ATM) Trunking

If ATM trunking is enabled, multiple ISDN-PRI T1 or E1 trunks can be emulated on one ATM pipe. Can only be enabled if the **ISDN-PRI** field is set to **y**. Enables circuit emulation service (CES).

ATM WAN Spare Processor

An ATM WAN spare processor acts as a PPN in the event of network failure, and can function as an SPE if the main PPN is not functional. Cannot be set to **y** if the **Async. Transfer Mode ATM) Trunking** field is **n**

.

ATMS

Provides for voice and data trunk facilities to be measured for satisfactory transmission performance.

Attendant Vectoring

Allows you to use attendant vectoring. Cannot be set to **y** if the **CAS Main** and **CAS Branch** fields are **y**.

Audible Message Waiting

Provides audible message waiting.

Authorization Codes

Permits you to selectively specify levels of calling privileges that override in-place restrictions. In addition to facilities access, authorization codes are used for unique identification for billing security purposes.

CAS Branch

Provides Centralized Attendant Service - Branch. See **CAS Main** for more information. Cannot be set to **y** if the **Attendant Vectoring** and **Centralized Attendant** fields are **y**.

CAS Main

Provides multi-location customers served by separate switching vehicles to concentrate attendant positions at a single, main Avaya Communication Manager location. The main Avaya Communication Manager is served by an attendant queue that collects calls from all locations (main and branch). Each branch location switches all of its incoming calls to the centralized attendant positions over release link trunks (RLTs). The calls are then extended back to the requested extension at the branch server/switch over the same RLT. When the call is answered, the trunks to the main server are dropped and can be used for another call. Cannot be set to **y** if the **Centralized Attendant** and **CAS Branch** fields are **y**.

Change COR by FAC

Provides certain users the ability to change the class of restriction of local extensions and local attendants via a telephone by using a feature access code (FAC). Cannot be set to **y** if the **Tenant Partitioning** field is **y**.

Computer Telephony Adjunct Links

Provides linkage between Avaya Communication Manager and adjuncts. Includes both the ASAI Link Core and ASAI Link Plus capabilities, plus the Phantom Calls and CTI Stations.

Cvg Of Calls Redirected Off-net

Provides continued monitoring for calls redirected to off-network (remote) coverage points. Uses call classification via call classifier circuit pack or ISDN trunk signaling.

DCS (Basic)

Provides transparent operation of selected features across a Distributed Communications System (DCS). Users on one server running Communication Manager can use features located on another server. Includes 4- and 5-digit uniform dialing and 1 to 4 digit steering. Does not support a 6/7-digit dial plan.

DCS Call Coverage

Provides DCS-based transparency of the call coverage feature across a DCS network of media servers or switches.

DCS with Rerouting

Provides for rerouting calls transferred among DCS nodes, enabling rerouting of the call for more effective use of facilities. Cannot be set to **y** if the **ISDN PRI** field is **n**.

Digital Loss Plan Modification

Allows you to customize the digital loss and digital tone plans.

DS1 MSP

Provides the ability to change fields on **DS1 Circuit Pack** screen without removing the related translations of all trunks from the trunk group.

DS1 Echo Cancellation

Removes perceivable echo from the system.

Field descriptions for page 4

```
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                          OPTIONAL FEATURES


     Emergency Access to Attendant? y                          IP Stations? Y
             Enable 'dadmin' Login? y           Internet Protocol (IP) PNC? y
             Enhanced Conferencing? y                    ISDN Feature Plus? y
                    Enhanced EC500? y        ISDN Network Call Redirection? y
      Enterprise Survivable Server? y
         Enterprise Wide Licensing? y                       ISDN-BRI Trunks? y
                ESS Administration? y                             ISDN-PRI? y
           Extended Cvg/Fwd Admin? y           Local Survivable Processor? y
       External Device Alarm Admin? y                 Malicious Call Trace? y
           Extended Cvg/Fwd Admin? y Mode Code for Centralized Voice Mail? y
       External Device Alarm Admin? y
    Five Port Networks Max per MCC? y              Multifrequency Signaling? y
             Flexible Billing? y   Multimedia Appl. Server Interface(MASI)? y
     Forced Entry of Account Codes? y     Multimedia Call Handling (Basic)? y
        Global Call Classification? y  Multimedia Call Handling (Enhanced)? y
                Hospitality (Basic)? y
   Hospitality (G3V3 Enhancements)? y
                           IP Trunks? y
             IP Attendant Consoles? Y

              (NOTE: You must logoff & login to effect the permission changes.)
```

System Parameters Customer-Options screen

Emergency Access to Attendant

Provides for emergency calls to be placed to an attendant. These calls can be placed automatically by Avaya Communication Manager or dialed by users.

Enable 'dadmin' Login

Provides business partners the ability to install, administer, and maintain Avaya media servers and DEFINITY switches. The dadmin login has access to all the same commands as other logins with the exception of `Go` and `WP`. `Go` is used for `go tcm` and `go debug` as well as `go server`. WP is for writing memory.

Enhanced Conferencing

Enhanced Conferencing allows the customer to use the Meet-me Conference, Expanded Meet-me Conference, Selective Conference Party Display, Drop, and Mute, and the No Hold Conference features. Must be **y** to enable the Enhanced Conferencing features.

Enhanced EC500

Enables Extension to Cellular for administration. "EC500" refers to the Extension to Cellular feature. When this field is set to **y**, all screens under the **`off-pbx-telephone`** commands are available.

Enterprise Survivable Server

This display-only field is activated through the license file. When this field is set to **y**, this server is an Enterprise Survivable Server (ESS).

ESS Administration

This display-only field enables administration of Enterprise Survivable Servers (ESS) on the **System Parameters - ESS** screen.

External Device Alarm Admin

Provides for analog line ports to be used for external alarm interfaces. Allows identification of
port location, adjunct associated with port location, and the alarm level to report.

Enterprise Wide Licensing

Enterprise Wide Licensing. See your Avaya representative for more information.

Five Port Networks Max Per MCC

Available only for S8700 Series Multi-Connect. Allows system administrator to create five port networks in a multi-carrier cabinet. If there are any cabinets with more than 2 PNs assigned, this field cannot be set to **n**.

Flexible Billing

Provides an internationally accepted standard interface for end-to-end digital connectivity. Used with a T1 interface and supports twenty-three 64-KBPS voice or data B-Channels and one 64-Kbps signaling D Channel for total bandwidth of 1.544 Mbps.

Forced Entry of Account Codes

Allows system administration to force account users to enter account codes based on user or trunk class of restriction, or by an option on the Toll Analysis table. FEAC provides an easy method of allocating the costs of specific calls to the correct project, department, etc.

Global Call Classification

Provides call classification outside of North America. Listens for tones and classifies tones detected. Required for Call Coverage Off Net and Outgoing Call Management.

Hospitality (Basic)

Provides access to basic features including: Attendant Crisis Alert, Attendant Room Status, Automatic Wakeup, Custom Selection of VIP DID Numbers, Do Not Disturb, Names Registration, Single-Digit Dialing, and Mixed Station Numbering.

Hospitality (G3V3 Enhancements)

Software required for Property Management System and Automatic Wakeup. Property Management System Interface activates Forward PMS Messages to INTUITY Lodging and PMS Protocol Mode (transmit in ASCII mode). Cannot be set to **y** if the **Hospitality (Basic)** field is **n**.

Internet Protocol (IP) PNC

This field is only for the Avaya S8700 Series IP-Connect-only port network connections and only then should be set to **y**. If the configuration calls for fiber-connected only PNs, or if it includes some port networks that are connected together with fiber and some port networks that are IP-connect, this field should be set to **n**. This field cannot be set to **y** if:

- the **Asynch. Transfer Mode (ATM) PNC** field is **y**
- the **PNC Duplication** field is **y**
- If fiber is administered

If the field is **n**, you can still add IP-connect port networks to the network. This field must be **n** if you have or are integrating IP-connected port networks with direct/multi-connect configurations.

> **! CAUTION:**
> If this field is set to **y**, then you cannot add direct-connect, CSS-connected, or ATM-connected port networks to the configuration. To integrate IP-connected port networks with direct/multi-connect configurations, this field must be set to **n**.

IP Attendant Consoles

Controls permission to administer the IP Attendant Console.

IP Stations

Controls permission to administer H.323 and/or SoftPhone stations. Must be **y** for IP Telephones

.

IP Trunks

Controls permission to administer H.323 trunks. Must be **y** for IP trunks.

ISDN Feature Plus

Provides ISDN Feature Plus signaling. This option is enabled when either the **ISDN-BRI Trunks** field or the **ISDN-PRI** field is **y**.

ISDN Network Call Redirection

Administrable if the **ISDN-PRI** or **ISDN-BRI Trunk** field is **y**. Network Call Redirection (NCR) redirects an incoming ISDN call from a server running Avaya Communication Manager to another PSTN endpoint. It is used in Call Centers with Best Service Routing and Lookahead Interflow.

ISDN-BRI Trunks

Provides the capability to add ISDN-BRI trunks to Communication Manager. If enabled, can add isdn trunk groups and the following screens are accessible:

- **network-facilities**
- **private-numbering**
- **public-unknown- numbering**

ISDN-PRI

Provides Integrated Services Digital Network (ISDN-PRI) software for either a switching-hardware platform migration only or a switching-hardware platform migration in combination with a software release upgrade. Also provides signaling support for H.323 signaling. Must be **y** for IP trunks.

Local Survivable Processor

This display-only field indicates that the server is a Local Survivable Processor (LSP). When this field is set to **y**, the LSP server is configured to provide standby call processing in case the primary media server is unavailable.

Malicious Call Trace

Provides ability to retrieve certain information related to a malicious call.

Mode Code for Centralized Voice Mail

This feature provides the ability to share a Voice Mail System (VMS) among several servers/ switches using the Mode Code - Voice Mail System Interface.

Multifrequency Signaling

Provides for a screen of number signaling used between Communication Manager and the central office.

Multimedia Appl. Server Interface (MASI)

Allows users of the Multimedia Communications Exchange (MMCX) to take advantage of certain Avaya Communication Manager telephony features.

Multimedia Call Handling (Basic)

Allows administration of desktop video-conferencing systems as data modules associated with Avaya Communication Manager voice stations in a multimedia complex. Users can dial one number to reach either endpoint (voice or data) in the complex. Also provides support for IP SoftPhones.

Multimedia Call Handling (Enhanced)

Allows a multifunction telephone to control a multimedia call like a standard voice call.

Field descriptions for page 5

```
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                         OPTIONAL FEATURES


                  Multinational Locations?            Station and Trunk MSP? n
Multiple Level Precedence and Preemption?       Station as Virtual Extension? n
                       Multiple Locations?
                                                System Management Data Transfer? n
          Personal Station Access (PSA)? y
                         Posted Messages? n              Tenant Partitioning? n
                         PNC Duplication? n       Terminal Trans. Init. (TTI)? y
                    Port Network Support? y              Time of Day Routing? y
               Processor and System MSP? n             Uniform Dialing Plan? y
                     Private Networking? y     Usage Allocation Enhancements? y
                     Processor Ethernet? y       TN2501 VAL Maximum Capacity? y
                          Remote Office? n              Wideband Switching? y
          Restrict Call Forward Off Net? y                         Wireless? n
                  Secondary Data Module? Y
```

System Parameters Customer-Options screen

Multinational Locations

The Multinational Locations feature provides you with the ability to use a single Enterprise Communication Server (ECS) with stations, port networks, remote offices, or gateways in multiple countries. With this feature enabled, you can administer location parameters such as companding, loss plans, and tone generation per location, instead of system-wide.

Multiple Level Precedence and Preemption

Multiple Level Precedence and Preemption (MLPP) provides users the ability to assign levels of importance to callers, and when activated, to give higher-priority routing to individual calls based on the level assigned to the caller.

Multiple Locations

Allows you to establish numbering plans and time zone and daylight savings plans that are specific for each cabinet in a port network.

Personal Station Access (PSA)

Provides basic telecommuting package capability for Personal Station Access.

Posted Messages

Supports users being able to post messages, which they select from among a set of as many as 30 (15 fixed, 15 administrable), to be shown on display telephones.

PNC Duplication

If set to **y**, the **Enable Operation of PNC (Port Network Connectivity) Duplication** field appears on the **System Parameters Duplication** screen. The **Enable Operation of PNC Duplication** field is set with the **Enable Operation of SPE (Switch Processing Element) Duplication** field to provide non-standard reliability levels (high, critical, or ATM PNC Network Duplication).

Port Network Support

Indicates that the server is operating as a stand-alone Internal Communications Controller (ICC) when set to **n** and is used to disable traditional port networking. Set to **y** to indicate that traditional Avaya DEFINITY port networks are in use.

Private Networking

Upgrades PNA or ETN software RTU purchased with earlier systems.

Processor Ethernet

Appears only on S8300, S8400, and S8500 Media Servers. Used to enable use of the Ethernet card resident in the processor cabinet for use by the DEFINITY Call Processing software in place of a C-LAN card (located in a port network). The Processor Ethernet interface is always enabled for S8700 series media servers.

Processor and System MSP

Allows the customer administrator or technician to maintain processor and system circuit packs.

Remote Office

Allows administration of a remote office.

Restrict Call Forward Off Net

The system can monitor the disposition of an off-call and, if it detects busy, bring the call back for further processing, including call coverage.

Secondary Data Module

Provides ability to use any data module as a secondary data module

.

Station and Trunk MSP

Provides the customer administrator or technician to maintain station and trunk circuit packs.

Station as Virtual Extension

Allows **virtual** to be entered in the **Type** field of the **Station** screen, which allows multiple virtual extensions to be mapped to a single physical analog telephone. The user can also administer a specific ringing pattern for each virtual extension. Useful in environments such as college dormitories, where three occupants can have three different extensions for one physical telephone.

System Management Data Transfer

Indicates Communication Manager is accessible by Network Administration.

Tenant Partitioning

Provides for partitioning of attendant groups and/or stations and trunk groups. Typically this is used for multiple tenants in a building or multiple departments within a company or organization.

Terminal Trans. Init. (TTI)

Allows administrators of Terminal Translation Initialization (TTI) to merge an station administered with **X** in the **Port** field, to a valid port by dialing a system-wide TTI security code and the extension from a terminal connected to that port. Must be set to **y** for Automatic Customer Telephone Rearrangement.

Time of Day Routing

Provides AAR and ARS routing of calls based on the time of day and day of the week. You can take advantage of lower calling rates during specific times.

TN2501 VAL Maximum Capacity

If this is enabled, you have the Enhanced offer, which allows up to 60 minutes storage capacity per pack and multiple integrated announcement circuit packs.

Uniform Dialing Plan

Provides 3- to 7-digit Uniform Dial Plan (UDP) and 1 to 7 digit steering. Also allows you to use Extended Trunk Access and Extension Number Portability features.

Usage Allocation Enhancements

Provides for assigning ISDN-PRI or ISDN-BRI Services/Features for Usage Allocation Plans. To use this enhancement, first set either the **ISDN-PRI** or **ISDN-BRI Trunks** fields to **y**.

Wideband Switching

Provides wideband data software for switching video or high-speed data. You can aggregate DSO channels up to the capacity of the span. Wideband supports H0, H11, and H12 standards, where applicable, as well as customer-defined data rates.

Wireless

Provides right to use for wireless applications in certain Network Systems sales. You can purchase it from Avaya Network Wireless Systems.

Field descriptions for Call Center Optional Features

```
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                    CALL CENTER OPTIONAL FEATURES

                                 Call Center Release:

                                       ACD? y    PASTE (Display PBX Data on Phone)? n
                            BCMS (Basic)? y                          Reason Codes? n
                                                            Service Level Maximizer? y
           BCMS/VuStats Service Level? n             Service Observing (Basic)? y
                     Business Advocate? n    Service Observing (Remote/By FAC)? n
                       Call Work Codes? y              Service Observing (VDNs)?
        DTMF Feedback Signals For VRU? y                              Timed ACW?
                     Dynamic Advocate? n                     Vectoring (Basic)? y
         Expert Agent Selection (EAS)? y                 Vectoring (Prompting)? y
                                EAS-PHD? n             Vectoring (G3V4 Enhanced)?
                     Forced ACD Calls? n    Vectoring (ANI/II-Digits Routing)? n
                 Least Occupied Agent?      Vectoring (G3V4 Advanced Routing)? n
             Lookahead Interflow (LAI)?                      Vectoring (CINFO)? n
  Multiple Call Handling (On Request)? n     Vectoring (Best Service Routing)? n
      Multiple Call Handling (Forced)? n                 Vectoring (Holidays)?
```

Call Center Optional Features screen

ACD

Automatic Call Distribution (ACD) automatically distributes incoming calls to specified splits or skills. Provides the software required for the Call Center Basic, Plus, Deluxe, and Elite features for the number of agents specified. Cannot be set to **n** if the **Call Work Codes** field is **y**.

BCMS (Basic)

Provides real-time and historical reports about agent, ACD split, Vector Directory Number (VDN) and trunk group activity.

BCMS/VuStats Service Level

Allows you to set up hunt groups or Vector Directory Numbers (VDNs) with an acceptable service level. An acceptable service level defines the number of seconds within which a call must be answered to be considered acceptable

Business Advocate

Software that provides an integrated set of advanced features to optimize call center performance. If set to **n**, the **Least Occupied Agent** field displays.

Call Center Release

Displays the call center release installed on the system.

Call Work Codes

Allows agents to enter digits for an ACD call to record customer-defined events such as account codes or social security numbers. Cannot be set to **y** if the **ACD** field is **n**.

DTMF Feedback Signals For VRU

Provides support for the use of C and D Tones to VRUs.

Dynamic Advocate

Software that provides an integrated set of advanced features to optimize call center performance.

EAS-PHD

Increases the number of skills an agent can log in to from four to 20. Increases the number of agent skill preference levels from two to 16.

Expert Agent Selection (EAS)

Provides skills-based routing of calls to the best-qualified agent.

Forced ACD Calls

See **Multiple Call Handling**.

Least Occupied Agent

Appears only if the **Business Advocate** field is **n**. Allows call center calls to be routed to the agent who has been the least busy, regardless of when the agent last answered a call. Cannot be set to **y** if the **Expert Agent Selection (EAS)** field is **n**.

Lookahead Interflow (LAI)

Provides Look-Ahead Interflow to balance the load of ACD calls across multiple locations. Cannot be set to **y** if the **Vectoring (Basic)** field is **n**.

Multiple Call Handling (On Request)

Allows agents to request additional calls when active on a call.

Multiple Call Handling (Forced)

Forces an agent to be interrupted with an additional ACD call while active on an ACD call. Splits or skills can be one forced, one per skill, or many forced. Cannot be set to **y** if the **ACD** field is **n** and the **Forced ACD Calls** field is **y**.

PASTE (Display PBX Data on Phone)

Provides an interface between the display of a DCP telephone set and PC-based applications.

Reason Codes

Allows agents to enter a numeric code that describes their reason for entering the AUX work state or for logging out of the system. Cannot be set to **y** if the **Expert Agent Selection (EAS)** field is **n**.

Service Level Maximizer

Allows an administrator to define a service level whereby X% of calls are answered in Y seconds. When Service Level Maximizer (SLM) is active, the software verifies that inbound calls are matched with agents in a way that ensures that the administered service level is met. SLM is used with Expert Agent Selection (EAS), and without Business Advocate. Call Center Release must be 12 or later.

Service Observing (Basic)

Allows a specified user to observe an in-progress call on a listen-only or listen-and-talk basis.

Service Observing (Remote/By FAC)

Allows users to service observe calls from a remote location or a local station using this feature's access codes.

Service Observing (VDNs)

Provides the option of observing and/or monitoring another user's calls.

Timed ACW

Places an auto-in agent in ACW for an administered length of time after completion of the currently active ACD call.

Vectoring (Basic)

Provides basic call vectoring capability.

Vectoring (Prompting)

Allows flexible handling of incoming calls based on information collected from the calling party or from an ISDN-PRI message.

Vectoring (G3V4 Enhanced)

Allows the use of enhanced comparators, wildcards in digit strings for matching on collected digits and ANI or II-digits, use of Vector Routing Tables, multiple audio/music sources for use with wait-time command and priority level with the oldest-call-wait conditional.

Vectoring (ANI/II-Digits Routing)

Provides for ANI and II-Digits vector routing

Vectoring (G3V4 Advanced Routing)

Provides for Rolling Average Speed of Answer Routing, Expected Wait Time Routing, and VDN Calls Routing.

Vectoring (CINFO)

Provides the ability to collect ced and cdpd from the network for vector routing. To use this enhancement, first set either the **ISDN-PRI** or **ISDN-BRI Trunks** fields to **y**.

Vectoring (Best Service Routing)

Enables the Best Service Routing feature. Through special vector commands, Best Service Routing allows you to compare splits or skills at local and remote locations and queue a call to the resource that will give the caller the best service.

Vectoring (Holidays)

Enables the Holiday Vectoring feature. It simplifies vector writing for holidays.

Field descriptions for Call Center Optional Features (page 2)

```
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              CALL CENTER OPTIONAL FEATURES


      VDN of Origin Announcement? n                      VuStats? n
          VDN Return Destination? n     VuStats (G3V4 Enhanced)? N


                                      Used

                 Logged-In ACD Agents: 500
             Logged-In Advocate Agents: 500
           Logged-In IP Softphone Agents: 500
```

Call Center Optional Features screen

Logged-In ACD Agents

Number of ACD Agents contracted for. This field limits the number of logged-in ACD agents to a number no more than the maximum purchased. The value of this field indicates the total of ACD agents that can be logged-in simultaneously.

The limit applies to ACD agents on ACD and EAS calls. Auto-Available Split (AAS) agent ports are counted when they are assigned. AAS split or skill members are also counted. If the port for an AAS split/skill member is logged out, (for example, when a ringing call is redirected) the logged-in agent count is not updated. These counts are updated only during administration.

Logged-In Advocate Agents

Appears when the **Business Advocate** field is **y**. Number of Business Advocate Agents contracted for. The total number of logged-in Business Advocate agents must be equal to or less than the number allowed in the **Logged-In ACD Agents** field. The number of logged-in Business Advocate agents counts towards the total number of logged-in ACD agents.

Logged-In IP Softphone Agents

Number of IP Softphone Agents contracted for. This field limits the number of logged-in IP Softphone agents to a number no more than the maximum purchased. The value of this field indicates the total of IP Softphone agents that can be logged-in simultaneously.

VDN of Origin Announcement

Provides a short voice message to an agent indicating the city of origin of the caller or the service requested by the caller based on the VDN used to process the call.

VDN Return Destination

Allows an incoming trunk call to be placed back in vector processing after all parties, except the originator, drop.

VuStats

Allows you to present BCMS statistics on telephone displays.

VuStats (G3V4 Enhanced)

Allows you to use the G3V4 VuStats enhancements including historical data and thresholds.

Field descriptions for QSIG Optional Features

```
display system-parameters customer-options                    Page 8 of
x

                    QSIG OPTIONAL FEATURES

                                     Basic Call Setup? n
                         Basic Supplementary Services? n
                                Centralized Attendant? n
                                Interworking with DCS? n
                  Supplementary Services with Rerouting? n
                          Transfer into QSIG Voice Mail? n
                                     Value-Added (VALU)? n
```

QSIG Optional Features screen

Basic Call Setup

Provides basic QSIG services: basic connectivity and calling line ID number. To use this enhancement, either the **ISDN-PRI** or **ISDN-BRI Trunks** fields must be **y**.

Basic Supplementary Services

To use this enhancement, either the **ISDN-PRI** or **ISDN-BRI Trunks** fields must be **y**. Provides the following QSIG Supplementary Services:

- Name ID
- Transit Capabilities; that is, the ability to tandem QSIG information elements
- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections
- Call Forwarding (Diversion) by forward switching. No reroute capabilities are provided
- Call Transfer by join. No path replacement capabilities are provided.
- Call Completion (also known as Automatic Callback)

Centralized Attendant

Can be enabled only if the **Supplementary Services with Rerouting** field is **y**. Cannot be set to **y** if the **CAS Main** and **CAS Branch** fields are **y**. Allows all attendants in one location to serve users in multi locations. All signaling is done over QSIG ISDN lines. If this field is **y**, the **IAS** fields on the **Console Parameters** screen do not display.

Interworking with DCS

Allows the following features to work between a user on a DCS-enabled media server or switch in a network and a QSIG-enabled media server or switch:

- Calling/Called/Busy/Connected Name
- Voice Mail/Message Waiting
- Leave Word Calling

This field cannot be set to **y** if the **DCS (Basic)** field is **n**.

Supplementary Services with Rerouting

Provides the following QSIG Supplementary Services:

- Transit Capabilities; that is, the ability to tandem QSIG information elements.
- Support of Notification Information Elements for interworking between QSIG and non-QSIG tandemed connections.
- Call Forwarding (Diversion) by forward switching. In addition, reroute capabilities are provided.
- Call Transfer by join. In addition, path replacement capabilities are provided.

Transfer Into QSIG Voice Mail

Can be enabled only if the **Basic Supplementary Services** field is **y** and either the **ISDN-PRI Trunk** or **ISDN-BRI Trunk** field is **y**. Allows transfer directly into the voice-mail box on the voice-mail system when a QSIG link connects Avaya Communication Manager and the voice-mail system.

Value Added (VALU)

Provides additional QSIG functionality, including the ability to send and display calling party information during call alerting.

Field descriptions for ASAI

```
change system-parameters customer options                    Page 8 of
X

                              ASAI FEATURES

                                CTI Stations? n
                               Phantom Calls? N

                         ASAI PROPRIETARY FEATURES

                              Agent States? n
```

ASAI Features screen when the ASAI Link Plus Capabilities field is y

Agent States

Appears when the **Computer Telephony Adjunct Links** field is **y**. The **Agent States** field provides proprietary information used by Avaya applications.

Note:

The **Agent States** field only applies to links administered as type **adjlk**. This field was previously named **Proprietary Applications**.

CTI Stations

Appears when the **ASAI Link Plus Capabilities** field is **y**. This field needs to be enabled for any application (using a link of Type ASAI) that uses a CTI station to receive calls.

Phantom Calls

Appears when the **ASAI Link Plus Capabilities** field is **y**. Enables phantom calls. The **Phantom Calls** field only applies to links administered as type ASAI.

Field descriptions for Maximum IP Registrations by Product ID

```
                                                          Page 9 of x

                    MAXIMUM IP REGISTRATIONS BY PRODUCT ID

Product ID_Rel. Limit      Product ID_Rel. Limit      Product ID_Rel. Limit
_________ ___. _____       _________ ___. _____      _________ ___. _____
_________ ___. _____       _________ ___. _____      _________ ___. _____
_________ ___. _____       _________ ___. _____      _________ ___. _____
_________ ___. _____       _________ ___. _____      _________ ___. _____
_________ ___. _____       _________ ___. _____      _________ ___. _____
_________ ___. _____       _________ ___. _____      _________ ___. _____
_________ ___. _____       _________ ___. _____      _________ ___. _____
```

Maximum IP Registrations by Product ID screen

Limit

Maximum number of IP registrations allowed.

Valid entries	Usage
1000 or **5000**, depending on your server configuration	Maximum number of IP registrations allowed. For Avaya R300 Remote Office Communicator, defaults to the maximum allowed value for the **Concurrently Registered Remote Office Stations** on page 1 of this screen.

Product ID

Identifies the product using the IP (internet protocol) registration.

Valid entries*	Usage
Avaya_IR	Interactive Response product
IP_Agent	IP Agents
IP_eCons	SoftConsole IP attendant
IP_Phone	IP Telephones
IP_ROMax	R300 Remote Office telephones
IP_Soft IP	Softphones

*These are just a few examples of valid Product IDs. The valid Product IDs for your system are controlled by the license file.

Rel

Release number of the IP endpoint.

Valid entries	Usage
0 to **99** or blank	Release number of the IP endpoint

Feature-Related System Parameters

This screen implements system parameters associated with various system features.

Note:

This screen used to contain Call Coverage and Call Forwarding parameters. These fields have been moved to a separate screen, which you can access with the command **`change system-parameters coverage-forwarding`**.

Field descriptions for page 1

```
change system-parameters features                                page 1 of x

                    FEATURE-RELATED SYSTEM PARAMETERS

                              Self Station Display Enabled? n
                                  Trunk-to-Trunk Transfer? none
    Automatic Callback - No Answer Timeout Interval (rings): 4_
                     Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20_
                               AAR/ARS Dial Tone Required? y
                            Music/Tone On Hold: music Type: _______
            Music (or Silence) On Transferred Trunk Calls: all
                     DID/Tie/ISDN/SIP Intercept Treatment: attd

    Internal Auto-Answer of Attd-Extended/Transferred Calls? y
                 Automatic Circuit Assurance (ACA) Enabled? y
                                       ACA Referral Calls: local
                                  ACA Referral Destination: ________
              ACA Short Holding Time Originating Extension: ______
               ACA Long Holding Time Originating Extension: ______
           Abbreviated Dial Programming by Assigned Lists:
        Auto Abbreviated/Delayed Transition Interval(rings):
                  Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls?
```

Feature-Related System Parameters screen

AAR/ARS Dial Tone Required

A second dial tone provides feedback to the user that additional dialing can occur.

Valid entries	Usage
y/n	Enter **y** to indicate a second dial tone is to be given to the calling party on a incoming tie or DID trunk call that is to be routed via AAR/ARS.

Abbreviated Dial Programming by Assigned Lists

Valid entries	Usage
y	Enter **y** to allow programming by station's assigned list.
n	Enter **n** if using Program Access code to indicate which personal list is to be programmed.

ACA Referral Calls

Indicates where ACA referral calls generate. This field only appears when the **Automatic Circuit Assurance (ACA) Enabled** field is **y**.)

Valid entries	Usage
local	Local referral calls generate on and for the local switch.
primary	Primary referral calls generate on the local switch for remote servers/ switches as well as the local switch.
Remote	Remote referral calls generate at another server in a DCS network. In this case, the remote node number must also be entered. The remote node number is the same node number as defined on the **Dial Plan** screen. Also, **ACA** button status transmits to other servers/switches when in a DCS network.

ACA Referral Destination

The specified extension should be equipped with a display module. This field only appears if ACA Referral Calls is **local** or **primary**.

Valid entries	Usage
An extension	Enter the extension on a local server running Communication Manager that is to receive the ACA referral call.
attd	Enter **attd** for attendant.

ACA Remote PBX Identification

This field only appears if **ACA Referral Calls** is **remote**.

Valid entries	Usage
1 to **63**	Enter a number to identify the switch in a DCS network that makes the referral call. Do not define the remote server/switch identified in this field as **local** on the system's **Dial Plan** screen.

ACA Short Holding Time Originating Extension and ACA Long Holding Time Originating Extension

Valid entries	Usage
An unassigned extension	Do not use the same extension number for both fields. The specified extensions are assigned automatically by the system when the screen is submitted. These fields only display if **ACA Referral Calls** is **local** or **primary**.

Auto Abbreviated/Delayed Transition Interval (rings)

Valid entries	Usage
1 to **16**	Enter the number of rings before an automatic abbreviated/ delayed transition is triggered for a call.

Automatic Callback — No Answer Timeout Interval (rings)

Valid entries	Usage
2 to **9**	Enter the number of times the callback call rings at the calling station before the callback call is canceled.

Automatic Circuit Assurance (ACA) Enabled

If **Automatic Circuit Assurance (ACA) Enabled** is **n**, associated ACA fields will not display. Must have an **aca-halt** button administered on the user's station. If you enable this feature, complete the following ACA-related fields.

Valid entries	Usage
y	Enter **y** if ACA measurements will be taken.
n	Otherwise, enter **n**.

Call Park Timeout Interval (minutes)

Valid entries	Usage
1 to **90**	Enter the number of minutes a call remains parked before it cancels.

DID/Tie/ISDN/SIP Intercept Treatment

Valid entries	Usage
A recorded announcement. extension	Toll charges do not apply to DID and private network calls routed to an announcement

Note: If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur. The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.

attd	For system security, Avaya recommends entering **attd** in this field. This routes intercept calls to the attendant and, if the attendant receives several of these, they will know a problem exists.

Display Calling Number for Room to Room Caller ID Calls

Valid entries	Usage
y/n	Enter **y** to display the calling number for room to room hospitality calls.

Internal Auto-Answer of Attd-Extended/Transferred Calls

This only applies to digital telephones (except BRI) with a headset or speakerphone capability.

Valid entries	Usage
attd-extended	Enter **attd-extended** to enable IAA for only attendant extended calls.
both	Enter **both** to enable IAA for station transferred and attendant extended calls.
none	Enter **none** to disable IAA for all calls.
transferred	Enter **transferred** to enable IAA for

Music (or Silence) On Transferred Trunk Calls

Valid entries	Usage
all	Enter **all** to allow all transferred trunk calls to receive music until the call is answered if the Music-on-Hold feature is available.
no	Enter **no** if trunk callers are to hear music (or silence if Music-on-Hold is not administered) while waiting to be transferred, and then ringback as soon as the transfer is completed till the call is answered.
call-wait	Enter **call-wait** if trunk calls transferred to stations that require the call to wait hear music (if administered); all other transferred trunk calls receive ringback tone.

Music/Tone on Hold

If you use equipment that rebroadcasts music or other copyrighted materials, you might be required to obtain a copyright license from, or pay fees to, a third party. You can purchase a Magic OnHold system, which does not require such a license, from Avaya or our business partners. This field does not appear if **Tenant Partitioning** is **y** on the **System-Parameters Customer-Options** screen. In that case, use the **Tenant** screen to establish Music on Hold.

Valid entries	Usage
Music **Tone** **None**	Indicates what a caller hears while on hold. Default is **none**. When **music** is entered, the **Type** field appears to define the music type.

Off-Premises Tone Detect Timeout Interval (seconds)

Valid entries	Usage
5 to **25**	The number of seconds a call progress tone receiver (CPTR) tries to detect dial tone from a trunk during dialing. Once the time-out interval occurs, the call either outpulses on the trunk or gets intercept treatment depending on the setting of the **Outpulse Without Tone** field on page 6 of this screen.

Port

Appears when **Music/Tone on Hold** is **music** and **Type** is **port**. Enter the necessary characters to indicate the port number that provides Music-on-Hold access.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S8700/S8300 Media Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20** number	Fourth and fifth character are the slot
01 to **04** (Analog TIE trunks) number **01** to **31**	Six and seventh characters are the circuit

1 to **80** (DEFINITY CSI) or **1** to **250** (S8700/S8300 Media Servers)	Gateway
V1 to **V9**	Module
01 to **31**	Circuit

Protocol for Caller ID Analog Terminals

Determines the protocol/tones sent to a Caller ID telephone.

Valid entries	Usage
Bellcore	Enter **Bellcore** for Bellcore protocol with 212 modem protocol tones. Used in the U.S. and similar countries.
V23-Bell	Enter **V23-Bell** for Bellcore protocol with V.23 modem tones. Used in Bahrain and similar countries.

Self Station Display Enabled

Use this field to control the use of the **inspect** button for digital display telephones. **Self Station Display** allows a user to display the primary extension associated with a digital display telephone. There are two methods: (1) enter a feature access code (FAC), and (2) use the **inspect** button. In either case, the display shows the primary extension associated with the telephone where the FAC or **normal** or **exit** button is entered. In the case of the FAC, the display continues until a display-altering event occurs (for instance, going on-hook or receiving an incoming call). In the case of the **inspect** button, the display continues until the user presses the **normal** or **exit** button or until a display-altering event occurs.

Valid entries	Usage
y	The primary extension does display when the **inspect** button is pressed.
n	The extension does not display when the **inspect** button is pressed.

Trunk-to-Trunk Transfer

Regulations in some countries control the settings for this field. See your Avaya technical support representative for assistance.

Valid entries	Usage
all	Enter **all** to enable all trunk-to-trunk transfers. This allows elephone users to set up trunk-to-trunk transfer, go on-hook without disconnecting the call, and forward the call to a remote location. This value is required for SIP Enablement Services (SES) support.
restricted	Enter **restricted** (restricted public) to restrict all public trunks (CO, WATS, FX, CPE, DID, and DIOD).
none	Enter **none** to restrict all trunks (except CAS and DCS) from being transferred.

Type

This field appears when **Music/Tone on Hold** is set to **music**.

Note:
If the **Tenant Partitioning** field on the **Optional Features** screen is set to **y**, you cannot administer the **Music/Tone on Hold** field. If the **Tenant Partitioning** field on the **Optional Features** screen set to **y**, you must use the **Music Sources** screen to assign music to a port.

Valid entries	Usage
Ext **Group** **Port**	• Indicate whether the source for Music on Hold is an announcement extension, an audio group, or a port on a VAL board. • Type **ext** and the corresponding extension number of the integ-mus announcement/audio source. • Type **group** and the corresponding Music-on-Hold analog group number. • Type **port** and the corresponding location of the Music-on-Hold analog/aux-trunk source.

Note: After a valid value is entered, a blank field appears for entry of the appropriate source identifier (extension number, audio group number, or port number).

Field descriptions for page 2

```
change system-parameters features                          page 2 of x

            FEATURE-RELATED SYSTEM PARAMETERS
LEAVE WORD CALLING PARAMETERS

                     Maximum Number of Messages Per Station: 10
          Maximum Number of External Calls Logged Per Station: 0
                Message Waiting Indication for External Calls? n
     Stations with System-wide Retrieval Permission (enter extension)

         1: 34430          3: attd          5:        7:        9:
         2: 34412          4:               6:        8:       10:

                                     TTI/PSA PARAMETERS MOVED TO PAGE 3


            Prohibit Bridging Onto Calls with Data Privacy? _
                          Enhanced Abbreviated Dial Length (3 or
            4)? _
              Record All Submission Failures in History Log? _
                  Record PMS/AD Transactions in History Log?_
                     Record IP Registrations in History Log?
       Default Multimedia Outgoing Trunk Parameter Selection: 2x64
```

Feature-Related System Parameters screen

LEAVE WORD CALLING PARAMETERS

Maximum Number of Messages Per Station

Valid entries	Usage
0 to **125**	The maximum number of LWC Messages that can be stored by the system for a telephone at a given time.

Maximum Number of External Calls Logged Per Station

When an external call is not answered, the server running Communication Manager keeps a record of up to 15 calls (provided information on the caller identification is available) and the telephone's message lamp lights. The telephone set displays the names and numbers of unsuccessful callers

Valid entries	Usage
0 to **15**	The maximum number of calls that can be logged for each user. The assigned number cannot be larger than the entry in the **Maximum Number of Messages Per Station (when MSA not in service)** field.

Message Waiting Indication for External Calls

Provides a message waiting indication when external calls are logged.

Valid entries	Usage
y	The message waiting indication for a particular station is on whenever an external call is logged.
n	The log of external calls has no impact on the message waiting indication.

Default Multimedia Outgoing Trunk Parameter Selection

Does not appear on S8700 Series IP-Connect.

Valid entries	Usage
2x56	Sets default parameter for bandwidth and bearer for all video ` calls.
2x64	

Enhanced Abbreviated Dial Length (3 or 4)

The administrator might not be able to use all entry slots because of system capacity constraints.

Valid entries	Usage
3	A value of **3** makes 1000 Enhanced List entries available to the administrator
4	A value of **4** makes 10,000 entries available.

Prohibit Bridging Onto Calls with Data Privacy

Valid entries	Usage
y/n	Enter **y** to protect calls from bridge-on by any party, including Service Observing, Intrusion, Verify, and Bridging.

Record All Submission Failures in History Log

Allows submission failures to be recorded on the history log.

Valid entries	Usage
y/n	Enter **y** to record submission failures on the history log.

Record IP Registrations in History Log

Allows the logging of IP registrations in the history log.

Valid entries	Usage
y/n	Enter **y** to record IP registrations on the history log.

Record PMS/AD Transactions in History Log

Allows PMS and abbreviated dialing button transactions to be recorded on the history log.

Valid entries	Usage
y/n	Enter **y** to record PMS or abbreviated dialing button transactions on the history log.

Stations With System-wide Retrieval Permission (enter extension)

An extension must be removed from this list before the station is removed from the system. The server running Communication Manager refers to the extensions on this list as "super-retrievers."

Valid entries	Usage
An assigned Extension	Enter up to 10 telephone extension numbers that can retrieve LWC Messages or External Call Log records for all other telephones. A VDN extension is not allowed.
attd	An entry of **attd** gives retrieval permission to all attendants.

Field descriptions for page 3

```
change system-parameters features                                  page 3 of x

            FEATURE-RELATED SYSTEM PARAMETERS


TTI/PSA PARAMETERS

               WARNING! SEE USER DOCUMENTATION BEFORE CHANGING TTI STATE

                   Terminal Translation Initialization (TTI) Enabled? __
                               TTI State: ________ TTI Security Code:

________                 Record CTA/PSA/TTI Transactions in History Log? __
                   Enhanced PSA Location/Display Information Enabled? __
                                  Default COR for Dissociated Sets: ___
                                     CPN, ANI for Dissociated Sets:
                         Customer Telephone Activation (CTA) Enabled? __



CALL PROCESSING OVERLOAD MITIGATION
Restrict Calls:
Don't Answer Criteria for Logged off IP/PSA/TTI Stations? n
```

Feature-Related System Parameters screen

TTI/PSA PARAMETERS

CPN, ANI for Dissociated Sets

Appears when the **Default COR for Dissociated Sets** field is non-blank. Specifies the ISDN calling party number (CPN), R2-MFC ANI, and CAMA CESID applied to calls made from PSA dissociated sets, if no system-wide calling party information has been administered for those protocols on their respective administration screens.

Valid entries	Usage
1 to 20 digits	Enter the calling party number or automatic number identification for calls made from dissociated telephones.

Customer Telephone Activation (CTA) Enabled

Valid entries	Usage
y	Enter **y** if you want the Customer Telephone Activation feature for your system.
n	Enter n if you do not want the Customer Telephone Activation feature for your system.

Default COR for Dissociated Sets

Appears when the **Terminal Translation Initialization (TTI) Enabled?** field is **y**.

Valid entries	Usage
0 to **995** or blank	Specify the Class of Restriction (COR) that the system uses for calls made from dissociated telephones.

Enhanced PSA Location/Display Information Enabled

Appears when the **Terminal Translation Initialization (TTI) Enabled?** field is **y**.

Valid entries	Usage
y	Enter **y**, if you want the system to display: • PSA login and associated station information when a station is PSA associated. • PSA logout and the port when a station is PSA dissociated.
n	Enter n if you do not want the system to display PSA information.

Record CTA/PSA/TTI Transaction in History Log

Appears when the **Terminal Translation Initialization (TTI) Enabled?** field is **y**.Use this field to record when extensions and physical telephones move between ports without additional administration from the administrator of Avaya Communication Manager.

Valid entries	Usage
y	Enter **y** if you want the system to record Customer Telephone Activation (CTA), Personal Station Activation (PSA), and TTI transactions in the system history log.
n	Enter **n** if you do not want the system to record Customer Telephone Activation (CTA), Personal Station Activation (PSA), and TTI transactions in the system history log.

Terminal Translation Initialization (TTI) Enabled

Valid entries	Usage
Y	Enter **y** to start ACTR, TTI, and PSA transactions (extension and telephone moves between ports). You can administer this field only if the **Terminal Trans. Init. (TTI)** field on the **Customer Options** screen is **y**.
n	Enter **n** to remove existing TTI port translations and make sure no new TTI port translations are generated.

TTI Security Code

Appears when the **Terminal Translation Initialization (TTI) Enabled?** field is **y**.

Valid entries	Usage
1 to 7 digits	Enter a one-digit to seven-digit number that a TTI user must use when the user accesses TTI from a telephone or data terminal. The system displays this field only when the **Terminal Translation Initialization (TTI)** field is **y**.

TTI State

Appears when the **Terminal Translation Initialization (TTI) Enabled?** field is **y**. Enter the type of port translation that you want for the system to use for unadministered digital ports. The default is **voice**.

Valid entries	Usage
data	Enter **data**, if you want a stand-alone data module to the be the TTI port translation for your system. The activation and deactivation sequence is entered at data terminal.
resume	Enter **resume**, if you want TTI to be available after TTI has been manually suspended. The state of TTI returns to the state that it was in before TTI was manually suspended.
suspend	Enter **suspend** to make TTI voice or TTI data translations temporarily unavailable. The system does not remove existing TTI translations.
Voice	Enter **voice**, if you want voice or voice/data terminal to be the TTI port ranslation for the system. The activation and deactivation sequence is entered from a telephone.

CALL PROCESSING OVERLOAD MITIGATION

Don't Answer Criteria for Logged Off IP/PSA/TTI Stations

Valid entries	Usage
y .	
n	

Restrict Calls

Indicate the type of calls to block first during overload traffic conditions on the system.

Valid entries	Usage
stations-first	Deny new traffic generated by internal stations, allowing inbound calls only (works best in call center environments).
all-trunk-first	Deny all out-bound calls to trunks, tie-lines and stations, and all station-originated calls.
public-trunks-first	Deny all in-bound calls from trunks and tie-lines

Field descriptions for page 4

```
change system-parameters features                                  page 4 of x

              FEATURE-RELATED SYSTEM PARAMETERS
                    Reserved Slots for Attendant Priority Queue: 5_
                                  Time Before Off-Hook Alert: 10__
                       Emergency Access Redirection Extension: ________
         Number of Emergency Calls Allowed in Attendant Queue: __
                                     Call Pickup Alerting? n
                    Temporary Bridged Appearance on Call Pickup? y
                                Call Pickup on Intercom Calls? y
                                         Directed Call Pickup? n
                                  Extended Group Call Pickup: flexible
            Deluxe Paging and Call Park Timeout to Originator? n
           Controlled Outward Restriction Intercept Treatment: tone
          Controlled Termination Restriction (Do Not Disturb): tone
                     Controlled Station to Station Restriction: tone
     AUTHORIZATION CODE PARAMETERS Authorization Code Enabled? y
                                   Authorization Code Length: 7
                       Authorization Code Cancellation Symbol? #
                                     Attendant Time Out Flag? n
                                  Display Authorization Code? _
                 Controlled Toll Restriction Replaces: station-station
              Controlled Toll Restriction Intercept Treatment: extension 3000
```

Feature-Related System Parameters screen

Call Pickup Alerting

This provides pickup group members with a visual indication on the Call Pickup status lamp of calls eligible to be answered via Call Pickup

Valid entries	Usage
y/n	Enter **y** to enable Call Pickup Alerting on a system-wide basis.

Call Pickup on Intercom Calls

Valid entries	Usage
y	Enter **y** to allow a user's or Agent LoginID's call, ringing as an intercom call, to be picked up using the Call Pickup or Directed Call Pickup features. This field controls the use of this feature throughout the system.
n	Enter **n** to prevent the use of these features to pickup an intercom call.

Controlled Outward Restriction Intercept Treatment

Enter the type of intercept treatment the caller receives when the call is outward restricted.

Valid entries	Usage
announcement	Provides a recorded announcement to calls that cannot be completed as dialed. You select and record the message. The calling party receives indication that the call is receiving Intercept treatment. Enter the extension number for the announcement in the associated field.

Note: If entering a Multi-Location Dial Plan shortened extension, note the following: When entering a Multi-Location Dial Plan shortened extension in a field designed for announcement extensions, certain administration end validations that are normally performed on announcement extensions are not done, and resultant warnings or submittal denials do not occur.
The shortened extensions also do not appear in any display or list that shows announcement extensions. Extra care should be taken to administer the correct type of announcement for the application when assigning shortened extensions.

attendant	Allows attendants to provide information and assistance to outgoing calls that cannot be completed as dialed or that are transferred to incomplete or restricted stations.
extension	Enter the extension number for the extension in an associated field. Cannot be a VDN extension.
tone	Provides a siren-type tone to internal calls that cannot be completed as dialed.

Controlled Station-to-Station Restriction

Enter the type of intercept treatment the caller receives when the call is placed to a restricted telephone.

Valid entries	Usage
announcement	If announcement is entered, an associated extension number field displays. Enter the extension of the restricted telephone in the field.
attendant	Intercepted calls are redirected to the attendant.
Extension (cannot be a VDN extension)	If extension is entered, an associated extension number field displays. Enter the extension of the restricted telephone in the field
tone	Intercepted calls receive intercept (siren) tone.

Controlled Termination Restriction (Do Not Disturb)

Enter the type of intercept treatment the caller receives when the call is placed to a termination restricted telephone.

Valid entries	Usage
announcement	If **announcement** is entered, complete an associated extension number field.
attendant	Redirects intercepted calls to the attendant.
coverage	Redirects intercepted calls to coverage.
extension	If **extension** is entered, complete an associated extension number field. Cannot be a VDN extension,
tone	Provides a siren-type tone to calls that cannot be completed as dialed.

Deluxe Paging and Call Park Timeout to Originator

Paged calls that are to be parked require separate activation of the Call Park feature. All parked calls that time out return to the attendant.

Valid entries	Usage
y	Enter **y** to enable the Loudspeaker Paging - Deluxe feature that essentially integrates the Loudspeaker Paging and Call Park features. All parked calls that time out (not answered by paged party) return to the parking party.
n	Enter **n** to enable the Loudspeaker Paging feature.

Directed Call Pickup

Feature use by individual stations, attendants, or EAS agents can be controlled by COR.

Valid entries	Usage
y	Enter **y** to allow use of the Directed Call Pickup feature across the system.
n	Enter **n** to prevent feature use.

Emergency Access Redirection Extension

Valid entries	Usage
An assigned Extension	Enter the assigned extension number (can be a VDN) where Emergency queue overflow will redirect.

Extended Group Call Pickup

Enables call pickup groups to answer calls directed to another call pickup group.

Valid entries	Usage
flexible	Flexible feature version supporting a one-to-n (pickup group-to-extended pickup group) mapping.
simple	Simple feature version with a one-to-one pickup group-to-extended pickup group mapping supported.
none	Extended group call pickup not supported.

Number of Emergency Calls Allowed in Attendant Queue

Valid entries	Usage
0 to **75**	Enter the number of emergency calls allowed in the attendant queue before additional calls are routed to the backup extension.

Reserved Slots for Attendant Priority Queue

Valid entries	Usage
2 to **75**	Enter the number of calls that can go in to the emergency queue

Temporary Bridged Appearance on Call Pickup

Valid entries	Usage
y	Enter **y** to allow a temporary bridged appearance for calls answered with the Call Pickup or Directed Call Pickup features. This field controls this capability on a system-wide basis.
n	Enter **n** to prevent the temporary bridged appearance of calls answered with these features.

Time Before Off-Hook Alert

Valid entries	Usage
1 to **3000** seconds	Enter the time in seconds that a telephone with an Off-Hook Alert Class of Service can remain off-hook (after intercept tone has started) before an emergency call is sent to the attendant.

AUTHORIZATION CODE PARAMETERS

Attendant Time Out Flag

If this field is not enabled, the caller receives Intercept tone. This flag affects only remote users or incoming calls over trunks requiring an authorization code. This field only appears if **Authorization Codes Enabled** is **y**.

Valid entries	Usage
y/n	Enter **y** if a call is to be routed to the attendant if the caller does not dial an authorization code within 10 seconds or dials an invalid authorization code.

Authorization Code Cancellation Symbol

Enter the symbol a caller must dial to cancel the 10-second wait period during which the user can enter an authorization code. This field only appears when **Authorization Code** is **y**.

Valid entries	Usage
#	Enter the cancellation code **#** if the main and tandem servers/switches are both of the same type.
1	Enter the cancellation code **1** if an Avaya System 85 or DIMENSION PBX switch is part of the complex/network.

Authorization Code Length

This field only appears and must be completed if **Authorization Codes Enabled** is **y**. This is the number of digits that must be assigned to the **Authorization Code (AC)** field on the **Authorization Code** screen.

SECURITY ALERT:
You enhance your system's security by using the maximum length for your authorization code.

Valid entries	Usage
4 to 13 digits	Enter a number that defines the number of digits (length) in the **Authorization Code** field.

Authorization Codes Enabled

This field cannot be administered if Authorization Codes is not enabled on the **System-Parameters Customer-Options** screen.

SECURITY ALERT:
To maintain system security, Avaya recommends that Authorization Codes be used.

Valid entries	Usage
y/n	Enter **y** to enable Authorization Codes on a systemwide basis.

Controlled Toll Restriction Intercept Treatment

Appears when the **Controlled Toll Restriction Replaces** field is **outward** or **station-to-station**. This field applies an intercept treatment to a toll call during the call processing.

Valid entries	Usage
announcement	A sub-field appears to the right if **announcement** is used. If the entry is **announcement**, enter the assigned announcement extension.
attendant	Intercepted calls are redirected to the attendant.
extension	A sub-field appears to the right if **extension** is used. If the entry is **extension**, enter the extension assigned to station or individual attendant.
tone	Intercepted calls receive intercept (siren) tone.

Controlled Toll Restriction Replaces

This field activates the Controlled Toll Restriction feature.

Valid entries	Usage
Outward **station-station** **none**	The value that you choose for this field will be replaced by controlled toll restriction. In other words, if you choose station-station, you will not be able to use station-station restrictions unless you reset this field.

Display Authorization Code

This field applies only to DCP, not to BRI or hybrid sets.

SECURITY ALERT:

To enhance your system's security, set **Display Authorization Code** to **n**.

Valid entries	Usage
y	Enter **y** to allow authorization code digits to display on the set during the dialing.
n	Enter **n** if these digits should not display.

Field descriptions for page 5

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                    FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
Endpoint: ____            Lines Per Page: 60          EIA Device Bit
Rate:


SYSTEM-WIDE PARAMETERS
                                     Switch Name: ________________
            Emergency Extension Forwarding (min): 10
          Enable Inter-Gateway Alternate Routing? N

MALICIOUS CALL TRACE PARAMETERS
       Apply MCT Warning Tone? n            MCT Voice Recorder Trunk
Group: ___
     Delay Sending Release (seconds)?
SEND ALL CALLS OPTIONS
               Send All Calls Applies to: station
          Auto Inspect on Send All Calls? n

UNIVERSAL CALL ID
                Create Universal Call ID (UCID)? n
                                                UCID Network Node ID: ___
```

Feature-Related System Parameters screen

SYSTEM PRINTER PARAMETERS

The system printer is the printer dedicated to support scheduled reports.

Endpoint

Valid entries	Usage
Data module Extension	Does not appear for S8700 Series IP-Connect. Associated with the System printer.
eia	Does not appear for S8700 Series IP-Connect. If the DCE jack is used to interface the printer.
SYS_PRNT	Use this value if the system printer is connected over a TCP/IP link, and the link is defined as SYS_PRNT on the **IP Services** screen.
blank	

Lines Per Page

Valid entries	Usage
24 to **132**	Enter the number of lines per page required for the report.

SYSTEM-WIDE PARAMETERS

Emergency Extension Forwarding (min)

If an emergency call should drop (get disconnected), the public safety personnel will attempt to call back. If the ELIN that was sent was not equivalent to the caller's extension number, the return call would ring some other set than the one that dialed 911. To overcome that limitation, you can automatically forward that return call to the set that placed the emergency call for an administered period of time.

This Emergency Extension Forwarding only applies if the emergency location extension number is an extension on the same PBX as the extension that dialed 911. Customers who have several PBXs in a campus should assign emergency location extensions accordingly.

This field sets the Emergency Extension Forwarding timer for all incoming trunk calls if an emergency call gets cut off (drops).

Valid entries	Usage
0 to **999**	Type a number between 0 and 999 that represents the time (in minutes) that an incoming trunk call will forward to the extension that made the initial 911 call. The default value for both new installs and upgrades is **10**.

Note:
If a user at the emergency location extension (the extension that made the initial 911 call) manually turns off the Call Forwarding feature, the feature is off no matter how many minutes might remain on the timer.

Enable Inter-Gateway Alternate Routing

Valid entries	Usage
y/n	Enter **y** to enable the Inter-Gateway Alternate Routing feature. Default is **n**.

EIA Device Bit Rate

This field does not appear for S8700 Media Servers

Valid entries	Usage
1200 **2400** **4800** **9600**	Enter the required printer speed setting.

Switch Name

Valid entries	Usage
Any keyboard character	Enter up to 20 alpha-numeric characters for identification.

MALICIOUS CALL TRACE PARAMETERS

Apply MCT Warning Tone

Valid entries	Usage
y/n	Enter **y** to provide an audible tone to the controlling station when an MCT recorder is actively recording a malicious call.

Delay Sending Release (seconds)

This field specifies the amount of time DEFINITY waits before sending an ISDN release message in response to receiving an ISDN disconnect message. This field appears only if, on the **System-Parameters Customer-Options** screen, the **Malicious Call Trace** field is **y**.

Valid entries	Usage
0 to **30**	Enter the number in increments of 10.

MCT Voice Recorder Trunk Group

Assign the trunk group for MCT voice recorders.

Valid entries	Usage
1 to **666** or blank	group number for DEFINITY CSI
1 to **2000** or blank	group number for S8700 Series IP-Connect

SEND ALL CALLS OPTIONS

Auto Inspect on Send All Calls

Valid entries	Usage
y	If set to **y**, allows you to be presented automatically with Calling Party information for calls which are silently alerting their station because of the Send-All-Calls feature.
n	If set to **n**, you are not guaranteed a Calling Party display for calls sent directly to Coverage by the Send-All-Calls feature.

Send All Calls Applies to

Valid entries	Usage
station	If set to **station**, any call to that station, regardless of the number dialed, causes calls to that station's own extension to be sent immediately to Coverage, or causes calls to different extensions assigned to the station as bridged appearances to become Ring-Ping notification if **Redirect Notification** field is **y**.
extension	When set to **extension**, only the calls sent to that extension are placed to coverage.

UNIVERSAL CALL ID

Create Universal Call ID (UCID)

Valid entries	Usage
y	If set to **y**, DEFINITY will generate UCID for each call when necessary.
n	If set to **n**, the DEFINITY will not generate a UCID for any call.

UCID Network Node ID

Enter a number unique to this server/switch in a network of switches.

Valid entries	Usage
1 to **32767** or blank	This number is an important part of the UCID tag and must be unique to the server/switch.

Field descriptions for page 6

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                FEATURE-RELATED SYSTEM PARAMETERS

        Public Network Trunks on Conference Call: 5              Auto Start? n
    Conference Parties with Public Network Trunks: 6             Auto Hold? n
 Conference Parties without Public Network Trunks: 6        Attendant Tone? y
          Night Service Disconnect Timer (seconds): 180       Bridging Tone? n
                   Short Interdigit Timer (seconds): 3       Conference Tone? n
                Unanswered DID Call Timer (seconds): ____     Intrusion Tone? n
                Line Intercept Tone Timer (seconds): 30  Special Dial Tone? n
                   Long Hold Recall Timer (seconds):   Mode Code Interface? n
                        Reset Shift Timer (seconds): 0
     Station Call Transfer Recall Timer (seconds): 0
                              DID Busy Treatment: tone
        Invalid Number Dialed Intercept Treatment: Announcement _____
                 Allow AAR/ARS Access from DID/DIOD? _
                   Allow ANI Restriction on AAR/ARS? _
      Use Trunk COR for Outgoing Trunk Disconnect? _
                    7405ND Numeric Terminal Display? n              7434ND? n
DISTINCTIVE AUDIBLE ALERTING
                   Internal: 1 External: 2 Priority: 3
                         Attendant Originated Calls:
    DTMF Tone Feedback Signal to VRU - Connection: _       Disconnection: _
```

Feature-Related System Parameters screen

7405ND Numeric Terminal Display

Valid entries	Usage
y/n	If enabled, this allows you to use 7405ND in the **Type** field of the **Station** screen. This is not an actual telephone type, but you can use this to define ports for certain types of Octel Messaging Division voice messaging systems. This numeric display setting sends only numbers, and not names, to the Octel system.

7434ND

Valid entries	Usage
y/n	If enabled, this allows you to use 7434ND in the **Type** field of the **Station** screen. This is not an actual telephone type, but you can use this to define ports for certain types of Octel Messaging Division systems. Use this value if your voice messaging system operates in Bridged Mode.

Allow AAR/ARS Access from DID/DIOD

Valid entries	Usage
y/n	Enter **y** to allow calls for DID and DIOD type trunk groups to complete calls using ARS or AAR.

Allow ANI Restriction on AAR/ARS

(For Russia only) If a call is placed over a Russian shuttle trunk or a Russian rotary trunk via an AAR or ARS entry with the **ANI Req** field set to **r**, then ANI is requested just like a **y** entry. However, if the ANI request fails, the call immediately drops. All other trunk types treat the **r** entry as a **y**.

Valid entries	Usage
y	The **ANI Req** field on the AAR and ARS Digit Analysis Table and the AAR and ARS Digit Conversion Table permits the additional value of **r**(estricted).
n	The **ANI Req** field on the two screens takes only the current values of **n** and **y**.

Attendant Originated Calls

Valid entries	Usage
Internal external priority	Indicate which type of ringing applies to attendant-originated calls.

Attendant Tone

Valid entries	Usage
y/n	Enter **y** to provide call progress tones to the attendants.

Auto Hold

Valid entries	Usage
y/n	Enter **y** to enable the Automatic Hold feature on a system-wide basis.

Auto Start

If this field is enabled, the **Start** buttons on all attendant consoles are disabled.

Valid entries	Usage
y/n	Enter **y** to enable the Automatic Start feature.

Bridging Tone

Valid entries	Usage
y/n	Enter **y** to apply a bridging tone when calls are bridged on primary extensions.

Conference Parties with Public Network Trunks

If the value of the **Public Network Trunks on Conference Call** field is **0**, this field will not appear on the screen.

Valid entries	Usage
3 to **6**	Specify the maximum number of parties allowed in a conference call involving a public network subscriber.

Conference Parties without Public Network Trunks

Valid entries	Usage
3 to **6**	Enter a number to specify the maximum number of parties allowed in a conference call involving no public network trunks.

Conference Tone

Note:
Bridging and Conference Tones are not supported by all countries. If these tones are enabled for countries other than Italy, Belgium, United Kingdom, or Australia, the tones will be equivalent to no tone (silence) unless the tone is independently administered or customized on the **Tone Generation** screen.

Valid entries	Usage
y/n	Enter **y** to provide conference tone as long as three or more calls are in a conference call.

DID Busy Treatment

Specifies how to handle a direct inward dialing (DID) call to a busy station.

Valid entries	Usage
attendant	Call is routed to attendant.
tone	Caller hears a busy tone.

Intrusion Tone

Valid entries	Usage
y/n	Enter **y** to apply an intrusion tone (executive override) when an attendant intrudes on the call.

Invalid Number Dialed Intercept Treatment

Enter the type of intercept treatment the end-user hears after dialing an invalid number.

Valid entries	Usage
announcement	Provides a recorded announcement when the end-user dials an invalid number. You select and record the message. Enter the extension number for the announcement in the associated field.
tone	Provides intercept tone when the end-user dials an invalid number. This is the default.

Line Intercept Tone Timer (seconds)

Valid entries	Usage
0 to **60**	Enter a number to specify how long an analog station user can wait after hearing warning tone without going on hook, before the station is placed in the lockout state.

Long Hold Recall Timer (seconds)

You can administer the system to remind a user that a call has been on hold for too long.

Valid entries	Usage
0 to **999**	Enter a number between **0** and **999**; **0** deactivates the timer. This value is the number of seconds a call can be on hold before the system re-alerts the user to remind them of the call.

Mode Code Interface

Note:
If you make a change to this field, you must log off and log back on to effect the permission changes to get to the Mode Code Related System Parameters .

Valid entries	Usage
y/n	A **y** entry allows you to use the Mode Code Voice Mail System Interface to connect the server running Communication Manager over a DTMF interface to INTUITY AUDIX or other vendors' voice-mail systems.

Night Service Disconnect Timer (seconds)

Valid entries	Usage
10 to **1024** or blank	Enter a number or blank to indicate how long a trunk call can be unanswered during night service before being disconnected. The trunk must not have Disconnect Supervision for this timer to apply.

Public Network Trunks on Conference Call

Indicates the number of public network trunks allowed on a conference call

Valid entries	Usage
0 to **5**	If this field is **0**, the **Conference Parties with Public Network Trunks** field will not appear on the screen.

Reset Shift Timer (seconds)

Used only for station-to-station calls or private network calls using ISDN trunks.

Valid entries	Usage
0 to **255**	Specifies the number of seconds that reset shift dial tone is audible before busy tone is heard. Reset shift dial tone allows the user to dial a new extension by dialing one new digit that replaces the last digit of the extension previously dialed. The new digit replaces the last digit of the extension previously dialed. Enter **0** to disable this feature.

Short Interdigit Timer (seconds)

Valid entries	Usage
3 to **9**	Enter a number to limit the time that digit analysis will wait for the next digit when it has predicted that all the digits have already been collected.

Special Dial Tone

Special dial tone notifies an analog-telephone user if certain features are still active when the user goes off-hook. These features include:

- Call Forwarding
- Send All Calls
- Do Not Disturb

Valid entries	Usage
y/n	Enter **y** to use the Special Dial Tone. You must have a TN2182 circuit pack.

Station Call Transfer Recall Timer (seconds)

Allows a user-transferred call (station-to-station, a trunk call, or a DCS call) to re-terminate with priority ringing back to the station user who initiates the transfer operation if the transfer-to party does not answer the call within the administered Station Call Transfer Recall timer.

Valid entries	Usage
0 to **999**	Enter the time in seconds before a call redirects back to the station user who initiated the transfer operation. Enter **0** to disable this feature.

Unanswered DID Call Timer (seconds)

Enter number or blank to limit how long a DID call can remain unanswered before routing to the DID/TIE/ISDN Intercept Treatment feature. This timer interacts with the nonadministrable 50 second Wait for Answer Supervision Timer (WAST). The WAST timer overrides this field. Thus if this field is set to a value equal to or greater than 50 seconds, the caller receives intercept tone instead of the normal attendant or announcement treatment that is given when the Unanswered DID Call Timer expires before the WAST. If the Unanswered DID Call Timer expires while the DID call is being processed by call vectoring, the timer is ignored.

Valid entries	Usage
A number between **10** and **1024**	Enter a number to indicate how long a DID call can remain unanswered before routing to the DID/TIE/ISDN Intercept Treatment feature.
blank	Disables the timer.

Use Trunk COR for Outgoing Trunk Disconnect

Use this field to indicate whether the Outgoing Trunk Disconnect Timer is set based on the COR of the originating station or of the trunk group. If enabled, the timer is based on the COR of the trunk, not the originating caller's station.

Valid entries	Usage
n	Default. The Outgoing Trunk Disconnect Timer to be set based on the COR of the originating station. This is the default.
y	Enter **y** to enable the Outgoing Trunk Disconnect Timer to be set based on the COR of the trunk instead of the originating station.

DISTINCTIVE AUDIBLE ALERTING

Attendant Originated Calls

Valid entries	Usage
Internal **External** **priority**	Indicates which type of ringing (defined above) applies to attendant-originated calls. Default is **external**.

Distinctive Audible Alerting (Internal, External, Priority)

This is also known as Distinctive Ringing. Enter the number of rings for **Internal**, **External**, and **Priority** calls. For virtual stations, this applies to the mapped-to physical telephone. Defaults are as follows:

- **1**: Internal calls
- **2**:External and attendant calls
- **3**:Priority calls

Note:

SIP Enablement Services (SES) messaging includes the ring types internal, external, intercom, auto-callback, hold recall, transfer recall, or priority. In Communication Manager, types intercom, auto-callback, hold recall, and transfer recall are treated as priority.

Valid entries	Usage
4.	1 burst, meaning one burst of ringing signal per period
5.	2 bursts, meaning two bursts of ringing signal per period
3	3 bursts, meaning two bursts of ringing signal per period

DTMF Tone Feedback Signal to VRU - Connection, Disconnection

This field appears only if **DTMF Feedback Signals for VRU** on the **Customer-Options System Parameters** screen is **y**.

Valid entries	Usage
0 to **9, *, #, A, B, C, D**	Enter the code to connect or disconnect the VRU. This can be a single digit, or a combination such as *99 to connect, #99 to disconnect. The tones must be programmed at the VRU as well.
blank	Blank means that no tone is to be sent to the VRU.

Field descriptions for page 7

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              FEATURE-RELATED SYSTEM PARAMETERS

CONFERENCE/TRANSFER

                 Abort Transfer?                      No Dial Tone
Conferencing?
          Transfer Upon Hang-Up?            Select Line Appearance
Conferencing?
 Abort Conference Upon Hang-Up?
Unhold?
    No Hold Conference Timeout:       Maximum Ports per Expanded Meet-me
Conf:


ANALOG BUSY AUTO CALLBACK
                    Without Flash?                      Announcement:
                                          Voice Mail Hunt Group Ext:


AUDIX ONE-STEP RECORDING
                              Recording Delay Timer (msec):
  Apply Ready Indication Tone To Which Parties In The Call?
```

Feature-Related System Parameters screen

CONFERENCE/TRANSFER

Abort Conference Upon Hang-Up

Allows DCP, hybrid, IP, wireless, or ISDN-BRI telephone users to abort the conference operation when they hang up.

Valid entries	Usage
y/n	Enter **y** to change a call placed on soft-hold in the conference-pending status to hard-held status if the user hangs up.

Abort Transfer

Stops the transfer operation whenever a user presses a non-idle call appearance button in the middle of the transfer operation, or when they hang up. If both the **Abort Transfer** and **Transfer Upon Hang-Up** fields are **y** and you press the **transfer** button and then dial the complete transfer-to number, hanging up the telephone transfers the call. You must select another non-idle call appearance to abort the transfer. If the **Transfer Upon Hang-Up** field is **y**, hanging up completes the transfer. Requires DCP, Hybrid, IP, ISDN-BRI or wireless telephones.

Valid entries	Usage
y/n	Enter **y** to abort the transfer a call by pressing the **Transfer** button, dialing the desired extension, and then hanging up or selecting another non-idle call appearance. The user must press the **Transfer** button again to complete the process unless **Transfer Upon Hang-up** is also set to **y**.

Maximum Ports per Expanded Meet-me Conf

This field allows you to administer the maximum number of conferees in an Expanded Meet-me Conference. This is a system-wide limit (i.e., not administrable on a per Expanded-Meet-me VDN basis). This field is hidden if **Maximum Number of Expanded Meet-me Conference Ports** is **0** on the **System Parameters Customer Options** screen.

Valid entries	Usage
3 to **300**	Enter the maximum number of parties allowed for each conference on your system.

No Dial Tone Conferencing

When another line is on hold or alerting, No Dial Tone Conferencing eliminates dial tone while setting up a conference.

Valid entries	Usage
y/n	Enter **y** to activate No Dial Tone Conferencing.

No Hold Conference Timeout

Controls the timeout of No Hold Conference call setup. The system Answer Supervision timer should be set to a value less than this.

Valid entries	Usage
20 to **120**	Enter the number of seconds.

Select Line Appearance Conferencing

Use this field to specify that the user can use the line appearance rather than the **Conf** button to include a call in a conference. If a user is on a call, and another line is on hold or an incoming call alerts on another line, the user can press the **Conf** button to bridge the calls together. Using the select line appearance capability, the user can press a line appearance button to complete a conference instead of pressing the **Conf** button a second time.

Valid entries	Usage
y/n	Enter **y** to activate Select Line Appearance Conferencing.

Transfer Upon Hang-Up

Allows DCP, hybrid, IP, wireless, or ISDN-BRI telephone users to complete a transfer operation by hanging up.

Valid entries	Usage
y/n	Enter **y** so users can transfer a call by pressing the **Transfer** button, dialing the desired extension, and then hanging up. The user can also wait to hang up, speak with the other party, then press **Transfer** again to complete the process. With this field set to **y**, users of the Call Park FAC can park a call without having to press the **Transfer** button a second time.

Unhold

Allows the user to press the **hold** button on a telephone to release a hold (if no other line appearance is on hold or alerting). This does not apply to BRI telephones or attendant consoles.

Valid entries	Usage
y/n	Enter **y** to activate the unhold capability

ANALOG BUSY AUTO CALLBACK

With the Analog Busy Auto Callback Without Flash (ACB) feature enabled, when a caller places a call through an analog station, and the called station is busy and has no coverage path nor forwarding, then an announcement plays, announcing that the station is busy and prompting the caller to enter **1** for ACB or **2** to cover to a voice mail hunt group extension.

Announcement

Appears only if the **Without Flash** field is **y**.

Valid entries	Usage
Extension number	Enter the extension of the announcement you want to play for the ACB feature. This field cannot be left blank.

Voice Mail Hunt Group Ext

Appears only if the **Without Flash** field is **y**.

Valid entries	Usage
Extension number	Enter a voice mail hunt group extension to which the call is to be forwarded if the user enters 2 at the ACB announcement prompt.

Without Flash

Provides automatic callback for analog stations without flashing the hook. It is applied only when the called station is busy and has no other coverage path or call forwarding. The caller can enable the automatic callback without flashing the hook or entering the feature access code.

Note:
If the **Analog Busy Auto Callback Without Flash** field is set to **y**, the **Busy Auto Callback without Flash** field on the **Station** screen defaults to **y** (enabled) for all analog station types that allow Analog Auto Callback.

Valid entries	Usage
y/n	Enter **y** to provide automatic callback for a calling analog station without flashing the hook.

AUDIX ONE-STEP RECORDNG

On stations administered with this feature button, this feature allows users to activate and deactivate the recording of active calls to their Audix with the press of one button.

Apply Ready Indication Tone To Which Parties In The Call

This field is for administering who hears the Audix recording ready tone.

Valid entries	Usage
all, initiator, or **none**	Enter a value for which party or parties on the call should hear the ready-to-record indication tone. The default is **all**. This field cannot be left blank.

Interval For Applying Periodic Alerting Tone (seconds)

Appears only if the **Apply Ready Indication Tone To Which Parties In The Call** field is set to **all**.

Valid entries	Usage
0 to **60**	Enter a number from zero to 60 for the number of seconds desired between alerting tones, where zero disables the tone. The default value is a 15 second interval.

Recording Delay Timer (msecs)

Valid entries	Usage
0 to **4000** in increments of 100	Use this field to administer a delay interval before starting audix recording.

Field descriptions for page 8

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                FEATURE-RELATED SYSTEM PARAMETERS


ISDN PARAMETERS

           Send Non-ISDN Trunk Group Names as Connected Name?
            Display Connected Name/Number for ISDN DCS Calls?
                  Send ISDN Trunk Group Name on Tandem calls?

                              Send Custom Messages Through QSIG?
                                                 QSIG TSC Extension:
            MWI - Number of Digits Per Voice Mail Subscriber:

                                          National CPN Prefix:
                                     International CPN Prefix:
                                    Pass Prefixed CPN to ASAI:
               Unknown Numbers Considered Internal for AUDIX?
                        UNSI Calling Name for Outgoing Calls?
                          Path Replacement with Measurements?
                             QSIG Path Replacement Extension:
                       Path Replace While in Queue/Vectoring?
```

Feature-Related System Parameters screen

ISDN PARAMETERS

Display Connected Name/Number for ISDN DCS Calls

Valid entries	Usage
y/n	Enter **y** to display the connected name/number (if received) for ISDN DCS calls.

Feature Plus Ext

Valid entries	Usage
A valid extension	Administration of this field is required for proper termination of some Feature Plus signaling. For example, Message Waiting Indication (MWI) requires this extension in order to send the indication to the appropriate server running Communication Manager. Appears only if the **ISDN Feature Plus** field is **y** on the **System-Parameters Customer-Options** screen.

International CPN Prefix

Allows you to apply prefixes to international calling numbers for display at receiving telephones. This is useful for those telephones that use or implement call back features based on incoming call numbers. When an ISDN-PRI call arrives, the incoming call setup is analyzed for: (1) whether the Type of Address (TOA) is national or international, and (2) whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony. This administered prefix is applied to international calls. Prefixing applies to any subsequent display on the same server when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Valid entries	Usage
1 to 5 digits, (**0** to **9**), * and **#** or blank	Enter a number that allows you to apply prefixes to international calling numbers for display.

Maximum Length

Appears only if the **Unknown Numbers Considered Internal for AUDIX** field is **y**. Indicates the maximum length of an unknown private number. Any unknown number longer than the administered value is considered external. This field cannot be blank when it appears.

Valid entries	Usage
1 to **20**	Enter a number for the maximum length of an unknown private number.

MWI - Number of Digits Per Voice Mail Subscriber

Appears only if the **Basic Supplementary Services** field or the **ISDN Feature Plus** field on the **System-Parameters Customer-Options** screen is **y**. This field provides an indication of the number of digits per AUDIX subscriber.

Note:

For QSIG-MWI, these routing digits and inserted digits must screen a digit string that, when analyzed and processed, routes to a Signaling Group supporting QSIG-TSCs. Once a QSIG TSC is established (from a message- center server/ switch to a served-user switch), then every lamp update message places the **Inserted Digits** field (from the **Message Waiting Indication Subscriber Number Prefixes** screen) in front of the AUDIX subscriber number to screen a complete QSIG network number for the served user.

For Feature Plus MWI, the routing digits and inserted digits must screen a digit string that routes over an SSF trunk to the Feature Plus extension on the remote (served user) switch. The **Inserted Digits** field must include the Feature Plus extension.

Valid entries	Usage
3 to **7**	Enter a value that corresponds to the digit string length of subscribers translated in the Message Center entity. For instance, if the Message Center entity is AUDIX, the value in this field must match the value of the **Extension Length** field on the **Switch Interface Administration** screen of AUDIX.

National CPN Prefix

Allows you to apply prefixes to national calling numbers for display at receiving telephones. This is useful for those telephones that use or implement call back features based on incoming call numbers. When an ISDN-PRI call arrives, the incoming call setup is analyzed for: (1) whether the Type of Address (TOA) is national or international, and (2) whether the Numbering Plan Identifier (NPI) is Unknown or ISDN/Telephony. This administered prefix is applied to national calls. Prefixing applies to any subsequent display on the same server when the call is transferred, covered, or forwarded. The same prefixing applies to outgoing ISDN-PRI calls when the connected number information is returned and meets the same TOA and NPI criteria. The prefix plus the calling/connected number digit string is limited to 15 digits, with truncation occurring at the least significant digits.

Valid entries	Usage
1 to 5 digits, (**0** to **9**), * and **#** or blank	Enter a number that allows you to apply prefixes to national calling numbers for display.

Pass Prefixed CPN to ASAI

Passes Calling Party Number information (CPN) to ASAI. The prefixed number is not passed on to other adjuncts, Call Detail Recording, or servers/switches.

Valid entries	Usage
y/n	Enter **y** to pass CPN information to ASAI.

Path Replacement While in Queue/Vectoring

Valid entries	Usage
y/n	Enter **y** to allow Path Replacement after queue/vector processing has started. Depending on the version of CMS you are using, some calls can go unrecorded if you enable this capability.

Path Replacement with Measurements

Valid entries	Usage
y/n	Allows QSIG path replacement or DCS with Reroute to be attempted on measured calls.

QSIG Path Replacement Extension

Enter the extension for the system to use as part of the complete number sent in the Path Replacement Propose message.

Valid entries	Usage
Extension	Enter an unused extension that conforms to your dial plan.

QSIG TSC Extension

Valid entries	Usage
Enter any valid, unassigned extension	This is the phantom endpoint extension for QSIG Call Independent Signaling Connections (CISCs), which are similar to NCA Temporary Signaling Connections (TSCs) (both incoming and outgoing).

Send Custom Messages Through QSIG?

Valid entries	Usage
y/n	Enter **y** to provide appropriate display information, for example for the Posted Messages feature, over QSIG links.

Send ISDN Trunk Group Name on Tandem Calls

Valid entries	Usage
y/n	Enter **y** to provide consistent display information regardless of trunk type. If set to **y**, provides only trunk group name.

Send Non-ISDN Trunk Group Name as Connected Name

Valid entries	Usage
y/n	Enter **y** to send a name of the non-ISDN trunk group as the connected name when a call routes from ISDN to non-ISDN and the call is answered.

Unknown Numbers Considered Internal for AUDIX

Appears when, on the **System-Parameters Customer-Options** screen, either the **ISDN-PRI** or **ISDN-BRI Trunks** field is **y**. This field controls the treatment of an ISDN number whose numbering plan identification is "unknown" in a QSIG centralized AUDIX arrangement. This field works in conjunction with the Calling Party Number to INTUITY AUDIX field on the **Hunt Group** screen. The **Calling Party Number to INTUITY AUDIX** field on the **Hunt Group** screen must be **y** for this field to have an effect.

Valid entries	Usage
y	The unknown number is considered "internal" and AUDIX tries to find a calling party name match for the digit string. If a name match is found, AUDIX provides the calling party's name. If no name is found, AUDIX provides the calling party's telephone number.
n	The unknown number is considered "external" and AUDIX provides the calling party's telephone number.

USNI Calling Name for Outgoing Calls?

Valid entries	Usage
y/n	Enter **y** to send a name on outgoing calls over NI PRI trunks. Important: Be sure you have validated that your service provider's central office is capable of accepting calling name information from Communication Manager in this way. For example, if the central office has a 5ESS, it must be a generic 5EXX or later. Failure to validate the central office capability might cause the central office to drop outgoing calls from your media server. In this case, change the value in this field to **n**. Enter **n** to prevent sending calling name information with outgoing calls over NI PRI trunks. **n** is this field overrides a **y** in the **Send Name** field of the outgoing **Trunk Group** screen.

Field descriptions for page 9

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            FEATURE-RELATED SYSTEM PARAMETERS


CPN/ANI/ICLID PARAMETERS

            CPN/ANI/ICLID Replacement for Restricted Calls:
           CPN/ANI/ICLID Replacement for Unavailable Calls:

INTERNATIONAL CALL ROUTING PARAMETERS
                     Local Country Code: 1
                International Access Code: 011

ENBLOC DIALING PARAMETERS
Enable Enbloc Dialing without ARS FAC?

CALLER ID ON CALL WAITING PARAMETERS
Caller ID on Call Waiting Delay Timer (msec): 200
```

Feature-Related System Parameters screen

CPN/ANI/ICLID PARAMETERS

CPN/ANI/ICLID Replacement for Restricted Calls

Valid entries	Usage
up to 15 characters	Enter a text string to replace the restricted numbers on the display.

CPN/ANI/ICLID Replacement for Unavailable Calls

Valid entries	Usage
up to 15 characters	Enter a text string to replace the unavailable numbers on the display.

INTERNATIONAL CALL ROUTING PARAMETERS

Local Country Code

Valid entries	Usage
1 to 3 digits or blank	Enter a valid country code for this node. The default is blank (no SBS signaling trunk groups are administered). For example, for an SBS node in the United States, enter **1**.

International Access Code

Valid entries	Usage
1 to 5 digits or blank	Enter the access code required by the PSTN to route calls out of the country. This code will be included with the telephone number received from the SBS terminating node if the Local Country Codes of the originating and terminating nodes are different. The default is blank (no access code is needed).

Note:
Once administered, these fields cannot be cleared until all trunk groups administered for SBS signaling have been removed.

ENBLOC DIALING PARAMETERS

Enable Enbloc Dialing without ARS FAC

Valid entries	Usage
y/n	Enter **y** to enable Enbloc Dialing without the need to dial a FAC. Default is **n**.

Minimum Digit Length

This field appears only when **Enable Enbloc Dialing without ARS FAC** is **y**.

Valid entries	Usage
1 to **20**	Enter the number of digits before Enbloc Calling Treatment is activated. Default is extension length plus 1.

CALLER ID ON CALL WAITING PARAMETERS

Caller ID on Call Waiting Delay Timer (msec)

Valid entries	Usage
5 to **1275** in increments of 5	Enter the desired delay in 5-millisecond intervals. Default is **200**.

Field descriptions for page 10

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change system-parameters features                                page 10 of x

                FEATURE-RELATED SYSTEM PARAMETERS

                   Pull Transfer: n          Update Transferred Ring Pattern?
n
          Outpulse Without Tone? y             Wait Answer Supervision Timer?
n
          Misoperation Alerting? n              Repetitive Call Waiting Tone?
y
    Allow Conference via Flash? y    Repetitive Call Waiting Interval (sec):
_
Vector Disconnect Timer (min): __   Network Feedback During Tone Detection?
y
  Hear Zip Tone Following VOA? y   System Updates Time On Station Displays?
n
                  Intercept Treatment on Failed Trunk Transfers? n
                                Station Tone Forward Disconnect: silence
                                     Level Of Tone Detection: precise
                      Charge Display Update Frequency (seconds): 30
                   Date Format on 607/2400/4600/6400 Terminals: mm/dd/yy
           On-hook Dialing on 607/2400/4600/6400/8400 Terminals? N

ITALIAN DCS PROTOCOL
```

Feature-Related System Parameters screen

Allow Conference via Flash

Valid entries	Usage
y	Enter **y** to allow an analog station to use flash to conference calls together.
n	Enter **n** to prevent this.

Charge Display Update Frequency (seconds)

This applies only if you use Advice of Charge or Periodic Pulse Metering with display functions.

Valid entries	Usage
10 to **60** or blank	The amount of time (in seconds) between charge-display updates. Frequent display updates might have considerable performance impact. If the duration of a call is less than the Charge Display Update Frequency, the display will not automatically show charge information. To see charge information for a call, the user must have a **disp-chrg** button and must press the button before the call drops.

Date Format on 607/2400/4600/6400 Terminals

The format of the date as displayed on the telephones.

Valid entries	Usage
mm/dd/yy	month/day/year
dd/mm/yy	day/month/year
yy/mm/dd	year/month/day

Hear Zip Tone Following VOA?

This tone alerts a telephone user that the announcement has completed and a caller is now connected. CallMaster set and attendant console users hear double zip tone following the announcement. All other telephone users hear single zip tone.

Note:
This field does not effect auto-answer zip tone heard prior to the VOA.

Valid entries	Usage
y	Enter **y** to play zip tone following a VDN of Origin Announcement (VOA).
n	Enter **n** if you do not want zip tone following a VOA.

Intercept Treatment on Failed Trunk Transfers

Valid entries	Usage
y	Enter **y** to provide intercept treatment to calls failing trunk transfers.
n	Enter **n** to drop these calls.

Level of Tone Detection

For the most part, this option is no longer required in today's switching environment. It might be useful if your users are having difficulty placing outgoing calls due to inaccurate detection of network dial tone.

Valid entries	Usage
broadband	This is the least exact of the levels of tone detection. If Avaya Communication Manager detects any tone at all, it interprets this as dial tone.
medium	The server running Avaya Communication Manager interprets any tone which has a continuous "on" period of longer than 1 second as dial tone. Otherwise, the server accepts whatever the tone detector circuit pack reports.
precise	Communication Manager accepts whatever the tone detector circuit pack reports.

Misoperation Alerting

Misoperation Alerting should not be enabled if Call Prompting is optioned.

Valid entries	Usage
y	Enter **y** for misoperation recall alerting on multi-appearance stations, analog stations, and attendant consoles.
n	Enter **n** for standard misoperation handling without recall alerting.

Network Feedback During Tone Detection

Valid entries	Usage
y/n	Enter **y** to provide audible feedback to the user while the system attempts to detect dial tone.

On-hook Dialing on 607/2400/4600/6400/8400 Terminals

For 6400/8400, 607, 2420, 2410, and 4600 telephone users with speakerphones.

Valid entries	Usage
y/n	Enter **y** allows users to dial without lifting the handset. If you enable this, users hear dial tone when they press the **Speaker** button, even if the handset is on-hook.

The next four fields control station-to-switch recall signal timing. If a flashhook interval (recall window) is required, the upper and lower bounds of the interval can be administered. An on-hook that lasts for a period of time greater than or equal to the lower bound and less than or equal to the upper bound will be treated as a recall flash. If an interval is not required, the **Disconnect Timing** value must be administered. An on-hook that lasts for a period of time less than this value will be ignored; greater than or equal to this value will be regarded as a disconnect. Regardless, an on-hook lasting 50 to 150 ms coming from a 2500-type set will always be treated as a digit pulse unless **Ignore Rotary Digits** is **y** for that station.

Outpulse Without Tone

Valid entries	Usage
y	Enter **y** to indicate the server will outpulse digits even when a dial tone has not been received.
n	Enter "**n**" if the calling party should receive intercept tone if no dial tone is detected.

Pull Transfer

Valid entries	Usage
y/n	Enter **y** to enable the Pull Transfer feature on a system-wide basis. This allows either the transferring or transferred-to party to press the **Transfer** button to complete the transfer operation

Repetitive Call Waiting Interval (sec)

This field appears when the **Repetitive Call Waiting Tone** field is **y**.

Valid entries	Usage
1 to **99**	Enter a number to specify the number of seconds between call waiting tones.

Repetitive Call Waiting Tone

Valid entries	Usage
y/n	Enter **y** to indicate that a repetitive call waiting tone be provided to the called party for all types of call waiting access.

Station Tone Forward Disconnect

Tone Forward Disconnect applies to any station other than one administered as a data endpoint, an attendant console, a BRI telephone, an auto answer, or as an Outgoing Call Management (OCM) agent.

Valid entries	Usage
Busy **Intercept** **Silence**	When a station is the last party remaining off-hook on a call, that station receives the indicated tone or silence until that station is placed on-hook, or until the tone has played for 45 seconds and is followed by silence.

System Updates Time On Station Displays

This does not apply to telephones (such as BRI telephones) where the user sets the time.

Valid entries	Usage
y/n	Enter **y** to have the system automatically update the time on display telephones when background maintenance is run (for example, when the set is plugged in).

Update Transferred Ring Pattern

Valid entries	Usage
y/n	Enter **y** to change the ringing pattern from internal to external when an internal station transfers an external call. If most of your calls go through an attendant, you might want to set this to **y**, so your users will be able to distinguish an external call.

Vector Disconnect Timer (min)

Enter the number of minutes, or blank that a trunk should remain connected to a vector.

Valid entries	Usage
1 to **240**	The number of minutes that you enter determines when the trunk will be disconnected if the **Disconnect Supervision-In** or **Disconnect Supervision-Out** fields on the **Trunk Group** screen are **n**.
blank	Enter blank if you do not want Avaya Communication Manager to initiate a disconnect.

Wait Answer Supervision Timer

Valid entries	Usage
y	Enter **y** to enable this feature on a systemwide basis. When **y** is entered in this field, calls to stations unanswered after 50 seconds are dropped.
n	When **n** is entered in this field, unanswered calls drop only when the calling party goes on-hook.

ITALIAN DCS PROTOCOL

The next three fields control the Italian DCS Protocol feature.

Valid entries	Usage
y/n	Enter **y** to indicate that DID/CO intercept treatment will be applied locally instead of on the originating server/switch.

Apply Intercept Locally

This field appears if the **Italian Protocol Enabled** field is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that DID/CO intercept treatment will be applied locally instead of on the originating server/switch.

Enforce PNT-to-PNT Restrictions

This field appears if the **Italian Protocol Enabled** field is **y**.

Valid entries	Usage
y/n	Enter **y** to indicate that restrictions and denial of PNT-to-PNT connections will be enforced when the EDCS message is unavailable. A **y** in this field means restrictions will be enforced.

Italian Protocol Enabled

Valid entries	Usage
y/n	Enter **y** to enable the Italian DCS feature on a systemwide basis

Field descriptions for page 11

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                FEATURE-RELATED SYSTEM PARAMETERS

CALL CENTER SYSTEM PARAMETERS
  EAS
              Expert Agent Selection (EAS) Enabled? n
             Minimum Agent-LoginID Password Length:
               Direct Agent Announcement Extension: _______       Delay: ___
         Message Waiting Lamp Indicates Status For: station
VECTORING
                         Converse First Data Delay: 0     Second Data Delay: 2
                    Converse Signaling Tone (msec): 100        Pause (msec):
70_
                          Prompting Timeout (secs): 10
                      Interflow-qpos EWT Threshold: 2
        Reverse Star/Pound Digit For Collect Step? n
              Available Agent Adjustments for BSR? n
                                  BSR Tie Strategy? 1st_found

SERVICE OBSERVING
                   Service Observing: Warning Tone? n    or Conference Tone? n
         Service Observing Allowed with Exclusion? N
```

Feature-Related System Parameters screen

CALL CENTER SYSTEM PARAMETERS

EAS

Direct Agent Announcement Delay

Only appears if **Expert Agent Selection (EAS)** or **ASAI Link Core Capabilities** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
0 to **99** or blank	Enter the number of seconds the caller will hear ringback before the Direct Agent Announcement is heard by the calling party.

Direct Agent Announcement Extension

Valid entries	Usage
Valid extension	Enter the extension of the direct agent announcement.

Expert Agent Selection (EAS) Enabled

To enable this field, either no ACD or vectoring hunt groups might exist or, existing ACD or vectoring hunt groups must be "skilled." Only appears if **Expert Agent Selection (EAS)** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
y/n	Enter **y** to enable Expert Agent Selection.

Message Waiting Lamp Indicates Status For

Only appears if **Expert Agent Selection (EAS)** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
station	Since you only have one message waiting lamp on a telephone, you need to indicate if the message is for at the telephone extension or the loginID.
loginID	Expert Agent Selection (EAS) must be enabled to use this option.

Minimum Agent-LoginID Password Length

Enter the minimum number of digits that must be administered as an EAS Agent's LoginID password. Only appears if **Expert Agent Selection (EAS)** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
0 to **9**	Entering a **0** or blank indicates no password is required.

VECTORING

Available Agent Adjustments for BSR

Controls the use of BSR available agent adjustments. The **Vectoring (Best Service Routing)** field must be **y** on the **System-Parameters Customer-Options** screen.

Valid entries	Usage
y/n	Enter **y** to allow adjustments to available agents.

BSR Tie Strategy

This field appears only when **Vectoring (Best Service Routing)** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
1st-found	BSR uses the first selection for routing. This is the default.
alternate	Allows alternating the BSR selection algorithm when a tie in EWT or available agent criteria occurs. Every other time a tie occurs for calls from the same active VDN, the selection from the consider step with the tie is used instead of the first selected split/skill or location to send the call. This helps balance the routing over the considered local splits/skills and remote locations when cost of routing remotely is not a concern.

Converse First Data Delay/Second Data Delay

The First Data Delay prevents data from being outpulsed (as a result of a converse vector step) from the system to CONVERSANT before CONVERSANT is ready. The delay commences when the CONVERSANT port answers the call. The Second Data Delay is used when two groups of digits are being outpulsed (as a result of a converse vector step) from the system to CONVERSANT. The Second Data Delay prevents the second set from being outpulsed before CONVERSANT is ready. The delay commences when the first group of digits has been outpulsed. Only appears if **Vectoring (Basic)** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
0 to **9**	Number of seconds for the delay.

Converse Signaling Tone/Pause

Only appears if **Vectoring (Basic)** and **DTMF** on the **System-Parameters Customer-Options** screen are **y**. In the **Signaling Tone** field, enter the length in milliseconds of the digit tone for digits being passed to the CONVERSANT. In the **Pause** field, enter the length in milliseconds of the delay between digits being passed. The optimum timer settings for the CONVERSANT or IR are 60 msec tone and 60 msec pause.

Valid entries	Usage
40 to **2550** (in increments of 10). **100**	Values entered in the **Tone/Pause** fields are rounded up or down depending upon the type of circuit pack used to outpulse the digits. • **TN742B or later suffix analog board** — Tone and pause round up or down to the nearest 25 msecs. For example, a 130 msec tone rounds down to 125 msecs, a 70 msec pause rounds up to 75 msec for a total of 200 msecs per tone. • **TN464F, TN767E or later suffix DS1 boards** — Tone and pause round up to the nearest 20 msecs. For example, a 130 msec tone rounds up to 140 msecs, a 70 msec pause rounds up to 80 msecs for a total of 220 msecs per tone. If a circuit pack has been used for end-to-end signalling to the CONVERSANT, and has then been used to send digits to a different destination, the CONVERSANT timers might stay in effect. To reset your timers to the system default, pull and reseat the circuit pack.

Interflow-qpos EWT Threshold

Displays only if, on the **System-Parameters Customer-Options** screen, the **Lookahead Interflow (LAI)** field is **y**. Part of enhanced Look-Ahead Interflow. Any calls predicted to be answered before this threshold will not be interflowed (therefore saving CPU resources).

Valid entries	Usage
0 to **9** or blank	Number of seconds for this threshold.

Prompting Timeout (secs)

Only appears if **Vectoring (Prompting)** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
4 to **10**	Enter the number of seconds before the `Collect Digits` command times out for callers using rotary dialing.

Reverse Star/Pound Digit for Collect Step

The "*" is interpreted as a "caller end-of-dialing indicator and the "#" is an indicator to clear all digits previously entered by the caller for the current "collect" vector step.

Valid entries	Usage
y/n	Enter **y** to reverse the star and pound digits by the "collect" vector step. If set to **y**, it does not affect any other DEFINITY vector step or other non-ACD DEFINITY feature (such as ARS) in that the "*" and "#" digit-processing is unchanged.

SERVICE OBSERVING

Service Observing: Warning Tone

Service Observing (Basic) on the **System-Parameters Customer-Options** screen must be **y** before this field can be administered.

! CAUTION:
The use of Service Observing features might be subject to federal, state, or local laws, rules or regulations or require the consent of one or both of the parties to the conversation. Customers should familiarize themselves and comply with all applicable laws, rules, and regulations before using these features.

Valid entries	Usage
y/n	Enter **y** to assign a warning tone to be given to telephone users and calling parties whenever their calls are being monitored using the Service Observing feature. This field cannot be set to **y** when **or Conference Tone?** is set to **y**.

or Conference Tone

Service Observing (Basic) on the **System-Parameters Customer-Options** screen must be **y** before this field can be administered.

Valid entries	Usage
y/n	Enter **y** to assign a conference tone to be given to telephone users and calling parties whenever their calls are being monitored using the Service Observing feature. This field cannot be set to **y** when **or Warning Tone?** is set to **y**.

Service Observing Allowed with Exclusion

Allows Service Observing of a station with Exclusion active, either by Class Of Service or by manual activation of Exclusion. Default is **n**.

Valid entries	Usage
y	Enter **y** to allow Service Observing of a station with Exclusion active, either by COS or by manual activation of Exclusion.
n	Observing towards a station with Exclusion active is denied, or if Exclusion is activated by a station while being observed, all bridged parties including the observer are dropped. This is the default.

Field descriptions for page 12

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                FEATURE-RELATED SYSTEM PARAMETERS

AGENT AND CALL SELECTION
                         MIA Across Splits or Skills? n
                          ACW Agents Considered Idle? y
                          Call Selection Measurement: current-wait-time
   Service Level Supervisor Call Selection Override? y
                                   Auto Reserve Agents:
           Copy ASAI UUI During Conference/Transfer?

ASAI
Call Classification After Answer Supervision? n           Send UCID to ASAI? N

CALL MANAGEMENT SYSTEM
                             Reporting Adjunct Release:
                       ACD Login Identification Length: 0
                                 BCMS/VuStats LoginIDs?
                   BCMS/VuStats Measurement Interval: hour
           BCMS/VuStats Abandon Call Timer (seconds):
                       Validate BCMS/VuStats Login IDs? n
                              Clear VuStats Shift Data: on-login
                 Remove Inactive BCMS/VuStats Agents? n
```

Feature-Related System Parameters screen

AGENT AND CALL SELECTION

ACW Agents Considered Idle

Valid entries	Usage
y/n	Enter **y** to have agents who are in After Call Work included in the Most-Idle Agent queue. This means that ACW is counted as idle time. Enter **n** to exclude ACW agents from the queue.

Auto Reserve Agents

When a critical skill is not meeting its service level, auto-reserve puts agents in standby for their other skills to ensure that there is an available agent when the next call arrives for the critical skill. When an agent becomes available, all of his or her assigned skills are checked to see if any auto-reserve skills are not meeting their target service level. If so, the agent is made available only in those skills.

Valid entries	Usage
all	Puts an agent on stand-by for all skills.
none	Agent is not on stand-by for any additional skills.
secondary-only	Puts an agent on stand-by only for secondary skills.

Call Selection Measurement

This field determines how Avaya Communication Manager selects a call for an agent when the agent becomes available and there are calls in queue.

Valid entries	Usage
current-wait-time	**Current Wait Time** selects the oldest call waiting for any of the agent's skills.
predicted-wait-time	**Predicted Wait Time** is a feature of Business Advocate.

.

Copy ASAI UUI During Conference/Transfer

Displays when, on the **System-Parameters Customer-Options** screen, either the **ASAI Interface** or **ASAI Proprietary Adjunct Links** field is **y**.

Valid entries	Usage
y/n	Enter **y** to copy user-to-user (UUI) information during a conference or transfer calls.

Note:

When this field is set to **y**, the system actually copies *all* UUI information, not just ASAI UUI. Copying only occurs during a human-initiated conference or transfer. Communication Manager does not copy the UUI if the conference or transfer is initiated by ASAI.

MIA Across Splits or Skills

Valid entries	Usage
y/n	Enter **y** to remove an agent from the MIA queue for all the splits/skills/hunt groups that he or she is available in when the agent answers a call from any of his or her splits/skills/hunt groups.

Service Level Supervisor Call Selection Override

This field determines whether Avaya Communication Manager changes agents' call handling preferences when a skill using Service Level Supervisor exceeds its Level 1 threshold.

Valid entries	Usage
y	Enter **y** if you want to override the normal call handling preferences of a skill's assigned agents in this situation.
n	Enter **n** if you do not want to override agents' normal call handling preferences when the skill exceeds its Level 1 threshold. Service Level Supervisor requires Expert Agent Selection and Business Advocate.

ASAI

Call Classification After Answer Supervision?

For use with ASAI Outbound Call Management (OCM).

Valid entries	Usage
y/n	Enter **y** to force the server running Communication Manager to rely on the network to provide answer/busy/drop classification to the server. After the call has been answered, a call classifier can be added to perform answering machine, modem, and voice answering detection. The default value **n** always connects a classifier after call setup for determining call progress and answer. ISDN progress messages generally take precedence.

Send UCID to ASAI

Valid entries	Usage
y/n	Enter **y** to enable transmission of Universal Call ID (UCID) information to ASAI.

CALL MANAGEMENT SYSTEM

ACD Login Identification Length

Enter the number of digits for an ACD Agent Login ID if **Expert Agent Selection (EAS)** on the **System-Parameters Customer-Options** screen is **n**. If **BCMS/VuStats Login IDs** is **y**, the ACD Login ID length must be greater than 0. This field identifies an ACD agent to CMS. The number you enter in this field must equal the number of characters in the agent's login ID

.

Valid entries	Usage
0 to **9**	For CMS, this field cannot be **0**.

BCMS/VuStats Measurement Interval

You can enter **half-hour** or **hour** for polling and reporting measurement data if the **BCMS (Basic)** and/or the **VuStats** on the **System-Parameters Customer-Options** screen is **y**.

Valid entries	Usage
half-hour	There are a maximum of 25 time slots available for measurement Intervals. If **hour** is specified, an entire day of traffic information will be available for history reports; otherwise, only half a day will be available.
Hour	This does not affect daily summaries as they always reflect traffic information for the entire day. The interval can be changed at any time, but will not go into effect until the current interval completes.

BCMS/VuStats LoginIDs

This feature can be used when EAS is not optioned, or in addition to EAS login IDs. When this field is **y**, both BCMS and CMS use the same login ID for an agent.

Valid entries	Usage
y/n	Enter **y** to administer valid agent login IDs to monitor call activity by agent.

BCMS/VuStats Abandon Call Timer (seconds)

Valid entries	Usage
1 to **10** or blank	Specify the number of seconds before calls are considered abandoned. Calls with talk time that is less than this number (and that are not held) are tracked by BCMS and displayed by VuStats as ABAND calls.

Clear VuStats Shift Data

Valid entries	Usage
on-login	Enter **on-login** to clear shift data for an agent when the agent logs in.
at-midnight	Enter **at-midnight** to clear shift data for all agents at midnight.

Remove Inactive BCMS/VuStats Agents

Valid entries	Usage
y	Agents are removed from reports when they have no staff time during the previous 7 days.
n	Agents remain on the report even if they have no staff time for any period of time.

Reporting Adjunct Release

Valid entries	Usage
R3V6 .	Enter the release of the management and reporting adjunct in use. For CMS, this field cannot be blank.
R3V8	
R3V9	
R3V11	
R12	
R13	
R13.1	
3.1	
4.0	

Validate BCMS/VuStats Login Ids

Valid entries	Usage
y	Enter **y** to allow entry only of login-IDs that have been entered on the **BCMS Login-ID** screen.
n	Enter **n** to allow entry of any ACD login of the proper length.

Field descriptions for page 13

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            FEATURE-RELATED SYSTEM PARAMETERS

CALL CENTER MISCELLANEOUS
                                          Clear Callr-info:
                        Allow Ringer-off with Auto-Answer?
```

Feature-Related System Parameters screen

Allow Ringer-off with Auto-Answer

Valid entries	Usage
y/n	Enter **y** to allow a user to use the **ringer-off** feature button to prevent ringing on EAS auto-answer calls.

Clear Callr-info

Use this field to specify when the collected digits Callr-Infor display is to be removed from the agent/station display.

Valid entries	Usage
leave-ACW	Leaves the display up while the agent is in ACW (After-call work mode).
next-call	Clears the display when the next call is received. This is the default.
on-call-release	Clears the display on the 2nd line of a two-line display as soon as the call is released, either because of receiving call disconnect or the agent/ station user pressing the release button.

Field descriptions for page 14

```
change system-parameters features                                  page 14 of x

            FEATURE-RELATED SYSTEM PARAMETERS

REASON CODES
                                Aux Work Reason Code Type:
                                  Logoff Reason Code Type:
                        Two-Digit Aux Work Reason Codes?:

REDIRECTION ON IP CONNECTIVITY FAILURE

                            Switch Hook Query Response Timeout:
                        Auto-answer IP Failure AUX Reason Code:

MAXIMUM AGENT OCCUPANCY PARAMETERS
                            Maximum Agent Occupancy Percentage:
                       Maximum Agent Occupancy AUX Reason Code:

FORCED AGENT LOGOUT PARAMETERS
              Maxiumum Time Agent in ACW before Logout (sec.):
                                  ACW Forced Logout Reason Code:
```

Feature-Related System Parameters screen

REASON CODES

Aux Work Reason Code Type

Valid entries	Usage
none	Enter **none** if you do not want an agent to enter a Reason Code when entering AUX work.
requested	Enter **requested** if you want an agent to enter a Reason Code when entering AUX mode but do not want to force the agent to do so. To enter **requested** the **Reason Codes** and **EAS** on the **System-Parameters Customer-Options** screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the **Reason Codes** and **EAS** on the **System-Parameters Customer-Options** screen must be **y**.

Logout Reason Code Type

Valid entries	Usage
none	Enter **none** if you do not want an agent to enter a Reason Code when logging out.
requested	Enter **requested** if you want an agent to enter a Reason Code when logging out but do not want to force the agent to do so. To enter **requested** the **Reason Codes** and **EAS** on the **System-Parameters Customer-Options** screen must be **y**.
forced	Enter **forced** to force an agent to enter a Reason Code when logging out. Enter **forced** to force an agent to enter a Reason Code when entering AUX mode. To enter **forced**, the **Reason Codes** and **EAS** on the **System-Parameters Customer-Options** screen must be **y**.

Two-Digit Aux Work Reason Codes

Valid entries	Usage
y/n	Enter **y** to enable two-digit Reason Codes for agent state changes to Aux Work. Default is **n**.

REDIRECTION ON IP CONNECTIVITY FAILURE

Switch Hook Query Response Timeout

Valid entries	Usage
500 to **5000** (msec)	Assign the time on a system basis that the call processing will wait for a response to the switch hook query before Return on IP Connectivity Failure (ROIF) is triggered.

Auto-answer IP Failure AUX Reason Code

Valid entries	Usage
0 to **99**	Enter the reason code assigned for auto-answer IP failure, as the reason the agent was put into Aux Work.

MAXIMUM AGENT OCCUPANCY PARAMETERS

The Maximum Agent Occupancy (MAO) threshold is a system-administered value that is applied across all administered agents and is based on the total percentage of agent time in call service. MAO data is derived from the same calculations that are used to derive Least Occupied Agent (LOA). When an agent who exceeds the specified MAO threshold attempts to become available, he or she is automatically placed in AUX mode for the reason code administered for this purpose. When the occupancy for such pending agents drops below the MAO, they are released from AUX mode and made available. To use MAO, Expert Agent Selection (EAS) must be enabled.

Maximum Agent Occupancy Percentage

Valid entries	Usage
0 to **100**	Enter the percentage for MAO. Default is **100**.

Maximum Agent Occupancy AUX Reason Code

Valid entries	Usage
0 to **99**	Enter a reason code value. Default is **9**. A different reason code can be used for this purpose, but Avaya recommends that you do *not* use reason code 0.

FORCED AGENT LOGOUT PARAMETERS

Maxiumum Time Agent in ACW before Logout (sec.)

This field is used for setting a maximum time the agent can be in ACW on a per system basis. You can only change the default if **Expert Agent Selection (EAS) enabled?** is set to **y** on the **Feature-Related System Parameters** screen, and the **Call Center Release** field on the **System-Parameters Customer-Options** screen is set to **3.0** or later. When this timer expires, the agent is logged out. This system option applies only to EAS configurations.

Valid entries	Usage
30 to **9999** or blank	Indicate the maximum time an agent can be in ACW before being automatically logged out. Default is blank, meaning no timeout.

ACW Forced Logout Reason Code

This field is used to specify the reason for logging out the agent due to timeout in ACW when the Reason Codes feature is active. You can only change the default if, on the **System-Parameters Customer-Options** screen, **Reason Codes** is set to **y**, and the **Call Center Release** field is set to 3.0 or later. Additionally, the **Expert Agent Selection (EAS) enabled?** field on the **Feature-Related System Parameters** screen must be set to **y**.

Valid entries	Usage
0 to **9**	Enter a reason code value. Default is **0**.

Field descriptions for page 15

```
change system-parameters features                                Page 15 of x

                FEATURE-RELATED SYSTEM PARAMETERS
SPECIAL TONE
                                    Special Dial Tone? y
```

Feature-Related System Parameters screen

Special Dial Tone

Valid entries	Usage
y/n	Default is **n**.

Field descriptions for page 16

```
change system-parameters features                                    Page 16 of
x

              FEATURE-RELATED SYSTEM PARAMETERS

AUTOMATIC EXCLUSION PARAMETERS

                                     Automatic Exclusion by COS? y
                                Automatic Exclusion Coverage/Hold? y
                           Automatic Exclusion with Whisper Page? y
                                          Recall Rotary Digit: 2
                                  Password to Change COR by FAC: *
                       Duration of Call Timer Display (seconds): 3

WIRELESS PARAMETERS
       Radio Controllers with Download Server Permission (enter board
location)
         1.          2.          3.          4.          5.

IP PARAMETERS
                                 Direct IP-IP Audio Connections? n
                                            IP Audio Hairpinning? n
RUSSIAN MULTI-FREQUENCY PACKET SIGNALING
                                                        Re-try?
         T2 (Backward Signal) Activation Timer (secs):
```

Feature-Related System Parameters screen

AUTOMATIC EXCLUSION PARAMETERS

Automatic Exclusion by COS

Activates automatic exclusion automatically by class of service when a user goes off-hook on a station with an assigned **Exclusion** button. This works only for stations on the local server running Communication Manager.

Valid entries	Usage
y	Enables automatic exclusion by a class of service.
n	Exclusion operates normally.

Automatic Exclusion Coverage/Hold

Appears when the **Automatic Exclusion by COS** field is **y**.

Valid entries	Usage
y	The principal can bridge onto the call by pressing the appropriate bridged appearance button. And, if the coverage point places the exclusion call on hold, the principal can retrieve the call.
n	If a coverage point has answered a call and there is active exclusion on he call, the principal cannot bridge onto the call. And, if the coverage point places the exclusion call on hold, the principal cannot retrieve the call.

Automatic Exclusion with Whisper Page

Appears when the **Automatic Exclusion by COS** field is **y**.

Valid entries	Usage
y	The whisper page goes through to an excluded call.
n	The whisper page is denied when a station attempts to whisper page to a station that is on an excluded call.

Duration of Call Timer Display

Administer a call timer button on the **Station** screen.

Valid entries	Usage
3 to **30**	Enter the length of time (in 3 second increments) that the call information remains on display after the call is terminated.

Password to Change COR by FAC

Appears if, on the **System-Parameters Customer-Options** screen, the **Change COR by FAC** field is **y**.

Valid entries	Usage
4 to 8 digits	Requires the password option. blank Disables the password option.

Recall Rotary Digit

This establishes the digit to use for rotary telephones to receive recall dial tone. Dialing this digit simulates switch-hook flash so that users of rotary telephones can use features such as conference and transfer. The telephone must also be administered to use the recall rotary digit.

Valid entries	Usage
0 to **9**	Enter the digit users can dial to generate recall dial tone. Use a number that is not the first digit in normal dialing patterns.

WIRELESS PARAMETERS

Radio Controllers with Download Server Permission

Enter the necessary characters for the port location of the circuit pack containing the radio controllers with download server permission.

Valid entries	Usage
01 to **03** (DEFINITY CSI) or **1** to **64** (S8700/S8300 Media Servers)	First and second characters are the cabinet number
A to **E**	Third character is the carrier
0 to **20**	Fourth and fifth characters are the slot number

IP PARAMETERS

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter to **y** to save on bandwidth resources and improve sound quality of voice over IP transmissions.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack in the server.

Valid entries	Usage
y/n	Enter **y** to allow IP endpoints to be connected through the IP circuit pack in the media server in IP format, without going through the Avaya DEFINITY TDM bus. Default is **n**.

RUSSIAN MULTI-FREQUENCY PACKET SIGNALING

Re-try

The **Re-try** field applies to outgoing Russian MFP trunks. It allows the server running Communication Manager to resend Russian MFP calling party number and dialed number information to the CO. The server resends the information only once over another outgoing trunk port of the same trunk group if Communication Manager receives a message that the information was received incorrectly by the CO. The switch also sends Russian MFP information over another trunk port if Communication Manager does not receive a timely response for the information.

Valid entries	Usage
y/n	Enter **y** to resend address information on outgoing Russian MFP trunks.

T2 (Backward Signal) Activation Timer (secs)

The **T2 (Backward Signal) Activation Timer (secs)** field applies to outgoing Russian MFP trunks. This field sets the number of seconds that Communication Manager waits for confirmation after sending calling party number and dialed number information on outgoing Russian MFP trunks

.

Valid entries	Usage
5 to **20**	Enter the number of seconds the system waits to receive confirmation after sending the address information on outgoing Russian MFP trunks.

Field descriptions for page 17

```
change system-parameters features                                  page 17 of x

                FEATURE-RELATED SYSTEM PARAMETERS


INTERCEPT TREATMENT PARAMETERS
Invalid Number Dialed Intercept Treatment: announcement 7700
Invalid Number Dialed Display: Invalid Number
Restricted Number Dialed Intercept Treatment: announcement 7701
Restricted Number Dialed Display: Restricted No.
Intercept Treatment On Failed Trunk Transfers? N

WHISPER PAGE
Whisper Page Tone Given To: all

DIGITAL STATION LINE APPEARANCE LED SETTINGS
                         Station Putting Call On Hold: green wink
                          Station When Call is Active: green solid
               Other Stations When Call Is Put On Hold:
                    Other Stations When Call Is Active:
                                             Ringing:
                                                Idle:
                 Display Information With Bridged Call?
                                   Pickup On Transfer?
```

Feature-Related System Parameters screen

INTERCEPT TREATMENT PARAMETERS

Invalid Number Dialed Intercept Treatment

Enter the type of intercept treatment the end-user hears after dialing an invalid number.

Valid entries	Usage
announcement	Provides a recorded announcement when the end-user dials an invalid number. You select and record the message. Enter the extension number for the announcement in the associated field.
tone	Provides intercept tone when the end-user dials an invalid number. This is the default.

Invalid Number Dialed Display

This field shows a name in either Latin or Asian characters for an invalid number calling in.

Valid entries	Usage
Letters, spaces, numerals, and special characters.; maximum 15 characters	This field supports both a NAME1 and a NAME2 value. A NAME1 value directs the system to use the table of names that contains Latin characters, which can be displayed. Type a value of NAME2 to direct the system to use the UTF-8 table of names, which contains non-ASCII characters suitable for Asian language names.

Restricted Number Dialed Intercept Treatment

This field controls whether an announcement or an intercept tone is played when an end-user dials an number restricted from them due to COS, COR, or FRL restrictions. Enter the type of intercept treatment the end-user hears after dialing a restricted number.

Valid entries	Usage
tone	Provides intercept tone when the end-user dials an restricted number. This is the default.
announcement	Provides a recorded announcement when the end-user dials a restricted number. You select and record the message. Enter the extension number for the announcement in the associated field.

Restricted Number Dialed Display

This field controls whether the system displays any string of alphanumeric characters assigned for calls that are denied because of COS/COR, or FRL restrictions.

Valid entries	Usage
Letters, spaces, numerals, and special characters.; maximum 15 characters	This field supports both a NAME1 and a NAME2 value. A NAME1 value directs the system to use the table of names that contains Latin characters, which can be displayed. Type a value of NAME2 to direct the system to use the UTF-8 table of names, which contains non-ASCII characters suitable for Asian language names.

Intercept Treatment on Failed Trunk Transfers

Valid entries	Usage
y	Enter **y** to provide intercept treatment to calls failing trunk transfers.
n	Enter **n** to drop these calls.

WHISPER PAGE

Whisper Page Tone Given To

Use this field to indicate who should hear a Whisper Page.

Valid entries	Usage
all	All parties hear the Whisper Page.
paged	The whisper page feature sends a beep to the paging and the paged party.

DIGITAL STATION LINE APPEARANCE LED SETTINGS

WARNING:

The following fields only change the LED operation for 84xx and 64xx model telephones. When the LED operation is changed using any of these fields, then IP Agent and IP Softphone using a station type of 84xx or 64xx does not work. For station types other than 84xx or 64xx, a change to the LEDs using these fields does not affect either IP Agent or IP Softphone.

Note:

The system generates a warning if the default values of the LED Settings field are changed. The warning message states "WARNING: Avaya Softphone will not operate correctly if this value is changed." You will see this warning message if you are running Avaya Communication Manager 3.1 or higher.

Station Putting Call On Hold

Use this field to control the LED color and flash rate on the 8400 and 6400 series telephones for a call held on a Primary or Bridged Appearance. The LED for the color not selected is turned OFF. The default values are **green** and **wink**.

Valid entries	Usage
green or **red**	Indicate whether the LED is green or red.
Off **wink** **inverse-wink** **flash** **flutter** **broken-flutter** **steady**	Select the flash rate for a call on hold.

Station When Call is Active

Use this field to control the red LED on the 8400 and 6400 series telephones, for a station active on a call. The default value is **steady**.

Valid entries	Usage
steady	When the value is steady, Communication Manager controls the red LED.
off	When the value is off, the red LED is always OFF.

Other Stations When Call Is Put On Hold

Use this field to control LED options for the other stations with a Bridged Appearance that has been placed on hold (e.g. the user of this station has not pushed the hold button). The default values are **green** and **wink**.

Note:

This field is for a DCP bridged appearance LED color and flash rate when a call on a bridged appearance is put on hold by another party on the DCP bridged appearance. Additionally, this field only applies to 8400 and 6400 series telephones. The 2400 series phone uses icons rather then LEDs. Correctoperation in the Japanese environment requires the administrator to select the values **red** and **flash** for this field.

Valid entries	Usage
green or **red**	Indicate the color of the LED. Default is **green**.
Off **wink** **inverse-wink** **flash** **flutter** **broken-flutter** **steady**	Select the flash rate for the LED. Default is **wink**.

Other Stations When Call Is Active

Use this field to control a DCP bridged appearance LED for those non-active parties with a bridged appearance that is active. The default value is **green**.

Note:
This field only applies to 8400 and 6400 series telephones. The 2400 series phone uses ICONs rather then LEDs. Correct operation in the Japanese environment requires the administrator to select the value **red** for this field.

Valid entries	Usage
green or **red**	Select the LED color. Default is **green**.

Ringing

Use this field to control the LED color and flash rate while a call is ringing.

Note:
This field only applies to 8400 and 6400 series telephones. The 2400 series phone uses icons rather then LEDs. Correct operation in the Japanese environment requires the administrator to select the values **red** and **wink** for this field. The default values are **green** and **flash**.

Valid entries	Usage
green or **red**	Indicate the LED color.
Off **wink** **inverse-wink** **flash** **flutter** **broken-flutter** **steady**	Indicate the flash rate.

Idle

Use this field to control the LED of a station that is idle. The default value is **steady**.

Note:
This field only applies to 8400 and 6400 series telephones. The 2400 series phone uses icons rather then LEDs. This value controls the red LED. Correct operation in the Japanese environment requires the administrator to select the value **off** for this field.

Valid entries	Usage
steady	LED is on. This is the default.
off	LED is off.

Display Information With Bridged Call

Use this field to control whether or not name and number for a bridged call are displayed on the telephone of the called party. A **y** entry indicates that the information is to be displayed; this field does not in any way control the content of the display.

Valid entries	Usage
y/n	Type **y** to display the name and number for an incoming call to the bridged appearance. Default value is **n**.

System Parameters Call Coverage / Call Forwarding

Field descriptions for page 1

```
change system-parameters coverage-forwarding (page 1)

   SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

CALL COVERAGE/FORWARDING PARAMETERS
          Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings): _
        Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings): _
                          Coverage - Caller Response Interval (seconds): _
    Threshold for Blocking Off-Net Redirection of Incoming Trunks Calls: 1

COVERAGE
                                     Keep Held SBA at Coverage Point? _
       External Coverage Treatment for Transferred Incoming Trunk Calls? _
        Immediate Redirection on Receipt of PROGRESS Inband Information? _
                                         Maintain SBA At Principal? _
            QSIG VALU Coverage Overrides QSIG Diversion with Rerouting? _
                                    Station Hunt Before Coverage? n

FORWARDING
                                             Call Forward Override? _
                                           Coverage After Forwarding? _
```

System-Parameters — Call Coverage / Call Forwarding

Local Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

This field specifies:

- the number of rings applied at a local coverage point before a call redirects to the next coverage point
- the number of rings applied at the principal before a call forwards when Call Forwarding Busy/Don't Answer is activated.

Off-Net Cvg Subsequent Redirection/CFWD No Ans Interval (rings)

This field specifies:

- the number of rings applied at an off-net coverage point before a call is redirected to the next coverage point
- the number of rings applied at an off-net forwarded-to destination before the call is redirected to coverage.

NOTE:
When ringing local destinations (say in an office environment), a short interval often is appropriate because the intended party either is near the phone or not present. However, when ringing off-net locations, you cannot assume how near the intended party is to the phone. If the call is left at an off-net destination for only a short interval, the call may be redirected to the next destination before the intended party has any real chance of answering the call.

Coverage - Caller Response Interval (seconds)

The time in seconds the caller (internal caller only) has before a call redirects to the called party's first coverage point. The calling party either can hang up, use Leave Word Calling, or press the GO TO COVER button during this time interval.

Valid entries

0 through **10**

Threshold for Blocking Off-Net Redirection of incoming Trunk Calls

This field applies for those occasions when an incoming call to a station redirects off-net. At that time, the Call Forward timer activates to block any further incoming calls to that station from being redirected off-net until the timer expires.

Valid entries	Usage
1-7	The number of allowed calls to be routed off-net before blocking commences.
All	Call processing never activates the Call Forward timer. Therefore, any number of calls to a principal may be redirected off-net.

Keep Held SBA at Coverage Point

This field governs how a covering user who has placed an answered coverage call on hold is treated if the original principal bridges onto the call.

Valid entries	Usage
y	Keeps the coverage party on the call. The coverage party remains on hold, but may enter the call along with the principal and the calling party.
N	Drops the coverage party from the call.

External Treatment for Transferred Incoming Trunk Calls

This field governs how an transferred incoming trunk call is handled if the call redirects to coverage.

Valid entries	Usage
Y	Enter y to allow external coverage treatment for incoming trunk calls that redirect to coverage.
N	Enter n to allow internal coverage treatment for incoming trunk calls that redirect to coverage

Immediate Redirection on Receipt of PROGRESS Inband Information

This field appears only if one of the following is true:

- The Coverage of Calls Redirected Off-Net Enabled field on the System Parameters Coverage/Forwarding screen is **y**.
- The Value-Added Avaya (VALU) field on the System Parameters Customer Options, Page 6, screen is **y**.

This field pertains only to CCRON and QSIG VALU coverage calls redirected over end-to-end ISDN facilities. Some cellular phone providers send an ISDN PROGRESS message with the Progress Indicator field set to 'inband information' when a cellular phone is unavailable to receive a call. In these circumstances, the message indicates that an announcement is being played to the originating party and we should move the call immediately to the next available coverage point.

However, a PROGRESS message with a Progress Indicator of 'inband information' may be received for reasons other than an unavailable cellular phone. In this case, you probably do not want to redirect the call to the next coverage point.

There is no way for the DEFINITY ECS to know the actual intent of such a PROGRESS message, yet you may choose how the system should handle this message. It is essentially an educated, but blind, choice and you should be aware that there will be instances when this choice leads to inappropriate call handling.

DEFINITY ECS queries this field on receipt of a qualifying PROGRESS message and acts according to your instruction on how to treat it.

As a guide, users in European countries following the ETSI standard and redirecting to GSM cellular phones may want to consider setting this field to **y**.

In the United States, PROGRESS messages with the Progress Indicator field set to 'inband information' are sent for a variety of reasons not associated with unavailable cell phones and you should set this field to **n**.

Valid entries	Usage
y	Immediately redirect an off-net coverage/forwarded call to the next coverage point.
n	Do not immediately redirect an off-net coverage/forwarded call to the next coverage point.

Maintain SBA At Principal

Allows a user to maintain a simulated bridged appearance when a call redirects to coverage.

Valid entries	Usage
y	Enter y to maintain a simulated bridged appearance (SBA) on the principal's phone when a call redirects to coverage. DCS with rerouting will not be attempted after coverage.
n	When set to n, no SBA is maintained on the principal's phone. DCS with rerouting will be attempted, and if successful, the principal will lose the bridged appearance and the ability to bridge onto the coverage call.

QSIG VALU Coverage Overrides QSIG Diversion with Rerouting

This field specifies whether, with both QSIG Diversion with Rerouting and QSIG VALU turned on, the Coverage After Forwarding option on the Station screen will work for a user for calls that go to remote coverage. Normally, with QSIG Diversion with Rerouting turned on, the local system passes control of a forwarded call to the remote QSIG switch on which the forwarding destination resides. In this case, the forwarded call cannot return to coverage for the user who originally received the call, even when the remote destination is busy or does not answer.

However, you can enter **y** in this field to have QSIG VALU call coverage take precedence. In this case, if the Coverage After Forwarding option on the Station screen is enabled for a user, then QSIG Diversion with Rerouting is not be attempted.

Valid entries	Usage
y/n	Enter **y** to allow Coverage After Forwarding to work when it is activated on a user's Station screen and Diversion with Rerouting is also turned on.

Station Hunt Before Coverage

This field allows you to choose whether a call to a busy station performs station hunting before going to coverage.

Valid entries	Usage
y/n	Enter **y** to use Station Hunt Before Coverage.

Call Forward Override

This field specifies how to treat a call from a forwarded-to party to the forwarded-from party.

Valid entries	Usage
y	Overrides the Call Forwarding feature by allowing a forwarded-to station to complete a call to the forwarded-from station
n	Directs the system to forward calls to a station even when they are from the forwarded-to party.

Coverage After Forwarding

This field governs whether an unanswered forwarded call is provided coverage treatment.

Valid entries	Usage
y	Coverage treatment is provided to unanswered forwarded calls.
n	No coverage treatment is provided to unanswered forwarded calls. The call remains at the forwarded-to destination.

Field descriptions for page 2

```
change system-parameters coverage-forwarding  (page 2)

          SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)
                          Coverage Of Calls Redirected Off-Net Enabled? y
     Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point? y
                                     Ignore Network Answer Supervision? y
                   Disable call classifier for CCRON over ISDN trunks? n
```

Coverage Of Calls Redirected Off-Net Enabled

This field allows you to enable/disable the Coverage of Calls Redirected Off-Net (CCRON) feature. This field provides a quick means of disabling this feature if the demand on the call classifier port resources degrades other services provided by the switch. The Coverage of Calls Redirected Off-Net field on this screen must be **y** to administer this field.

Valid entries	Usage
y	DEFINITY ECS monitors off-net coverage/forwarded calls and provides further coverage treatment for unanswered calls.
n	DEFINITY ECS does not monitor offnet coverage/forwarded calls. No further coverage treatment is provided if such calls are unanswered.

Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point

This field appears only if the Coverage of Calls Redirected Off-Net Enabled field on this screen is **y**.

When the system redirects a call off-net at the final coverage point in a coverage path, the system can apply no further coverage treatment even if the call is unanswered. The only reason for activating answer detection on such a call is to maintain the simulated bridged appearance (SBA) on the principal's phone that allows the principal to answer or bridge onto the call. However, when the system monitors the call through use of a call classifier port, there is an inherent cut-through delay following the detection of answer at the far end. This field has no consequence when the off-net call is carried end-to-end by ISDN facilities; the SBA is maintained and there is no cut-through delay.

Valid entries	Usage
y	Directs the system to maintain a simulated bridged appearance on the principal when redirecting to a final off-net coverage point.
n	Allows the system to drop the SBA on the principal's phone when the call redirects off-net at the last coverage point, eliminating the cut-through delay inherent in CCRON calls, but sacrificing the principal's ability to answer the call.

Ignore Network Answer Supervision

This field appears only if the Coverage of Calls Redirected Off-Net Enabled field on this screen is **y**.

CCRON may use a call classifier port to determine whether an off-net coverage or forwarded call has been answered, discarding other information that may indicate an answered state. However, some customers pay the operating company to provide network answer supervision on their trunks and desire that CCRON not discard that information. They may preserve this service by setting this field to **n**.

On the other hand, beware when you tandem a call over a tie trunk through another switch node from where it redirects to the public network over non-ISDN facilities. If the trunk on the far-end node sends a timed answer supervision, that may get tandemed back to the originating switch as a network answer. In such a scenario, the originating switch interprets the call as answered, leading to some undesirable behavior. To avoid these calls from mistakenly be treated as answered, set this field to **y**. An unfortunate consequence is that a short cut-through delay that is inherent to call classification is introduced when the call is answered.

Valid entries	Usage
y	Ignore network answer supervision and rely on the call classifier to determine when a call is answered.
n	Treat network answer supervision as a true answer.

Disable call classifier for CCRON over ISDN trunks

When a CCRON call routes offnet over ISDN end-to-end facilities, no call classifier is attached to the call. If, subsequently during the call, an ISDN PROGRESS or ALERT message is received that indicates that interworking has occurred, a call classifier is normally attached to the call and assumes precedences over ISDN trunk signalling. This field provides a customer the means of directing the switch to dispense with the call classifier on interworked calls and rely on the ISDN trunk signalling messages.

Valid entries	Usage
y	Use **y** to disable the call classifier for CCRON calls over interworked trunk facilities.
n	Use **n** to enable the call classifier for CCRON calls over interworked trunk facilities.

Understanding the Dial Plan (v10 and above)

Your dial plan tells your system how to interpret dialed digits. For example, if you dial 9 on your system to access an outside line, it is actually the dial plan that tells the system to find an external trunk when a dialed string begins with a 9.

The dial plan also tells the system how many digits to expect for certain calls. For example, the dial plan might indicate that all internal extensions are 4-digit numbers that start with 1 or 2. Let us take a look at an example dial plan so you'll know how to read your system's dial plan.
The following figure shows an example of a simple dial plan.

Dial Plan Analysis Table screen

```
display dialplan analysis                                    Page 1 of x

                    DIAL PLAN ANALYSIS TABLE

                                                        Percent Full: 7

Dialed Total Call                Dialed Total Call            Dialed Total
Call
String Length Type              String Length Type           String Length
Type
00     2      attd
1      3      dac
2      4      ext
3      4      ext
3      1      aar
4      1      ars
4      5      ext
5      5      ext
5      7      ext
6      5      ext
7210   7      ext
8      7      ext
9      1      fac
*      3      fac
#      3      fac
```

The **Dial Plan Analysis Table** defines the dialing plan for your system.
The **Call Type** column in the **Dial Plan Analysis Table** indicates what the system does when a user dials the digit or digits indicated in the **Dialed String** column. The **Total Length** column indicates how long the dialed string will be for each type of call. For example, this dial plan shows that when users dial a 5-digit number that starts with 3, they are dialing an extension.

The **Dial Plan Analysis Table** in our example contains the following call types:

• Attendant (attd) — Defines how users call an attendant. Attendant access numbers can be any number from 0 to 9 and contain 1 or 2 digits.

In our example figure, the system calls an attendant when users dial 0.

• Dial access code — Allows you to use trunk access codes (TAC) and feature access codes (FAC) in the same range. For example, you could define the group 100 to 199, which would allow both FAC and TAC in that range. Dial access codes can start with any number from 1 to 9, * and #, and contain up to 4 digits.

In our example figure, dial access codes begin with 1 and must be 3 digits long.

Note:

The **Dial Plan Analysis Table** does not allow you to enter a range specifically for trunk access codes. However, the **Trunk Group** screen still allows you to assign a TAC to a trunk group. The TAC you enter on the **Trunk Group** screen must match the format you have administered for a DAC on the **Dial Plan Analysis Table**.

• Extensions (ext) — Defines extension ranges that can be used on your system. In our figure, extensions must be in the ranges 30000 to 39999, 40000 to 49999 and 50000 to 59999.

• Feature access codes (fac) only — FAC can be any number from 1 to 9 and contain up to 4 digits. You can use * or #, but only as a first digit. In our example, feature access codes can begin with * or # and are 3-digits long.

The **Dial Plan Analysis Table** works with the **Dial Plan Parameters Table** for fully defining your dial plan. The **Dial Plan Parameters Table** allows you to set system-wide parameters foryour dial plan.

Modifying your dial plan

It is easy to make changes to your dial plan. For example, we will add a new range of dial access codes to the dial plan. We want to be able to assign both FAC and TAC in the 700 to 799 range.

1. Type **`change dialplan analysis`**. Press **Enter**.
   The **Dial Plan Analysis** appears.

2. Move the cursor to an empty row.

3. Type **7** in the **Dialed String** column. Press **Tab** to move to the next field.

4. Type **3** in the **Total Length** column. Press **Tab** to move to the next field.

Adding extension ranges

You might find that as your needs grow you want a new set of extensions. Before you can assign a station to an extension, the extension must belong to a range that is defined in the dial plan. We will add a new set of extensions that start with 3 and are 4 digits long (3000 to 3999).

To add this set of extensions to the dial plan:

1. Type **`change dialplan analysis`**. Press **Enter**.
   The **Dial Plan Analysis Table** screen appears.

2. Move the cursor to an empty row.

3. Type **3** in the **Dialed String** column. Press **Tab** to move to the next field.

4. Type **4** in the **Total Length** column. Press **Tab** to move to the next field.

5. Type **ext** in the **Call Type** column.

6. Press **Enter** to save your changes.

Understanding the dial plan (v9 and older)

Your dial plan tells your system how to interpret dialed digits. For example, if you dial 9 on your system to access an outside line, it is actually the dial plan that tells the system to find an external trunk when a dialed string begins with a 9.

The dial plan also tells the system how many digits to expect for certain calls. For example, the dial plan may indicate that all internal extensions are 4-digit numbers that start with 1 or 2

```
change dialplan                                                    Page 1 of 1
                                  DIAL PLAN RECORD
                                                      Local Node Number: _
                                                        ETA Node Number: _
          Uniform Dialing Plan: _____                 ETA Routing Pattern: _
    UDP Extension Search Order: ____________________
FIRST DIGIT TABLE
  First                                Length
  Digit   -1-        -2-        -3-        -4-        -5-        -6-
    1:  _________  _________  _________  _________  _________  _________
    2:  _________  _________  _________  _________  _________  _________
    3:  _________  _________  _________  _________  _________  _________
    4:  _________  _________  _________  _________  _________  _________
    5:  _________  _________  _________  _________  _________  _________
    6:  _________  _________  _________  _________  _________  _________
    7:  _________  _________  _________  _________  _________  _________
    8:  _________  _________  _________  _________  _________  _________
    9:  _________  _________  _________  _________  _________  _________
    0:  _________  _________  _________  _________  _________  _________
    *:  _________  _________  _________  _________  _________  _________
    #:  _________  _________  _________  _________  _________  _________
```

If you look at the lower half of the Dial Plan Record screen, you see the First Digit Table. This table defines the dialing plan for your system.

The rows in the First Digit Table indicate what the system does when the row's first digit is dialed. The columns indicate how long the dialed string will be for each type of call. For example, this dial plan shows that when users dial a 4-digit number that starts with 2, they are dialing an extension.

The first digit table may have any of the following codes:

- Attendant **(attd)** — Defines how users call an attendant. Attd access numbers can be any number from 0 to 9 and contain 1 or more digits. In our example figure, the system calls an attendant when users dial 0.

- Dial access codes **(dac)** — Allows you to use trunk access codes (TAC) and feature access codes (FAC) in the same range. For example, you could define the group 600–699 for DAC, which would allow both FAC and TAC in that range. Dial access codes can start with any number from 1 to 9 and contain up to 4digits, * and #. In our example figure, dial access codes begin with 6 and must be 3 digits long, so this company can have a feature access code set to 633 and a trunk access code assigned to 634.

- Extensions **(ext)** — Defines extension ranges that can be used on your system. In our figure, extensions must be in the ranges: 1000–1999, 2000–2999, and 5000–5999.

- Feature access codes **(fac)** only — FAC can be any number from 1 to 9 and contain up to 4 digits. You can use * or #, but only as a first digit. In our example, this company can use *21 to activate a feature and use #21 to deactivate the same feature. Our example also shows that one FAC can be set to 9 (first digit 9, only one digit long).

- Miscellaneous code **(misc)** — these codes are used if you want to have more than one kind of code start with the same digit. Using a misc code requires that you also define a second digit table.

Local Node Number

Enter a number to identify a specific node in a switch network. This entry must match the DCS switch node number and the CDR node number if they are specified.

ETA Node Number

Enter the number of the destination switch for Extended Trunk Access (ETA) calls. ETA calls are unrecognized numbers you can send to another switch for analysis and routing. Such numbers can be Facility Access Codes, Trunk Access Codes, or extensions that are not in the UDP table.

Uniform Dialing Plan

The Uniform Dialing Plan field must be y on the System-Parameters Customer-Option screen before you can administer this field.

The Uniform Dialing Plan is a separate screen that must be administered if **4-digit** or **5-digit** is entered in this field. The UDP provides a common 4- or 5-digit dial plan that can be shared among a group of switches. Additionally, UDP can be used alone to provide uniform 4- or 5-digit dialing between two or more private switching systems without ETN, DCS, or Main/Satellite/Tributary configurations.

ETA Routing Pattern

Enter the number of the routing pattern to reach the destination switch.

UDP Extension Search Order

Appears only when Uniform Dialing Plan is **4-digit** or **5-digit**. Specifies the first table to search to match a dialed extension.

First Digit Table

This table defines the dialing plan for your system. The rows in the First Digit Table indicate what the system does when the row's first digit is dialed. The columns indicate how long the dialed string will be for each type of call.

Second Digit Table

You must complete the Second Digit Table each time you enter **misc** in the digit length of 1 column on the Dial Plan Record screen. The second digit table is named for the row where the **misc** appears. In addition, a second digit table can exist for every first digit value.

```
change second-digit x                                          Page 1 of 1
                          SECOND DIGIT TABLE FOR DIGIT _
 SECOND DIGIT TABLE
 Digit  Identification   Number of        Digit  Identification   Number of
                         Digits                                   Digits
     0: _________         0                 5: _________           0
     1: _________         0                 6: _________           0
     2: _________         0                 7: _________           0
     3: _________         0                 8: _________           0
     4: _________         0                 9: _________           0
```

Second Digit Table

Digit

A display-only field that marks the value of the second digit.

Identification

Enter an abbreviation to identify the dialed number when a second digit is dialed. **(attd, extension, tac, fac, ars)**

Number of Digits

This field tells the system how many digits to collect if the first and second digits match. Enter the length of the dialed number.

Feature Access Codes

This screen assigns feature access codes (FACs) that, when dialed, activate or cancel the system features. Each field on this screen has the same valid values, which must conform to feature access codes or dial access codes as defined by your dial plan.

Valid entries	Usage
1–4 digit number, * #	* and # may be used as first digit. However, analog rotary dial phones cannot use the "*" and "#" symbols.

Field descriptions for page 1

```
change feature-access-codes                                          Page 1 of X
                                FEATURE ACCESS CODE (FAC)
          Abbreviated Dialing List1 Access Code: ____
          Abbreviated Dialing List2 Access Code: ____
          Abbreviated Dialing List3 Access Code: ____
   Abbreviated Dial - Prgm Group List Access Code: ____
                     Announcement Access Code: ____
                      Answer Back Access Code: ____
       Auto Alternate Routing (AAR) Access Code: ____
        Auto Route Selection (ARS) Access Code1: ____   Access Code 2: ____
                 Automatic Callback Activation: ____    Deactivation: ____
   Call Forwarding Activation Busy/DA: ____  All: ____  Deactivation: ____
                        Call Park Access Code: ____
                      Call Pickup Access Code: ____
   CAS Remote Hold/Answer Hold-Unhold Access Code: ____
                 CDR Account Code Access Code: ____
                       Change COR Access Code: ____
                  Change Coverage Access Code: ____
                 Data Origination Access Code: ____
                     Data Privacy Access Code: ____
            Directed Call Pickup Access Code: ____
      Emergency Access To Attendant Access Code: ____
```

Feature Access Code (FAC) screen

Abbreviated Dialing List1 Access Code

Used to access AD list 1.

Abbreviated Dialing List2 Access Code

Used to access AD list 2

.

Abbreviated Dialing List3 Access Code

Used to access AD list 3.

Abbreviated Dial - Prgm Group List Access Code

Used to enter a group list from a telephone. The user's extension must be entered on the Abbreviated Dial Group List screen in order to program the group list.

Announcement Access Code

Used to record announcements.

Answer Back Access Code

Used to retrieve parked calls.

Auto Alternate Routing (AAR) Access Code

Used to access AAR.

Auto Route Selection (ARS) Access Code1

Used to access ARS. You can have one ARS access code for local and one for long distance, and route accordingly.

(ARS) Access Code 2

Also used to access ARS.

Automatic Callback Activation/Deactivation

Used to activate/cancel Automatic Callback.

Call Forwarding Activation Busy/DA

Used to forward calls to an administered number if the user is busy or does not answer.

(Call Forwarding Activation) All

Used to forward calls to an administered number for all calls.

(Call Forwarding) Deactivation

Used to deactivate call forwarding.

Call Park Access Code

Used to park an active call, which can then be retrieved from a different station using the answer back access code. Do not administer to have the same first digit as another feature access code that is longer in length.

Call Pickup Access Code

Used to answer a call directed to a pickup group.

CAS Remote Hold/Answer Hold-Unhold Access Code

Used by a Centralized Attendant Service (CAS) attendant to place calls on hold and answer calls held at a remote switch.

CDR Account Code Access Code

Used prior to entering an account code for CDR purposes.

Change COR Access Code

Used to allow users to change their class of restriction (COR) from a phone. This field can only be used if the Change COR by FAC field is enabled on the System-Parameters Customer-Options screen.

Change Coverage Access Code

Used to change a coverage path from a telephone or remote station.

Data Origination Access Code

Used to originate a data call from a voice station.

Data Privacy Access Code

Used to isolate a data call from call waiting or other interruptions.

Directed Call Pickup Access Code

Used to establish directed call pickup.

Emergency Access To Attendant Access Code

Used to gain access to the attendant in an emergency situation. Such calls alert as emergency calls. This field cannot be used if the Emergency Access to Attendant field is not enabled on the System-Parameters Customer-Options screen.

Extended Call Fwd Activate Busy D/A

Used to activate call forwarding from a telephone or remote location.

Extended Call Fwd Activate All

Used to activate call forwarding from a telephone or remote location.

Extended Call Fwd Deactivation

Used to deactivate call forwarding from a telephone or remote location.

NOTE:
An extension must have Station Security Codes administered to use the following FACs:

- Extended Call Forward All Activate
- Extended Call Forward Busy/Don't Answer Activate
- Extended Call Forward Deactivate
- Change Coverage

Field descriptions for page 2

```
change feature-access-codes                                          Page  2  of X
                              FEATURE ACCESS CODE (FAC)
     Extended Call Fwd Activate Busy D/A_  All:        Deactivation:
        Extended Group Call Pickup Access Code:
                Facility Test Calls Access Code:
                             Flash Access Code:
             Group Control Restrict Activation:        Deactivation:
                    Hunt Group Busy Activation:        Deactivation:
                              ISDN Access Code:
                 Last Number Dialed Access Code:
    Leave Word Calling Message Retrieval Lock:
   Leave Word Calling Message Retrieval Unlock:
             Leave Word Calling Send A Message: #66
           Leave Word Calling Cancel A Message: *66
                Malicious Call Trace Activation:        Deactivation:
PASTE  (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:
        Per Call CPN Blocking Code Access Code:
      Per Call CPN Unblocking Code Access Code:        Dissociate Code:
                   Print Messages Access Code:
                 Priority Calling Access Code:
                          Program Access Code:
```

Extended Group Call Pickup Access Code

The feature access code (FAC) users enter when a call directed to another pickup group is to be answered. Users must enter a valid "Pickup Number" following the Extended Group Call Pickup Access Code to complete the operation.

Facility Test Calls Access Code

Used to place activate a facility test call.

SECURITY ALERT:
To ensure the security of your system, leave Facility Test Calls Access Code blank except when actually testing trunks.

Flash Access Code

Used to generate trunk flash. This code ensures that the flash signal is interpreted by the central office switch, rather than the DEFINITY ECS.

Group Control Restrict Activation / Deactivation

Used to change the restriction level for all users with a given class of restriction. Requires console permissions.

Hunt Group Busy Activation/Deactivation

Hunt Group members can dial these codes to place themselves in a busy state, and to become available again.

ISDN Access Code

Used to place an ISDN call without using ARS, AAR, or UDP.

Last Number Dialed Access Code

Used to redial the last number dialed from this station.

Leave Word Calling Message Retrieval Lock

Used to lock the display module on telephones. The lock function activates at a telephone by dialing this system-wide lock access code. This prevents unauthorized users from displaying, canceling, or deleting messages associated with the telephone. The Lock Messages field on the Station screen also must be enabled.

Leave Word Calling Message Retrieval Unlock

Used to unlock a telephone's display module. The lock function is canceled at the telephone by dialing this unlock FAC followed by the SCC.

Leave Word Calling Send A Message

Used to send a leave word calling message.

Leave Word Calling Cancel A Message

Used to cancel a leave word calling message.

Malicious Call Trace Activation

Used to activate a trace request on a malicious call.

PASTE (Display PBX data on Phone) Access Code

Allows users to view call center data on display phones. PASTE is used in conjunction with Avaya IP Agent.

Personal Station Access (PSA) Associate Code

Used to associate a telephone with the phone features assigned to a user's extension.

Per Call CPN Unblocking Code Access Code

If CPN blocking is on for a trunk group, users can turn it off for a call by using this code. When they dial this code, the calling party number is sent to the public network.

Dissociate Code

Used to remove the association between a physical phone and an extension number. You cannot provide the code until Personal Station Access (PSA) on the System Parameters Customer-Options screen is **y**.

Print Messages Access Code

Allows users to print undelivered messages without having to call the message center.

Program Access Code

Used to program abbreviated dial buttons on an individual phone.

Refresh Terminal Parameters Access Code

Used to update terminal parameters on an individual phone when system settings have changed.

Remote Send All Calls Activation/Deactivation

Used to activate or deactivate the Send All Calls feature. Requires console permissions.

Self Station Display Activation

The self station field is not active. If set to a valid FAC, a digital station displays its primary extension number when the FAC is entered.

Send All Calls Activation/Deactivation

Used to activate or deactivate sending all calls to coverage with minimal or no alerting at the station.

Field descriptions for page 3

```
                                                          Page 3 of X
                        FEATURE ACCESS CODE (FAC)

      Refresh Terminal Parameters Access Code:
             Remote Send All Calls Activation:       Deactivation:
             Self Station Display Access Code:
                    Send All Calls Activation:       Deactivation:
                      Station Lock Activation:       Deactivation:
    Station Security Code Change Access Code:
                         Telephone Activation: #*
            Terminal Dial-up Test Access Code:
Terminal Translation Initialization Merge Code:      Separation Code:
           Transfer to Voice Mail Access Code:
          Trunk Answer Any Station Access Code:
             User Control Restrict Activation:       Deactivation:
  Voice Coverage Message Retrieval Access Code:
```

Station Lock Activation/Deactivation

Used to activate or deactivate Station Lock.

Station Security Code Change Access Code

Enter the code the user must dial to change their Station Security Code. The SCC must be administered before the user can change it using this FAC. That is, a user cannot change a blank SCC.

Terminal Dial-Up Test Access Code

Used to perform tests on digital telephones to make sure that the telephone and the buttons are communicating properly with the switch. See your Maintenance documentation for information about Digital Terminal Remote Looparound Test.

Terminal Translation Initialization Merge Code

Enter the digits that must be dialed to install (merge) a station without losing any of its previous feature settings. The Terminal Translation Initialization Separation Code must have been used, or an X administered in the Port field of the Station screen, when the telephone was removed from its former location in order for the Terminal Translation Initialization Merge Code to be effective. (If you try to use this and it doesn't work, check the station screen for this extension. If there is still a port assigned, type X in the port field, then try the TTI merge again.)

Terminal Translation Initialization Separation Code

Enter the digits that must be dialed to remove (separate) a station from a location without losing any of its feature settings.

Transfer to Voice Mail Access Code

Enter the digits that must be dialed to allow coverage to transfer the caller to the original call recipient's AUDIX mail where the caller can leave a message. Do not administer this code to have the same first digit as another feature access code that is longer in length.

Trunk Answer Any Station Access Code

Enter the access code that station users must dial to answer calls alerting on night bells.

User Control Restrict Activation/Deactivation

Used to change the restriction level for a specific extension. Requires console permissions.

Voice Coverage Message Retrieval Access Code

Allows users to retrieve voice messages for another user (for whom they are a coverage point) via a digital display module.

Voice Principal Message Retrieval Access Code

Allows users to retrieve their own voice messages for another user via a digital display module.

Whisper Page Activation Access Code

Allows users to place a page to another user's phone, when active on a call. The paged user, and not the other parties on the call, hears the page.

Field descriptions for page 4

The feature access codes on this page pertain only to ACD call centers.

```
change feature-access-codes                                          Page 4 of X
                        FEATURE ACCESS CODE (FAC)

                  Automatic Call Distribution Features

                   After Call Work Access Code: ___
                            Assist Access Code: ___
                          Auto-In Access Code: ___
                         Aux Work Access Code: ___
                            Login Access Code: ___
                           Logout Access Code: ___
                        Manual-In Access Code: ___
     Service Observing Listen Only Access Code: ___
     Service Observing Listen/Talk Access Code: ___
                     Add Agent Skill Access Code: ___
                  Remove Agent Skill Access Code: ___
              Remote Logout of Agent Access Code: ___

                 Call Vectoring/Call Prompting Features

                      Converse Data Return Code: ___
```

After Call Work Access Code

Enter the code the agent must dial when the agent will be performing work-related ACD activities.

Assist Access Code

Enter the digit the agent must dial to request assistance from the split supervisor.

Auto-In Access Code

Enter the code the agent must dial to become automatically available to receive another ACD call each time a call is released.

Aux Work Access Code

Enter the code the agent must dial when the agent will be performing non-ACD activities.

Login Access Code

Enter the code the agent must dial to gain access to the ACD functions. This is a system-wide code for all ACD agents.

Logout Access Code

Enter the logout code the agent must enter to exit ACD. This is a system-wide logout code for all ACD agents.

Manual-In Access Code

Enter the code the agent must dial to receive a single, new ACD call upon the completion of an ACD call.

NOTE:
The following two fields appear only if Service Observing (Remote/By FAC) on the System Parameters Customer-Options screen is **y**.

Service Observing Listen Only Access Code

Enter the code that must be dialed to allow a station with Service Observing permission (COR) to listen to other agent ACD calls without being heard on the ACD call.

Service Observing Listen/Talk Access Code

Enter the code that must be dialed to allow a station with Service Observing permission (COR) to both listen and be heard on an ACD call.

NOTE:
The following two fields appear only if Expert Agent Selection (EAS) Enabled is optioned on the Feature-Related System-Parameters screen.

Add Agent Skill Access Code

Enter the digits an agent must dial to be able to add a skill to their current skill set.

Remove Agent Skill Access Code

Enter the digits an agent must dial to be able to remove a skill from their current skill set.

NOTE:
The next field is available only if Vectoring (Basic) and Vectoring (Prompting) have been enabled on the System-Parameters Customer-Options screen.

Remote Logout of Agent Access Code

Enter the digits you need to dial to remotely logout an idle ACD or EAS agent.

Converse Data Return Code

Enter the access code the CONVERSANT must outpulse prior to outpulsing the digits being returned to the system. This FAC must match the code administered on CONVERSANT.

Field descriptions for page 5

The feature access codes on this page pertain only to Hospitality features.

```
change feature-access-codes                                    Page 5 of X
                          FEATURE ACCESS CODE (FAC)

                            Hospitality Features

                     Automatic Wakeup Call Access Code: *11___
          Housekeeping Status (Client Room) Access Code: _____
          Housekeeping Status (Client Room) Access Code: _____
          Housekeeping Status (Client Room) Access Code: _____
          Housekeeping Status (Client Room) Access Code: _____
          Housekeeping Status (Client Room) Access Code: _____
          Housekeeping Status (Client Room) Access Code: _____
              Housekeeping Status (Station) Access Code: _____
              Housekeeping Status (Station) Access Code: _____
              Housekeeping Status (Station) Access Code: _____
              Housekeeping Status (Station) Access Code: _____
                 Verify Wakeup Announcement Access Code: _____
                       Voice Do Not Disturb Access Code: _____
```

Automatic Wakeup Call Access Code

Enter the access code the user must dial to schedule or cancel a wakeup call.

Housekeeping Status (Client Room) Access Code

Enter the access code the housekeeper dials from the client's room to provide room status. These codes are transmitted to the Property Management System (PMS) for processing. You can assign a definition to the six codes on the Hospitality screen.

Housekeeping Status (Station) Access Code

Enter the access code the housekeeper must dial to provide room status. This access code must be dialed from designated telephones. There are four codes.

Verify Wakeup Announcement Access Code

Enter the access code the user can dial to verify a wakeup announcement.

Voice Do Not Disturb Access Code

Enter the access code the user must dial to enter or cancel a do not disturb request without using a display — through the use of voice prompting.

Field descriptions for page 6

The feature access codes on this page pertain only to Multimedia Call Handling (MMCH).

```
change feature-access-codes                                          Page   6 of   6
                              FEATURE ACCESS CODE (FAC)
                                 Multimedia Features

                      Basic Mode Activation:
                   Enhanced Mode Activation:
                Multimedia Call Access Code:
     Multimedia Data Conference Activation:  Deactivation:
      Multimedia Multi-Address Access Code:
          Multimedia Parameter Access Code:
```

Basic Mode Activation

If you enter this FAC when your system is an Enhanced multimedia complex, it will revert to a Basic multimedia complex. If you enter this FAC when your system is a Basic mode station it will do nothing.

Enhanced Mode Activation

If you enter this FAC when your system is a Basic multimedia complex, it will become an Enhanced multimedia complex. If you enter this FAC when your system is an Enhanced mode station it will do nothing.

Multimedia Call Access Code

If you enter this FAC from any voice station, it indicates to the DEFINITY ECS that you are making an Enhanced mode multimedia call. If you originate a multimedia call with the multimedia call access code, it will originate a call according to the Default Multimedia Parameters selected on the Feature Related System Parameters screen.

Multimedia Data Conference Activation

If you enter this FAC from any voice station that is participating in a multimedia call, it will alert the DEFINITY ECS that you want to enable data collaboration with the other parties on the call. If you enter this FAC a second time, it will give denial treatment (since a collaboration session is already active). This FAC only applies to voice stations on a DEFINITY ECS switch equipped with an ESM adjunct.

Multimedia Data Conference Deactivation

If you enter this FAC from the phone that enabled data collaboration on a multimedia mode call, it will deactivate the data session and revert to a voice and video call. If a user enters this FAC while participating in a data-collaboration multimedia call that the user did not initiate, the system responds with denial treatment.

Multimedia Multi-Address Access Code

The multimedia multi-address access code is similar to the multimedia call access code. It allows origination of a multimedia call from a voice station. It is used when the destination being dialed requires a different address for each of the 2 B-channels. For example, ISDN-BRI provided by a Central Office is provisioned with separate listed directory numbers for each B-channel. In order to make a 2B multimedia call to such a device, two sets of addresses must be entered. Originating a multimedia call with the multimedia multi-address access code will originate a call according to the Default Multimedia Parameters selected on the System Parameters Features screen.

Multimedia Parameter Access Code

This FAC can be entered by any voice station to indicate to the DEFINITY ECS that you want to initiate a multimedia mode call with a specific bearer capability. This FAC would be followed by a 1 or 2 to indicate the following parameter selections respectively: 2x64 (unrestricted initial system default), 2x56 (restricted).

Understanding your configuration

At a very basic level, the DEFINITY ECS consists of hardware to perform call processing, and the software to make it run. You use the administration interface to let the system know what hardware you have, where it is located, and what you want the software to do with it.

You can find out which circuit packs are in the system and which ports are available by entering the command **list configuration all**. There are variations on this command that display different types of configuration information. Use the help function to experiment, and see which command works for you.

1. To view a list of port boards on your system, type **list configuration port-network** and press **RETURN.**

The System Configuration screen appears.

```
                          SYSTEM CONFIGURATION
                                                   Assigned Ports
Board
Number      Board Type        Code     Vintage u=unassigned t=tti p=psa

01A05       DIGITAL LINE       TN754B 000002  01 u  03 u  05 u  07 08
01A06       ANALOG LINE        TN742  000010  01 02 03 04 u  u  u  u
01B05       ANALOG LINE        TN746B 000008  u  u  u  u  u  u  u  u
                                              u  u  u  u  u  u  u  u
01C04       ANALOG LINE        TN746B 000008  u  u  u  u  u  u  u  u
                                              u  u  u  u  u  u  u  u
01C05       DIGITAL LINE       TN2224 000004  01 u  u  04 u  u  07 08
                                              u  u  u  u  u  u  u  u
                                              u  u  u  u  u  u  u  u
01C06       HYBRID LINE        TN762B 000004  01 02 u  u  u  u  u  u
01C09       MET LINE           TN735  000005  01 u  u  u
01C10       DIGITAL LINE       TN754  000004  u  u  u  u  u  u  u  u
```

The System Configuration screen shows all the boards on your system that are available for connecting phones, trunks, data modules and other equipment. You can see the board number, board type, circuit-pack type, and status of each board's ports. The **u** entries on this screen indicate unused ports that are available for you to administer. These may also appear as **p** or **t**, depending on settings in your system.

You will find many places in the administration interface where you are asked to enter a port or slot. The port or slot is actually an address that describes the physical location of the equipment you are using.

A port address is made up of four parts:

- **cabinet** — the main housing for all the switch equipment. Cabinets are numbered starting with 01.
- **carrier** — the rack within the cabinet that holds a row of circuit packs. Each carrier within a cabinet has a letter, A–E.
- **slot** — the space in the carrier that holds an individual circuit pack. Slots are numbered 01-16.
- **port** — the wire that is connected to an individual piece of equipment (such as a phone or data module). The number of ports on a circuit pack varies depending on the type.

So, if you have a single-carrier cabinet, the circuit pack in slot 06 would have the address 01A06. If you want to attach a phone to the 3rd port on this board, the port address is 01A0603 (01=cabinet, A=carrier, 06=slot, 03=port).

Phone Types

This describes many of the telephones and adjuncts that you can connect to the DEFINITY ECS.

- determine where to connect a phone—is it analog, digital, or hybrid? is it designed for a 2-wire or 4-wire environment?
- determine whether a phone will fit your users' needs—does it accommodate enough feature buttons? does it include a display?
- understand how to assign the user's feature buttons—how do the buttons on the Station screen map to buttons on the physical phone.

500 telephones

The 500 telephones are single appearance analog rotary-dial telephones which provides cost-effective service wherever it is located. It provides limited access to features because the rotary dial has no * or # positions.

2500-series telephones

The 2500-series telephones consist of single appearance analog telephones with conventional touch-tone dialing. You can allow 2500-series phones users to access features by giving them the appropriate feature access codes.

6400-series telephones

The 6400-series telephones are DCP 2-wire telephones that work with the DEFINITY ECS. The last two digits of the 6400-series model number identify the number of call appearances (2-lamp buttons) for that model. For example, the 6424D has 24 call appearances. The 6400 series includes two single-line sets (6402 and 6402D), 8-button sets, a 16-button set, a 24-button set, and a 24-button expansion module (for the 6416D+ and 6424D+ telephones).

Each of the 6400-series phones includes 6 standard feature buttons:

- SPEAKER button, which can access a 2-way speakerphone or allow group listen
- MUTE button, which mutes the handset or speakerphone microphone
- HOLD button
- REDIAL button
- TRANSFER/TEST button for transferring a call or testing the lights and display on the telephone
- CONF/RING button for setting up a conference call and for selecting a personalized ringing pattern.

These phones do not have a standard Drop button, but you can assign a drop button to any feature button. The 6400-series display phones show the date and time in Normal mode, so you do not have to assign a Date/Time button to these phones.

6408D Telephone

The 6408 is a multi-appearance digital telephone with eight call appearance/feature buttons.

The 6408 telephone is available in the following four models
:

- 6408 telephone—includes a 1-way, listen-only speaker, and no display
- 6408+ telephone—includes a 2-way speakerphone, and no display
- 6408D telephone—includes a 1-way, listen-only speaker, and a 2-line by 24-character display
- 6408D+ telephone—includes a 2-way speakerphone, and a 2-line by 24-character display.

With the 6408D and 6408D+ telephones, the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

The 6408, 6408+, 6408D, and 6408D+ telephones can work only in 2-wire environments.

6416D Telephone

The 6416D+ telephone is a multi-appearance digital telephone with 16 call appearance/feature buttons.

With the 6416D+ telephone the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

NOTE:
You can connect an XM24 expansion module to the 6416D+ telephone to expand the number of buttons you can assign. However, when the expansion module is connected, you must connect an auxiliary power supply to the telephone.

6424D Telephone

The 6424D+ telephone is a multi-appearance digital telephone with 24 call appearance/feature buttons. With the 6424D+ telephone the end-user can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features you assign to the call appearance/feature buttons.

The 6424D+ telephone can work in both 4-wire and 2-wire environments.

NOTE:
You can connect an XM24 expansion module to a 6424D+ phone to expand the number of buttons you can assign. However, when the expansion module is connected, you must connect an auxiliary power supply to the telephone.

7400-series telephones

7405D telephone

The 7405D telephone is a multi-appearance digital telephone which allows features to be added as the user needs them. The Digital Display can be added to provide access to the Message Center. A Digital Terminal Data Module or 7400B can be added to enable the user of a 7405D telephone to transmit or receive data with an associated data terminal.

The basic 7405D provides 10 call appearance/feature buttons with lights that can be assigned to call appearances or system features. It has 24 programmable feature buttons and six fixed-feature buttons. The 7405D can also have a function key module which adds 24 feature buttons and a call coverage module (when no display module is used) which adds 20 call appearance/feature buttons.

7406 telephones

The 7406D telephone (7406D01A, 7406D02A, 7406D03A, and 7406D04A models) has five call appearance/feature buttons, each with a red in-use light and a green status light, seven shiftable (2-level) programmable feature buttons with no lights, four programmable feature buttons with a green light, four fixed feature buttons (CONFERENCE, TRANSFER, DROP, and HOLD), a SHIFT button with a green light, a SPEAKER button, and a green Message light.

The 7406BIS telephone (7406D05A and 7406D06A models) has five call appearance/feature buttons, each with a red in-use light and a green status light, seven shiftable (2-level) programmable buttons with no lights, two programmable feature buttons with a green light, four fixed feature buttons (CONFERENCE, TRANSFER, DROP, and HOLD), a SHIFT button with a green light, a SPEAKER button with a green light, a MUTE button with a red light, a SPEAKER VOLUME "arrow" button, and a red Message light.

The 7406+ telephone (7406D07A and 7406D08A models) has five call appearance/feature buttons, each with a red in-use light and a green status light, three shiftable (2-level) programmable feature buttons with a green light, six shiftable (2-level) programmable feature buttons without lights, four fixed feature buttons (CONFERENCE, TRANSFER, DROP, and HOLD), a SELECT button with a green light, a SPEAKER/RESET SPKR button with a green light, a MUTE button with a red light, a VOLUME "arrow" button, and a red Message light.

7407+ telephone

The 7407D, Enhanced 7407D, and 7407+ telephones are multi-appearance digital telephones which provide digital voice, display, and data capabilities (the latter with the 7400B+ Data Module).

7434D telephone

The 7434D is a multi-appearance digital telephone that offers 34 call appearance/feature buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with one light each (SELECT, SPEAKER/RESET SPKR, and MUTE), seven display feature buttons with one light each, a Message light, personalized ringing, a built-in speakerphone with a reset option, and a built-in 2-line by 40-character display.
You can connect this phone to a digital line port. The 7434D telephone supports an adjunct display module or a call coverage module.

7444D telephone

The 7444 telephone is a multi-appearance digital telephone that offers 34 call appearance/feature buttons, each with a red in-use light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), three fixed feature buttons with one light each (SELECT, SPEAKER/RESET SPKR, and MUTE), seven display feature buttons with one

light each, a Message light, personalized ringing, a built-in speakerphone with a reset option, and a built-in 2-line by 40-character display. You can connect this telephone to a digital line port. It is powered from the switch.

NOTE:
The 7444 is powered by the switch, however, to use the display, you must connect an auxiliary power supply to the telephone.

ISDN telephones (7500s & 8500s)

The Integrated Services Digital Network (ISDN) phones include both the 7500-series and the 8500-series telephones. Each of these phones uses the ISDN communications through a 4-wire "T"-interface.

8110 telephones

The basic 8110 (8110A01A, 8110A01B, and 811A01C) and the modified 8110M (8110A01D) telephones are single-line analog telephones. These telephones are exactly the same in appearance: each contains 12 programmable dialing buttons with PROGRAM and PAUSE buttons, automatic redial, a flashing red Message light, and a Hold button. They also have built-in speakerphones with Mute capability and the Automatic Answer feature.

8400-series telephones

8403B telephones

The 8403 telephone is a multi-appearance digital telephone with three call appearance buttons, Conference, Transfer, Drop, and Hold buttons, a TEST button, a blue FEATURE button which allows you to access 12 system features assigned by the System Manager and to choose from among eight different ringing patterns, a MUTE button, a SPEAKER button which accesses a 1-way, listen-only speaker, a red Message light, and a Volume control button.

The 8403 can be used in either a 4-wire or 2-wire environment.

8405B telephone

There are four varieties of the 8405 telephone: the 8405B and 8405B+, the 8405D and 8405D+. All four varieties are multi-appearance digital telephones with five call appearance/feature buttons.The 8405 telephones also have four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a MUTE button, a SPEAKER button, a TEST button, and a Volume control button. The 8405D and 8405D+ allow you to administer 12 softkey feature buttons in addition to the call appearance and feature buttons.

The four 8405 variations have the following differences:

- The 8405B has a 1-way, listen-only speaker, with NO display.
- The 8405B+ has a 2-way speakerphone, without a display
- The 8405D has a 1-way, listen-only speaker and a 2-line by 24-character display.
- The 8405D+ has a 2-way speakerphone and a 2-line by 24-character display.

The 8405 telephones work in 4-wire or 2-wire environments.

8410B telephone

The 8410 telephone is a multi-appearance digital telephone with 10 call appearance/feature buttons, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a MUTE button, a SPEAKER button which can access either a 2-way speakerphone or a 1-way, listen-only speaker, a TEST button, and a Volume control button.

- The 8410B is the basic set, without a display.
- The 8410D (8410D03A) has a built-in 2-line by 24-character display. Those users who have an 8410D with display can access 12 features with the softkeys and display control buttons. These 12 features can be used *in addition to* the features on the call appearance/feature buttons.

The 8410 telephone can work in both 4-wire and 2-wire environments.

8411B and 8411D telephones

The 8411 telephone is a multi-appearance digital telephone with 10 call appearance/feature buttons, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a blue SHIFT button, a MUTE button, a SPEAKER button which can access either a 2-way speakerphone or a 1-way, listen-only speaker, a TEST button, and a Volume control button.

The rear of the 8411 telephone has two jacks: The Analog Adjunct jack can be used for connecting answering machines, fax machines, PC or laptop data/fax modem cards, data sets or modems, audio teleconferencing equipment, and TTY machines commonly used by the hearing impaired. The RS-232-D Jack can be used for connecting the telephone to a COM port on an IBM®-compatible personal computer on which you can load PassageWay Solution software.

There are two varieties of the 8411 telephone: the 8411B (8411D02A) is the basic set, without a display; the 8411D (8411D01A) has a built-in 2-line by 24-character display. Those users who have an 8411D with display can access 12 features with the softkeys and display control buttons. These 12 features can be used in addition to the features on the call appearance/feature buttons.

The 8411 telephone can work in both 4-wire and 2-wire environments.

8434D telephone

The basic 8434 (8434D01A) and the enhanced 8434DX (8434D02A) telephones are multi-appearance digital telephones which offer 34 call appearance/feature buttons, each with a red light and a green status light, four standard fixed feature buttons (CONFERENCE, DROP, HOLD, and TRANSFER), a MUTE button, a SPEAKER button which accesses either a 2-way speakerphone or a 1-way listen-only speaker, a TEST button, a SHIFT button (some 8434DX telephones will have a RING button instead), a red Message light, personalized ringing, a built-in speakerphone with a reset option, and a built-in 2-line by 40-character VFD display. The 8434 and 8434DX also have five softkeys and four display control buttons which allow the user to access 15 features. These softkey features can be used *in addition to* the features on the call appearance/feature buttons.

The 8434 and 8434DX telephones can be used in both a 4-wire and a 2-wire environment.

NOTE:
In order to use the display on the 8434 or 8434DX telephone and to use an 801A expansion module connected to the 8434DX, you must connect an auxiliary power supply to the telephone.

You can connect an 801A Expansion Module to the 8434DX telephone to provide 24 additional call appearance/feature buttons.

CALLMASTER telephones

There are several types of CALLMASTER telephones:

602A and 602D CALLMASTER

The 602 CALLMASTER models have a display, a Message light, a Mute button, and four fixed feature buttons: Conference, Drop, Hold, and Transfer. You can administer its 10 call appearance/feature (2-lamp) buttons and its 17 feature-only (1-lamp) buttons.

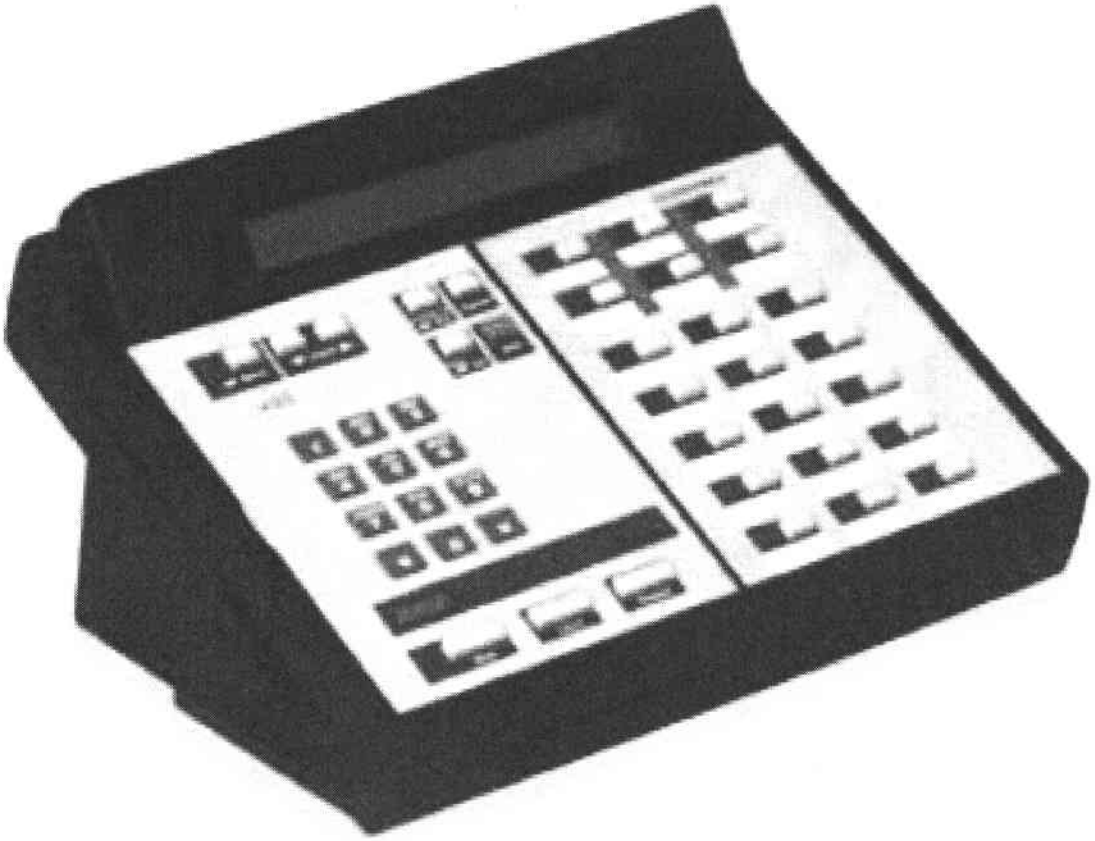
603D (CALLMASTER II)

The CALLMASTER II model has a display, a Message light, and the Mute, Select, Log In, and Release buttons. It also has four fixed features: Conference, Drop, Hold, and Transfer. You can administer its 6 call appearance/feature (2-lamp) buttons and its 15 feature-only (1-lamp) buttons.

603E (CALLMASTER III)

The CALLMASTER III model has a display, a Message light, and the Select, Mute, Log In, and Release buttons. It also has four fixed features: Conference, Drop, Hold, and Transfer. You can administer its 6 call appearance/feature (2-lamp) buttons and its 15 feature-only (1-lamp) buttons. Note that you can assign any feature to the Log In and Release buttons.

You can connect the CALLMASTER III to either a standard 4-wire DCP or a 2-wire circuit pack.



603F (CALLMASTER IV)

The CALLMASTER IV model has a display, a Message light, and the Select, Mute, Log In, and Release buttons. It also has four fixed features: Conference, Drop, Hold, and Transfer. You can administer its 6 call appearance/feature (2-lamp) buttons and its 15 feature-only (1-lamp) buttons. Note that you can assign any feature to the Log In and Release buttons.

You can connect the CALLMASTER IV to either a standard 4-wire DCP or a 2-wire circuit pack.

606A (CALLMASTER VI)

The CALLMASTER VI model is a miniature, 8-button, 2-headset jack, digital telephone that is controlled by the user's personal computer (PC) through an RS-232 serial-port connection.

(607A (CALLMASTER V)

The CALLMASTER V model has a display, softkeys, and the display control buttons (Menu, Exit, Previous, and Next). This model does not have a standard handset, but you can connect a handset to one of its headset jacks. The CALLMASTER V has six fixed feature buttons: Speaker, Mute, Hold, Redial, Conference, and Transfer. You can administer its 16 call appearance/feature (2-lamp) buttons, however, one of these buttons must be administered as a Headset On/Off button and a second one must be administered as a Release button. You can also administer the 12 softkey buttons.

The 4620/4620SW/4621SW IP Telephone

The 4620/4620SW/4621SW IP Telephone is an innovative telephone that gives you access to the World Wide Web while offering the latest features and applications. The large display area allows up to 12 application-specific buttons to be presented and labeled at one time. Additionally, 12 Line/Feature buttons, 4 softkeys, and other fixed buttons provide access to powerful capabilities such as:

- call server-based features,
- speed dialing,
- a Call Log, and
- a WML (Wireless Markup Language, a Web development protocol) browser.

The WML browser provides access to Web sites tailored specifically for devices with smaller display screens like cell phones and Personal Data Assistants. In addition to these features and applications, the 4620/4620SW/4621SW provides a menu of options to customize your phone preferences. Your telephone's display area coincides with how your System Administrator sets up the Line/ Feature buttons. The 4 softkeys assist you in using 4620/4620SW/4621SW applications and features. The 14 standard (labeled) buttons assist in telephone operation and call handling. A built-in, two-way Speaker, and an infrared interface combine to provide ease of use and flexibility. The telephone has an adjustable stand that moves to optimize your viewing position via the button on the back. The figure shows the face of the 4620/4620SW/4621SW IP Telephone. This diagram contains numbered "callouts" identifying the phone's primary features and buttons. If you are viewing this guide online, you can click the callout to jump to the corresponding feature or button description. Otherwise, each callout is described in detail in Table 1 following the diagram.

4620 IP TELEPHONE

4620/4620SW/4621SW IP Telephone Button/Feature Descriptions

1 Message Waiting Lamp

When lit, indicates you have a message waiting on your voice messaging system. This indicator can also be optioned to flash for incoming calls.

2 Display

The display screen is 4 inches by 2.9 inches. Information displayed varies according to the application/function currently active. When the phone is idle, the top area displays the current date and time. When someone is calling you, the name/phone number of that person displays in the top area. The display has eight lines. Six display lines are devoted to the current application. One line shows softkey labels for the current application and one line shows Help and other procedural messages. Four grayscale colors are used to indicate activity.

3 Line/Feature buttons

Twelve Line/Feature buttons provide both call appearances (lines for incoming and outgoing calls) and application-specific functionality.

4 Softkeys

Used to navigate to, or start application-specific actions, such as **Call** a number, **Cancel** the current activity, **Save** entered data such as a Speed Dial label.

5 Phone/Exit

Displays the Phone application main screen or, if applicable, exits the current call server-based feature and normalizes the display.

6 Options

Displays the Options main screen, from which display and application settings can be updated.

7 Page Left/Right

Shifts from one page to another in the same application.

8 Speaker LED Indicator

Lights steadily when the Speaker is active.

9 Speaker

Accesses the Speaker feature.

10 Headset LED Indicator

Lights steadily when the headset is active.

11 Headset

With a headset connected, changes audio control from the handset or Speaker to the headset.

12 Mute LED Indicator

Lights steadily when the handset, headset or Speaker is muted.

13 Mute

Turns off the active Speaker, handset, or headset microphone, to prevent the other person from hearing you.

14 Volume Control

Adjusts the handset, Speaker, headset, or ringer volume, depending on which item is in use. When you increase or decrease the volume, the top display area shows an icon to indicate the item for which you are adjusting the volume. A visual "volume meter" that shows the volume level follows the icon. This button also controls the volume of the key click sounds. Key clicks sound when you press fixed buttons on the phone such as the dialpad or softkeys.

15 Headset Jack

Provides a port for connecting a headset on the underside of the phone.

16 Hold

Red button used to place a call on hold.

17 Transfer

Transfers a call to another phone.

18 Conference

Sets up conference calls with more than one other person.

19 Drop

Drops the last person added to a Conference call or ends the current call, if you are not on a Conference call.

20 Redial

Redials the last number dialed from the phone or displays a list of the last six numbers dialed for selection.

21 Numeric (Dialing) Pad

Standard 12 button pad for dialing phone number

4620/4620SW/4621SW IP Telephone Applications

Your 4620 IP Telephone provides four applications (Phone, Speed Dial, Call Log, and Web). Additionally, use the Options function to define settings, personalize your phone, and troubleshoot certain functions. All applications appear in the display area, accessed by the softkeys appearing below the display.

A brief description of each application follows.

Phone Application

The Phone application is the primary application. Use this application to make and receive calls, and perform call-handling operations like conference calling or transferring calls to another phone.

Speed Dial Application

The Speed Dial application provides access to the Speed Dial buttons that facilitate automatic dialing. Use this application to:

- set up buttons for speed dialing,
- update Speed Dial button labels, or
- remove the label from a Speed Dial button.

Call Log Application

The Call Log application provides three lists, each showing up to 30 outgoing, incoming answered or missed calls. Use this application to call a person directly from the Call Log or to add a log entry's name and phone number to a Speed Dial button.

Functionality

This section describes how to make calls, receive calls, retrieve voice mail messages, and log off your 4620/4620SW/4621SW IP Telephone. The telephone Line/Feature buttons that assist in call handling are also covered in this chapter.

Call Appearances

In the Phone application, a call appearance, by default, takes up the entire display width, while administered Feature buttons take up half the display width. When a call appearance is full-width, use the Line/Feature buttons on either side of that row to select that call appearance, and usually, all associated messages. When a call appearance is half-width, for example, the full width default has been changed, use only the Line/Feature buttons on the appropriate side of that row to select that call appearance. In the case of half-width call appearances, call-associated messages show on the top display line instead of on that row.

Redialing a party

Depending on how you have set up your Redial option, selecting **Redial** automatically dials the most recent number dialed, or displays a list of the last six outgoing calls for selection of the number to be dialed.

Dialing a party using a Speed Dial button

Automatic dialing of pre-stored numbers is the most common method of automatic dialing. You can set up Speed Dial buttons and select the party you want to call by pressing that button.

Because the 4620 IP Telephone's advanced capabilities allow up to 108 speed dial entries, speed dialing as described here is convenient and efficient.

1. Press the **SpDial** softkey at the bottom of the display screen. *The first twelve Speed Dial buttons display, one name/number per button, and the prompt "Select entry to dial" appears at the top of the display area.*

2. If the party you want to call appears on the display, proceed to Step 3. If the party you want to call is not shown, press the **Page Right** ( ) button to display the next page of entries, Continue until the number/party you want displays. You can also press the **Page Left** button to display the preceding page of entries.

3. Press the Line/Feature button associated with the name/number of the person you want to call. *The number of the selected person dials automatically.*

4. Pick up the handset, activate the headset, or use the Speaker to proceed with the call.

5. Hang up the handset, deactivate the headset, or press the **Speaker** button to end the call.

Automatically dialing a party using an administered Line/Feature button

Your System Administrator can program individual numbers on Line/Feature buttons (this is called Abbreviated Dialing). If so, such numbers display on the Phone application screen (or the Feature Key Expansion Unit, if this optional device is attached to your phone) with labels assigned by the System Administrator. If the label for the number you want to call appears in the display area, press the appropriate button,

Calling a party from the Call Log

Your 4620 IP Telephone maintains a log of up to 90 outgoing, incoming answered, and incoming unanswered calls to/from your phone. Each log can have up to 30 calls.

1. To call a party listed in the Call Log, press the **Log** softkey at the bottom of the display area. *The first six missed calls display, one name/number per button, and the prompt "Select entry for details" appears at the top of the display area.*

2. If the party you want to call is in a different Call Log, press the softkey that represents the Call Log where that entry appears (either **Outgo** or **InAns**). *The selected Call Log displays.*

3. If the party you want to call appears, proceed to Step 4. If the party you want to call is not displayed, press the **Page Right** ( ) button to display the next page of entries. Continue until the number/party you want displays. You can also press the **Page Left** button to display the preceding page of entries.

4. Press the Line/Feature button associated with the name/number of the person you want to call. *The associated Call Detail screen displays.*

5. Select **Call**. *The phone goes off-hook and the selected party's number is dialed.*

6. Proceed with the call as usual.

Call Handling Features

The features described in this section are available while calls are in progress. Use the dedicated Feature buttons on the telephone itself, or administered Feature buttons available using the 4620's softkeys, as applicable.

Conference

The Conference feature allows you to conference up to the maximum number of parties set by your System Administrator.

Adding another party to a call

1. Dial the first party, then press the **Conference** button. *The line's display area changes to white text with a dark gray background. The current call is placed on hold, the Soft Hold icon displays, and you hear a dial tone.*

2. Dial the number of the next party and wait for an answer.

3. Press the **Conference** button again to add the new party to the call.

4. Repeat Steps 1- 3 for each party you want to conference in to the call.

Adding a held call to the current call

1. Press the **Conference** button. *The icon on the current line changes to the Soft Hold icon.*
2. Press the Line/Feature button of the held call.
3. Press the **Conference** button again. *All parties are now connected.*

Dropping the last person added to the call

Press the **Drop** ( ) button. *The last party connected to the conference call is dropped from the call.*

Hold

The Hold feature puts a call on hold until you retrieve it.

Placing a call on hold

Press the **Hold** button. *The line's display area changes to white text with a dark gray background, and the Hold icon displays.*

Mute

During an active call, the Mute feature prevents the party with whom you are speaking from hearing you. You generally use this feature in conjunction with the Speaker, but you can hold an off-line conversation at any time during a call.

Preventing the other person on the line from hearing you

1. Press the **Mute** button. *The other party cannot hear you. The indicator next to the Mute button lights when Mute is active.*
2. To reinstate two-way conversation, press the **Mute** button again.

Speaker

A two-way, built-in Speaker allows you to place and answer calls without lifting the handset.

Placing or answering a call without lifting the handset, or using the Speaker with any feature

1. Press the **Speaker** button. *The indicator next to the Speaker button lights and the Speaker handles voice control. The first available call appearance line activates.*

2. Place or answer the call, or access the selected feature.

3. Adjust the Speaker volume if needed by pressing the **Volume Control** button until you reach the desired volume.

Transfer

The Transfer feature allows you to transfer a call from your telephone to another extension or outside number.

Sending a call to another telephone

1. With the call active (or with only one held call and no active calls), press the **Transfer** button. *The call is placed on hold. The Hold icon displays and you hear a dial tone while the next available line activates.*

2. Dial the number to which you want to transfer the call.

3. If you do not want to announce the call, press the **Transfer** button again and proceed to Step 6. If you wish to wait for an answer and announce the call, see Step 4. *The call is sent to the extension or number you dialed. A two-second display message indicates the transfer is complete.*

4. Remain on the line and announce the call. If the line is busy or if no one answers, return to the call by pressing the Line/Feature button on which the call is being held.

5. Press the **Transfer** ( ) button again. *The call is sent to the extension or number you dialed. A two-second display message indicates the transfer is complete.*

6. Hang up your handset.

Station Programming

This section provides descriptions of all of the fields that may appear on Station screens. Some of the fields are used for specific phone types; others are used for all phone types. To make it easier to find a specific field, they are listed in alphabetical order by field name.

```
change station 1014                                                Page 1 of X
                                STATION

Extension: 1014                 Lock Messages? n                      BCC:
     Type:                      Security Code:                         TN:1
     Port:                    Coverage Path 1:                        COR: 1
     Name:                    Coverage Path 2:


STATION OPTIONS
              Loss Group: 2                 Personalized Ringing Pattern: 3
            Data Module? n                            Message Lamp Ext: 1014
           Speakerphone: 2-way                     Mute button enabled? y
      Display Language? English                      Media Complex Ext:
                                                           IP Softphone? y
                                                  Remote Office Station? n
                                                     IP Emergency calls:
extension
```

Station

```
change station 75001                                                Page 2 of X
                                        STATION

FEATURE OPTIONS
           LWC Reception? msa-spe                Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
            CDR Privacy? n                                  Data Restriction? n
   Redirect Notification? y                       Idle Appearance Preference? n
 Per Button Ring Control? n
   Bridged Call Alerting? n                         Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n                 Per Station CPN - Send Calling Number? _
       Service Link Mode: as-needed       Special Character for Restricted Number? n
         Multimedia Mode: basic
    MWI Served User Type: _____                     Display Client Redirection? n
        Automatic Moves:
              AUDIX Name:                          Select Last Used Appearance? n
   Messaging Server Name: _____                      Coverage After Forwarding? _
     Recall Rotary Digit? n                            Multimedia Early Answer? n
      IP Emergency Calls: extension             Direct IP-IP Audio Connections? n
  Emergency Location Ext: 75001                            IP Audio Hairpinning? n
```

Station

```
add station 1014                                                    Page 3 of X
                                        STATION

     SITE DATA                                                  Headset? n
          Room: _______                                         Speaker? n
          Jack: ___                                            Mounting: d
         Cable: ___                                          Cord Length: 0_
         Floor: _______                                        Set Color: _____
      Building: _______

ABBREVIATED DIALING
     List1: _________          List2: _________          List3: _________

BUTTON ASSIGNMENTS
 1: call-appr                          5: _______
 2: call-appr                          6: _______
 3: call-appr                          7: _______
 4: _______                            8: _______
```

Station

The standard station fields are organized alphabetically for easy access.

1-Step Clearing

Valid entries	Usage
y/n	If set to **y**, the call terminates again at the WCBRI terminal when the user drops from the call.

Abbreviated Dialing List1, List2, List3

You can assign up to 3 abbreviated dialing lists to each phone.

Valid entries	Usage
enhanced	Allows the phone user to access the enhanced system abbreviated dialing list.
group	Allows the phone user to access the specified group abbreviated dialing list. If you enter **group**, you also must enter a group number.
personal	Allows the phone user to access and program their personal abbreviated dialing list. If you enter **personal**, you also must enter a personal list number.
system	Allows the phone user to access the system abbreviated dialing list.

Active Station Ringing

Defines how call rings to the phone when it is off-hook. This field does not affect how calls ring at this phone when the phone is on-hook.

Valid entries	Usage
continuous	Enter **continuous** to cause all calls to this phone to ring continuously.

single	Enter **single** to cause calls to this phone to receive one ring cycle and then ring silently.
if-busy-single	Enter **if-busy-single** to cause calls to this phone to ring continuously when the phone is off-hook and idle and calls to this phone to receive one ring cycle and then ring silently when the phone is off-hook and active.
silent	Enter **silent** to cause all calls to this station to just ring silently.

Adjunct Supervision

Adjunct Supervision appears when the Type field is **500**, **2500**, **k2500**, **8110**, **ops**, **ds1fd**, **ds1sa**, **VRU**, **VRUFD**, or **VRUSA**.

Valid entries	Usage
y	Enter **y** if an analog disconnect signal is sent automatically to the port after a call terminates. Analog devices (such as answering machines and speakerphones) use this signal to turn the devices off after a call terminates.
n	Set this field to **n** so hunt group agents are alerted to incoming calls. In a hunt group environment, the disconnect signal blocks the reception of zip tone and incoming call notification by an auto-answer station when a call is queued for the station.

Assigned Member — Ext

The system automatically assigns this extension. This is the extension of the user who has an associated Data Extension button and shares the module.

Assigned Member — Name

Display-only field that shows the na
me associated with the extension shown in the
Assigned Member - Ext field.

Att. Call Waiting Indication

Attendant call waiting allows attendant-originated or attendant-extended calls to a busy single-line phone to wait and sends distinctive call-waiting tone to the single-line user.

Valid entries	Usage
y/n	Enter **y** to assign Attendant Call Waiting to the phone. You should not set this field to y, if the Data Restriction field is **y** or the Switchhook Flash field is **n**, or if Data Privacy is enabled for the phone's class of service (COS). If any of these conditions are true, the phone cannot accept or handle call waiting calls.

Audible Message Waiting

The Audible Message Waiting tone indicates that the user has a waiting message. This field appears only if Audible Message Waiting is set to **y** on the System-Parameters Customer-Options screen.
Note that this field does not control the Message Waiting lamp.

Valid entries	Usage
y/n	Enter **y** if you want the phone user to receive stutter dial tone when they have a waiting message and they go off-hook.

Audix Name

Specifies which AUDIX is associated with the station.

Valid entries	Usage
Names assigned to an AUDIX Adjunct	Must contain a user-defined adjunct name that was previously administered in the Node-Names screen.

Auto Answer

In EAS environments, the auto answer setting on the Agent LoginID screen can override a station's setting when an agent logs in there.

NOTE:
For analog stations, if Auto Answer is set to acd and the station is off-hook and idle, only the ACD split/skill calls and direct agent calls auto answer; non-ACD calls receive busy treatment. If the station is active on an ACD call and a non-ACD call arrives, the Agent receives call-waiting tone.

Valid entries	Usage
all	Enter all to allow all calls (ACD and non-ACD) terminated to an idle station to be cut through immediately. Does not allow automatic hands-free answer for intercom calls.
acd	Enter acd to allow only ACD split /skill calls and direct agent calls to auto answer. If this field is set to acd, Non-ACD calls terminated to a station ring audibly.
none	Enter none to cause all calls terminated to this station to receive an audible ringing treatment.
icom	Enter icom to allow a phone user to answer an intercom call from the same intercom group without pressing the intercom button.

Automatic Moves

Automatic Moves allows a phone to be unplugged from one location and moved to a new location without additional switch administration. The switch automatically associates the extension to the new port.

CAUTION:
When a phone is unplugged and moved to another physical location, the Emergency Location Extension field must be changed for that extension or the USA Automatic Location Identification data base must be manually updated. If the Emergency Location Extension field is not changed or if the USA Automatic Location Identification data base is not updated, the DID number sent to the Public Safety Network could send emergency response personnel to the wrong location.

Valid entries	Usage
always	Enter always and the phone can be moved anytime without additional administration by unplugging from one location and plugging into a new location.
once	Enter once and the phone can be unplugged and plugged into a new location once. After a move, the switch sets the field to done the next time routine maintenance runs on the phone.

	Use once when moving a large number of phones so each extension is removed from the move list. Use once to prevent automatic maintenance replacement.
no	Enter no to require administration in order to move the phone.
done	Done is a display-only value. The switch sets the field to done after the phone is moved and routine maintenance runs on the phone.
error	Error is a display-only value. The switch sets the field to error, after routine maintenance runs on the phone, when a non-serialized phone is set as a movable phone.

Auto Select Any Idle Appearance

Valid entries	Usage
y/n	Enter y to allow automatic selection of any idle appearance for transferred or conferenced calls. The system first attempts to find an idle appearance of the call being transferred or conferenced. If that attempt fails, the system selects the first idle appearance.

Automatic Selection of DID Numbers

The switch chooses a 2- to 5-digit extension from a predetermined list of numbers and assigns the extension to a hotel room phone.

Valid entries	Usage
y/n	Enter y to use the Automatic Selection of DID Numbers for Guest Rooms feature.

BCC

Appears when ISDN-PRI or ISDN-BRI Trunks is enabled on the System-Parameters Customer-Options screen. Display-only field set to 0 for stations (that is, indicates voice or voice-grade data).

The BCC value is used to determine compatibility when non-ISDN facilities are connected to ISDN facilities (ISDN Interworking).

Bridged Call Alerting

If Bridged Call Alerting is n and Per Button Ring Control is n, audible ringing is suppressed for incoming calls on bridged appearances of another phone's primary extension.

Valid entries	Usage
y/n	Enter **y** to enable audible ringing for TEG, PCOL, bridged appearances, or Data Extension calls.

Building

Enter a valid building location.

Button Assignments

Enter the abbreviated software name to assign a feature button.

NOTE:
If you want to use Terminal Translation Initialization (TTI), you must assign a call appearance (call-appr) to the first button position. TTI needs the button on the first call appearance to get dial tone.

Cable

You can use this field to identify the cable that connects the phone jack to the system. You also can enter this information in the Blank column on the Port Assignment Record.

Caller ID Message Waiting Indication

Appears when the Type field is CallrID. For CallrID type phones or analog phones with Caller ID adjuncts only.

Valid entries	Usage
y/n	Enter **y** to allow aliasing of various non-Avaya phones and adjuncts.

NOTE:
The Caller ID Message Waiting Indication administration is independent of the administration of LED or NEON lamp DEFINITY ECS Message Waiting Indication (MWI). For example, it is possible to administer a Caller ID phone with the Caller ID Message Waiting Indication field set to **n** and the Message Waiting Indicator field set to **neon**.

Call Waiting Indication

This allows user, attendant-originated, and outside calls to busy single-line phone to wait and sends a distinctive call-waiting tone to the single-line user. This feature is denied if Data Restriction is **y** or Switchhook Flash is **n**, or if Data Privacy is active via the phone COS assignment.

Valid entries	Usage
y/n	Enter **y** to activate Call Waiting Termination for the phone.

CDR Privacy

This option allows digits in the called number field of an outgoing call record to be blanked, on a per-station basis. You administer the number of blocked digits system-wide in the Privacy - Digits to Hide field on the CDR System Parameters screen.

Valid entries	Usage
y/n	Enter **y** to enable Call Privacy for each station.

COR

Enter a Class of Restriction (COR) number to select the desired restriction.

Cord Length

Enter a number to specify the length (in feet) of the cord attached to the receiver.

COS

Enter the desired Class of Service (COS) number to select allowed features.

Country Protocol

Enter the protocol that corresponds to your supported initialization and codesets. The Country Protocol must match any previously-administered endpoint on the same port.

Valid entries	Usage
1	United States (Bellcore National ISDN)
2	Australia
etsi	ETSI (Europe)
3	Japan
6	Singapore

Coverage After Forwarding

This field governs whether an unanswered forwarded call is provided coverage treatment.

Valid entries	Usage
y	Coverage treatment is provided after forwarding regardless of the value of the Coverage After Forwarding field on the System Parameters - Call Coverage/Call Forwarding screen.
n	No coverage treatment is provided after forwarding regardless of the value of the Coverage After Forwarding field on the System Parameters - Call Coverage/Call Forwarding screen.
s(ystem)	Indicates that call processing uses the Coverage After Forwarding field on the "System Parameters Call Coverage / Call Forwarding" screen. To override the system-wide parameter for a given station, set this field to **y** or **n**.

Coverage Msg Retrieval

Applies if the phone is enabled for LWC Reception.

Valid entries	Usage
y/n	Enter **y** to allow users in the phone's Coverage Path to retrieve Leave Word Calling (LWC) messages for this phone.

Coverage Module

Valid entries	Usage
y	Enter **y** to indicate that a coverage module is connected to the station. Once you enter y, the system displays an additional page that allows you to assign the buttons for the module.

Coverage Path 1 or Coverage Path 2

Enter a coverage-path number or time-of-day table number from a previously-administered Call Coverage Path screen or Time of Day Coverage Table screen.

NOTE:
If Modified Misoperation is active (Misoperation Alerting is **y** on the Feature-Related System Parameters screen), you must assign a Coverage Path to all stations on the switch.

CRV Length

Only for ASAI stations. Enter **1** or **2** to indicate the length of CRV for each interface.

Custom Selection of VIP DID Numbers

Custom Selection of VIP DID numbers allows you to select the DID number assigned to a room when a guest checks in.

Valid entries	Usage
y/n	Enter y to allow you to select the DID number assigned to a room when a guest checks in.

Data Extension

Enter the extension assigned to the data module.

Data Module

Valid entries	Usage
y/n	Enter **y** to indicate that you want to administer a data module with this phone. Entering **y** displays the Data Module screen.

Data Restriction

Data restriction provides permanent protection and cannot be changed by the phone user. Do not assign a Data Restriction if Auto Answer is **all** or **acd**. If y, whisper page to this station is denied.

Valid entries	Usage
y/n	Enter **y** to prevent tones, such as call-waiting tones, from interrupting data calls.

Default Dialing Abbreviated Dialing Dial Code

Appears only when the Special Dialing Option is set to default. Enter a list number associated with the abbreviated dialing list.

When the user goes off-hook for a data call and presses the Return button following the DIAL prompt, the system dials the AD number. The data call originator also can perform data-terminal dialing by specifying a dial string that may or may not contain alphanumeric names.

Direct IP-IP Audio Connections

Allows direct audio connections between IP endpoints.

Valid entries	Usage
y/n	Enter to y to save on bandwidth resources and improve sound quality of voice over IP transmissions.

Display Caller ID

Appears when the Type field is CallrID. For CallrID type phones or analog phones with Caller ID adjuncts only.

Valid entries	Usage
y/n	Enter **y** to allow transmission of calling party information to the Caller ID phone or adjunct.

Display Cartridge

For 7404 D phones only. Enter **y** to indicate there is a display cartridge associated with the station. This displays an additional page to allow you to assign display buttons for the display cartridge.

Display Client Redirection

Only administrable if Hospitality is enabled on the System-Parameters Customer-Options screen. This field affects the phone display on calls that originated from a station with Client Room Class of Service.

NOTE:
For stations with an audix station type, AUDIX Voice Power ports, or ports for any other type of messaging that needs display information, Display Client Redirection must be set to y.

Valid entries	Usage
y	When set to **y**, the redirection information for a call originating from a Client Room and terminating to this station displays.
n	When set to **n**, this station's display does not show the redirection information for all calls originating from a Client Room (even redirected calls) that terminate to this station. Only the client name and extension (or room, depending on what is administered on the "Hospitality" screen) display.

Display Language

Specifies the display language.

Valid entries	Usage
english	Enter the language you want users to see on their displays.
French	
Italian	
Spanish	
user-defined	

Display Module

Valid entries	Usage
y/n	Enter **y** if this phone has a display module.

Distinctive Audible Alert

Valid entries	Usage
y/n	Enter **y** so the phone can receive the 3 different types of ringing patterns which identify the type of incoming calls. Distinctive ringing may not work properly for off-premises telephones.

Emergency Location Ext

The Emergency Location Ext field defaults to the phone's extension. This extension identifies the street address or nearby location when an emergency call is made.

Valid entries	Usage
1-8 digits	Enter the Emergency Location Extension for this station

Endpt ID

Appears only if Endpt Init is **y**. Enter a unique 2-digit number (**00–62**) for this endpoint. Each Endpt ID field must have a unique value for each endpoint on the same port.

This field provides for multipoint configuration conformance to the Bellcore Terminal Initialization procedures. In these procedures, a multipoint configuration requires the last 2 digits of the Service Profile Identifier (SPID) be between 00 and 63 and be binary unique for each endpoint. This field, combined with the SPID, gives the effective SPID administered into the terminal. Bellcore ISDN-1 requires the SPID programmed into the endpoint contain at least 9 digits.

For example, if the SPID is **1234**, and Endpt ID is **01**, then the SPID administered on the terminal is 000123401. The three leading zeros are necessary to create a 9-digit SPID.

Endpt Init

Endpoint initialization is a procedure, required for multipoint operation, by which User Service Order Profile (USOP) is associated with an endpoint on the ISDN-BRI. This association is made via the SPID, administered into the system, and entered into the ISDN-BRI terminal. For an ISDN-BRI terminal to be operational in a multipoint configuration, both the administered SPID and the SPID programmed into the ISDN-BRI terminal must be the same. Therefore, the SPID of new or reused terminals must be programmed to match the administered SPID value. Appears only if MIM Support is **y** and indicates the terminal's endpoint initialization capability.

Valid entries	Usage
y	Enter **y** if the terminal supports Bellcore ISDN-1 terminal initialization procedures.
n	Enter **n** for all other country protocols.

Event Minimization

Allows you to minimize events sent on a link. This field appears only if you set Type to **asai**.

Valid entries	Usage
y/n	Enter **y** when an application or library does not want to receive identical event reports over different associations. When minimization is enabled, the switch sends a single event report on only one association and discards any remaining reports. It is up to the library or application to report this event to other interested parties or applications.

Expansion Module

Valid entries	Usage
y/n	Enter **y** if this phone has an expansion module. This will enable you to administer the buttons for the expansion module.

Extension

Displays the extension for this station—this is either the extension you specified in the station command or the next available extension (if you used add station next).

Feature Module

Enter **y** to indicate the station is connected to a feature module. Entering **y** displays an additional page to allow you to assign feature buttons to the module.

Fixed TEI

This field appears only for ISDN-BRI data modules and ASAI links. It indicates that the endpoint has a fixed Terminal Endpoint Identifier (TEI).
The TEI identifies a unique access point within a service. You must administer TEIs for fixed TEI terminals. However, for terminals with the automatic TEI capability, the system dynamically assigns the TEI.

Valid entries	Usage
y/n	Entering **y** displays the TEI field. For ASAI, enter **y**.

Floor

Enter a valid floor location.

H.320 Conversion

Allows H.320 compliant calls made to this phone to be converted to voice-only. Because the system can only handle a limited number of conversion calls, you may need to limit the number of telephones with H.320 conversion.

Headset

Enter **y** if the telephone has a headset.

HOT LINE DESTINATION — Abbreviated Dialing Dial Code

Appears only when Special Dialing Option is **hot-line.**

Use Hot Line Service when very fast service is required and when you use a telephone only for accessing a certain facility. Hot Line Service allows single-line telephone users, by simply lifting the handset, to automatically place a call to a preassigned destination (extension, telephone number, or feature access code).

The Hot Line Service destination number is stored in an Abbreviated Dialing List. When the user goes off-hook on a Data Hot Line call, the system dials the AD number.

A Direct Department Calling (DDC), a Uniform Call Distribution (UCD), a Terminating Extension Group (TEG) extension, or any individual extension within a group can be a Hot Line Service destination. Also, any extension within a DDC group, UDC group, or TEG can have Hot Line Service assigned.

Loudspeaker Paging Access can be used with Hot Line Service to provide automatic access to paging equipment.

HOT LINE DESTINATION — Abbreviated Dialing List Number

Enter the abbreviated dialing list where you stored the hotline destination number.

HOT LINE DESTINATION — Dial Code

Enter the dial code in the specified abbreviated dialing list where you stored the hotline destination number.

Hunt-to Station

Enter the extension the system should hunt to for this phone when the phone is busy. This field allows you to create a station hunting chain (by assigning a hunt-to station to a series of phones).

Idle Appearance Preference

Indicate which call appearance is selected when the user lifts the handset and there is an incoming call.

Valid entries	Usage
y	If you enter **y**, the user connects to an idle call appearance instead of the ringing call.
n	If you enter **n**, the Alerting Appearance Preference is set and the user connects to the ringing call appearance.

Ignore Rotary Digits

If this field is **y**, the short switch hook flash (50 -150) from a 2500-type set is ignored.

Valid entries	Usage
y	Enter **y** to indicate that rotary digits from the set should be ignored.
n	Enter **n** to make sure they are not ignored.

IP Audio Hairpinning

Allows IP endpoints to be connected through the IP circuit pack on the switch.

Valid entries	Usage
y/n	Enter y to allow IP endpoints to be connected through the IP circuit pack on the switch in IP format, without going through the DEFINITY TDM bus.

IP Emergency calls

Use this field to tell the switch how to handle emergency calls from the IP phone. This field appears when either the IP Softphone field or the Remote Office Station field is set to y on the Station screen.

CAUTION:
An Avaya IP endpoint can dial emergency calls (for example, 911 calls in the U.S.). It reaches solely the local emergency service in the Public Safety Answering Point area where the telephone system is located. Please be advised that an Avaya IP endpoint does not dial to and connect with local emergency service when dialing from remote locations. You should not use an Avaya IP endpoint to dial emergency numbers for emergency services when dialing from remote locations. Avaya Inc. is not be responsible or liable for any damages resulting from misplaced emergency calls made from an Avaya endpoint. Your use of this product indicates that you have read this advisory and agree to use an alternative telephone to dial all emergency calls from remote locations.

Valid entries	Usage
extension	Enter **extension** to send the extension entered in the Emergency Location Extension field, to the Public Safety Answering Point (PSAP).
block	Enter **block** to prevent the completion of emergency calls. Use this entry for users who move around but always have a circuit-switched phone nearby, and for users who are farther away from the switch than an adjacent area code served by the same 911 Tandem office. When users attempt to dial an emergency call from an IP Telephone and the call is blocked, they can dial 911 from a nearby circuit-switched phone instead.
cesid	Enter **cesid** to allow the switch to send the CESID information supplied by the IP Softphone to the PSAP. The end user enters the emergency information into the IP Softphone.

Use this entry for IP Softphones with road warrior service that are near enough to the switch that an emergency call routed over the switch's trunk reaches the PSAP that covers the switch.

If the switch uses ISDN trunks for emergency calls, the digit string is the telephone number, provided that the number is a local direct-dial number with the local area code, at the physical location of the IP Softphone. If the switch uses CAMA trunks for emergency calls, the end user enters a specific digit string for each IP Softphone location, based on advice from the local emergency response personnel.

option

Enter **option** to allow the user to select the option (extension, block, or cesid) that the user selected during registration and the IP Softphone reported. Use this entry for extensions that are swapped back and forth between IP Softphones and a phone with a fixed location.

The user chooses between **block** and **cesid** on the softphone. A DCP or IP phone in the office automatically selects **extension**.

IP Station

Appears only for DCP station types and IP Telephones.

Valid entries	Usage
y/n	Enter **y** indicate that this extension is either a PC-based multifunction station or part of a telecommuter complex with a call-back audio connection. The type of IP softphone depends on the value of the Media Complex Ext field.

ITC (Information Transfer Capability)

Indicates the type of transmission facilities to be used for ISDN calls originated from this endpoint. The field does not display for voice-only or BRI stations.

Valid entries	Usage
restricted	If you set to **restricted**, either restricted or unrestricted transmission facilities are used to complete the call. A restricted facility is a transmission facility that enforces 1's density digital transmission (that is, a sequence of 8 digital zeros are converted to a sequence of 7 zeros and a digital 1).
unrestricted	If you set to **unrestricted**, only unrestricted transmission facilities are used to complete the call. An unrestricted facility is a transmission facility that does not enforce 1's density digital transmission (that is, digital information is sent exactly as is).

Lock Messages

Valid entries	Usage
y/n	Enter **y** to restrict other users from reading or canceling the voice messages or retrieving messages via Voice Message Retrieval.

Loss Group

This field determines which administered 2-party row in the loss plan applies to each station. Does not appear for stations that do not use loss (for example, x-mobile stations and MASI terminals).

Valid entries	Usage
1-17	Shows the index into the loss plan and tone plans.

LWC Activation

Valid entries	Usage
y/n	Enter **y** to allow internal telephone users to leave short LWC messages for this extension. If the system has hospitality, enter **y** for guest-room telephones if the extension designated to receive failed wakeup messages should receive LWC messages that indicate the wakeup calls failed. Enter **y** if LWC Reception is **audix**.

LWC Log External Calls

Appears only where the LWC Reception field is available. When an external call is not answered, the switch keeps a record of up to 15 calls (provided information on the caller identification is available) and the phone's message lamp lights. The phone set displays the names and numbers of unsuccessful callers.

Valid entries	Usage
y/n	Enter **y** to make unanswered external call logs available to end-users. Each record consists of the latest call attempt date and time.

LWC Reception

Valid entries	Usage
audix	Enter **audix** if the messages are stored on the Audio Information Exchange System.
None	
msa-spe	Enter **msa-spe** if LWC messages are stored in the system.
msa	Enter **msa** if LWC messages are stored in the system or on the Messaging Server Adjunct - (for r models).

spe	Enter **spe** if LWC messages are stored in the system or on the Switch Processor (for r models).

Map-to Station

This is the physical phone used for calls to a virtual extension. Do not use an xmobile, xdid or another virtual extension.

Valid entries	Usage
Valid extension number	Enter the extension of the physical phone used for calls to a virtual extension.

Media Complex Ext

When used with Multi-media Call Handling, indicates which extension is assigned to the data module of the multimedia complex. Users can dial this extension to place either a voice or a data call, and voice conversion, coverage, and forwarding apply as if the call were made to the 1-number.

For an IP Telephone or an IP Softphone, this is the extension already administered as an H.323 station type. You can administer this field if the IP Station field on the System-Parameters Customer-Options screen is y.

Valid entries	Usage
A valid BRI data extension	For MMCH, enter the extension of the data module that is part of this multimedia complex
H.323 station extension	For 4600 series IP Telephones, enter the corresponding H.323 station. For IP Softphone, enter the corresponding H.323 station. If you enter a value in this field, you can register this station for either a road-warrior or telecommuter/Avaya IP Agent application.
blank	For IP Softphone, if you leave this field blank, the station can be registered only as a telecommuter/Avaya IP Agent application.

Softphone

Enter the extension of the station you want to track with the message waiting lamp. This field appears only when Type is **7101A**, **7103A**, **8110**, or **VRU**.

Message Server Name

Specifies which Message Server is associated with the station.

Valid entries	Usage
Names administered in alphabetical order	Must contain a user-defined adjunct name that was previously administered in the Node-Names screen.

Message Waiting Indicator

This field appears only for ISDN-BRI data modules and for 500, 2500, K2500, 7104A, 6210, 6218, 6220, 8110, and VRU telephones. (This field is independent of the administration of the Caller ID Message Waiting Indication for CallrID phones.)

Valid entries	Usage
led	Enter **led** if the message waiting indicator is a light-emitting diode (LED).
neon	Enter **neon** if the indicator is a neon indicator.

MIM Mtce/Mgt

Indicates if the telephone supports MIM Maintenance and Management capabilities other then endpoint initialization. Appears only if MIM Support is **y**.

MIM Support (Management Information Message Support)

This field appears only for ISDN-BRI data modules and ASAI. This field supports MIM endpoint initialization (SPID support) and other Maintenance or Management capabilities.

Valid entries	Usage
y	Enter **y** to display Endpt Init and MIM Mtce/Mgt.
n	Enter **n** for ASAI.

Multimedia Early Answer

Allows you to establish multimedia early answer on a station-by-station basis.

Valid entries	Usage
y/n	If this station will receive coverage calls for multimedia complexes, but is not multimedia-capable, enter **y** to ensure that calls are converted and talk path is established before ringing at this station.

Mute Button Enabled

Valid entries	Usage
y/n	Enter **y** to allow the user to use the Mute button on this phone.

MWI Served User Type

Controls the auditing or interrogation of a served user's message waiting indicator (MWI).

Valid entries	Usage
fp-mwi	Use if the station is a served user of an fp-mwi message center.
qsig-mwi	Use if the station is a served user of a qsig-mwi message center.
blank	Leave this field blank if you do not want to audit the served user's MWI or if the user is not a served user of either an fp-mwi or qsig-mwi message center.

Name

Enter a name for the person associated with this phone or data module. The system uses the Name field to create the system Directory.

Off Premises Station

Analog phones only.

Valid entries	Usage
y	Enter **y** if this phone is not located in the same building with the system. If you enter **y**, you must complete R Balance Network.
n	Enter **n** if the phone is located in the same building with the system.

PCOL/TEG Call Alerting

Appears only for 510 telephones.

Valid entries	Usage
y/n	Enter **y** to alert the station for Personal CO Line/Terminating Extension Group calls.

Per Button Ring Control

Valid entries	Usage
y	Enter **y** to allow users to select ring behavior individually for each call-appr, brdg-appr, or abrdg-appr on the station and to enable Automatic Abbreviated and Delayed ring transition for each call-appr on the station. Also, enter **y** if you do not want the system to automatically move the line selection to a silently alerting call unless that call was audibly ringing earlier.
n	Enter **n** if you want calls on call-appr buttons to always ring the station and calls on brdg-appr or abrdg-appr buttons to always ring or not ring based on the Bridged Call Alerting field value. Also, enter **n** if you want the system to move line selection to a silently alerting call if there is no call audibly ringing the station.

Personalized Ringing Pattern

Personalized Ringing allows users of some telephones to have one of 8 ringing patterns for incoming calls. Users working closely in the same area can each specify a different ringing pattern. This enables the users to distinguish their own ringing telephone from other telephones in the same area. For virtual stations, this field dictates the ringing pattern on its mapped-to physical phone.
Enter a Personalized Ringing Pattern. (L = 530 Hz, M = 750 Hz, and H = 1060 Hz)

Valid entries	Usage
1	MMM (standard ringing)
2	HHH
3	LLL
4	LHH
5	HHL
6	HLL
7	HLH
8	LHL

Per Station CPN - Send Calling Number

Valid entries	Usage
y	All outgoing calls from the station will deliver the Calling Party Number (CPN) information as "Presentation Allowed."
n	No CPN information is sent for the call.
r	Outgoing non-DCS network calls from the station will deliver the Calling Party Number information as "Presentation Restricted."

blank	The sending of CPN information for calls is controlled by any administration on the outgoing trunk group the calls are carried on.

Port

Enter 7 characters to specify a port, or an x. If this extension is registered as an IP softphone endpoint, this field will display S00000.

Valid entries	Usage
01 through **44** (G3r)	First and second numbers are the cabinet number
01 through **03** (G3si)	
A through **E**	Third character is the carrier
01 through **20**	Fourth and fifth characters are the slot number
01 through **32**	Sixth and seventh characters are the circuit number
x	Indicates that there is no hardware associated with the port assignment. Use for IP softphones and IP Telephones. Use for AWOH and CTI stations.

For DCP sets, the port can only be assigned once. ISDN-BRI provides a multipoint configuration capability that allows a previously assigned port to be specified more than once as follows: 2 stand-alone voice endpoints, 2 stand-alone data endpoints, or 1 integrated voice and data endpoint. However, for the following cases, the port is assumed to be fully assigned:

— Maximum number of users (currently 2) are assigned on the port.

— One of the users on the port is a fixed TEI station.

— One of the users on the port has B-channel voice and B-channel data capability.

— One of the users on the port has no SPID assigned, which includes telephones that have no SPID initialization capability.

— This field is display-only for H.323 set types and 4600 set types. The system assigns an "X" when the station is first administered.

R Balance Network

Valid entries	Usage
y	Enter **y** to select the R Balance Capacitor network. In all other cases except for those listed under **n**, enter **y**.
n	Enter **n** to select the standard resistor capacitor network. You must complete this field if Off-Premise Station is **y**. Enter **n** when the station port circuit is connected to terminal equipment (such as SLC carriers or impedance compensators) optioned for 600-ohm input impedance and the distance to the equipment from the system is less than 3,000 feet.

Recall Rotary Digit

This field only appears if type is 500 or 2500.

Valid entries	Usage
y/n	Enter **y** to allow the user of a rotary phone to dial the administered Recall Rotary Digit to receive recall dial tone. This will enable this user to perform conference and transfer operations. You establish the Recall Rotary Digit on the Feature-Related System Parameters screen.

Redirect Notification

Valid entries	Usage
y/n	Enter **y** to give a half ring at this phone when calls to this extension are redirected (via Call Forwarding or Call Coverage). Enter **y** if LWC Reception is **audix**.

Remote Office Phone

Valid entries	Usage
y/n	Enter y to use this station as an endpoint in a remote office configuration.

Restrict Last Appearance

Valid entries	Usage
y/n	Enter **y** to restrict the last idle call appearance for incoming priority calls and outgoing call originations only.

Room

Valid entries	Usage
Up to 10 characters	To identify the phone location.
Up to 5 characters	To identify the guest room number, if this station is one of several to be assigned a guest room and the Display Room Information in Call Display is **y** on the Hospitality-Related System Parameters screen.

Security Code

Enter the security code required by users for specific system features and functions, including Personal Station Access, Redirection of Calls Coverage Off-Net, Leave Word Calling, Message Retrieval, and Demand Printing. The required security code length is determined by Minimum Security Code Length on the Feature-Related System Parameters screen.

Select Last Used Appearance

Valid entries	Usage
y	Enter **y** to indicate a station's line selection is not to be moved from the currently selected line button to a different, non-alerting line button. If you enter **y**, the line selection on an on-hook station only moves from the last used line button to a line button with an audibly alerting call. If there are no alerting calls, the line selection remains on the button last used for a call.
n	Enter **n** so the line selection on an on-hook station with no alerting calls can be moved to a different line button, which may be serving a different extension.

Service Link Mode

The service link is the combined hardware and software multimedia connection between an Enhanced mode complex's H.320 DVC system and the DEFINITY ECS which terminates the H.320 protocol. When the user receives or makes a call during a multimedia or IP Softphone or IP Telephone session, a "service link" is established.

Valid entries	Usage
as-needed	Use this setting for most multimedia, IP Softphone, or IP Telephone users. Setting the Service Link Mode to as-needed leaves the service link connected for 10 seconds after the user ends a call so that they can immediately place or take another call. After 10 seconds the link is dropped and a new link would have to be established to place or take another call.

permanent	Use **permanent** for busy call center agents and other users who are constantly placing or receiving multimedia, IP Softphone, or IP Telephone calls. In permanent mode, the service link stays up for the duration of the multimedia, IP Softphone, or IP Telephone application session.

Set Color

Enter a valid set color as defined in the Signaling Group screen (page 2) screen. Valid entries include the following colors: beige, black, blue, brown, burg (burgundy), gray, green, ivory, orng (orange), red, teak, wal (walnut), white, and yel (yellow).

Speakerphone

This field controls the behavior of speakerphones for the 6400-series and 8400-series phones.

Valid entries	Usage
1-way	Enter **1-way** to indicate that you want the speakerphone to be listen-only.
2-way	Enter **2-way** to indicate that you want the speakerphone to be both talk and listen.
grp-listen	Group Listen works with only 6400-series phones. Group Listen allows a phone user to talk and listen to another party with the handset or headset while the phone's 2-way speakerphone is in the listen-only mode. Others in the room can listen, but cannot speak to the other party via the speakerphone. The person talking on the handset acts as the spokesperson for the group. Group Listen provides reduced background noise and improves clarity during a conference call when a group needs to discuss what is being communicated to another party.
none	

Special Character for Restricted Number

Appears when the Type field is **asai** and, on the System-Parameters Country-Options screen, the ASAI Link Core Capabilities field is **y**.

Valid entries	Usage
y/n	Enter **y** to allow an ASAI station to identify restricted numbers.

Special Dialing Option

This field identifies the type of dialing for calls when this data module originates calls.

Valid entries	Usage
hot-line	
default	
blank	For regular (normal) keyboard dialing.

SPID — (Service Profile Identifier)

Enter a variable length parameter. This field appears only if Endpt Init is **y**. The SPID is a numeric string, which means that the value of 00 is different from 000. The SPID must be different for all terminals on the BRI and from the Service SPID. The SPID should always be assigned. If the SPID is not assigned for the first BRI on a port, any other BRI assignment to that port are blocked.

NOTE:
If you have set the Port field to X for an ISDN-BRI extension and intend to use Terminal Translation Initialization (TTI) to assign the port, then the SPID number must equal the station number.

Station Lock Active

Valid display	Usage
yes/no	Shows the phone's status of Station Lock.

Switchhook Flash

Valid entries	Usage
y	Enter **y** to allow users to use the switchhook flash function to activate Conference/Transfer/Hold and Call Waiting.
n	Enter **n** to disable the flash function so that when the switchhook is pressed while active on a call, the call drops. If this field is **n**, you must set Call Waiting Indication to **n**.

TEI

Appears only when Fixed TEI is **y**.

Valid entries	Usage
0 to **63**	1- or 2-digit number

Tests

Valid entries	Usage
y	Enter **y** to enable port maintenance tests.
n	If the equipment (dictaphone) connected to the port does not support these tests, you must enter **n**.

TN

Enter the Tenant Partition number.

Type

For each station that you want to add to your system, you must specify the type of telephone in the Type field. This is how you distinguish between the many different types of telephones.
The following table lists the telephones, virtual phones, and personal computers that you can administer on the DEFINITY ECS. To administer telephones that are not in the table, use the Alias Station screen.

NOTE:
You cannot change an analog phone administered with hardware to a virtual extension if TTI is y on the Feature-Related System Parameters Customer Options screen. Contact your Avaya representative for more information.

XID

Appears only for an ISDN-BRI data module or an ASAI link. Used to identify Layer 2 XID testing capability.

Adding new phones

When you are asked to add a new phone to the phone system, what do you do first? To connect a new phone you need to do three things:

- find an available port
- wire the port to the cross-connect field or termination closet
- tell the telephone system what you're doing

Before you can determine which port to use for the new phone, you need to determine what type of phone you are installing, what ports are available, and where you want to install the phone.

Gathering necessary information

1. Determine whether the phone is an analog, digital, ISDN, or hybrid set.

You need this information to determine the type of port you need, because the port type and phone type must match.

2. Record the room location, jack number, and wire number.

You may find this information on the jack where you want to install the phone, recorded in your system records, or from the technician responsible for the physical installation.

3. Display the available boards (cards) and ports.

To view a list of boards on your system, type **list configuration station** and press RETURN.

The System Configuration screen shows all the boards on your system that are available for connecting phones. You can see the board number, board type, circuit-pack type, and status of each board's ports.

4. Choose an available port and record its port address.

Each port that is available or unassigned is indicated by a 'u.' Choose an available port from a board type that matches your phone type (such as a port on an analog board for an analog phone).

Every phone must have a valid port assignment, also called a port address. The combined board number and port number is the port address. So, if you want to attach a phone to the 3rd port on the 01C05 board, the port address is 01C0503 (01=cabinet, C=carrier, 05=slot, 03=port).

5. Choose an extension number for the new phone.

The extension you choose must not be assigned and must conform to your dial plan. You should also determine whether this user needs an extension that can be directly dialed (DID) or reached via a central phone number.

Be sure to note your port and extension selections on your system's paper records.

Completing the station screens

The information that you enter on the station screen advises the system that the phone exists and indicates which features you want to enable on the phone.

Swapping phones

You will often find that you need to move or swap phones. For example, employees moving from one office to another may want to bring their phones. In this case, you can use X ports to easily swap the phones.

In general, to swap one phone (phone A) with another phone (B), you change phone A's port assignment to **x**, change phone B's port assignment to A's old port, and, finally, change the **x** for phone A to B's old port. Note that these swapping instructions work only if the two phones are the same type (both digital or both analog, etc.).

For example, to swap phones for extension 4567 (port 01C0505) and extension 4575 (port 01C0516), complete the following steps:

1. Type **change station 4567** and press RETURN.
2. Record the current port address (01C0505) and type **x** in the Port field.
3. Press ENTER to save your changes.
4. Type **change station 4575** and press RETURN.
5. Record the current port address (01C0516).
6. Type **01C0505** in the Port field.
7. Update the Room and Jack fields.
8. Press ENTER to save your changes.
9. Type **change station 4567** again and press RETURN.
10. Type **01C0516** in the Port field.

    This is the port that used to be assigned to extension 4575.
11. Update the Room and Jack fields.
12. Press ENTER to save your changes.
13. Physically unplug the phones and move them to their new locations.

When you swap phones, the system keeps the old button assignments. If you are swapping to a phone with softkeys, the phone could have duplicate button assignments, because softkeys have default assignments. You may want to check your button assignments and modify them as necessary.

Using TTI to move phones

Terminal Translation Initialization (TTI) allows you to merge an x-ported station to a valid port by dialing a TTI merge code, a system-wide security code, and the x-port extension from a telephone connected to that port. TTI also allows you to separate an extension from its port by dialing a similar separate digit sequence. This action causes the station to revert to an x-port.

TTI can be used for implementing telephone and data module moves from office to office. That is, you can separate a telephone from its port with TTI, unplug the telephone from the jack, plug in the telephone in a jack in a different office, and merge the telephone to its new port with TTI.

Instructions

TTI merge from a voice TTI port

! CAUTION:

You can destroy your hardware if you attempt to connect an analog telephone to a digital port.

To merge an extension to a telephone with TTI, complete the following steps from the telephone you want to merge:

1. Dial the TTI merge FAC.

—If the code is correct, you receive dial tone.

—If the code is not correct, you receive intercept (Turkey) tone (Nerd Alert).

2. Dial the TTI security code from the telephone you want to merge.

—If the code is correct, you receive dial tone.

—If the code is not correct, you receive intercept tone.

3. Dial the extension of the telephone you want to merge.

—If the extension is valid, you receive confirmation tone, which may be followed by dial tone. (It is possible to receive intercept tone immediately following the confirmation tone. If this happens, you need to attempt the merge again.)

Alias Station

This screen allows you to configure the system so that you can administer new phone types that are not supported by your system software. This screen maps new telephone models to a supported telephone model. This mapping does not guarantee compatibility, but allows nonsupported models to be administered and tracked by their own names

```
change alias station
                                                          Page 1 of 1
                              ALIAS STATION
                     Alias Set Type    Supported Set Type
                          _____             _____
                          _____             _____
                          _____             _____
                          _____             _____
                          _____             _____
                          _____             _____
                          _____             _____
                          _____             _____
                          _____             _____
                          _____             _____
          '#' indicates previously aliased set type is now native
```

Adding feature buttons

Once you add a phone to the system, you can use the station screen to change the settings for the phone, such as adding or changing feature button assignments. The system allows you to assign features or functionality to each programmable button. It is up to you to decide which features you want for each phone and which feature you want to assign to each button.

Instructions

To assign feature buttons:

1. Type **change station *nnnn*** and press ENTER, where *nnnn* is the extension for the phone you want to modify.

   The Station screen appears.

2. Press NEXT PAGE until you locate the Feature Button Assignment fields.

   Some phones have several feature button groups. Make sure that you are changing the correct button.
3. Move the cursor to the field you want to change.
4. Type the button name that corresponds to the feature you want to add.

   To determine feature button names, press HELP.
5. Press ENTER to save your changes.

Features and technical reference

AAR and ARS partitioning

You can use Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) partitioning to change the call routing plan for up to 8 different user groups within a single DEFINITY ECS. You assign a Partition Group Number (PGN) to each user group and identify different call routing treatment for each PGN.

For example, you can partition hotel employees and guests into separate groups (PGN) and route the calls differently. When a guest makes a long-distance call, the guest's PGN and digit analysis tables route the call to a telephone-billing system that allocates long-distance charges. A similar call placed by an employee routes over a direct-distance dialing (DDD) trunk.

Detailed description

Partition user groups are used only with AAR, and ARS, and Uniform Dial Plan (UDP). You can assign AAR and ARS partitioning to phones, attendant consoles, remote-access users, data endpoints, and incoming trunks.

Use partitioning for:

- groups with different routing due to special billing needs
- groups that have dedicated use of a particular network facility
- groups in different businesses serviced by a single system
- data users who require special facility types on outgoing calls

You can assign a route pattern to just one partitioned user group or you can assign a route pattern to all your partitioned user groups.

You assign the PGN on the Class of Restriction (COR) screen, and then assign the COR on each station screen. When a user dials an AAR or ARS feature access code and a number, the switch uses the PGN of the caller's COR to determine the route pattern. The PGN field appears on the COR screen only if Time of Day Routing is **n** on the System Parameters Customer Options screen.

If Time of Day Routing is **y** on System Parameters Customer Options, you specify the partition group number (PGN) on the Time of Day Routing Plan screen.

Interactions

- Bridged Call Appearance

  If a Bridged Call Appearance is used for an AAR or ARS call, the system uses the bridged extension's PGN instead of the caller's PGN.

- DCS

  When a call routes over DCS, the far-end switch uses the incoming trunk's PGN to route the call.

- Remote Access

  When a remote-access user dials barrier code or authorization code and an ARS feature access code, the barrier code or authorization code's COR determines the PGN.

- Straightforward Outward Completion and Through Dialing

  If the attendant assists or extends a call and dials an ARS feature access code, the attendant's COR determines the PGN if the individual extension is assigned. Otherwise, the COR set on the console parameter determines the PGN.

AAR/ARS shortcut dialing

(Not available with Offer B) Use AAR/ARS shortcut dialing to modify your dial plan and expand the capabilities of automatic routing. With shortcut dialing, users can make AAR and ARS calls without dialing the Feature Access Code (FAC), usually 8 for AAR and 9 for ARS in the US. Dial plans with 5-digit extensions can be expanded to include 6- or 7- digit extensions (without DCS transparency), and you can use a 6- to 7-digit private network dial plan to convert a large DCS network to QSIG. You can apply shortcut dialing to dial plans that use 2-digit through 20-digit dialing, depending on your digit analysis tables.

Detailed description

Shortcut dialing must be enabled on the System Parameters Customer Options screen, along with either AAR (private networking), ARS, or both. See your Avaya representative for more details.

Shortcut dialing simplifies dialing in the following ways:

- Public-network dialing (AAR) — The feature access code (usually 9 in the U.S.) does not need to be dialed to access an outside line.
- 5-digit Uniform Dial Plan networks — You can add 6- or 7-digit numbers to a 5-digit Uniform Dial Plan (UDP) in a DCS network to provide additional extensions to your network.

NOTE:

Network nodes that are accessed using shortcut dialing (6 or 7 digits) in a 5-digit UDP lose DCS transparency.

- QSIG networks — Providing a 6- to 7-digit private-network dialing plan makes it easier to convert a large DCS network to QSIG. You can implement QSIG networks with all the QSIG Supplementary Services installed in your network. It also makes it easier to integrate DEFINITY ECS into existing networks composed of non-Avaya systems.
- 3-digit Uniform Dial Plan — This feature provides a 3-digit dialing scheme similar to uniform dial plan.

NOTE:
DCS will not work using a 3-digit dialing plan. 3-digit dialing requires QSIG for feature transparency.

If the caller dials enough digits to match the **aar** or **ars** field entry, the dialed digits are analyzed as though the AAR or ARS FAC had been dialed.

Displays

On display phones, the internal extension dialed appears instead of the shortcut number. Malicious Call Trace (MCT) displays the short, internal extension of the internal caller. On display phones with directory service, the CALL DISPLAY button cannot be used in networks where 6 or more digits are required for internal calls.

Emergency numbers

Shortcut dialing allows users to make emergency calls (for example, 911 in the U.S. and 112 in Europe) without first dialing an access code.When this is set up correctly, other 4- or 5-digit extensions that begin with the same leading digit (9 in the U.S. and 1 in Europe) cannot be dialed.

Interactions

- Abbreviated Dialing

  You can store AAR/ARS shortcut numbers on Abbreviated Dial buttons.

- Attendant direct extension selection with busy lamp

  On third-party call forwarding, attendants must dial the full extension of the phone to be forwarded, and may use the Direct Extension Selection (DXS) button and the shortcut dialing number for the destination phone.

- Attendants cannot use DXS modules in a network where users are required to dial 6 or more digits for internal calls.

- AUDIX mailboxes allow a maximum of 5 digits when AUDIX is connected to a switch that is not administered as a central office. If an internal extension is 6 or more digits, the corresponding voice mailbox must be set up and accessed with 5 or less digits. AUDIX cannot associate a subscriber's name with 6- or more digit number.

NOTE:
If AUDIX is connected to a switch in a remote location or administered as a central office, mailbox IDs can be up to 10 digits and AUDIX can look up a calling party number longer than 5 digits.

- DCS AUDIX (Centralized Messaging)

  With shortcut dialing, DCS is not used and AUDIX does not associate the name with the calling number. AUDIX associates the name with the calling number only when DCS is used in the network.

- ISDN - QSIG/AUDIX

  A QSIG/AUDIX network can perform name lookup on 3-, 4-, and 5-digit calling numbers, as well as on longer calling numbers that are be truncated to a unique 3-, 4-, or 5-digit mailbox extension.

- AUDIX transfer

  Shortcut dialing allows AUDIX Enhanced Transfer to a 3-digit extension, but not to a 6- or 7-digit extension.

- Coverage paths for call coverage

  Use the short, internal extensions on local coverage paths, even when you must dial 6 or more digits for on-switch calls.

- Call center

  To call an agent, users must dial the agent's shortcut dialing number. However, the agent must use the agent ID (extension), not the shortcut dialing number, to login.

- Call Detail Recording (CDR)

  Intra-switch CDR records show the short, internal extensions of the calling and called parties when shortcut dialing is used.

  Because the AAR or ARS access code is not used in shortcut dialing, the Access Code Dialed and Access Code Used fields recorded

by CDR are blank for trunk calls made with shortcut dialing.

- When a caller dials a CDR account code before the shortcut dialing number, the CDR account code is recorded in CDR.

- Call Management Systems (CMS)

  Because the AAR or ARS access code is not used in shortcut dialing, there is no feature access code on CMS records to indicate outgoing trunk calls.

- Call vectoring

  In a 5-digit dialing plan, a message step in a vector cannot be updated to reflect a 6-digit shortcut number.

- Coverage of Calls Redirected Off Net

  You can use shortcut dialing for an off-net destination on a coverage path. On networks where 6 or more digits are required, you can use the Remote Call Coverage Table even for on-switch coverage points. On a large system, however, you may reach the maximum number of remote coverage points.

- Distributed Communications System (DCS)

  DCS feature transparency is lost on a shortcut dialing call.

- Deluxe paging

  Dial the full extension to send a deluxe page. The page may be answered with a shortcut dialing number.

- Interdigit timers

  The short interdigit timer tracks calls to internal extensions if

- AAR/ARS shortcut numbers and internal extensions share the same first digit extensions are shorter than the minimum-length shortcut number

Abbreviated Dialing

Abbreviated Dialing (AD) provides easy access to selected numbers by reducing the number of digits users have to dial to place a call. Instead of dialing the entire number, the user dials a short code to access the number. The system then dials the stored number automatically. You can assign abbreviated dialing buttons to phones, so users can dial frequently-called numbers by pressing a single button.

You can also assign privileged numbers to abbreviated dialing lists, so you can allow a user to place calls to numbers that might otherwise be restricted.

List types

You can store phone numbers in 4 different types of abbreviated dialing lists:

personal, group, system, enhanced.

Your switch type and version determines which lists you have available and how many entries you can have on each list. You can assign up to 3 AD lists to each user (extension). The 3 lists can be made up of any combination of a system list, an enhanced list, up to 3 personal lists, or up to 3 group lists. Each abbreviated dialing entry can have up to 24 characters.

Personal lists

You can provide personal lists to phone users who need their own set of stored numbers. You determine which users have access to a personal list and the size of each list. Either you or the user can assign phone numbers to personal lists. A personal list is created automatically when you assign the list to an individual phone. Each user can have up to 3 personal lists. Note that you cannot assign personal list to the attendant.

Group lists

You can define group lists for groups or departments, such as purchasing or human resources, where members of the group have the need to frequently dial the same numbers. You determine which users have access to group lists and each user may have access to up to 3 group lists. You can program the list or you can designate a user in each group to program the list. You specify this designated user on the Abbreviated Dialing Group List screen.

System lists

You can define one system list for the entire organization. Most administrators assign this list to every phone and allows everyone in the organization to use the list. If you choose to let everyone use the system list, you should only add numbers to the list that anyone in your organization may call. For example, you may want to add an emergency phone number or phone numbers for other office locations to this list.

The system list can contain up to 100 entries and can only be changed by a system administrator.

Enhanced lists

Enhanced-number lists are used by telephone users, data-terminal users, and attendants who need more list entries than those allowed in group-number and system-number lists. One enhanced-number list is allowed per system in addition to the system-number list. The enhanced-number list can contain any number or dial-access code. You administer the enhanced-number lists and determine which users can access the list.

Considerations

- You cannot remove a telephone or attendant if it is designated as the extension number that is permitted to program a group-number list.
- When using an AD button to access a messaging system, the user's login and password should not be assigned to the button. The system ignores button entries after the messaging access number.
- You can use an abbreviated dialing list at any time during incoming or outgoing calls.

Interactions

- Last Number Dialed

  The Last Number Dialed feature redials the same number a user just dialed, even if the user accessed an abbreviated dialing list for the previous call. However, if the last dialed string includes any special characters (such as indefinite wait,

mark, pause, suppress, or wait) these characters are ignored by last-number-dialed call.

If the previously-called number was in an AD privileged list, and if the user is not normally allowed to dial the number because of his or her class of restriction, they cannot redial the number using Last Number Dialed. To redial the number, the user must again access the AD privileged list.

Attendant Features

Attendant Call Waiting

Attendant Call Waiting allows an attendant-originated or attendant-extended call to a busy single-line telephone to wait at the called telephone so that the attendant can handle other calls.

If you want the attendant to be able to send calls to busy single-line phones, set the Att. Call Waiting Indication field to Y on the Station screen for each single-line phone.

When the single-line phone receives a waiting call, the phone user hears a call-waiting signal. You can administer the number of bursts (1, 2, or 3) in the call-waiting signal by changing the Attendant Originated Calls field on the Feature-Related System Parameters screen.

If the attendant activates Attendant Call Waiting, and the Timed Reminder on Hold interval or the Return Call Timeout interval expires without the call being answered, the call returns to the console. You can modify these intervals on the Console Parameters screen.

Interactions

- Automatic Callback

  Activating Automatic Callback at the called telephone denies Attendant Call Waiting.

- Call Coverage

Attendant Call Waiting calls redirect to coverage if the called phone has Data Privacy or Data Restriction activated. If one of these conditions exists, and you assign call coverage to a telephone, and the user activates Send All Calls or coverage criteria is met, the call redirects to coverage.

— The Coverage Don't Answer interval specifies how long a call remains directed to the called telephone before redirecting to coverage. Attendant Call Waiting if applicable on the call, is active for the duration of the Don't Answer interval only. At the end of this interval, the call redirects to coverage.

  — If the Return Call Timeout (Timed Reminder) interval 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, but can still return to the console if a coverage point does not answer the call before the Return Call Timeout.

  — If the Station Hunting field is assigned and the called telephone is busy, the call redirects to the Hunt To Station Assignment.

- Data Privacy, Data Restriction

  Activating Data Privacy or Data Restriction at the called telephone denies Attendant Call Waiting.

- DDC and UCD

  Calls to a DDC or UCD group do not wait. However, they can enter the group queue, if provided.

- Loudspeaker Paging Access

  Activating Loudspeaker Paging Access at the called telephone denies Attendant Call Waiting.

- Music-on-Hold Access

  The calling party hears music if the call is a trunk-transferred call administered to receive Music-on-Hold. Otherwise, the calling party hears ringing.

- Recorded Telephone Dictation Access

  Activating Recorded Telephone Dictation Access at the called telephone denies Attendant Call Waiting.

Attendant Control of Trunk Group Access

Attendant Control of Trunk Group Access allows the attendant to control trunk groups, and prevents telephone users from directly accessing a controlled trunk group. The attendant gains direct access to an outgoing trunk group by pressing the button assigned to that trunk group.

Each attendant console has 12 designated Trunk Hundreds Select buttons that can be administered for Attendant Control of Trunk Group Access. You can also administer each console with up to 12 feature buttons for Trunk Hundreds Select buttons, which gives you up to a total of 24 buttons.

Each Trunk Hundreds Select button has busy lamps that light when all the members of the associated trunk group are busy. If you administer one of the 2-lamp feature buttons on a basic console as a Trunk Hundreds Select button, use the bottom lamp as the busy lamp. These buttons have 2 additional lamps for Attendant Control of Trunk Group Access. The 2 lamps are:

- Warn (warning) lamp

  Lights when the administered number of trunks are busy in the associated trunk group. You administer the Busy Threshold field on the associated Trunk Group screen to determine when to light this warning lamp.
- Cont (control) lamp

  Lights when the attendant activates Attendant Control of Trunk Group Access for the associated trunk group. Assign act-tr-grp and deact-tr-g buttons on the Attendant Console screen to allow the attendant to activate and deactivate control of the trunk group access.

Interactions

- Authorization Codes

  Authorization codes do not collect when a trunk group has an incoming destination set to the attendant.
- Automatic Route Selection and Automatic Alternate Routing (ARS/AAR) Activating Attendant Control of Trunk Group Access removes the controlled trunk groups from the ARS and AAR patterns. Deactivating the feature reinserts the groups into the patterns. ARS calls do not route to the attendant.
- QSIG

  QSIG trunks do not support Attendant Control of Trunk Group Access.
- Uniform Dial Plan

  Activating Attendant Control of Trunk Group Access removes the controlled trunk groups from preferences. Deactivating the feature enables the UDP to access the trunk groups.

Attendant Direct Extension Selection

Attendant Direct Extension Selection (DXS) with busy lamp field allows the attendant to track extension status (idle or busy) and to place or extend calls to extension numbers without having to dial the extension.

Standard DXS Tracking

The basic selector console has 8 Hundreds Select buttons and 100 DXS buttons. The enhanced selector console has 20 Hundreds Select buttons and 100 DXS buttons. You can assign 12 additional Hundreds Select buttons to feature buttons on the attendant console.

However, as you assign these feature buttons, note that the total number of Hundreds Select buttons per attendant (including both attendant-console feature buttons and selector-console buttons) cannot exceed 20.

Enhanced DXS Tracking

Enhanced DXS Tracking can help you if you have more than 100 telephones, but you use a console that does not have Hundreds Select buttons administered. It can also help if you have more telephones than you do Hundreds Select buttons (and thus have hundreds groups that are administered with Hundreds Select buttons).

To use Enhanced DXS, assign a Group Select button on the Attendant Console screen. This button allows the attendant to track and extend calls to telephones that do not have associated Hundreds Select buttons. You can not use Enhanced DXS Tracking if your extensions have fewer than three digits.

Group Display button

You can administer a Group Display button on the Attendant Console screen to help the attendant track extension status. When the attendant presses this button, the system displays the range of extensions currently tracked by the selector console. Administer the Group Display button for either the feature area or the display area of the console.

If the attendant selects this button, the system identifies the digits associated with a Hundreds Select button — unless it finds no Hundreds Select button is lit, in which case it identifies the digits last entered with the Group Select button. The system continues to track the selected group of extensions until the attendant selects a new group of extensions.

Attendant Intrusion

The attendant intrusion (Call Offer) button allows an attendant to intrude on an existing call to offer a new call or message to the intruded party.

When the attendant releases the intruded call, the source party waits at the intruded party's analog telephone or holds on an available line appearance on a digital telephone.

Interactions

- Intrusion is denied in the following cases:

  A telephone is on a conference call with administered maximum number of conferees

  A call is established with Data Privacy activated

  Establish a call with Data Restriction activated

  A telephone is a forward-to point of another telephone

  A telephone is busy talking to another attendant
- If a call is already call waiting for the intruded party, the source (split from attendant) party cannot wait for the intruded party using Call Waiting.
- The attendant display shows the character '1 wait' or '1 busy' if an intrusion is possible. Otherwise, the display shows 'wait' or 'busy'.
- The system provides Attendant Intrusion on remote telephones via TGU/TGE trunks (Italy only).

Attendant Override of Diversion Features

Attendant Override of Diversion Features (override button) allows an attendant to bypass call-diversion features activated by a called extension. A diversion feature is any feature that, when activated, causes a call to redirect from the called telephone. Send All Calls, Call Coverage, and Call Forwarding are diversion features.

You should explain to your attendants that they can use this feature with the Attendant Intrusion to place an emergency or urgent call to a telephone user.

Attendant Serial Calling

Attendant Serial Calling enables the attendant to transfer trunk calls that return to the same attendant after the called party hangs up. Once outside callers reach an attendant, they can use the same line into the switch for multiple calls. Attendant Serial Calling is useful if trunks are scarce and Direct Inward Dialing services are unavailable.

To allow your attendant to use serial calling, assign a serial-cal button on the Attendant Console screen. The Attendant Serial Calling feature is valid only on calls that have only one trunk on the connection.

You can define a priority queue for Serial Calls on the Console Parameters screen.

Interactions

- Centralized Attendant Services

  Attendant Serial Calling does not work with Centralized Attendant Services.

- DCS

  Attendant Serial Calling works in a DCS environment only if the attendant activates it on the same node as the trunk to which the attendant is connected. Do not conference the incoming trunk call with a DCS party when activating. This would put two trunks on the connection.

Attendant Vectoring

Attendant Vectoring allows you to establish an attendant vector directory number (VDN) and send attendant group calls through vector processing. This is useful when you want more flexibility with how calls are routed when the system is in Night Service mode. For more information, see *DEFINITY ECS Call Vectoring/EAS Guide*.

Auto Start and Don't Split

Auto Start allows the attendant to initiate a call by pressing any key on the keypad without having to first press the Start button.

If an attendant enables Auto Start and dials an AAR number where the min and max in the AAR analysis table are not equal, the attendant must dial a **#** after the digit string or the call cannot process.

You can assign a Dont-Split button on the Attendant Console screen which allows attendants to deactivate Auto Start. To deactivate

Auto Start, the attendant presses the Don't Split button. When Don't Split is active, keys pressed on the keypad are heard by the parties on the call.

To reactivate Auto Start, and allow end-to-end signaling, the attendant again presses the Don't Split button, presses Cancel, or lets the current call terminate.

Interactions

- CDR — Account Code Dialing

  If the system is using Call Detail Recording Account Code Dialing, Auto Start and Don't Split is not activated.

- Visually Impaired Attendant Service

  If VIAS is activated or deactivated while Don't Split is active, Don't Split deactivates.

Attendant Timers

Attendant timers automatically alert the attendant after an administered time interval. The attendant can reenter the call and decide whether to terminate the call or permit the waiting to continue. You administer the timers on the Console Parameters screen.

Attendant Timers include:

- Unanswered DID Call Timer — Specifies how long a DID call can go unanswered before it routes to the administered DID/TIE/ISDN Intercept Treatment.
- Attendant Return Call Timer — For unanswered calls that were extended by the attendant, they are returned to the same attendant who released them if the attendant is available. Otherwise they return to the attendant-group queue. The Attendant Return Call Timer is not set for calls extended from one attendant to another individual attendant. A transferred call that times out redirects to an attendant after an interval equal to the Attendant Return Call timer.
  - Attendant Timed Reminder of Held Call Timer — Specifies how long a call is held. When the timer expires, the held call alerts the attendant. The message hc appears on the

- attendant display. You can administer either a high-pitched ring or a primary alert.
- Attendant No-Answer Timer — Specifies how long a call that terminates at an attendant console can ring with primary alerting. When the call reaches this interval setting, it rings with a secondary, higher-pitch ring. A disabled Attendant No Answer Timer's ringing pattern does not change over from the primary to the secondary pattern. If the call remains unanswered during this interval, it routes to the attendant group and console where the call was placed in a Position Busy state. This feature does not apply to calls placed to the attendant's extension or to calls originated by the attendant.
- Attendant Alerting Interval (Timed Reminder) — Specifies how long a call that terminates at an attendant console can ring with secondary alerting. When the call reaches this interval, the attendant console is placed into position busy mode and the call forwards to the attendant group. If the console where the alerting interval is reached is the last active day console, then the system goes into night service if night service is enabled. This feature does not apply to calls placed to the attendant's extension or to calls originated by the attendant.

  You can disable the alerting interval. In this case, a call continues to ring at the original attendant's extension until the caller hangs up or another feature disconnects the call (for example, reaching the timeout limit for unanswered DID calls during night service.)

- Line Intercept Tone Timer — Specifies how long line intercept can be. For example: LITT:10 seconds means that line intercept stops after 10 seconds.

Interactions

- Call Coverage

  If a telephone user transfers a call to an on-premises telephone and the call remains unanswered at the expiration of the Timed Reminder Interval, the call redirects to an attendant. Redirection occurs even if the call redirects via Call Coverage or Call Forwarding from the transferred-to telephone.

An attendant-extended call redirects to coverage instead of returning to an attendant if the coverage criteria are met before the Timed Reminder Interval expires. However, unanswered calls return to an attendant at the expiration of the interval.

If a call alerts an attendant as a coverage call (unanswered station-to-station call with the "attd" (attendant) in the called telephone's coverage path screen), the secondary alerting tone does not sound.

- Centralized Attendant Service

  If an attendant at the main location transfers a call from a branch location to an extension at the main location, the timed reminder does not apply and the call does not return to the attendant if unanswered.

Audible Message Waiting

Audible Message Waiting places a stutter at the beginning of a station dial tone on a station that has a message waiting. Audible Message Waiting is particularly useful for visually impaired people who may not be able to see a message light.

Messages for a station can be waiting in system memory (to be accessed via display or voice synthesizer), Property Management System (PMS), Message Servicing Adjunct (MSA), or AUDIX. When the system loses synchronization between telephones and message-status data, use Clear Message Waiting Indicators to turn off message-waiting indicators.

You typically assign Audible Message Waiting on phones without message-waiting lights, such as analog telephone Audible Message Waiting requires a separate software right-to-use fee. Audible Message Waiting may not be applicable in countries that restrict the characteristics of dial tones provided to users.

Automated Attendant

Automated Attendant uses vector commands to allow a caller to enter the extension of the party that he or she would like to reach. The call is routed by the vector to that extension.

Refer to *DEFINITY ECS Call Vectoring/EAS Guide* for a detailed description of Automated Attendant and for a sample vector that can be used for Automated Attendant. The guide contains information that is critical to the effective and efficient use of Automated Attendant.

You can administer any display-equipped phone or attendant console with a Caller Information CALLR-INFO button. The button displays digits collected for the last **collect digits** command.

Automated Attendant competes with several features for ports on the call classifier — detector circuit pack or equivalent.

Interactions

- AUDIX

Automated Attendant gives the caller the option of leaving a message or waiting in queue for an attendant. Refer to "Message Collection" in Chapter 5 of the *DEFINITY Enterprise Communications Server Call Vectoring/EAS Guide*.

- Authorization Codes

If authorization codes are enabled, and a *route-to* command in a prompting vector accesses AAR or ARS, if the VDN's FRL does not have the permission to use the chosen routing preference, then the system does not prompt for an authorization code and the *route-to* command fails.

- CallVisor ASAI

ASAI-provided digits can be collected by the Call Vectoring feature via the *collect* vector command as dial-ahead digits. CINFO is passed to CallVisor ASAI.

- Hold

If a call is put on hold during the processing of a *collect* command, the command restarts, beginning with the announcement prompt, when the call is taken off hold. All dialed-ahead digits are lost. Similarly, if a call to a vector is put on hold, vector processing is suspended when a *collect* command is encountered. When the call becomes active, the *collect* command resumes.

- Inbound Call Management (ICM)

You can use Automated Attendant to collect information that may later be used by an adjunct to handle a call.

- Transfer

If a call to a VDN is transferred during a *collect* command, the *collect* command restarts when the transfer is complete, and all dialed-ahead digits are lost. Similarly, if a call to a vector is transferred, vector processing is suspended when a *collect* command is encountered. When the transfer is complete, the *collect* command resumes. Attendant extended calls do suspend vector processing in the same way as transferred calls.

Automatic Callback

Automatic Callback allows internal users who placed a call to a busy or unanswered internal telephone to be called back automatically when the called telephone becomes available.

When a user activates automatic callback, the system monitors the called telephone. When the called telephone becomes available to receive a call, the system originates the automatic callback call. The originating party receives priority ringing. The calling party then lifts the handset and the called party receives the same ringing provided on the original call.

A single-line telephone user activates this feature by pressing the Recall button or flashing the switchhook and then dialing the automatic callback access code. A single-line user can activate automatic callback for only one call at a time.

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

If the calling telephone user answers an automatic callback call, and for some reason the called extension cannot accept a new call, the calling user hears confirmation tone and then silence. The call is still queued.

Users cannot activate automatic callback for calls to:

- A telephone assigned Termination Restriction
- An extension with automatic callback already activated toward it
- A data terminal (or data module)
- An attendant console group
- A Terminating Extension Group

- An extension for a hunt group, split, or skill
- An EAS agent's Login ID
- A VDN Extension

Automatic circuit assurance

Automatic circuit assurance (ACA) helps you identify possible trunk malfunctions. With ACA enabled, the system measures the holding time of each trunk call. If the measurements show calls with either extremely long or extremely short holding times, DEFINITY ECS places a referral call to an attendant or telephone.

The system records holding time from when a trunk is accessed to when it is released. You set short-holding-time and long-holding-time limits for each trunk group. The system then compares the recorded holding times against these limits.

You enable ACA for the entire system, and administer thresholds for individual trunk groups. You can have all trunks or only certain trunks measured.

DEFINITY ECS deals with long-holding and short-holding calls differently. For every call that is shorter than the administered short-holding time, the system increases the short-holding counter by 1. For calls over the same trunk that are within the normal range, it decreases the short-holding counter by 1. Thus, trunks that handle a normal variety of call lengths are not singled out as faulty. If the counter reaches the administered short-holding threshold, the system places a referral call.

If one long call exceeds the long-holding time, the system makes a referral call.

You cannot measure personal CO lines, out-of-service trunks, or trunks undergoing maintenance testing.

The referral call

The display or voice-synthesized message that accompanies an ACA call contains the following information:

- The fact that this is an ACA call
- The trunk access code, trunk group number, and trunk group member number
- The type of referral (short or long holding time)

Once the referral call is answered, this information is displayed and remains displayed until the call is released. If the call is not answered within three minutes, the call stops. The system places the call again after one hour, and continues to place the call hourly until someone answers.

The attendant or telephone user who receives the referral call can stop further calls by pressing the aca-halt button, if one is provided. This is a toggle button, and turns off the feature until the user presses the button again.

Automatic routing — general

DEFINITY ECS automatically routes outgoing calls using the most preferred (normally the least expensive) route available at the time the call is placed. Generally, Automatic Alternate Routing (AAR) routes calls over a private network and Automatic Route Selection (ARS) routes calls using the public network numbering plan. However, both AAR and ARS support public and private networks.

AAR

AAR routes calls over private networks. When a user dials the AAR feature access code (normally 8 in North America) and phone number, AAR selects the least expensive route for the call in the private network and performs any digit conversion. If the first-choice route is not available, another route is chosen automatically.

AAR routes private-network numbers, public-network numbers, service codes, an international number, operator access code, or an operator-assisted dialing number. AAR routes calls route as far as possible over the private network, and then accesses the public network. This saves long-distance charges and allows you to use your private network as much as possible.

ARS

ARS routes calls over the public network. When a user dials the ARS feature access code (normally 9 in the US and 0 outside of the US) and phone number, ARS selects the least expensive route for the call when there are one or more long-distance carriers or services.

ARS, like AAR, routes private-network numbers, public-network numbers, service codes, an international number, operator access code, or an operator-assisted dialing number, and also routes to Inter-exchange carriers (IXC). These are your long-distance providers.

You can route ARS calls to a variety of types of public-network and private-network trunk groups including Central Office (CO), Foreign Exchange (FX), Integrated Services Digital Network (ISDN), Tie, and Wide Area Telecommunications Service (WATS).

Bridged Call Appearance

Bridged Call Appearance allows single-line and multiappearance telephone 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.

An appearance of a telephone's primary extension number at another telephone is called a bridged call appearance. A bridged call appearance can be used to originate, answer, or bridge onto an existing call to or from the primary telephone user's extension number.

On single-line telephones, Bridged Call Appearance is used by going off-hook. On multiappearance telephones, Bridged Call Appearance is used by going off-hook and pressing the bridged appearance button. In both cases, the user is then bridged onto the primary telephone's extension number and can handle calls on that extension number.

An incoming call rings the primary extension number's telephone and all telephones that have a bridged call appearance of the telephone's primary extension number. Each telephone is visually alerted for all bridged appearances on the telephone, but has the option of audible ringing.

On multiappearance telephones, 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, Conference, Transfer, Hold, Drop, and Priority Calling can be used from a bridged appearance, just as they are used from a regular call appearance.

You can administer a telephone with zero call appearances of its primary extension. In this way, a telephone can be administered to have only bridged appearances.

Extension administrable buttons and lamps for multiappearance telephones

You can administer the message lamp and some feature buttons to apply to a specified extension rather than the extension of the telephone they reside on.

- You can administer the message lamp to light when messages are waiting for the extension specified on the Station screen. In this way, the bridged user's telephone can be set up to indicate when messages are waiting for the primary extension.

- You can administer the call forwarding all calls and call forwarding busy/don't answer buttons to activate Call Forwarding for any extension that is on the telephone, even if this extension is a bridged appearance. In addition, you can administer the lamp associated with the call forwarding button to track the call forwarding status of any extension. In this way, a bridged user can activate or deactivate Call Forwarding for all primary and bridged appearances of the extension from the bridged appearance telephone, and the bridged appearance telephone shows the call forwarding status of the specified extension.

- You can administer the send all calls button to activate Send All Calls for any administered extension. The lamp associated with Send All Calls tracks the status of the administered extension. In this way, a bridged user can activate Send All Calls for the primary extension user.

Busy Indicator

The Busy Indicator button provides multiappearance telephone users and attendants with a visual indicator of the busy or idle status of one of the following system resources:

- An extension number
- A trunk group
- A terminating extension group
- A hunt group—either direct department calling (DDC) or uniform call distribution (UCD)
- Any loudspeaker paging zone, including all zones

The Busy Indicator button provides the attendant or user with direct access to the extension number, trunk group, or paging zone.

You can assign extension numbers, trunk group access codes, and Loudspeaker Paging access codes to a Busy Indicator button.

The Facility Busy lamp indication for a vector directory number (VDN) does not light when the VDN is being used. The associated button may be used to place a call to a VDN.

Call Coverage

Call Coverage provides automatic redirection of calls to alternate answering positions in a Call Coverage path. Call Coverage allows you to:

- Establish coverage paths with up to 6 alternate answering positions
- Establish redirection criteria that govern when a call redirects
- Redirect calls to a local switch location
- Redirect calls to a remote (off-net) location
- Redirect calls based on time-of-day
- Allow users to change back and forth between two coverage choices (either specific lead coverage paths or time-of-day tables).

Hardware requirement

The Coverage of Calls Redirected Off-Net (CCRON) generally requires call classification hardware. Both the Call Classifier - Detector and Tone Clock/Call Classifier - Detector circuit packs provide tone detection ports including the capability to do call classification. There are 8 ports on each circuit pack.or Tone Clock/Call Classifier - Detector circuit pack is sufficient to provide call classification.

For countries not using the USA tone plan, the Call Classifier - Detector and Tone Clock with Call Classifier - Tone Detector circuit packs must be configured appropriately to provide call classification.

The number of simultaneous monitored calls depends on the: total amount of outbound call traffic, number of call classification ports available, and use of other switch applications that make use of call classification ports.

Coverage of Calls Redirected Off-Net competes with the following switch applications for ports on the Call Classifier - Detector and Tone Clock with Call Classifier - Tone Detector circuit packs:

- Answer Detection
- Call Prompting
- CallVisor ASAI
- Multi-Frequency Compelled (MFC) signaling

Serious degradation of switch performance, including the inability to launch new calls, can result from an insufficient resource of call classifier ports.

Detailed description

When a call meets the redirection criteria of the principal, the call attempts to route to one of up to 6 points in the coverage path. 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 phone 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 the Coverage Subsequent Redirection interval. At the end of this interval, 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 remains queued or continues ringing the current coverage point.

Coverage Path

A Call Coverage path is a list of up to six alternate answering positions (covering users/points) that are accessed, in sequence, when the called party or group is not available to answer the call.

You can assign any of the following entities a coverage path so they are eligible to have calls redirected to coverage:

- ACD split
- Agent LoginID
- PCOL group
- TEG
- Hunt group
- Phone (on-net or off-net)

You establish the coverage paths and set the redirection criteria. If a coverage path is not assigned to a particular facility, calls are not redirected from that facility, unless another feature is assigned. A coverage path can include any of the following:

- Announcement
- Attendant group
- AUDIX
- Coverage answer group
- Hunt group
- Public network number (off-net)
- Vector directory number (VDN)
- Phone (on-net or off-net)

DEFINITY ECS allows for multiple coverage paths. However, for any particular call only one coverage path is used. The "lead" coverage path is the first coverage path in a chain that is considered when a call redirects to coverage. The chain is defined in the Next Path Name field on the Coverage Path screen.

When a call redirects to coverage, the lead coverage path at that time is checked to determine whether its coverage redirection criteria match the call status. If there is a match the lead coverage path is used. If the lead coverage path's redirection criteria does not match, the system moves down the path chain until it finds a coverage path with redirection criteria that matches the call status. If the chain is exhausted before the system finds a match, the call does not redirect to coverage. Once a coverage path is selected, it is used exclusively through the duration of the call.

You can assign lead coverage paths directly in the Coverage Path 1 or Coverage Path 2 fields on the appropriate screens. For example, to assign a lead path for a TEG, set the Coverage Path field on the Terminating Extension Group screen. You can also assign the lead paths indirectly by assigning a Time-of-Day Coverage Table to the Coverage Path 1 and Coverage Path 2 fields. Then, the system selects the lead path according to the time of day

Subsequent redirection interval

The number of times a call rings at a particular coverage point before the switch moves the call to the next coverage point depends on the type of ringing coverage point (for example, local, Distributed Communications System (DCS), CCRON, and so forth). For each type of coverage point, the following table shows which subsequent redirection interval on the System-Parameters Call Coverage/Call Forwarding screen is used.

- Local — On the System-Parameters Coverage/Forwarding screen, the Local Subsequent Redirection/CFWD Don't Answer Interval field.
- Off-net — On the System-Parameters Coverage/Forwarding screen, the Offnet Subsequent Redirection/CFWD Don't Answer Interval field.
- — The call is left off-net.

Call redirection criteria

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

- Active

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

- Busy

Redirects calls to coverage when all available call appearances at the principal extension are in use. For multiappearance phones, one call appearance can be 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) redirects 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 remains at that idle appearance.

A Terminating Extension Group (TEG) is considered busy if any phone in the group is active on a call.

Each phone in a UCD or DDC group must be active on at least one call appearance for the call to redirect to coverage. If any phone in the group is idle the call directs to that phone. If no phone is available, the call can queue if queuing is provided. If queuing is not provided, then the call routes to coverage. If the queue is full or all agents are in an auxiliary state, the group is considered busy and the call routes to coverage. Queued calls remain in queue for the Don't Answer Interval.

A call will not cover to a hunt group if no agents are logged in, or if all agents are in AuxWork mode.

- Don't Answer

Redirects calls to coverage if unanswered during the assigned Don't Answer Interval. A call rings for the Don't Answer Interval and then redirects to coverage.

- Cover All Calls

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

- Send All Calls/Go to Cover

Allows users to activate Send All Calls or Go to Cover as an overriding coverage criteria. This redirection criteria must be assigned before a user can activate Send All Calls or Go to Cove (discussed later).

- No Coverage

Occurs when none of the above criteria are assigned. Calls redirect to coverage only 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. For example, you can combine Active/Don't Answer and Busy/Don't Answer. Other combinations are not possible or do not provide any useful function. For example, Active/Busy does not accomplish anything. A busy phone is always active.

Redirection criteria are assigned separately for internal and external calls. By linking the coverage paths, Busy/Don't Answer can be assigned for internal calls and Active can be assigned for external calls. Similarly, Busy/Don't Answer can apply for external calls and No Coverage can apply for internal calls. In the latter case, internal calls remain directed to the called phone or group.

All calls extended by the attendant are treated as external.

Time-of-Day Coverage

The Time-of-Day Coverage Table allows you to redirect calls to different lead-coverage paths at different times of the day and on different days of the week.

For example, an employee may want incoming calls to cover to a co-worker (office) during normal business hours, to cover to an off-net destination (home) in the early evening, and to cover to AUDIX at all other times. By specifying the appropriate lead-coverage paths in the Time-of-Day Coverage Table, the employee can have the call redirection flexibility shown in the following table. (If you were actually administering a Time-of-Day Coverage Table, you would provide the lead-coverage path numbers that redirect the calls to the employee's office, to their home, and to AUDIX.

Off-Net Call Coverage

Call Coverage allows a call to be redirected to a destination on the public network. The remote (off-net) number is administered on the Remote Call Coverage Table screen and may have up to 16 digits including either the outgoing trunk access code (TAC) or the feature access code (FAC) specifying ARS or AAR. Any coverage point can be an off-net destination.

Whenever an incoming trunk call is redirected off-net (coverage or forwarded), a timer is set that precludes any other incoming trunk call from redirecting off-net until the timer either expires or is cancelled. The rationale for this mechanism is to prevent calls that were redirected off-net from being re-routed back to the original principal from the off-net destination, effectively creating a round-robin loop that continuously seizes trunks until they are exhausted.

Call Detail Recording

CDR tracks call information on a per-trunk-group or station-to-station basis. For every trunk group (including auxiliary trunks) that you administer for CDR reports, the system keeps track of incoming, outgoing and tandem calls. You can also receive reports on temporary signaling connections (TSCs) that involve trunks, and calls made using loudspeaker paging or code calling access.

Call Forwarding

Call Forwarding allows users to redirect calls to designated destinations. The forwarded-to destination can be an internal extension, external (off-net) number, an attendant group, or a specific attendant.

Call Forwarding provides these functions:

- Call Forwarding-All Calls — Allows a user to redirect every incoming call to the forwarded-to destination.
- Call Forward Busy/Don't Answer — Allows a user to redirect incoming calls to a forwarded-to destination only when the user is busy or when the call is not answered after an administrable interval. If the extension is busy, the call forwards immediately. If the extension is not busy, the incoming call rings the called extension, then forwards only if it remains unanswered longer than the administered interval.
- Call Forwarding Off Net— Allows a user to forward calls to an off-net destination.

Detailed description

You assign Call Forwarding All Calls and Call Forwarding Busy/Don't Answer to extensions on a Class of Service basis. You assign Call Forwarding Override and Call Forwarding Off-Net on a system-wide basis. You can also restrict Call Forwarding Off-Net with the Class of Service.

Call Forwarding All Calls

Phone users and data-terminal users can activate or deactivate Call Forwarding All Calls for their own terminals with a feature-access code or Call Forward-All feature button. An attendant or phone user with console permission can activate or deactivate the feature for another extension, TEG, DDC, UCD group, or ACD split (but not vector-controlled splits).

Call Forwarding Busy/Don't Answer

The feature is activated or deactivated with a feature-access code or Call Forward Busy/Don't Answer feature button. An attendant or phone with console permission can also activate or deactivate the feature for another extension by using a feature-access code.

Call Forward Busy/Don't Answer cannot be activated for hunt groups, data extensions, or terminating extension groups (TEG). Calls to an attendant or EAS agent cannot be forwarded.

Call Forwarding Off Net

When a call is forwarded off net, the forwarded-to number can have up to 16 digits. When counting the 16-digit limit, count the digits in the Trunk Access Code or AAR/ARS feature access code. Do not count the "#" used to terminate a forwarded-to number if the "#" is used.

Call Park

Call Park allows users to put a call on hold and then retrieve the call from any other telephone within the system.

You can set a system-wide expiration interval for parked calls. If a call is not answered within the interval, the parked call redirects to an attendant or to the user who activated Call Park (the parking user). Calls redirect to the attendant if the default "Loudspeaker Paging" option is assigned and to the parking user if the Deluxe Paging and Call Park Timeout to Originator option is assigned.

If no attendant or night service extension is administered, and if Night Service — Trunk Answer from Any Station is not administered, the expiration interval is ignored and the call remains parked.

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 common, shared extensions used exclusively for Call Park. These extensions are not assigned to a telephone, but are stored in system translations and used to park a call. The extensions are particularly useful when one party is paged at the request of another party. The caller is parked on a common shared extension and the extension is announced. The status lamp associated with the extension identifies "call parked" or "no call parked" (instead of active or idle status).

Call Park allows telephone users to answer a call at one extension, but complete the call at another extension. Call Park also allows users to answer a call at any telephone after being paged by a telephone user or an attendant.

Considerations

- Only one call per extension can be parked at a time, even if the extension has multiple call appearances. Conference calls with up to five parties can be parked; the sixth position must remain open for the retrieving party.
- Calls cannot be parked on a group extension. If a group member places a call in Call Park, the call is parked on the member's extension. Group members can belong to the following:

— A coverage answer group

— A DDC group

— A terminating extension group

— A UCD group

Call Pickup

Call Pickup and Directed Call Pickup allow a telephone user to answer calls that alert at other extension numbers within the user's specified call pickup group. Directed Call Pickup allows telephone users to pick up any call on the DEFINITY ECS system.

Call Pickup

Establish a call pickup group so that when one member of a group is away, other members can answer the absent member's calls. A call pickup group usually consists of users who are located in the same area or who have similar functions.

To pick up another user's call, a user goes off-hook and dials the Call Pickup access code or presses a Call Pickup button.

If a user's telephone has a Call Pickup button and status lamp, then:

- The status lamp lights steadily when Call Pickup is used.
- If Call Pickup Alerting is activated, members' status lamps flash when a call comes in to any extension in the call pickup group. Group members other than the called party can answer using Call Pickup. The called party can answer on the ringing call or bridged appearance.

NOTE:

Call Pickup Alerting for a telephone takes effect only when the Call Pickup status lamp is not lit. If Call Pickup is used to answer a call, the status lamp lights steadily and does not flash if there are additional calls to the call pickup group.

Both Call Pickup and Call Appearance status buttons flash at the called party's telephone.

If calls ring at 2 or more telephones in a call pickup group and a group member presses the Call Pickup button, a distribution algorithm determines which call is answered. Thus, all call pickup group members are treated equally. Specifically, when a Call Pickup button is pressed, the system searches the group extension numbers until reaching an extension with a call eligible for Call Pickup. The next time a Call Pickup button is pressed, the system searches from the *next* extension number.

Directed Call Pickup

Directed Call Pickup functions like Call Pickup, except for the following:

- A user can answer an alerting call at any telephone on the system — the alerting and answering telephones need not be members of the same call pickup group.
- You grant users permission to have their calls answered or to answer others' calls with Directed Call Pickup on a per-telephone basis on the Class of Restriction screen.

Class of Restriction

You use Class of Restriction (COR) to define the types of calls your users can place and receive. Your system may have only a single COR, a COR with no restrictions, or as many CORs as necessary to effect the desired restrictions.

You will see the COR field in many different places throughout the DEFINITY System - when administering phones, trunks, agent logins, and data modules, to name a few. You must enter a COR on these screens, although you control the level of restriction the COR provides. You must administer a COR for the following objects:

- Agent LoginID
- Access Endpoint
- Announcements/Audio Sources
- Attendant Console
- Authorization Code — COR Mapping
- Console-Parameters
- Hunt Groups
- Loudspeaker Paging
- Data Modules
- Remote Access (each barrier code has a COR)
- Station
- Terminating Extension Group
- Trunk Groups
- Vector Directory Number

Called-party and calling-party restrictions

Called-party and calling-party restrictions are the basis for all CORs. When no restrictions are needed, assign a single COR with called-party and calling-party restrictions set to none. You can use this COR for unrestricted telephones, trunk groups, terminating extension groups, Uniform Call Distribution (UCD) groups, Direct Department Calling (DDC) groups, data modules, attendant groups, and individual attendant extensions.

The called-party restriction is checked only at the called terminal, module, attendant console, zone, or group, even if a call redirects from one telephone to another. For example, if a called terminal (with no terminal restrictions) has Call Forwarding active to a restricted terminal, the call still completes.

Conference

The Conference button allows multiappearance telephone users to make up to six party conference calls without attendant assistance. This button also allows single-line telephone users to make up to three party conference calls without attendant assistance.

Crisis Alert

Crisis Alert notifies designated extensions when an emergency call is made, and indicates the origin of the emergency call. This information allows the attendant or other user to direct emergency-service response to the caller.

When a user places an emergency call, the system notifies the designated extensions with audible and visual alerting. Audible alerting sounds like an ambulance siren. Visual alerting consists of flashing of the CRSS-ALERT button lamp and display of the caller name and extension.

When crisis alerting is active at the attendant console, the console is in position-busy mode so that no other incoming calls interfere with the emergency call. The console can still originate calls. The attendant must press the POSITION-BUSY button to unbusy the console and then the CRSS-ALERT button to deactivate audible and visual alerting.

Dial Plan

This is the system's guide to digit translation. When the system receives dialed digits, it must know what to expect next based on the digits received so far. For example, if a user dials 4, the system must know how many more digits to expect before the call is processed.

All feature access codes, extensions and trunk access codes must be consistent with the dial plan.

Detailed description

The dial plan provides information to the switch on what to do with dialed digits. Tables define the intended use of a code beginning with a specific first digit or pair of digits. These digits tell the system how many digits to collect before processing the full digit string.

For example, a digit string beginning with 8 may tell the system to wait for 4 more digits because this is the first digit of a 5-digit internal extension. The choices of a first digit are 0–9, *, and #. Permissible codes and the allowable number of digits are listed below.

You can also administer a Uniform Dial Plan (UDP) as part of the dial plan to be shared among a group of switches. If you establish a UDP, make all extensions the same length (4 or 5 digits). So that calls route to the desired switch, a UDP requires the following information:

- A PBX code, which represents the first 1 to 5 digits of a 4-digit or 5-digit extension and can range from 0 to 9xxxx with a maximum of 50,000 PBX codes on G3r or 20,000 PBX codes on G3si/csi.
- An RNX, which is associated with the PBX code and is used to select an AAR pattern for the call. This information is

  required for each PBX code. The 3-digit RNX can be an AAR location code or, for ENP calls, an ENP code.
- A PBX ID (1 to 63), which represents a specific switch (optional).
- Whether or not the PBX code is local to this system (optional).

Emergency Transfer

Emergency Transfer provides service to and from the local telephone company central office (CO) during a power failure or when service is impaired. Emergency Transfer is also called Power Failure Transfer; the terms are synonymous.

Detailed description

Emergency Transfer allows analog telephones (500- or 2500-type) to access the local CO and to answer incoming calls during a power failure.

Each DEFINITY ECS cabinet supports Emergency Transfer panels via the AUX connectors on the rear panel. The transfer is initiated when:

- a transfer panel or associated cabinet loses power.
- someone manually activates the Emergency Transfer switch on the associated maintenance circuit pack
- the software determines that service for that cabinet is severely impaired

Emergency Transfer panels are available in multiples of five telephones, which may be pulse-dialing or touch-tone phones. You must use pulse dialing if the CO accepts dial pulses only. Each telephone can be connected to a separate CO.

When your system is not in Emergency Transfer mode, transfer phones can be used as regular telephone.

Facility restriction levels

Facility Restriction Levels (FRL) allow certain calls to specific users, and deny the same calls to other users. For example, you can give certain users access to central office (CO) trunks to other corporate locations, and you can restrict other users to less expensive, private-network lines.

FRL

The switch compares the FRL of the outgoing phone to the FRL of either the terminating trunk group or, for AAR and ARS, the routing preference specified on the Routing Pattern Table. If the FRL of the originator is equal to or greater than the terminating or route pattern FRL, the call continues. Otherwise, the call is blocked.

FRL guidelines

You assign the FRL to the trunk group within the route pattern. You can use the same trunk group in more than one route pattern, and the same trunk group can have a different FRL in a different pattern. You can assign the same FRL to more than one trunk group.

Be consistent in FRL assignments. For ease of assignments, always use FRL 0 or 1 for a trunk group that everyone can access. If you use a range of 0–5 in one pattern, use the same range in another pattern if all users can access the first-choice route.

Assign a COR with an FRL of 0 to a group of users to restrict them from making outgoing calls. Use any other number for the FRL on your first choice route pattern. This denies access to any trunk group for the users, because all trunk-group FRLs are greater than 0.

You assign FRLs for remote access users through the remote-access barrier codes. You can assign up to 10 barrier codes, each with its own COR and FRL.

The simplest way to assign these FRLs is to duplicate the on-premises FRLs, then relate the appropriate barrier code to users who need remote access.

Group paging

Group paging allows users to make an announcement over a group of digital speakerphones.

- You can create up to 32 paging groups on one DEFINITY ECS.
- Each group can have up to 32 extensions in it.
- It's OK to assign the same extension to different groups.

Hunt Groups

A hunt group is a group of extensions that can handle multiple calls simultaneously to a single phone number. For each call to the phone number, the system hunts for an available extension in the group and connects the call to that extension.

A hunt group is especially useful when you expect a high number of calls to a particular phone number. A hunt group might consist of people trained to handle calls on specific topics. For example, the group might be:

- A benefits department within your company
- A service department for products you sell
- A travel reservations service
- A pool of attendants

Loudspeaker paging

You can connect DEFINITY ECS to loudspeaker systems and allow users to page from their phones. You can administer up to 9 separate zones (sets of loudspeakers) on DEFINITY ECS, so an announcement can be made to one group or location without disturbing people who don't need to hear the announcement. Auxiliary trunks connect the speakers in each zone to ports on an auxiliary trunk circuit pack.

Malicious Call Trace

Malicious Call Trace (MCT) allows you to trace malicious calls. MCT allows you to define a group of telephone users who can notify others in the group when they receive a malicious call. These users then can retrieve information related tothe call. Using this information, you can identify the malicious call source or provide information to personnel at an adjacent switch to complete the trace. MCT also allows you to record the malicious call.

You allow users in the group to activate MCT and/or to control malicious call trace. The controlling telephone user, or controller, receives the information that MCT collects on the call.

MCT Voice Recorder

The MCT Voice Recorder is any type of audio recorder (for example, a standard audio cassette player) that you can control via the DEFINITY Auxiliary Trunk board.

To record the call, manually place the MCT Voice Recorder in Record mode. The telephone user then activates the MCT feature which applies power to the recorder (via the connected Auxiliary Trunk's control signal interface).

Activating MCT

To activate MCT while on an active malicious call, perform one of the following:

- Push an MCT-Activate feature button
- Place the call on hold, get a second call appearance, and dial an MCT-Activate Feature Access Code (FAC). After the dial tone, the user then dials their own extension, presses #, or waits for a 10-second time-out.
- Signal another user in the defined group to activate MCT. The co-worker activates MCT, waits for the dial tone, and dials the call recipient's extension.
- Inform a controller, who can request that another switch continue tracing the call.

The switches must be tandemed. The controller on the first switch supplies the trunk member port id to be traced. The controller on the second switch activates MCT and presses *, followed by the trunk port id. The letters A through E of a port id are entered as 1 through 5 on the station keypad. For example, trunk port id 01C0401 would be entered as 0130401.

Once MCT is activated, information on the call is collected and alerts users in the group. The alert is not a call, so it is not affected by queues at the user's terminal. If an MCT Voice Recorder is connected, it begins recording the conversation.

Night Service

DEFINITY ECS provides the following Night Service features:

- Hunt Group Night Service
- Night Console Service
- Night Station Service
- Trunk Answer from Any Station
- Trunk Group Night Service

Off-Premises Station

Off-Premises Station allows a phone located outside the building where the switch is located to be connected to the system. If central office (CO) trunk circuits are used, the voice terminal must be analog and must be FCC-registered or, outside the US, registered by the appropriate governmental agency.

Digital communications protocol (DCP) sets can be used as off-premises terminals with the addition of the DEFINITY extender. Off-premises stations are useful when it is necessary to have a voice terminal located away from the main location. The maximum loop distance for off-premises stations is 20,000 feet (6093.34 meters) without repeaters. For cabling distance information for the various voice terminal types, refer to the *DEFINITY ECS System Description*.

Personal Station Access

Personal Station Access (PSA) allows users to associate the preferences and permissions of their telephone with any other telephone of the same type.

PSA makes it convenient for different users to use the same bank of phones at different times. For example, several telecommuting or hoteling employees can use the same office on different days of the week. The employees use PSA to "associate" with the office phone — that is, make the terminal "theirs" for the day. Calls an employee originates from the station are recognized and displayed as the employee's calls, and calls routed to the employee's extension route to the voice terminal "associated" with that extension.

Priority Calling

Priority Calling provides a special type of call alerting between internal telephone users, including the attendant. The called party receives a distinctive ring when the calling party uses Priority Calling.

You administer the priority-calling ringing-pattern system wide. Default is a 3-burst alerting signal. You allow feature use for each telephone user by administering the user's class of service.

The following types of calls are always priority-calling calls:

- Call coverage consult
- Automatic callback
- Ringback queuing
- Attendant intrusion
- Security violation notification

The system generates the call waiting ringback tone that a single-line telephone user hears even if the user is active on a call. In contrast, the system *does not* generate the pattern for a multiappearance telephone if there are no idle call appearances. In this case the caller hears busy tone. However, the system does generate the pattern if the telephone has an idle call appearance, including the one reserved for call origination.

Remote Access

Remote Access permits authorized callers to access the system via the public network from remote locations and then use its features and services. The Remote Access caller must use a touch-tone phone or equivalent equipment. Since the system does not have access to the calling (outside) number, Ringback Queuing and Automatic Callback cannot be used on a Remote Access call. Also, any feature requiring recall dial tone (for example, Hold and Transfer) cannot be accessed remotely.

! SECURITY ALERT:
Avaya has designed the Remote Access feature incorporated in this product that, when properly administered by the customer, enables the customer to minimize the ability of unauthorized persons to gain access to the network. It is the customer's responsibility to take the appropriate steps to properly implement the features, evaluate and administer the various restriction levels, protect access codes, and distribute them only to individuals who have been advised of the sensitive nature of the access information. Each authorized user should be instructed on the proper use and handling of access codes.

In rare instances, unauthorized individuals make connections to the telecommunications network through use of remote-access features. In such an event, applicable tariffs require that the customer pay all network charges for traffic. Avaya cannot be responsible for such charges, and does not make any allowance or give any credit for charges that result from unauthorized access.

Security violations notification

When a security violation occurs, security violations notification (SVN) notifies a designated referral point. This can be an attendant console, a display-equipped phone, or a phone without display for SVN referral calls with announcements.

The system monitors and reports on the following types of security violations:

- Login violations
- Remote access barrier code violations
- Authorization code violations
- Station security code violations

DEFINITY ECS provides the option to log a major alarm if a security violation occurs involving a Avaya services login ID. Avaya is responsible for retiring the alarm.

Refer to *DEFINITY ECS Reports* for more information on how to run reports, and respond to security violations.

To effectively monitor the security of your system, you need to know how often both valid and invalid attempts at system entry are normally made. Then you will know if the number of invalid attempts is unusually high. A significant increase in such attempts can mean the system is being compromised.

Service observing

Designated users, normally supervisors, can listen to other users' calls. This capability is often used to train agents and monitor service quality in call centers and other environments where employees serve customers over the phone. On DEFINITY, this is called "service observing" and the user observing calls is the "observer."

Terminal Translation Initialization

Terminal Translation Initialization (TTI) allows you to merge an x-ported extension to a valid port by dialing a system-wide TTI security code and the extension from a telephone connected to that port. TTI also allows you to separate an extension from its port by dialing a similar separate digit sequence. This action causes the extension to be administered as an X port.

When TTI is enabled for voice, all voice ports (except Basic Rate Interface (BRI) ports) become TTI ports or ports from which a TTI merge sequence can occur.

TTI is usually used to move phones, however, it also supports connecting and moving attendants, data modules, voice/data telephones, and ISDN-BRI telephones.

Time of Day Routing

You can use Time of Day Routing to select route patterns for calls according to the time of day and day of the week. You need to define the route pattern you want to use before you set up time of day routing.

You can route calls based on the least expensive route, and you can deny outgoing long-distance calls after business hours to help prevent toll fraud. You can use partition groups to assign different time of day route plans for different groups of users.

Automatic Alternate Routing (AAR) or Automatic Route Selection (ARS) must be administered on your switch before you use Time of Day Routing. Time of Day Routing applies to all AAR or ARS outgoing calls and trunks used for call forwarding to external numbers.

Voice Messaging Systems

DEFINITY ECS supports several Avaya voice or multimedia messaging systems. These systems allow users to send, retrieve, store, and forward messages, as well as perform many other tasks associated with messages. In addition to supporting multiple AUDIX systems, DEFINITY can have multiple hunt groups associated with a single AUDIX system. This allows partitioning of the voice ports into different hunt groups and different coverage paths to cover different voice ports. Thus voice ports can be reserved for particular users or groups of users (for example, those that use unique coverage paths).

The following features do not use coverage paths:

- Transfer into AUDIX with the feature access code or the GOTO COVER button
- Last Call

If a local AUDIX and a remote AUDIX use the same hunt-group numbers, calls route to the local hunt group.

DEFINITY ECS supports the following systems:

INTUITY:

INTUITY AUDIX runs on a separate MAP/5, MAP/40, or

AUDIX MAP/100 PC.

The switch communicates with INTUITY AUDIX via analog voice ports and a data link. The switch can also communicate with INTUITY AUDIX without the data link. In this case, the switch and INTUITY AUDIX communicate by sending and receiving special strings of touch-tone codes (dual tone multifrequency tones) via analog voice ports. These touch-tone codes are called *mode codes* and carry data such as calling party ID, called party ID, and type of call.

INTUITY AUDIX allows up to 64 ports. This means up to 64 people can be simultaneously retrieving or leaving messages. INTUITY AUDIX also supports fax and e-mail messaging.

DEFINITY

DEFINITY AUDIX runs on a AUDIX multifunction circuit pack assembly. This assembly fits into 2 contiguous slots in the DEFINITY switch. DEFINITY AUDIX communicates with the switch via analog voice ports with a data link. DEFINITY AUDIX can also communicate exclusively via analog voice ports when set up to emulate a digital phone set.

DEFINITY AUDIX allows up to 16 ports.

For more information, refer to *DEFINITY AUDIX System Feature Descriptions, DEFINITY AUDIX System Administration, Switch Administration for DEFINITY AUDIX System,* or *DEFINITY AUDIX System Forms Reference.*

Octel

Octel Serenade is a voice messaging system that supports Serenade DEFINITY systems via QSIG signaling protocols.

Octel 100

Octel 100 runs on personal computer running the OS2 operating system.

Other non-Avaya messaging systems may also use mode codes to work with DEFINITY ECS.

Whisper paging

Whisper paging allows one user to interrupt or "barge in" on another user's call and make an announcement. The paging user dials a feature access code or presses a feature button, then dials the extension they want to call.

Only the person on the paged extension can hear the page other parties on the call cannot hear it, and the person making the page cannot hear anyone on the call. If the paged user has a display phone, he or she can see who is making the whisper page.

For example, let's say users A and B are on a call. C has an urgent message for A and makes a whisper page. All 3 users hear the tone that signals the page, but only A hears the page itself. The person making the page, C, cannot hear A or B.

Allowing users to make whisper pages

To make a whisper page users dial a feature access code or press a feature button, then dial the extension of the user they are trying to reach.

- To assign a feature access code, enter a code in the Whisper Page Activation Access Code field on the Feature Access Code screen.
- To give users a feature button for making a whisper page, use the Station screen and administer a Whisper Page Activation button on users' phones.

Telephone feature buttons

The following table provides descriptions of the feature buttons that you can administer on multiappearance telephones. It also lists the administrable software names and recommended button label names. Display buttons support telephones equipped with alphanumeric displays. Note that some buttons may require 1-lamp or 2-lamp buttons. Some buttons are not allowed on some systems and on some phones.

Telephone feature buttons			
	Button		
Button name	label	Description	Maximum
abr-prog	AbrvDial	Abbreviated Dialing Program: allows	1 per
	Program	users to program abbreviated dialing and	station
		autodial buttons or to store or change	
		numbers in a personal list or group list	
		associated with the station.	
abr-spchar	AbrvDial	Abbreviated Dialing Special Character:	1 each per
	(char)	allows users to enter an associated special	station
		character [~, ~m (mark), ~p (pause), ~s	
		(suppress), ~w (wait for dial tone), or ~W	
		(wait forever)] when programming an	
		abbreviated dialing list entry.	
abrdg-appr	(extension)	Bridged Appearance of an analog phone:	Depends on
(Ext:)		allows the user to have an appearance of a	station type
		single-line telephone extension. Assign to	
		a 2-lamp appearance button.	
abrv-dial	AD	Abbreviated Dialing: dials the stored	1 per AD
(List: __		number on the specified abbreviated	list per dial
DC: __)		dialing list.	code
		List: specify the list number 1 to 3 where	
		the destination number is stored	
		DC: specify the dial code for the	
		destination number	
abrv-ring	AR	Abbreviated and Delayed Ringing: allows	
		the user to trigger an abbreviated or	
		delayed transition for calls alerting at an	
		extension.	

ac-alarm	AC Alarm	Administered Connection alarm	1 per
		notification: allows the user to monitor	station
		when the number of failures for an	
		administered connection has met the	
		specified threshold.	
aca-halt	Auto-Ckt	Automatic Circuit Assurance (*display*	1 per
	Assure	*button*): allows users of display telephones	system
		to identify trunk malfunctions. The system	
		automatically initiates a referral call to the	
		telephone when a possible failure occurs.	
		When the user presses ACA Halt, the	
		system turns off ACA monitoring for the	
		entire system. The user must press ACA	
		Halt again to restart monitoring.	
account	Acct	Account: allows users to enter Call Detail	
		Recording (CDR) account codes. CDR	
		account codes allow the system to	
		associate and track calls according to a	
		particular project or account number.	
admin	Admin	Administration: allows a user to program	
		telephone.	
after-call	After Call	After Call Work Mode: allows an agent to	1 per split
Grp:	Work	temporarily be removed from call	group
		distribution in order for the agent to finish	
		ACD-related activities such as completing paperwork.	
		Grp: specify the ACD split group number.	
alrt-agchg	Alert Agent	Alert Agent: indicates to the agent that	1 per
		their split/skill hunt group changed while	station
		active on a call. This button blinks to	
		notify the agent of the change.	

alt-frl		Alt FRL	Alternate Facility Restriction Level (FRL): activates or deactivates an alternate facility restriction level for the extension.	1 per system
ani-requst		ANI Request	Automatic Number Identification Request: allows the user to display the calling party's number from incoming trunks during the voice state of call. The trunk must support this functionality.	1 per station
assist (Group: __)		Assist	Supervisory Assistance: used by an ACD agent to place a call to a split supervisor. Group: specify the ACD split group number.	1 per split group
asvn-halt		asvn-halt	Authorization Code Security Violation Notification: activates or deactivates call referral when an authorization code security violation is detected.	1 per system
atd-qcalls		AQC	Attendant Queue Calls (display button): tracks the number of calls in the attendant group's queue and displays the queue status. Assign this button to any user who you want to backup the attendant.	1 per station
atd-qtime		AQT	Attendant Queue Time (display button): tracks the calls in the attendant group's queue according to the oldest time a call has been queued, and obtains a display of the queue status.	1 per station
aut-msg-wt (Ext:)		Message (name or ext #)	Automatic Message Waiting: associated status lamp automatically lights when an LWC message has been stored in the system for the associated extension (can be a VDN).	1 per aut-mst-ext

auto-cback	Auto	Automatic Call Back: when activated,	1 per
	CallBack	allows inside user who placed a call to a	station
		busy or unanswered telephone to be called	
		back automatically when the called	
		telephone becomes available to receive a call.	
auto-icom	Auto (name	Automatic Intercom: places a call to the station associated with the button	1 per group
(Group: __)	or ext #)	. The called user receives a unique alerting signal, and a status lamp associated with a Intercom button flashes. Grp Intercom — Auto-Icom group number. This extension and destination extension must be in the same group.	1per dial code
auto-in	Auto In	Auto-In Mode: allows the user to become	1 per split
(Group: __)		automatically available for new ACD calls	group
		upon completion of an ACD call. Grp:	
		The split group number for ACD.	
auto-wkup	Auto	Automatic Wakeup (*display button*):	1 per
	Wakeup	allows attendants, front-desk users, and	station
		guests to request a wakeup call to be	
		placed automatically to a certain extension(may not be a VDN extension) at a later time.	
autodial	Autodial	Allows a user to dial a number that is not	
		part of a stored list.	
aux-work	Auxiliary	Auxiliary Work Mode: removes agent	1 per split
(Group: __)	Work	from ACD call distribution in order to	group
		complete non-ACD-related activities.	
		Grp: The split group number for ACD.	

brdg-appr	(extension)	Bridged Call Appearance: provides an	Depends on
(Btn: __ Ext:)		appearance of another user's extension on this telephone.	station type
		For example, an assistant	
		might have a bridged appearance of their	
		supervisor's extension. The bridged	
		appearance button functions exactly like	
		the original call appearance, for instance it	
		indicates when the appearance is active or ringing.	
		You can assign brdg-appr buttons only to	
		2-lamp appearance buttons. You must	
		indicate which extension and which call	
		appearance button the user wants to	
		monitor at this phone.	
btn-view	Button View	Button View: allows users to view, on the	
		phone's display, the contents of any	
		feature button. Button View does more	
		than the "View" or "stored-num" feature	
		button; these only display what is	
		contained in abbreviated dialing and	
		autodial buttons.	
		When the user presses the btn-view button	
		and then a specific feature button, they see	
		the feature name and any auxiliary data for	
		that button. This allows users to review	
		the programming of their feature buttons.	
		You can assign this soft-key button to any	
		6400-, 7400-, or 8400-series display	
		telephone.	

	Telephone feature buttons — *Continued*		
	Button		
Button name	**label**	**Description**	**Maximum**
busyind(TAC/Ext: __)	Busy	Busy Indication: indicates the busy or idle time status of an extension, trunk group, terminating extension group (TEG), hunt group, or loudspeaker paging zone. Users	1 per TAC/Ext
		can press the busy-ind button to dial the specified extension. You can assign this button to any lamp	
		or button and must specify which Trunk extension the user wants to monitor	
call-appr	*extension*	Call Appearance: originates or receives	Depends on
		calls. Assign to a 2-lamp appearance	station type
		button.	
call-disp	Return Call	Call Displayed Number (*display button*):	1 per
		initiates a call to the currently displayed	station
		number. The number may be from a leave	
		word calling message or a number the user	
		retrieved from the Directory.	
call-fwd	Call	Activates or deactivates Call Forwarding	
(Ext-)	Forwarding	All Calls.	
call-park	Call Park	Allows the user to place the current call in	1 per
		the call park state so it can be retrieved	station
		from another phone.	
call-pkup	Call Pickup	Allows the user to answer a call that is	1 per
		ringing in the user's pickup group.	station
call-timer	CTime	Used only on the 6400 sets. Allows users	1 per
		to view the duration of the call associated	station
		with the active call appearance button.	
callr-info	Caller Info	(*display button*) Used with Call Prompting	1 per
		to allow users to display information	station
		collected from the originator.	

cas-backup	CAS Backup	Centralized Attendant Service Backup: used to redirect all CAS calls to a backup extension in the local branch if all RLTs The associated status lamp indicates if CAS is in the backup mode.	1 per station
cdr1-alrm	CDR 1 Failure	CDR Alarm: associated status lamp is used to indicate that a failure in the interface to the primary CDR output device has occurred.	1 per station
cdr2-alrm	CDR 2 Failure	CDR Alarm: associated status lamp is used to indicate that a failure in the interface to the secondary CDR output device has occurred.	1 per station
cfwd-bsyda	Call Forwarding bsyda (Ext)	Call Forward Busy/Don't Answer: activates and deactivates call forwarding for calls when the extension is busy or the user does not answer.	
check-in	Check In	Check In (*display button*): changes the state of the associated guest room to occupied and turns off the outward calling restriction for the guest room's station.	1 per station
check-out	Check Out	Check Out (*display button*): Changes the state of the associated guest room to vacant and turns on the outward calling restriction for the guest room's station. Also clears (removes) any wake-up request for the station.	1 per station
clk-overid	Clocked Override	Clocked Manual Override (*display button*): used in association with Time of Day Routing to override the routing plan in effect for the activating user. The routing plan is overridden for a specified period of time.	1 per station

Button name	Button label	Description	Maximum
consult	Consult	The Consult button allows a covering user, after answering a coverage call, to call the principal (called party) for private consultation. Activating Consult places the caller on hold and establishes a private connection between the principal and the covering user. The covering user can then add the caller to the conversation, transfer the call to the principal, or return to the caller.	1 per station
cov-cback	Coverage Callback	Allows a covering party to store a leave word calling message for the principal (called party)	1 per station
cov-msg-rt	Covr Msg Retrieve	Coverage Message Retrieval (*display button*): places a covering station into the message retrieval mode for the purposes of retrieving messages for the group.	1 per station
cpn-blk	CPN Block	Blocks the sending of the calling party number for a call.	1 per station
cpn-unblk	CPN Unblock	Deactivates calling party number (CPN) blocking and allows the CPN to be sent for a single call.	1 per station

crss-alert	Crisis	Crisis Alert (*display button*): provide this	1 per
		button to the telephones or consoles that	station
		you want to notify when any user makes	10 per
		an emergency call. (You define which	system
		calls are emergency calls on the	
		AAR/ARS Analysis screen by setting the Call Type to **alrt**.)	
		After a user receives an alert, they can	
		press the crss-alert button to disable the current alert. If tenant partitioning is active, the	
		attendants within a partition can receive emergency notification only from callers in the same partition.	
data-ext	Data (data	Data Extension: sets up a data call. May	1 per data-
	ext #)	be used to pre-indicate a data call or to	extension
		disconnect a data call. May not be a VDN or ISDN-BRI extension.	group
date-time	Date Time	Date and Time (*display button*): displays	1 per
		the current date and time. Do not assign	station
		this button to 6400-series display phones	
		as they normally show the date and time.	
delete-msg	Delete Message	Delete message (*display button*): deletes a stored message that is currently on the	1 per station
		display.	
dial-icom	Dial Icom	Dial Intercom: accesses the intercom	1 per group
(Grp:)		group assigned to the button. Grp:	
		Intercom — Dial (Dial Icom) group	
		number.	
did-view	DID View	DID View (*display button*): allows DID	1 per
		assignments to be displayed, changed, or removed.	station

directory	Directory	Directory (*display button*): allows users with display	1 per
		telephones to access the system directory, use the	station
		touch-tone buttons to key in a name, and retrieve an	
		extension from the directory. The directory contains the	
		names and extensions that you have assigned to the	
		telephones administered in your system.	
		If you assign a directory button, you should also assign a	
		Next and Call-Disp button to the phone. These buttons allow the user to navigate within the directory and call	
		an extension once they find the correct one.	
		Note that Vector Directory Numbers do not appear in the	
		Directory.	
dir-pkup	dir-pkup	Directed call pickup: allows the user to answer a call	
		ringing at another extension without having to be a	
		member of a pickup group.	
disp-chrg	Display	Provides your display phone with a visual	1 per
	Charge	display of accumulated charges on your	station
		current telephone call. Used exclusively	
		outside the U.S. and Canada.	
disp-norm	Local/	Normal (*display button*): Toggles between LOCAL display	1 per
	Normal	mode (displays time and date) and NORMAL mode (displays call-related data). LED off = LOCAL mode and	station
		LED on = NORMAL.	
dn-dst	Do Not Disturb	Places the user in the do not disturb mode.	1 per station
drop	Drop	Allows users to drop calls. Users can drop	
		calls from automatic hold or drop the last	
		party they added to a conference call.	

exclusion	Exclusion	Exclusion: allows multiappearance telephone users	1 per
		to keep other users with appearances of the same	station
		extension from bridging onto an existing call.	
		If the user press the EXCLUSION button while other	
		users are already bridged onto the call, the other	
		users are dropped.	
		There are two means of activating exclusion.	
		before dialing or during the call).	
		Automatic Exclusion — as soon as the user picks	
		up the handset. To turn off Automatic Exclusion during Automatic Exclusion during a call, the user	
		presses the EXCLUSION button. To use Automatic	
		Exclusion, set the Automatic Exclusion by COS field to **y** on the Feature-Related System Parameters	
		screen.	
ext-dn-dst	Do Not Disturb Ext	Extension — Do Not Disturb (*display button*): used by the attendant console or hotel front desk display phone to activate do not disturb and assign a corresponding deactivate time to an extension.	1 per station
flash	Flash	1) Allows a station on a trunk call with	1 per
		Trunk Flash to send a Trunk Flash signal	station
		to the far end (e.g., Central Office); 2)	
		allows a station on a CAS main call to	
		send a Trunk Flash signal over the	
		connected RLT trunk back to the branch to	
		conference or transfer the call.	

goto-cover	Go To	Go To Coverage: sends a call directly to	1 per
	Cover	coverage instead of waiting for the called inside-user to answer. Go to	station
		Cover forces intercom and priority calls to follow a coverage path.	
		NOTE:	
		Go to Cover cannot be activated for	
		calls placed to a Vector Directory	
		Number extension. Go to Cover can	
		be used to force a call to cover to a	
		VDN if the called principal has a	
		VDN as a coverage point.	
grp-dn-dst	Do Not	Group Do Not Disturb (*display button*):	1 per
	Disturb Grp	removes a group of users from the do not disturb mode.	station
grp-page	GrpPg	Allows users to make announcements to	
		groups of stations by automatically	
		turning on their speakerphones.	
headset	Headset	Signals onhook/offhook state changes	
		to the switch. The green LED is on for	
		offhook state and off (dark) for onhook	
		state.	
hunt-ns	Hunt Group	Hunt-Group Night Service: places a	3 per hunt
(Grp:		hunt-group into night service. Grp: Hunt	group
		group number.	
in-call-id	Coverage	The Coverage Incoming Call	1 per
(Type: __	(group #,	Identification (ICI) button allows a	group-type
Grp:	type, name,	member of a coverage answer group or	per group
	or ext #)	hunt group to identify an incoming call to	
		that group even though the member does not have a display telephone.	
		In the Type field, enter **c** for coverage	
		answer groups and type of **h** for a hunt	
		group. In the Grp field, enter the group	
		number.	

inspect	Inspect Mode	Inspect (*display button*): allows users on an active call to display the	1 per
		identification of an incoming call. Inspect also allows users to determine the identification of calls they placed on Hold.	station
int-aut-an	IntAutoAns	Internal Automatic Answer: causes any hybrid or digital station to automatically	1 per
		answer incoming internal calls.	station
last-numb	LastNumb	Last Number Dialed (redial): originates a	1 per
	Dialed	call to the number last dialed by the	station
		station.	
link-alarm	Link Failure	Link Alarm: associated status lamp	8 per
(link#)	(link #)	indicates that a failure has occurred on one of the Processor Interface circuit	station
		pack data links. Link: Link number — 1 to 8 for multi-carrier cabinets or 1 to 4 for single-carrier cabinets.	
lsvn-halt	Login SVN	Login Security Violation Notification:	1 per
		activates or deactivates referral call when a login security violation is detected.	system
lwc-cancel	Cancel	Leave Word Calling Cancel: cancels the	1 per
	LWC	last leave word calling message originated by the user.	station
lwc-lock	Lock LWC	Leave Word Calling Lock: locks the	1 per
		message retrieval capability of the display module on the station.	station
lwc-store	LWC	Leave Word Calling Store: leaves a	1 per
		message for the user associated with the last number dialed to return the call	station
		to the originator	
major-alrm	Major Hdwe	Major Alarm: assign to a status lamp to	1 per
	Failure	notify the user when major alarms occur.	station
		Major alarms usually require immediate	
		attention.	

man-msg-wt	Msg Wait	Manual Message Waiting: allows a	None
(Ext:	(name or	multiappearance telephone user to press a	
	ext #)	button on their telephone in order to light the Manual Message Waiting button at another telephone	
		You can administer this feature only to pairs of telephones, such as an assistant and an executive. For example, an assistant can press the man-msg-wt button to signal the executive that they have a call.	
man-overid	Immediate	Immediate Manual Override (*display*	1 per
(TOD: _)	Override	*button*): allows the user (on a system with Time of Day Routing) to temporarily override the routing plan and use the TOD: specify the routing plan the user wants to follow in override situations.	station
manual-in	Manual In	Manual-In Mode: prevents the user from	1 per split
(Group: __)		becoming available for new ACD calls	group
		upon completion of an ACD call by	
		automatically placing the agent in the after call work mode. Grp: The split	
		group number for ACD.	
mct-act	MCT	Malicious Call Trace Activation: sends a	
	Activation	message to the MCT control extensions	
		that the user wants to trace a malicious	
		call. MCT activation also starts recording	
		the call, if your system has a MCT voice	
		recorder.	

mct-contr	MCT Control	Malicious Call Trace Control: allows the user to take control of a malicious call trace request. Once the user becomes the MCT controller, the system stops notifying other MCT control extensions of the MCT request. **NOTE:** To add an extension to the MCT control group, you must also add the extension on the "Extensions Administered to have an MCT-Control Button" screen. When the user presses the MCT Control button, the system first displays the called party information. Pressing the button again displays the rest of the trace information. The MCT controller must dial the MCT Deactivate feature access code to release control.	
mf-da-intl	Directory Assistance	Multifrequency Operator International: allows users to call Directory Assistance.	1 per station
mf-op-intl	CO attendant	Multifrequency Operator International: allows users to make international calls to the CO attendant.	1 per station
mj/mn-alrm	Maj/Minor Hdwe Failure	Minor Alarm: assign to a status lamp to notify the user when minor or major alarms occur. Minor alarms usually indicate that only a few trunks or a few stations are affected.	1 per station
mm-basic	MM Basic	Multimedia Basic: used to place a multimedia complex into the "Basic" mode or to return it to the "Enhanced" mode.	1 per station

mm-call	MM Call	Multimedia Call: used to indicate a call is to be a multimedia call.	1 per station
mm-cfwd	MM CallFwd	Multimedia Call Forward: used to activate forwarding of multimedia calls as multimedia calls, not as voice calls.	1 per station
mm-datacnf	MM Datacnf	Multimedia Data Conference: used to initiate a data collaboration session between multimedia endpoints; requires a button with a lamp.	1 per station
mm-multnbr	MM MultNbr	Indicate that the user wants to place calls to 2 different addresses using the 2 B-channels.	1 per station
mm-pcaudio	MM PCAudio	Switches the audio path from the telephone (handset or speakerphone) to the PC (headset or speakers/microphone).	1 per station
msg-retr	Message Retrieve	Message Retrieval (*display button*): places the station's display into the message retrieval mode.	1 per station
mwn-act	Message Waiting Act.	Message Waiting Activation: lights a message waiting lamp on an associated station.	1 per station
mwn-deact	Message Waiting Deact	Message Waiting Deactivation: dims a message waiting lamp on an associated station.	1 per station
next	Next	Next (*display button*): steps to the next message when the phone's display is in Message Retrieval or Coverage Message Retrieval mode. Shows the next name when the phone's display is in the Directory mode.	1 per station
night-serv	Night Serv	Night Service Activation: toggles the system in or out of Night Service mode.	1 per Station

noans-alrt	RONA	Redirection on No Answer Alert: indicates occurred for the split.	1 per split
normal	Normal	Normal (*display button*): places the	1 per
	Mode	station's display into normal call	station
		identification mode.	
off-bd-alm	Off board alarm	Off board Alarm: associated status lamp lights if an off-circuit pack major, minor, or warning alarm is active on a circuit pack. Off-board alarms (loss of signal, slips, misframes) relate to problems on the facility side of the DS1, ATM, or	
		other interface.	
per-COline	CO Line	Personal CO Line: allows the user to	1 per group
(Grp:	(line #)	receive calls directly via a specific trunk.	
		Grp: CO line group number.	
pms-alarm	PMS Failure	Property Management System alarm:	1 per
		associated status lamp indicates that a	station
		failure in the PMS link occurred. A major	
		or minor alarm condition raises the alarm.	
pr-awu-alm	Auto	Automatic Wakeup Printer Alarm:	1 per
	Wakeup Alm	associated status lamp indicates that an automatic wakeup printer interface failure occurred.	station
pr-pms-alm	PMS Ptr	PMS Printer Alarm: associated status	1 per
	Alarm	lamp indicates that a PMS printer interface failure occurred.	station
pr-sys-alm	Sys Ptr	System Printer Alarm: associated status	1 per
	Alarm	lamp indicates that a system printer failure occurred	station
print-msgs	Print Msgs	Print Messages: allows users to print	1 per
		messages for any extension by pressing	station
		the button and entering the extension and a security code.	

priority	Priority Call	Priority Calling: allows a user to place priority calls or change an existing call to a priority call	1 per station
q-calls (Grp:	NQC	Queue Calls: associated status lamp flashes if a call warning threshold has been reached. Grp: Group number of hunt group.	1 per hunt group per station
q-time (Grp:	OQT	Queue Time: associated status lamp flashes if a time warning threshold has been reached. Grp: Group number of hunt group.	1 per hunt group per station
release	Release	Releases an agent from an ACD call.	1 per Station
ringer-off	R Cutoff inger	Ringer-Cutoff: silences the alerting ringer on the station.	1 per station
rs-alert	System Reset Alert	The associated status lamp lights if a problem escalates beyond a warm start.	1 per station
rsvn-halt	rsvn-halt	Remote Access Barrier Code Security Violation Notification Call: activates or deactivates call referral when a remote access barrier code security violation is detected.	1 per system
scroll	Scroll	Scroll (*display button*): allows the user to select one of the two lines (alternates with each press) of the16-character LCD display. Only one line displays.	1 per station
send-calls(Ext:)	Send All Calls	Send All Calls allows users to temporarily direct all incoming calls to coverage regardless of the assigned call-coverage redirection criteria. Assign to a lamp button.	
send-term	Send All Calls-TEG	Send All Calls For Terminating Extension Group: allows the user to forward all calls directed to a terminating extension group.	1 per TEG

serv-obsrv	Service Observing	Service Observing: activates Service Observing. Used to toggle between a listen-only and a listen-talk mode.	1 per station
signal (Ext:)	Signal (name or ext #)	Signal: allows the user to use one button to manually signal the associated extension. The extension cannot be a VDN extension.	1 per signal extension
ssvn-halt	ssvn-halt	Toggle whether or not station security code violation referrals are made to the referral destination.	1 per station
stored-num	Stored Number	(*display button*) Places the station's display into the stored number mode.	1 per station
stroke-cnt	ACD SD Stroke Count	Automatic Call Distribution Single Digit Stroke Count: sends a message to CMS to increment a stroke count number.	1 per station
term-x-gr (Grp:)	Term Grp (name or ext #)	Terminating Extension Group: provides one or more extensions. Calls may be received but not originated with this button. Grp: TEG number.	1 per TEG
timer	Timer	(*display button*) Starts a clock on the station to display elapsed time.	1 per station
trk-ac-alm	FTC Alarm	Facility Test Call Alarm: associated status lamp lights when a successful Facility Test Call (FTC) occurs.	
trk-id	Trunk ID	Trunk Identification (*display button*): identifies the tac (trunk access code) and trunk member number associated with a call.	1 per station
trunk-name	Trunk Name	(*display button*) Displays the name of the trunk as administered on the CAS Main or on a switch without CAS.	1 per station
trunk-ns (Grp:	Trunk Grp	Trunk-Group Night Service: places a trunk-group into night service. Grp: Trunk group number.	1 per trunk group

verify	Verify	Busy Verification: allows users to make	1 per
		test calls and verify a station or a trunk.	station
vip-retry	VIP Retry	VIP Retry: starts to flash when the user places a VIP wakeup call and continues to flash until the call is answered. If the VIP wakeup call is not answered, the user can press the VIP Retry button to drop the call	1 per station
		and reschedule the VIP wakeup call as a	
		classic wakeup call.	
		To assign this button, you must have both	
		Hospitality and VIP Wakeup enabled.	
vip-wakeup	VIP	VIP Wakeup: flashes when a VIP wakeup	1 per
	Wakeup	reminder call is generated. The user	station
		presses the button to place a priority (VIP)	
		wakeup call to a guest.	
		To assign this button, you must have both	
		Hospitality and VIP Wakeup enabled.	
voa-repeat	VOA repeat	VDN of Origin Announcement. VDN of	1 per
		Origin Announcement must be enabled.	station
vu-display	VuStats #	VuStats Display: allows the agent to	limited to
(format: __		specify a display format for the statistics.	the number
ID: __)		If you assign a different VuStats display	of feature
		format to each button, the agent can use	buttons on
		the buttons to access different statistics.	the phone
		You can assign this button only to display	
		phones.	

Button name	Button label	Description	Maximum
whisp-act	Whisper	Whisper Page Activation: allows a user to	
	Page	make and receive whisper pages. A	
	Activation	whisper page is an announcement sent to	
		another extension who is active on a call	
		where only the person on the extension	
		hears the announcement; any other parties	
		on the call cannot hear the announcement.	
		The user must have a class of restriction	
		(COR) that allows intraswitch calling to	
		use whisper paging.	
whisp-anbk	Answerback	Whisper Page Answerback: allows a user	
		who received a whisper page to respond to	
		the user who sent the page.	
whisp-off	Whisper	Deactivate Whisper Paging: blocks other	
	Page Off	users from sending whisper pages to this	
		phone.	
work-code	Work Code	Call Work Code: allows an ACD agent	1 per
		after pressing "work-code" to send up to	station
		16 digits (using the dial pad) to CMS.	

Attendant console feature buttons

The following table lists the feature buttons that you can assign to an attendant console.

Attendant console feature buttons

Feature or Function	Recommended Button Label	Name Entered on Station form	Maximum Allowed
Abbreviated Dialing	AD	abrv-dial (List: DC)	1 per List/DC
Administered Connection [status lamp]	AC Alarm	ac-alarm	1
Alert Agent of Change to Split/Skill Hunt Group	Alert Agent	alrt-agchg	1
Automatic Call Distribution (ACD)	After Call Work	after-call (Grp. No.__)	N
	Assist	assist (Grp. No:__)	1 per split group
	Auto In	auto-in (Grp. No.__)	1 per split group
	Auxiliary Work	aux-work (Grp. No.__)	1 per split group
	Manual-In	manual-in (Grp. No.__)	1 per split group
	Release	release	1
	Work Code	work-code	1
	Stroke (0-9)	stroke-cnt (Code:_)	1
Attendant Console (Calls Waiting)	CW Aud Off	cw-ringoff	1
Attendant Control of Trunk Group Access (Activate)	Cont Act	act-tr-grp	1

Attendant Control	Cont Deact	deact-tr-g	1
of Trunk Group			
Access (Deactivate)			
Attendant Direct	Local TG	local-tgs	12
Trunk Group Select	Remote TG	remote-tgs	
Attendant Crisis	Crisis Alert	crss-alert	1
Alert			
Attendant Display	Date/Time	date-time	1
[display buttons]			
	Inspect Mode	inspect	1
	Normal Mode	normal	1
	Stored Number	stored-num	1
Attendant Hundreds	Group Select _	hundrd-sel (Grp:__)	20 per
Group Select			console
Attendant Room	OccupiedRooms	occ-rooms	1
Status	Status		
	Maid Status	maid-stat	1
Attendant Override	Override	override	1
Automatic Circuit	ACA	aca-halt	1 per system
Assurance			
Automatic Wakeup	Auto Wakeup	auto-wkup	1
(Hospitality)			
Busy Verification	Busy Verify	verify	1
Call Coverage	Cover Cback	cov-cback	1
	Consult	consult	1
	Go To Cover	goto-cover	1
Call Coverage	Cover Msg Rt	cov-msg-rt	1
[display button]			
Call Offer	Intrusion	intrusion	1
(Intrusion)			

Call Prompting	Caller Info	callr-info	1
[display button]			
Call Type	Call Type	type-disp	1
Centralized	CAS-Backup	cas-backup	1
Attendant Service			
Check In/Out	Check In	check-in	1
(Hospitality)			
	Check Out	check-out	1
[display buttons]			
Class of Restriction	COR	class-rstr	1
[display button]			
Demand Print	Print Msgs	print-msgs	1
DID View	DID View	did-view	1
Do Not Disturb	Do Not Disturb	dn-dst	1
(Hospitality)			
Do Not Disturb	Do Not Disturb	ext-dn-dst	1
(Hospitality)	Ext		
[display buttons]			
	Do Not Disturb	grp-dn-dst	1
	Grp		
Don't Split	Don't Split	dont-split	1
Emergency Access	Emerg. Access	em-acc-att	1
To the Attendant	To Attd		
Facility Busy	Busy (trunk or	busy-ind (TAC/Ext:	1 per
Indication [status	extension#)		TAC/Ext.
lamp]			
Facility Test Calls	FTC Alarm	trk-ac-alm	1
[status lamp]			
Group Display	Group Display	group-disp	1
Group Select	Group Select	group-sel	1

Hardware Failure	Major Hdwe	major-alrm	10 per system
[status lamps]	Failure		
	Auto Wakeup	pr-awu-alm	1
	DS1 (facility)	ds1-alarm	10 per sys
	PMS Failure	pms-alarm	1
	PMS Ptr Alm	pr-pms-alm	1
	CDR 1 Failure	cdr1-alrm	1
	CDR 2 Failure	cdr2-alrm	1
	Sys Ptr Alm	pr-sys-alm	1
Hold	Hold	hold	1
Integrated Directory[display button]	Integrtd Directory	directory	1
Incoming Call Identification	Coverage(Group	in-call-id	N
	number, type, name, or		
	ext.#)		
Intrusion (Call Offer)	Intrusion	intrusion	1
Leave Word Calling	Cancel LWC	lwc-cancel	1
	LWC	lwc-store	1
Leave Word Calling	Delete Msg	delete-msg	1
[display buttons]			
	Next	next	1
	Call Display	call-disp	1
Leave Word Calling	Msg (name or	aut-msg-wt (Ext	N
(Remote Message	extension #)		
Waiting) [status lamp]			
Link Failure	Link Failure(Link No.)	link-alarm (Link No.)	1 per Link #

	lsvn-halt	lsvn-halt	1 per system
Violation			
Message Waiting	Message	mwn-act	1 per system
	Waiting Act.		
	Message	mwn-deact	1 per system
	Waiting Deact.		
Night Service	Trunk Grp. NS	trunk-ns (Grp. No.__)	1 per trunk
			group
PMS Interface	PMS display		
[display buttons]			
Priority Calling	Prior Call	priority	N
Position Busy	Position Busy	pos-busy	1
Queue Status	AQC	atd-qcalls	1
Indications (ACD)			
	AQT	atd-qtime	
[display buttons]			
Queue Status	NQC	q-calls (Grp:_)	1
Indications (ACD)			
	OQT	q-time Grp:_)	1 per hunt
[status lamps]			
			group
Remote Access	rsvn-halt	rsvn-halt	1 per system
Security Violation			
Ringing	In Aud Off	in-ringoff	1
Security Violation	ssvn-halt	ssvn-halt	1 per system
Notification Halt			
Serial Call	Serial Call	serial-cal	1
Split/Swap	Split-swap	split-swap	1
System Reset Alert	System Reset	rs-alert	1
	Alert [status		
	lamp]		
Station Security	ssvn-halt	ssvn-halt	1 per system

Feature or Function	Recommended Button Label	Name Entered on Station form	Maximum Allowed
Night Service (ACD)	Hunt Group	hunt-ns (Grp. No.__)	3 per hunt group
Time of Day Routing [display buttons]	Immediate Override	man-ovrid	1
	Clocked Override	clk-overid	1
Timed Reminder	RC Aud Off	re-ringoff	1
Timer	Timer	timer	1
Trunk Identification [display button]	Trunk-ID	trk-id	1
Trunk Group Name [display button]	Trunk-Name	trunk-name	1
Visually Impaired Service (VIAS)	VIS	vis	1
	Console Status	con-stat	1
	Display	display	1
	DTGS Status	dtgs-stat	1
	Last Message	last-mess	1
	Last Operation	last-op	1
VDN of Origin Announcement Repeat	VOA Repeat	voa-repeat	1
VuStats	VuStats	vu-display	1

N = any number of buttons on the phone can be assigned to this feature.

Managing your attendant consoles

Overview

The attendant console is the main answering position for your organization. The console operator is responsible for answering incoming calls and for efficiently directing or "extending" calls to the appropriate phone.

Console Parameters

This screen administers attendant console group parameters. This includes basic parameters for Centralized Attendant Service (CAS) and Inter-PBX Attendant Service (IAS). A list of the administered attendant consoles also displays on this screen.

Field descriptions for page 1

```
change console-parameters                                        Page   1 of   4
                              CONSOLE PARAMETERS
        Attendant Group Name: OPERATOR
                         COS: 0                                        COR: 0
     Calls in Queue Warning: 5                     Attendant Lockout? y
      Ext Alert Port (TAAS):
                         CAS: none
                                                 Night Service Act. Ext.:
              IAS (Branch)? n                    IAS Tie Trunk Group No.:
      IAS Att. Access Code:                        Alternate FRL Station:
           Backup Alerting? n          DID-LDN Only to LDN Night Ext? n
```

Console Parameters — Default Attendant Group

Attendant Group Name

Enter a name for the attendant group.

COS

Enter a class of service (COS) number that reflects the desired features for all your attendant consoles. You can override this COS, by assigning a different COS on the individual Attendant screen.

COR

Enter the class of restriction (COR) number that reflects the desired features for the attendant. You can override this COR, by assigning a different COR on the individual Attendant screen.

Calls In Queue Warning

Enter the number of incoming calls that can be in the attendant queue before the console's second Call Waiting lamp lights. The console's first Call Waiting lamp lights when any incoming calls are waiting to be answered. The second lamp lights when the number of calls waiting equals the value you entered in the Calls in Queue Warning field.

Attendant Lockout

Attendant Lockout prevents an attendant from re-entering a multiple-party connection held on the console unless recalled by a telephone user.

Attendant Lockout provides privacy for parties on a multiple-party call held on the console. The held parties can hold a private conversation without interruption by the attendant.

Valid entries	Usage
y/n	Enter **y** to activate Privacy — Attendant Lockout. If **y** is entered, the attendant is prohibited from reentering a conference call that has been placed on hold unless recalled by a phone user on the call.

Ext Alert Port (TAAS)

Enter the seven-digit port number assigned to the external alerting device. This supports the Night Service — Trunk Answer From Any Station feature.

NOTE:
Type an "X" in this field to indicate that there is no hardware associated with this port assignment. If an **X** is used here, you must also fill in the Ext Alert (TAAS) Extension field.

Ext Alert (TAAS) Extension

Appears only when an **X** is entered in the Ext Alert Port (TAAS) field. This extension is used by the Terminal Translation Feature (TTI) to assign a port to the

Ext Alert Port from a station on the Ext Alert port during system installation or provisioning. Once a port is assigned (either via TTI or by changing the Ext Alert Port field from the G3-MA or other manager terminal) the extension is automatically removed and treated as unassigned.

CAS

The CAS Main or Branch features must be enabled on the System- Parameters Customer-Options screen for either of these features to be functional here.

Valid entries	Usage
Main	
Branch	
None	
QSIG-main	Can be used if, on the System-Parameters Customer-Options screen, the Centralized Attendant field is **y**. Indicates all attendants are located on the main PBX.
QSIG-branch	Can be used if, on the System-Parameters Customer-Options screen, the Centralized Attendant field is **y**. Indicates there are no local attendants and routes to the main PBX.

RLT Trunk Group No.

Appears only when **branch** is entered in the CAS field. Enter the trunk group number corresponding to the Release Link Trunk (RLT) trunk group to the main location when supporting CAS Branch service.

CAS Back-Up Ext.

This field handles attendant-seeking calls if the RLT trunk group to the CAS Main switch is out of service or if CAS Back-Up is activated. This field must be explicitly defined as an extension in the dial plan. Neither a prefixed extension nor a VDN extension is allowed. Appears only when **branch** is entered in the CAS field.

Valid entries

An extension number for a station

individual attendant console

hunt group

TEG

AAR/ARS Access Code

Appears if the CAS field is **QSIG-branch**. An optional field that contains an AAR/ARS access code to route to the main PBX, if needed.

Valid entries	Usage
0 - 9, *, #	Enter up to 4 digits.
blank	

Night Service Act. Ext.

This is a display-only field. It contains the extension of the current night service activation station, if any. Such a station is administered by assigning it a "night-serv" button.

IAS (Branch)

Enables or disables Inter-PBX Attendant Service (IAS) Branch feature. Does not appear if, on the System-Parameters Customer-Options screen, the Centralized Attendant field is **y**.

NOTE:
CAS and IAS cannot both be active at the same time.

QSIG CAS Number

Appears if the CAS field is **QSIG-branch**. Contains the complete number of the attendant group at the main switch, or a vector directory number (VDN) local to the branch switch. This field cannot be left blank.

Valid entries	Usage
0 - 9	Enter up to 20 digits.

IAS Tie Trunk Group No.

Enter the number of the tie trunk group to the main for the IAS (Branch). This entry is required when IAS Branch is **y**. Does not appear if, on the System-Parameters Customer-Options screen, the Centralized Attendant field is **y**.

IAS Att. Access Code

Enter the extension number of the attendant group at the main switch. This entry is required when IAS Branch is **y**. Does not appear if, on the System-Parameters Customer-Options screen, the Centralized Attendant field is **y**.

Alternate FRL Station

This is a display-only field. It displays the extension of the alternate facility restriction level (FRL) activation station.

Backup Alerting

Indicates whether or not system users can pick up alerting calls if the attendant queue has reached its warning state.

DID-LDN Only to LDN Night Ext.

Valid entries	Usage
y	Enter **y** to allow only listed directory number (LDN) calls to go to the listed directory night service extension.
n	Enter **n** if you want all attendant seeking calls to route to the LDN night service extension.

Field descriptions for page 2

```
change console-parameters                                          Page   2 of   4
                              CONSOLE PARAMETERS

TIMING
  Time Reminder on Hold (sec): 10          Return Call Timeout (sec): 10
  Time in Queue Warning (sec):
  INCOMING CALL REMINDERS
      No Answer Timeout (sec): 20                     Alerting (sec): 40
                               Secondary Alert on Held Reminder Calls? y
ABBREVIATED DIALING
     List1: group    1        List2:                    List3:
            SAC Notification? n
                            COMMON SHARED EXTENSIONS
            Starting Extension:                  Count:
```

Timed Reminder on Hold (sec)

Enter the time in seconds that a call remains on hold at the console before the attendant is alerted. In a CAS arrangement, the main and the branch consoles (when administered) should be administered the same.

Return Call Timeout (sec)

Enter the time in seconds before a split away call (call extended and ringing a station or otherwise split away from the console) returns to the console. Be sure to allow five seconds for each ring at all points in a coverage path to ensure the entire path is completed before the call returns to the console.

Time In Queue Warning (sec)

Enter the number of seconds a call can remain in the attendant queue before activating an alert.

No Answer Timeout (sec)

Enter the number of seconds a call to the attendant can remain unanswered without invoking a more insistent sounding tone. Be sure to allow five seconds for each ring at all points in a coverage path to ensure the entire path is completed before the call returns to the console.

Alerting (sec)

Enter the number of seconds after which a held or unanswered call is disconnected from an attendant loop and routed to another attendant or night service

Secondary Alert on Held Reminder Calls?

Valid entries	Usage
y	Enter **y** to begin attendant alerting for Held Reminder Calls with secondary alerting.
n	Enter **n** to have held reminder calls alert the attendant the same as normal calls. Normal calls start with primary alerting and switch to secondary alerting when the No Answer Timeout expires.

List1, List2, List3

You can assign up to 3 abbreviated dialing lists to each attendant. However, you cannot assign a personal list to an attendant.

Valid entries	Usage
enhanced	Allows the attendant to access the enhanced system abbreviated dialing list.
group	Allows the attendant to access the specified group abbreviated dialing list. You also must enter a group number.
system	Allows the attendant to access the system abbreviated dialing list.

SAC Notification

Enables or disables Enhanced Attendant Notification for Send All Calls.

Common Shared Extension—Starting Extension

These extension numbers can be used by the attendant to park calls.

Common Shared Extension—Count

Enter a number to indicate the number of consecutive extensions, beginning with the Start Extension to be used as common, shared extensions. For example, if you enter a starting extension of 4300 and a count of 3, the system provides three consecutive extension numbers (4300, 4301, and 4302) for parking calls.

The extensions should be assigned to the optional Attendant Selector Console in the 00 through 09 block (bottom row) in any hundreds group for easy identification by the attendant. The lamp associated with the number will identify "call parked" or "no call parked", instead of busy or idle status.

Field descriptions for page 3

```
change console-parameters                                        Page 3 of 4
                                CONSOLE PARAMETERS
QUEUE PRIORITIES
            Emergency Access:1_
             Assistance Call:2_
                     CO Call:2_
            DID to Attendant:2_
                    Tie Call:2_
         Redirected DID Call:2_
             Redirected Call:2_
                 Return Call:2_
                 Serial Call:2_
 Individual Attendant Access:2_
             Interpositional:2_
    VIP Wakeup Reminder Call:2_
          Miscellaneous Call:2_

 Call-Type Ordering Within Priority Levels? n
```

Queue Priorities

Attendant Priority Queue allows attendants to answer calls by call category (for example, by trunk type). The Attendant Priority Queue handles incoming calls to an attendant when the call cannot be immediately terminated to an attendant. The calling party hears ringback until an attendant answers the call.

You may assign the same priority level to more than one call. Priority 1 is the highest priority and is the default for Emergency Access. Assign a priority level from **1** through **13** to each of the call types.

The attendant call categories are:

- Emergency Access — A call from a telephone user who dials the emergency access code (default is highest-priority level)
- Assistance Call— A call from a telephone user who dials the attendant-group access code, or from a telephone that has the Manual Originating Line Service feature activated
- CO Call — An incoming trunk call (CO/FX/WATS trunk) to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- DID to Attendant — An incoming DID trunk call to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- Tie Call — An incoming TIE trunk call (dial-repeating or direct types) to an attendant group. This does not include trunk calls that return to the attendant group after a timeout or deferred attendant recall.
- Redirected DID Call — A DID or ACD call that times out due to ring/no-answer, busy condition (if applicable), or Number Unobtainable and reroutes to the attendant group.
- Redirected Call — A call assigned to one attendant, but redirected to the attendant group because the attendant is now busy
- Return Call — A call returned to the attendant after it times out. If the attendant is now busy, the call redirects to the attendant group.
- Serial Call — A call from the Attendant Serial Call feature when an outside trunk call (designated as a serial call by an attendant) is extended to and completed at a telephone, and then the telephone user goes on-hook. If the attendant who extended the call is busy, the call redirects to the attendant group.
- Individual Attendant Access — A call from a telephone user, incoming trunk call, or a system feature to the Individual Attendant Access (IAA) extension of a specific attendant. If the attendant is busy, the call queues until the attendant is available.
- Interposition — A call from one attendant to the Individual Attendant Access (IAA) extension of another attendant

- VIP Wakeup Reminder Call — A VIP Wakeup reminder call.
- Miscellaneous Call — All other calls.

Call-Type Ordering Within Priority Levels?

If you use call-type ordering, calls to the attendant are first grouped by the queue priority level, then by call type, and, finally, in the order received.

Valid entries	Usage
y	Enter **y** if you want to present calls by call type. You can assign a type-disp button on the Attendant Console screen so that the attendant can review the call type for the active call.
n	Enter **n** if you wish the calls to be queued in chronological order by queue priority level.

The call types, in descending order of priority, are:

- Type 1 call: outgoing public-network calls receive answer supervision when the Answer Supervision Timer of the trunk group expires, even if the trunk is actually still ringing. Also, incoming calls when answered by the attendant.
- Type 2 call: incoming external public-network calls before they receive answer supervision or before the Answer Supervision Timer of the trunk group expires
- Type 3 call: all other calls (internal calls, conference calls, and tie-trunk calls of any type)

Note that external public-network calls have priority over all other calls including conference calls. And, answered public-network calls have priority over those calls not yet answered.

Field descriptions for page 4

```
change console-parameters                                          Page 4 of 4
                                CONSOLE PARAMETERS

  ASSIGNED MEMBERS ( Installed attendant consoles )
            Type          Grp  TN          Type          Grp  TN
            1:             1   1           9:             1   1
            2:             1   1          10:             1   1
            3:             1   1          11:             1   1
            4:             1   1          12:             1   1
            5:             1   1          13:             1   1
            6:             1   1          14:             1   1
            7:             1   1          15:             1   1
            8:             1              16:             1   1
```

ASSIGNED MEMBERS (Installed attendant consoles)

Display-only field that shows all attendants in the group. You administer the individual attendant consoles on the Attendant Console screen.

Grp

Display-only field that lists the Attendant Group number.

TN

Display-only field that lists the Tenant Partition number.

Adding an attendant console

Usually Avaya connects and administers your primary attendant console during cutover. However, you might find a need for a second attendant console, such as a backup console that is used only at night.

Type **add attendant 1** and press RETURN. The Attendant Console screen appears.

Field descriptions for page 1

```
change attendant                                                    Page   1 of   3
                              ATTENDANT CONSOLE 1

         Type: console            Name: 27 character attd cons name
    Extension: 1000              Group: 1                 Auto Answer: none
 Console Type: principal            TN: 1                 Data Module? n
         Port: 01C1106             COR: 1           Disp Client Redir? n
Security Code:                     COS: 1            Display Language: english
                                                    H.320 Conversion? n

DIRECT TRUNK GROUP SELECT BUTTON ASSIGNMENTS (Trunk Access Codes)
    Local Remote            Local Remote              Local Remote
 1: 9                    5:                       9:
 2: 82                   6:                      10:
 3:                      7:                      11:
 4:                      8:                      12:

HUNDREDS SELECT BUTTON ASSIGNMENTS
 1:           5:           9:           13:           17:
 2:           6:          10:           14:           18:
 3:           7:          11:           15:           19:
 4:           8:          12:           16:           20:
```

Attendant Console

If you give your attendants an individual extension, users can call the attendant directly by dialing the extension.

Individual attendant extensions also allow attendants to use features that an attendant group cannot use — for example, you can assign them to hunt groups.
In the Console Type field, **enter principal, day-only, night-only or day/night**.

This indicates how this console is used in your organization—as a principal, day only, night only, or day/night console. You can have only one night-time console (night only or day/night) in the system.

In the Port field, enter the port address for this console.

Type a name to associate with this console in the Name field.

In the Direct Trunk Group Select Button Assignments fields, enter trunk access codes for the trunks you want the attendant to be able to select with just one button.

If you are using the Enhanced Selector console, assign the Hundreds Select Buttons that you want this console to have.

If you want this console to be able to access extensions in the range 3500 to 3999, you need to assign them 5 Hundreds Select Buttons: **35** for extensions 3500 to 3599, **36**, **37**, **38**, and **39**.

Assign the Feature Buttons that you want the console to have.

Tip:
Feature buttons are not numbered top-to-bottom on the attendant console, as you might expect. Button numbers map to physical positions on the console.

Field descriptions for page 2 (non IP)

```
add attendant                                                    Page 2 of 4
                                   ATTENDANT
DATA MODULE
     Data Extension: ____            BCC: 2        ITC: restricted
               Name: ______________  COR: 1_       COS: 1_
                                      TN: 1_
ABBREVIATED DIALING
 List1: ________
SPECIAL DIALING OPTION: default
   DEFAULT DIALING
        Abbreviated Dialing Dial Code (From above list): _
ASSIGNED MEMBER (Station with a data extension button for this data module)
        Ext      Name
     1:
```

Field descriptions for page 3

```
change attendant                                                 Page Y of X
                              ATTENDANT CONSOLE
FEATURE BUTTON ASSIGNMENTS
  1: split_____                          13: _________
  2: _________                           14: _________
  3: _________                           15: _________
  4: _________                           16: _________
  5: _________                           17: _________
  6: hold _____ *                        18: _________
  7: _________                           19: forced-rel
  8: aux-work      RC:    Grp:           20: _________
  9: _________                           21: _________
 10: _________                           22: _________
 11: _________                           23: night-serv *
 12: _________                           24: pos-busy__ *
```

```
change attendant                                                    Page Y of X
                                   ATTENDANT CONSOLE
DISPLAY MODULE BUTTON ASSIGNMENTS
  1: normal____                                  5: delete-msg
  2: inspect___                                  6: call-disp_
  3: cov-msg-rt                                  7: date-time_
  4: next______                                  8: timer_____
```

Attendant Display Module Button Assignments

Class of Restriction (COR)

Use this screen to establish classes of restriction (COR). Classes of restriction control call origination and termination. Your system may use only one COR or as many as necessary to control calling privileges. You can assign up to 96 different (996 in v12)

Field descriptions for page 1

```
change cor 10                                                       Page 1 of 4
                          CLASS OF RESTRICTION

                 COR Number: 10
            COR Description: supervisor

                        FRL: 0                                          APLT? y
   Can Be Service Observed? n                  Calling Party Restriction: none
 Can Be A Service Observer? y                   Called Party Restriction: none
          Time of Day Chart: 1               Forced Entry of Account Codes? n
           Priority Queuing? n                         Direct Agent Calling? y
      Restriction Override: none                 Facility Access Trunk Test? n
      Restricted Call List? n                          Can Change Coverage? n
    Unrestricted Call List? __ __ __ __ __ __ __ __ __ __
             Access to MCT? y                   Fully Restricted Service? n
 Group II Category For MFC: 7                    Hear VDN of Origin Annc.? n
           Send ANI for MFE? n_                   Add/Remove Agent Skills? y
              MF ANI Prefix: _____              Automatic Charge Display? n
 Hear System Music on Hold? y        PASTE(Display PBX Data on telephone)? n
                           Can Be Picked Up By Directed Call Pickup? n
                                          Can Use Directed Call Pickup? n
                                          Group Controlled Restriction: inactive
```

Class of Restriction

COR Number

This is a display-only field when the screen is accessed via an administration command such as **change** or **display**.

COR Description

Valid entries	Usage
Up to 35	Enter a description of the COR.

FRL

Valid entries	Usage
0 to **7**	Enter an originating FRL number. AAR and/or ARS features use this entry to determine call access to an outgoing trunk group. Outgoing call routing is determined by a comparison of the FRLs in the AAR/ARS Routing Pattern and the FRL associated with the COR of the call originator (typically, a telephone user). An originating FRL of 0 has the least calling privileges. To enhance system security, assign the lowest possible FRL.

APLT

Valid entries	Usage
y/n	Enter **n** to allow access to APLT trunk group

Can Be Service Observed

Note that this field allows or denies service observing for not only physical extensions, but also for logical agent IDs and VDNs. If you want an observer to observe users, set the users' CORs to **y** on the observer's COR Service Observing Permission table.

Valid entries	Usage
y/n	Enter **y** if users with this COR can be service observed.

Calling Party Restriction

This field determines the level of calling restriction associated with this COR.

NOTE:
To enhance system security, limit calling permissions as much as possible.

Valid entries	Usage
Origination	Blocks the calling party from originating a call from the facility
Outward	Blocks the calling party from calling outside the private network.
All-toll	Blocks the calling party from making ARS and trunk access calls
Tac-toll	Blocks the calling party from making trunk access
None	No calling party restrictions.

Can Be a Service Observer

If you want an observer to observe users, set the users' CORs to **y** on the observer's COR Service Observing Permission table.

Called Party Restriction

Valid entries	Usage
Inward	Blocks the calling party from receiving incoming exchange
Manual	Blocks the called party from receiving all calls except for those originated or extended by the attendant.
Public	Blocks the called party from receiving public network calls.
Termination	Blocks the called party from receiving any calls at any time.
None	No called party restrictions.

Partitioned Group Number

This field appears only if AAR/ARS Partitioning is **y** and Time of Day Routing is **n** on the System Parameters Customer-Options screen.

Time of Day Chart

Appears only if Time of Day field is enabled on the System Parameters Customer-Options screen.

Forced Entry of Account Codes

FEAC must be enabled on the System Parameters Customer-Options screen and on the CDR System Parameters screen.

Priority Queuing

Enter **y** to allow the telephone user's calls to be placed ahead of non-priority calls in a hunt group queue If you do not use Automatic Call Distribution (ACD is not enabled on the System-Parameters Customer-Options screen), this field must be **n**.

Direct Agent Calling

If this is y, users may dial an ACD agent's extension directly, rather than anyone in the agent pool. If the system is in Night Service, the call routes to the Night Service extension. If the extension with this COR belongs to an agent, the agent may receive calls directly.

Restriction Override

Allows the specified users to bypass restriction on conference, transfer or call forwarding operations.

Facility Access Trunk Test

An associated feature button ("trk-ac-alm") status lamp lights when a successful test attempt occurs. Pressing one of the alarm buttons (ten maximum) when its associated status lamp is lit turns off all lamps on all buttons whether the access is still in progress or has completed.

Restricted Call List

This list can be used whether the COR is toll restricted. The Restricted Call List (RCL) has priority over the Toll Analysis Unrestricted Call List (UCL). A call attempt from a facility assigned a COR (with RCL field set to **y**),
whose dialed digit string is on the Toll Analysis screen and is marked as being associated with the RCL, will be denied.

Can Change Coverage

Unrestricted Call List

Any entry on the Toll Analysis screen with an "X" in the Toll List column is restricted, meaning that the system blocks any attempt to complete a call containing the Dialed String. However, this field overrides that restriction. For example, if the Toll Analysis screen shows a Dialed String entry of 538 and there is an "X" in the Toll List column, the 538 number is restricted. To override this restriction, in the Toll Analysis screen, enter **X** in the "5" column under the Unrestricted Call List heading. In the Class of Restriction screen, in this field, enter **5** to complete the restriction override.

Hear System Music on Hold

Can Be Picked Up By Directed Call Pickup

Can Use Directed Call Pickup

Field descriptions for page 3

```
change cor 10                                                    Page 3 of 4
                          CLASS OF RESTRICTION
CALLING PERMISSION (Enter y to grant permission to call specified COR)
  0? n     12? n     24? n     36? n     48? n     60? n     72? n     84? n
  1? n     13? n     25? n     37? n     49? n     61? n     73? n     85? n
  2? n     14? n     26? n     38? n     50? n     62? n     74? n     86? n
  3? n     15? n     27? n     39? n     51? n     63? n     75? n     87? n
  4? n     16? n     28? n     40? n     52? n     64? n     76? n     88? n
  5? n     17? n     29? n     41? n     53? n     65? n     77? n     89? n
  6? n     18? n     30? n     42? n     54? n     66? n     78? n     90? n
  7? n     19? n     31? n     43? n     55? n     67? n     79? n     91? n
  8? n     20? n     32? n     44? n     56? n     68? n     80? n     92? n
  9? n     21? n     33? n     45? n     57? n     69? n     81? n     93? n
 10? n     22? n     34? n     46? n     58? n     70? n     82? n     94? n
 11? n     23? n     35? n     47? n     59? n     71? n     83? n     95? n
```

CALLING PERMISSION

A **y** means an originating facility assigned this COR can be used to call facilities assigned this COR. Enter **n** for each COR number (0 through 95) that cannot be called by the COR being implement.

Field descriptions for page 4

```
change cor 10                                                    Page 4 of 4
                          CLASS OF RESTRICTION
SERVICE OBSERVING PERMISSIONS
(Enter y to grant permission to service observe specified COR)
  0? n     12? n     24? n     36? n     48? n     60? n     72? n     84? n
  1? n     13? n     25? n     37? n     49? n     61? n     73? n     85? n
  2? n     14? n     26? n     38? n     50? n     62? n     74? n     86? n
  3? n     15? n     27? n     39? n     51? n     63? n     75? n     87? n
  4? n     16? n     28? n     40? n     52? n     64? n     76? n     88? n
  5? n     17? n     29? n     41? n     53? n     65? n     77? n     89? n
  6? n     18? n     30? n     42? n     54? n     66? n     78? n     90? n
  7? n     19? n     31? n     43? n     55? n     67? n     79? n     91? n
  8? n     20? n     32? n     44? n     56? n     68? n     80? n     92? n
  9? n     21? n     33? n     45? n     57? n     69? n     81? n     93? n
 10? n     22? n     34? n     46? n     58? n     70? n     82? n     94? n
 11? n     23? n     35? n     47? n     59? n     71? n     83? n     95? n
```

SERVICE OBSERVING PERMISSION

A **y** grants permission to observe specific CORs. Enter **n** for each COR number (0 through 95) that cannot be observed by the COR being implemented.

Class of Service

This screen administers access permissions for call processing features that require dial code or feature button access.

NOTE:
Class of Service (COS) does not apply to trunk groups except for the Remote Access feature.

A COS assignment defines whether or not a telephone user may access or use the following features and functions. Up to 16 different COS numbers may be administered (0–15).

```
change cos                                                          Page 1 of 1
                              CLASS OF SERVICE

                                     0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15

    Auto Callback                    n y y n y n y n y n  y  n  y  n  y  n
    Call Fwd-All Calls               n y n y y n n y y n  n  y  y  n  n  y
    Data Privacy                     n y n n n y y y y n  n  n  n  y  y  y
    Priority Calling                 n y n n n n n n n y  y  y  y  y  y  y
    Console Permissions              n n n n n n n n n n  n  n  n  n  n  n
    Off-hook Alert                   n n n n n n n n n n  n  n  n  n  n  n
    Client Room                      n n n n n n n n n n  n  n  n  n  n  n
    Restrict Call Fwd-Off Net        n y y y y y y y y y  y  y  y  y  y  y
    Call Forward Busy/DA             n n n n n n n n n n  n  n  n  n  n  n
    Personal Station Access          n n n n n n n n n n  n  n  n  n  n  n
    Extended Forwarding All          n n n n n n n n n n  n  n  n  n  n  n
    Extended Forwarding B/DA         n n n n n n n n n n  n  n  n  n  n  n
    Trk-to-Trk Restriction Override  n n n n n n n n n n  n  n  n  n  n  n
    QSIG Call Offer Originations     n n n n n n n n n n  n  n  n  n  n  n
    Automatic Exclusion              n n n n n n n n n n  n  n  n  n  n  n
```

Class of Service screen

Automatic Callback

Allows this user to request Automatic Callback.

Call Forwarding All Calls

Allows this user to forward all calls to any extension.

Data Privacy

Allows this user to enter a feature access code to protect a data call from interruption by any of the system's override or ringing features.

Priority Calling

Allows user to dial a feature access code to originate a priority call. Such calls ring differently and override send all calls, if active.

Console Permissions

Console Permissions allow multiappearance telephone users to control the same features that the attendant controls. You might assign this permission to front-desk personnel in a hotel or motel, or to a call center supervisor. With console permission, a user can:

- Activate Automatic Wakeup for another extension
- Activate and deactivate controlled restrictions for another extension or group of extensions
- Activate and deactivate Do Not Disturb for another extension or group of extensions
- Activate Call Forwarding for another extension
- Add and remove agent skills
- Record integrated announcements

Off-Hook Alert

To enable this option, either the Hospitality (Basic) or Emergency Access to Attendant field must be enabled on the System-Parameters Customer-Options screen.

Client Room

Allows users to access Check-In, Check-Out, Room Change/Swap, and Maid status functions. In addition, Client Room is required at consoles or telephones that are to receive message-waiting notification. You can administer class of service for Client Room only when you have Hospitality Services and a Property Management System interface.

Restrict Call Fwd-Off Net

This restricts users from forwarding calls to the public network. For security reasons, this should be enabled for all classes of service except the ones you use for very special circumstances.

Call Forwarding Busy/DA

Allows this user to forward calls to any extension when the dialed extension is busy or does not answer.

Personal Station Access

Allows users to associate a telephone to their extension with their programmed services, using a feature access code. You cannot change this field to **y** if Personal Station Access (PSA) on the System Parameters Customer-Options screen is **n**.

Extended Forwarding All

Allows a user to administer call forwarding (for all calls) from a remote location. You cannot change a COS to **y** if Extended Cvg/Fwd Admin on the System Parameters Customer-Options screen is **n**.

Extended Forwarding B/DA

Allows this user to administer call forwarding (when the dialed extension is busy or does not answer) from a remote location. You cannot change this COS to **y** if Extended Cvg/Fwd Admin on the System Parameters Customer-Options screen is **n**.

Trk-to-Trk Restriction Override

Users with this COS override any system and/or COR-to-COR calling party restrictions that would otherwise prohibit the trunk-to-trunk transfer operation for users with this COS.

! **SECURITY ALERT:**
Use this COS capability with caution. The ability to perform trunk-to-trunk transfers greatly increases the risk of toll fraud.

QSIG Call Offer Originations

Allows this user to invoke QSIG Call Offer services.

Automatic Exclusion

Allows a user to activate automatically Exclusion when they go off hook on a station that has an assigned EXCLUSION button. If set to **n**, allows a user manual exclusion when they press the EXCLUSION button before dialing or during a call. Appears when, on the Feature-Related System Parameters screen, the Automatic Exclusion by COS field is **y**

Call Coverage

Call Coverage provides automatic redirection of calls to alternate answering positions in a Call Coverage path. Call Coverage allows you to:

- Establish coverage paths with up to 6 alternate answering positions
- Establish redirection criteria that govern when a call redirects
- Redirect calls to a local switch location
- Redirect calls to a remote (off-net) location
- Redirect calls based on time-of-day
- Allow users to change back and forth between two coverage choices (either specific lead coverage paths or time-of-day tables).

Field descriptions for page 1

```
change coverage path 2                                          Page  1 of  1
                          COVERAGE PATH

           Coverage Path Number: 2
                                          Hunt After Coverage: n
               Next Path Number: ___      Linkage: ___  ___

COVERAGE CRITERIA

  Station/Group Status   Inside Call   Outside Call
            Active?            n             n
              Busy?            y             y
      Don't Answer?            y             y Number of Rings:2
               All?            n             n
DND/SAC/Goto Cover?            y             y

COVERAGE POINTS

   Terminate to Coverage Pts. with Bridged Appearance? n

              Point1: ____ Point2: ____ Point3: ____
              Point4: ____ Point5: ____ Point6: ____
```

Coverage Path screen

Coverage Path Number

A display-only field indicating the coverage path being administered.

Hunt After Coverage

Y Coverage treatment continues by searching for an available station in a hunt chain that begins with the hunt-to-station assigned on the station screen of the last coverage point.

N Coverage treatment is terminated; the call is left at the last available location (principal or coverage point).

Next Path Number

Enter the next coverage path in a coverage path chain. Refer to "Call Coverage" on page 1210 for more information. If the coverage criteria of the current coverage path is not satisfied, the system steps down this chain until it finds a coverage path with redirection criteria that matches the call status. If the chain is exhausted before the system finds a match, the call does not redirect to coverage. No path number here indicates that this path is the only path for the principal.

Valid entries

1 to **999**

Linkage

Display-only fields that show the (up to) two additional coverage paths in the coverage path chain. (See above.)

Coverage Criteria

COVERAGE CRITERIA are the conditions that, when met, cause the call to redirect to coverage. Assign one of the following:

Valid entries	Usage
Active	Calls redirect if at least one call appearance is busy.
Busy	Calls redirect if all call appearances that accept incoming calls are busy
Don't Answer	Calls redirect when the specified number of rings has been exceeded.
All	Calls redirect immediately to coverage and overrides any other criteria with a y in this column.

DND/SAC/Go to Cover	Must be assigned before a user can activate Do Not Disturb (Hospitality Services), Send All Calls (SAC), or Go to Cover features. Allows a calling user, when calling to another internal extension, to redirect a call immediately to coverage by pressing a GO TO COVER button. Allows a principal temporarily to direct all incoming calls to coverage, regardless of the other assigned coverage criteria by pressing the SEND ALL CALLS (or DO NOT DISTURB)

COVERAGE POINTS

Terminate to Coverage Pts. with Bridged Appearances

Y Allows a call to alert as both a bridged call and a redirected call.

N The call skips the coverage point if it has already alerted as a bridged call.

Point1, Point2, Point3, Point4, Point5, Point6

The alternate destinations that comprise a coverage path. Coverage points must be assigned sequentially beginning with Point 1 (do not leave gaps). Each path can have up to six coverage points.

Valid entries	Usage
extension	Redirects the call to an internal extension or announcement
attd	Redirects the call to the attendant.
h1 to **h255**	Redirects the call to a hunt group.
c1 to **c750**	Redirects the call to a coverage answer group.
r1 to **r999**	Redirects the call to a remote coverage point
v + extension	Redirects the call to a VDN extension.

Time-of-Day Coverage

The Time-of-Day Coverage Table allows you to redirect calls to different lead-coverage paths at different times of the day and on different days of the week.

For example, an employee may want incoming calls to cover to a co-worker (office) during normal business hours, to cover to an off-net destination (home) in the early evening, and to cover to AUDIX at all other times. By specifying the appropriate lead-coverage paths in the Time-of-Day Coverage Table, the employee can have the call redirection flexibility shown in the following table. (If you were actually administering a Time-of-Day Coverage Table, you would provide the lead-coverage path numbers that redirect the calls to the employee's office, to their home, and to AUDIX.

Off-Net Call Coverage

Call Coverage allows a call to be redirected to a destination on the public network. The remote (off-net) number is administered on the Remote Call Coverage Table screen and may have up to 16 digits including either the outgoing trunk access code (TAC) or the feature access code (FAC) specifying ARS or AAR. Any coverage point can be an off-net destination.

Whenever an incoming trunk call is redirected off-net (coverage or forwarded), a timer is set that precludes any other incoming trunk call from redirecting off-net until the timer either expires or is cancelled. The rationale for this mechanism is to prevent calls that were redirected off-net from being re-routed back to the original principal from the off-net destination, effectively creating a round-robin loop that continuously seizes trunks until they are exhausted.

Coverage answer groups

You can create a coverage answer group so that up to 8 phones simultaneously ring when calls cover to the group. Anyone in the answer group can answer the incoming call.

Hunt Groups

A hunt group is a group of extensions that can handle multiple calls simultaneously to a single phone number. For each call to the phone number, the system hunts for an available extension in the group and connects the call to that extension.

A hunt group is especially useful when you expect a high number of calls to a particular phone number. A hunt group might consist of people trained to handle calls on specific topics. For example, the group might be:

- A benefits department within your company
- A service department for products you sell
- A travel reservations service
- A pool of attendants

Field descriptions for page 1

```
change hunt-group 4                                          Page 1 of X
                                   HUNT GROUP

           Group Number: 4__                                       ACD? _
             Group Name: ____________________________            Queue? _
        Group Extension: ____                                    Vector? _
             Group Type: _____                           Coverage Path: ____
                     TN: ___                   Night Service Destination: _____
                    COR: _                               MM Early Answer? _
     Security Code: ____
  ISDN Caller Disp: _____

           Queue Length: ___
Calls Warning Threshold: ___   Port: x____   Extension: ____
 Time Warning Threshold: ___   Port: x____   Extension: ____
```

Hunt Group screen

```
change hunt-group x                                                  Page 2 of X
                                    HUNT GROUP
                        Skill? _            Acceptable Service Level (sec): ___
                          AAS? _          Expected Call Handling Time (sec): ___
                     Measured: _____                     VuStats Objective: ____
          Supervisor Extension: ____                Timed ACW Interval (sec): ___
        Priority on Intraflow? _                   Service Level Supervisor? _
       Inflow Threshold (sec): ____                 Level 1 Threshold (sec): ___
          Controlling Adjunct: ____                 Level 2 Threshold (sec): ___
Adjunct Link Extension: ____
      Multiple Call Handling: __________     Redirect on No Answer (rings): ___
                                                          Redirect to VDN: _____
                         Forced Entry of Stroke Counts or Call Work Codes? _
```

Hunt Group screens

```
change hunt-group 1                                                  Page x of x
                                    HUNT GROUP

                              Message Center: rem-vm
                        Voice Mail Extension: ____
      Calling Party Number to INTUITY AUDIX? n
                               LWC Reception: none
```

Hunt Group screens

```
change hunt-group 1                                                  Page   4 of   39
                                    HUNT GROUP
           Group Number: 1        Group Extension: 3001        Group Type: ucd
  Member Range Allowed: 1 - 999          Administered Members (min/max): 1   /9
                                             Total Administered Members: 9
GROUP MEMBER ASSIGNMENTS
     Ext      Name                              Ext      Name
  1: 1022   station 1022                   14: 1023   station 1023
  2: 1010   bri 1010                       15:
  3: 1095   Station 1095                   16:
  4: 1002   station 1002                   17:
  5: 1001   Station 1001                   18:
  6: 1053   stat x1053                     19:
  7: 1094   Station 1094                   20:
  8: 311    stat x311                      21:
  9:                                       22:
 10:                                       23:
 11:                                       24:
 12:                                       25:
 13:                                       26:
                                           27:
  At End of Member List
```

Data Modules — general

A Data Module is a connection device between a basic-rate interface (BRI) or digital-communications protocol (DCP) interface of the switch and data-terminal equipment (DTE) or data-communications equipment (DCE).

The following types of data modules can be used with the system:

- Announcement data module
- Data line data module
- Processor/trunk data module (P/TDM)
- Netcon data module (G3si configurations only)
- Processor interface data module (G3si configurations only).
- System port data module (G3r configurations only)
- X.25 data module (G3r configurations only). Io
- 7500 data module
- World Class BRI data module
- Ethernet data module.
- Point-to-Point Protocol (PPP) data module..

```
change data-module 30                                          Page   1 of 2
                                DATA MODULE

  Data Extension: 30            Name: 27                          BCC:
            Type: data-line___   COS: 1
            Port: _______        COR: 1
             ITC: restricted__    TN: 1                  Connected to: dte


ABBREVIATED DIALING
  List1:

SPECIAL DIALING OPTION:


ASSIGNED MEMBER (Station with a data extension button for this data module)

          Ext      Name
      1: 1002    27 character   station name
```

Managing trunks

Adding a CO, FX, or TIE trunk group

In most cases, it is recommended you leave the default settings in fields that aren't specifically mentioned in the following instructions. Your settings in the following fields *must* match your provider's settings (or the setting on the far-end switch, if this is a private network trunk group):

- Direction
- Comm Type
- Trunk Type

Before you start

Before you can administer any trunk group, you must have one or more circuit packs of the correct type with enough open ports to handle the number of trunks you need to add. To find out what circuit packs you need, refer to the *DEFINITY ECS System Description.*

Instructions

1. Type **add trunk-group next** and press RETURN.

The Trunk Group screen appears. The system assigns the next available trunk group number to this group. In our example, we're adding trunk group 5.

The figure below shows a common configuration for page 1 of the Trunk Group screen when the Group Type field is **tie**. This screen is only an example, and the fields shown below may change or disappear according to specific field settings.

```
add trunk-group next                                              Page 1 of x
                                   TRUNK GROUP

 Group Number: __                    Group Type: _______        CDR Reports: _
   Group Name: ______________________________ COR: __      TN: ___      TAC: ___
    Direction: _______          Outgoing Display? _    Trunk Signaling Type: ___
  Dial Access? _                  Busy Threshold: __          Night Service: ____
 Queue Length: __                                     Incoming Destination: ____
    Comm Type: ____                     Auth Code? _
                                     Trunk Flash? _
          BCC: _                             ITC? ___
TRUNK PARAMETERS
    Trunk Type (in/out): __________     Incoming Rotary Timeout(sec): __
     Outgoing Dial Type: ________                 Incoming Dial Type: _______
                                          Disconnect Timing(msec): ___
        Digit Treatment: _________                          Digits: ___
                                                  Sig Bit Inversion: none
     Analog Loss Group: __                        Digital Loss Group: __
    Incoming Dial Tone? _
              Bit Rate: ____          Synchronization: ____     Duplex: ___
 Disconnect Supervision - In? _  Out? _
 Answer Supervision Timeout: __          Receive Answer Supervision? _
```

2 In the Group Type field, type **tie**.

This field specifies the kind of trunk group you're creating.

3. In the Group Name field, type **Outside calls**.

This name will be displayed, along with the group number, for outgoing calls if you set the Outgoing Display? field to y. You can type any name up to 27 characters long in this field.

4. Type **85** in the COR field.

This field controls which users can make and receive calls over this trunk group. Assign a class of restriction that's appropriate for the COR calling permissions administered on your system.

5. In the TAC field, type **105**.

This field defines a unique code that you or your users can dial to access this trunk group. The code also identifies this trunk group in call detail reports.

6. In the Direction field, type **two-way**.

This field defines the direction of traffic flow on this trunk group.

7. Type **1234** in the Night Service field.

This field assigns an extension to which calls are routed outside of business hours.

8. In the Incoming Destination field, type **attd**.

This field assigns an extension to which incoming calls are routed during business hours. By entering attd in this field, incoming calls go to the attendant and the system treats the calls as Listed Directory Number calls.

9. In the Comm Type field, type **voice**.

This field defines whether a trunk group can carry voice, data, or both. Analog trunks only carry voice and voice-grade data.

10. In the Trunk Type field, type **wink/wink**.

This field tells the system what kind of signaling to use on this trunk group. To prevent glare, ground start signaling is recommended for most two-way CO, FX, and WATS trunk groups and wink/wink for most tie trunk groups.

11. In the Outgoing Dial Type field, type **tone**.

This field tells the switch how digits are to be transmitted for outgoing calls. Entering tone actually allows the trunk group to support both DTMF and rotary signals, so Lucent recommends that you always put tone in this field.

T1 recommended settings

The table below shows recommended settings for standard T1 connections to your local exchange carrier.

Field	Value	Notes
Line Coding	**b8zs**	Use ami-zcs if b8zs is not
		available.
Signaling Mode	**robbed-bit**	Robbed-bit signaling gives you 56K bandwidth
		per channel. If you need a 64K clear channel
		for applications like ISDN or DCS
		use **common channel** signaling.
Framing	**esf**	Use d4 if esf is not available.

If you use b8zs line coding and esf framing, it will be easier to upgrade your T1 facility to ISDN should you want to. You can upgrade without reconfiguring external channel service units, and your service provider won't have to reconfigure your network connection.

Trunk Types

NOTE:
For trunk groups connecting 2 switches in Distributed Communication System networks, we recommend that you assign the same group number on both switches.

Access Use access trunks to connect satellite switches to the main switch in Electronic Tandem Networks (ETN). Access trunks do not carry traveling class marks (TCM) and thus allow satellite callers unrestricted access to out-dial trunks on the main switch. Allows Inband ANI.

Advanced Private Line Termination (APLT) trunks are used in private networks. APLT trunks allow inband ANI.

CAMA trunks route emergency calls to the local community's Enhanced 911 systems.

CO trunks typically connect your switch to the local central office, but they can also connect adjuncts such as external paging systems and data modules.

Use **CPE** trunks to connect adjuncts, such as paging systems and announcement or music sources, to the switch.

Use **DID** trunks when you want people calling your organization to dial individual users directly without going through an attendant or some other central point. Allows Inband ANI

.

DIOD trunks are two-way trunks that transmit dialed digits in both directions. In North America, use tie trunks for applications that require two-way transmission of dialed digits. Allows Inband ANI.

Digital Multiplexed Interface – Bit-Oriented Signaling **(DMI-BOS)** trunks allow communication with systems using DMI-BOS protocol. DMI-BOS trunks allow inband ANI.

An **FX** trunk is essentially a CO trunk that connects your switch directly to a central office outside your local exchange area. Use FX trunks to reduce long distance charges if your organization averages a high volume of long-distance calls to a specific area code.

Use **ISDN** trunks when you need digital trunks that can integrate voice, data, and video signals and provide the bandwidth needed for applications such as high-speed data transfer and video conferencing. ISDN trunks can also efficiently combine multiple services on one trunk group.

You cannot enter isdn unless the ISDN-PRI field, the **ISDN-BRI** Trunks field, or both have been enabled on the System-Parameters Customer-Options screen.

Release Link trunks work with Centralized Attendant Service in a private network.

Tandem trunks connect tandem nodes in a private network. Tandem trunks allow inband ANI.

Use **Tie** trunks to connect a switch to a central office or to another switch in a private network. Tie trunks transmit dialed digits with both outgoing and incoming calls, and allow inband ANI.

Use **WATS** trunks to reduce long-distance bills when your organization regularly places many calls to a specific geographical area in North America. Outgoing WATS service allows calls to certain areas ("WATS bands") for a flat monthly charge.

Incoming WATS trunks allow you to offer toll-free calling to customers and employees.

Group Name

Enter a unique name that provides information about the trunk group. Don't use the default entry or the group type (DID, WATS) here. For example, you might use names that identify the vendor and function of the trunk group: USWest Local; Sprint Toll, etc.

COR

Enter a class of restriction (COR). Classes of restriction control access to trunk groups, including trunk-to-trunk transfers.

Tip:
Remember that facility restriction levels (FRL) are assigned to classes of restriction. Even if 2 trunk groups have classes of restriction that allow a connection, different facility restriction levels may prevent operations such as off-net call forwarding or outgoing calls by remote access users.

TN

Enter a Tenant Partition number to assign this trunk group to the partition.

TAC

Enter the trunk access code (TAC) that must be dialed to access the trunk group. A different TAC must be assigned to each trunk group. CDR reports use the TAC to identify each trunk group.

CO Type

This field specifies whether the trunk group is connected to analog or digital facilities at the central office.

Protocol Type

This field specifies the type of line signaling protocol used for DID and DIOD trunk groups. This field appears when the Country field is 15 and is used only by trunk group members administered on a TN2199 or TN464D vintage 3 or later circuit pack.

Direction

This field appears for all trunk groups except DID and CPE.

Enter the direction of the traffic on this trunk group. The entry in this field affects which timers appear on the Administrable Timers page.(**incoming, outgoing, two-way**)

Outgoing Display

This field allows display phones to show the name and number of the trunk group used for an outgoing call before the call is connected. This information may be useful to you when you're trying to diagnose trunking problems.

CESID I Digits Sent

For emergency 911 service, your switch may send Caller's Emergency Service Identification (CESID) information to the central office or E911 tandem switch. This digit string is part of the E911 signaling protocol. This field appears when Group Type is cama.

Trunk Signaling Type

This field controls the signaling used by members in private network trunk groups, mainly in Italy, Brazil, and Hungary. This field appears if the Group Type is access, aplt, diod, rlt, tandem, or tie. Entries in this field affect which timers appear on the Administrable Timers page.

Dial Access

This field controls whether users can route outgoing calls through an outgoing or two-way trunk group by dialing its trunk access code. Allowing dial access does not interfere with the operation of AAR/ARS.

Busy Threshold

Use this field if you want attendants to control access to outgoing and two-way trunk groups during periods of high use. When the threshold is reached and the warning lamp for that trunk group lights, the attendant can activate trunk group control: internal callers who dial out using a trunk access code will be connected to the attendant, and the attendant can prioritize outgoing calls for the last remaining trunks. Calls handled by AAR and ARS route patterns go out normally.

Night Service

This field sets the destination to which incoming calls go when Night Service is in operation. If a Night field on the Group Member Assignments page is administered with a different destination, that entry will override the group destination for that trunk. CPE, DID, and DIOD trunk groups do not support night service.

Queue Length

Outgoing calls can wait in a queue, in the order in which they were made, when all trunks in a trunk group are busy. If you enter 0, callers receive a busy signal when no trunks are available. If you enter a higher number, a caller hears confirmation tone when no trunk is available for the outgoing call. The caller can then hang up and wait: when a trunk becomes available, the switch will call the extension that placed the original call. The caller will hear 3 short, quick rings. The caller doesn't need to do anything but pick up the handset and wait: the switch remembers the number the caller dialed and automatically completes the call. This field appears when the Direction field is outgoing or two-way.

Country

! **CAUTION:**
Don't change this field.

Incoming Destination

Use this field to set the destination for all incoming calls on trunk groups such as CO, FX, and WATS that must terminate at a single destination. The destination you enter here is also the default night service destination unless you enter a different destination in the Night Service field. Appears when the Direction field is **incoming** or **two-way**.

Comm Type

Use this field to define whether the trunk group carries voice, data, or both.

NOTE:
Comm Types of **avd**, **rbavd** and **data** require trunk member ports on a DS1 circuit pack.

Auth Code

This field affects the level of security for tandemed outgoing calls at your switch. This field appears if the Direction field is incoming or two-way, and it can only be **y** if the Authorization Codes field is **y** on the System-Parameters Customer-Options screen.

Digit Absorption List

This field assigns a digit absorption list, when used, to a trunk group that terminates at a step-by-step central office.

Prefix-1

Use this field for outgoing and two-way trunk groups handling long distance service. This field appears for CO, FX, and DIOD trunk groups.

Enter **y** to add the prefix "1" to the beginning of the digit string for outgoing calls. Do not enter **y** for trunk groups in AAR or ARS route patterns.

Trunk Flash

Some central offices allows users to activate special services (call waiting or 3-way conferencing, for example) by flashing the switch hook on their phone. If an outside caller does this while they're on a call with one of your users, your switch may interpret the flash as a disconnect signal and end the call. Use this field to prevent such accidental disconnections.

Toll Restricted

Enter **y** to prevent toll-restricted users from using a trunk access code to make restricted outgoing calls over this trunk group.

Enter **n** if the Outgoing Dial Type field is automatic or if you don't want to restrict access.

BCC

Generalized Route Selection uses the BCC to select the appropriate facilities for routing voice and data calls. Far-end tandem switches also use the BCC to select outgoing routing facilities with equivalent BCC classes.The entry in the Bearer Capability Class field is used to select the appropriate facilities for incoming ISDN calls. DEFINITY ECS compares the entry in the BCC field to the value of the Bearer Capability information element for the incoming call and routes the call over appropriate facilities. For example, a call with BCC 4 will only be connected through facilities that support 64 kbps data transmission.

ITC

The Generalized Route Selection feature, part of the automatic routing technology used in DEFINITY ECS, compares the line coding of available digital facilities and selects appropriate routes for voice and data calls. The Information Transfer Capability field appears when the Comm Type field is data, avd, or rbavd and the BCC field is not 0.

Trunk Type

Use this field to control the seizure and start-dial signaling used on this trunk group. Entries in this field vary according to the function of the trunk group and *must* match the corresponding setting on the far-end switch. This field appears for CO, DID, FX, and TIE trunk groups.

Use **ground-start** signaling for two-way trunks whenever possible: ground-start signaling avoids glare and provides answer supervision from the far end.

In general, try to use **loop-start** signaling only for one-way trunks. Loop-start signaling is susceptible to glare and does not provide answer supervision.

Valid entries are: **auto/auto, auto/delay, auto/immed, auto/wink**

The term before the slash tells the switch how and when it will receive incoming digits. The term after the slash tells the switch how and when it should send outgoing digits.

auto — Used for immediate connection to a single preset destination (incoming CO trunks, for example). No digits are sent, because all calls terminate at the same place.

delay — The sending switch does not send digits until it receives a delay dial signal (an off-hook signal followed by an on-hook signal) from the far-end switch, indicating that it's ready to receive the digits.

wink — The sending switch does not send digits until it receives a a wink start (momentary off-hook) signal from the far-end switch, indicating that it's ready to receive the digits.

immed — The sending switch sends digits without waiting for a signal from the far-end switch.

Trunk Type (in/out)

Use this field to control the seizure and start-dial signaling used on this trunk group. The setting of the Trunk Signaling Type field can affect the entries allowed in this field. In addition, settings may differ for incoming and outgoing trunks.

Incoming Rotary Timeout (sec)

Call setup at central offices that still use older switching equipment, such as step-by-step technology, is considerably longer then at central offices with more modern switches. If you're receiving digits with incoming calls from a central office that uses less efficient switching technology, your switch needs to allow more time to ensure it receives all the incoming digits. When the Incoming Dial Type field is rotary, use this field to set the maximum time your switch will wait to receive all incoming digits from the far-end switch.

Outgoing Dial Type

This field sets the method used to transmit digits for an outgoing call. Usually, you should match what your central office provides. This field appears for Access, APLT, CO, DIOD, DMI-BOS, FX, RLT, and WATS trunk groups. It also appears for Tie trunk groups when the Trunk Signaling Type field is blank, **cont**, or **dis**.

Enter **tone** to use Dual Tone Multifrequency (DTMF) addressing, also known as "touch tone" in the U.S. Entering **tone** actually allows the trunk group to support both DTMF and rotary signals.

For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone** or **mf**.

Enter **rotary** if you only want to allow the dial pulse addressing method used by non-touch tone phones. If you have a full touch tone system internally and a connection to a central office that only supports rotary dialing, for example, it would be appropriate to enter **rotary**.

Cut-Through

This field appears when the Outgoing Dial Type field is either **rotary** or **tone**.

Enter **y** to allow users to get dial tone directly from the central office. Outgoing calls over this trunk group will bypass AAR/ARS (if you're using it) and any of your administered restrictions (such as COR or FRL).

Enter **n** and the user will receive switch dial tone. Instead of digits being sent to the central office, they will be collected and checked against administered restrictions. If no restrictions apply, the digits are sent to the central office.

Incoming Dial Type

Indicates the type of pulses required on an incoming trunk group. Usually, you should match what your central office provides. This field appears for Access, APLT, DID, DIOD, DMI-BOS, FX, RLT, Tandem, and WATS trunk groups. It also appears for Tie trunk groups when the Trunk Signaling Type field is blank, **cont**, or **dis**.

Enter **tone** to use Dual Tone Multifrequency (DTMF) addressing, also known as "touch tone" in the U.S. Entering **tone** actually allows the trunk group to support both DTMF and rotary signals. Also, if you're using the Inband ANI feature, enter **tone**.

For pulsed and continuous E&M signaling in Brazil and for discontinuous E&M signaling in Hungary, use **tone**.

Enter **rotary** if you only want to allow the dial pulse addressing method used by non-touch tone phones. Though the tone entry supports rotary dialing as well, it's inefficient to reserve touch tone registers for calls that don't use DTMF.

Enter **mf** if the Trunk Signaling Type field is blank. The Multifrequency Signaling field must be enabled on the System-Parameters Customer-Options screen in order for you to enter **mf** here.

You may not enter **mf** if the Used for DCS field (Field descriptions for page 2) is **y**.

Digit Treatment

Use this field to modify an incoming digit string (as on DID and tie trunks, for example) by adding or deleting digits. You'll need to do this if the number of digits you receive doesn't match your dial plan.

Blank The incoming digit string is not changed.

Absorption Deletes digits, starting at the beginning of the string.

Insertion Adds digits, starting at the beginning of the string.

If you enter absorption or insertion, then you must enter a value in the Digits field.

Digits

This field is used with the Digit Treatment field, and its meaning depends on the entry in that field. If the Digit Treatment field is **absorption**, this field specifies *how many* digits are deleted. If the Digit Treatment field is **insertion**, this field identifies the *specific digits* that are added.

Expected Digits

NOTE:
Set this field to **blank** if the Digit Treatment field is set to **insert** and the Digits field contains a feature access code (for example, AAR or ARS) followed by digits. In this case, the number of digits expected are set on the AAR and ARS Digit Analysis Table and AAR and ARS Digit Conversion Table.

Auto Guard

This field controls ports only on TN438B, TN465B, and TN2147 circuit packs. TN438B ports have hardware support for detecting a defective trunk. TN465B and TN2147 ports consider a trunk defective if no dial tone is detected on an outgoing call, and the Outpulse Without Tone field is **n** on the Feature-Related System Parameters screen.

The figure below shows a common configuration for page 2 of the Trunk Group screen when the Group Type field is **co**. This screen is only an example, and the fields shown below may change or disappear according to specific field settings.

```
add trunk-group next                                          Page 2 of x
TRUNK FEATURES
         ACA Assignment? _      Measured: ____
                                                Maintenance Tests? _
                        Data Restriction? _

  Abandoned Call Search? _
  Suppress # Outpulsing? _

      Charge Conversion: _____
          Decimal Point: ______
        Currency Symbol: ___
            Charge Type: _______   Receive Analog Incoming Call ID: ________
                                        Per Call CPN Blocking Code: ___
                                      Per Call CPN Unblocking Code: ___
                                                   MF Tariff Free? _
          Outgoing ANI: _____________
```

ACA Assignment

Enter **y** if you want Automatic Circuit Assurance (ACA) measurements to be taken for this trunk group. If **y** is entered, complete the Long Holding Time, Short Holding Time, and Short Holding Threshold fields.

Measured

Indicates if the system will transmit data for this trunk group to the Call Management System (CMS). You cannot use **internal** and **both** unless either the BCMS (Basic) or the VuStats field is **y** on the System-Parameters Customer-Options screen. If the ATM field is set to **y** on the System-Parameters Customer-Option screen, this field accepts only **internal** or **none**. If this field contains a value other than **internal** or **none** when ATM is **y**, **none** appears.

Enter **internal** if the data can be sent to the Basic Call Management System (BCMS), the VuStats data display, or both.

Enter **external** to send the data to the CMS.

Enter **both** to collect data internally and to send it to the CMS.

Enter **none** if trunk group measurement reports are not required.

Long Holding Time (hours)

This field appears only when the ACA Assignment field is **y**.

Enter the length of time (in hours) that the system will consider as being a long holding time. If you enter **0**, the system will not consider long holding calls.

Internal Alert

Enter **y** if internal ringing and coverage will be used for incoming calls.

Maintenance Tests

Enter **y** if hourly maintenance tests will be made on this trunk group. Your entry is not saved when the screen is submitted unless one or more trunk members are administered.

Short Holding Time (seconds)

This field appears when the ACA Assignment field is **y**.

Enter the length of time (in seconds) that the system considers as being a short holding time. If **0** is entered, the system will not consider short holding calls.

Data Restriction

Enter **y** to prevent features from generating tones on a data call that would cause erroneous data transmission.

Short Holding Threshold

This field appears when the ACA Assignment field is **y**.

Enter the number of times the system will record a short holding call before alerting an attendant to the possibility of a faulty trunk.

Glare Handling

This field determines what the switch will do when glare occurs. This field appears when the Direction field is **two-way** and the outgoing side of the Trunk Type field is either **.../wink** or **.../delay**.

Abandoned Call Search

Use this field when the Trunk Type field is **ground-start**. Abandoned Call Search is designed to work with analog ground-start CO trunks that *do not* provide disconnect supervision. Your central office must support Abandoned Call Search for the feature to work properly. If your central office provides disconnect supervision, you do not need to use the Abandoned Call Search feature.

Used for DCS

Enter **y** if this trunk group will send and receive messages on a DCS signaling link.

PBX ID

This field, which appears when the Used for DCS field is **y**, identifies the remote switch in the network with which the trunk will communicate on a DCS signaling link.

Suppress # Outpulsing

Enter **y** if end-to-end signaling begins with (and includes) "#". The final "#" is suppressed in cases where the system would normally outpulse it. This field should be **y** when the Central Office (for example, rotary) or any other facility treats "#" as an error.

The figure below shows a common configuration for page 3 of the Trunk Group screen when the Group Type field is **co**. This screen is only an example, and the fields shown below may change or disappear according to specific field settings.

```
add trunk-group next                                                Page 3 of x
ADMINISTRABLE TIMERS
                Send Incoming/Outgoing Disconnect Timers to TN465 Ports? _
                                          Outgoing Dial Guard(msec): ____
  Incoming Glare Guard(msec): ____       Outgoing Glare Guard(msec): ____

                              Outgoing Rotary Dial Interdigit (msec): ___
       Ringing Monitor(msec): ____                Incoming Seizure(msec): ___
   Outgoing End of Dial(sec): __          Outgoing Seizure Response(sec): __
 Programmed Dial Pause(msec): ____          Disconnect Signal Error(sec): __
          Flash Length(msec): ___
                                                 Busy Tone Disconnect?

END TO END SIGNALING
    Tone (msec): ___     Pause (msec): 150

OUTPULSING INFORMATION
     PPS: 10    Make(msec): 40    Break(msec): 60  PPM? y  Frequency: 50/12k
```

! **CAUTION:**
Customers: Do not change fields on this page without assistance from your network service provider.

This screen appears when the Direction field on Page 1 is **outgoing** or **two-way** and the ATMS field is **y** on the Feature-Related System Parameters screen. The figure below shows a common configuration for page 4 of the Trunk Group screen when the Group Type field is **co**. This screen is only an example, and the fields shown below may change or disappear according to specific field settings.

The figure below shows a common configuration for page 4 of the Trunk Group screen when the Group Type field is **co**. This screen is only an example, and the fields shown below may change or disappear according to specific field settings.

```
add trunk-group next                                              Page 4 of x
                          ATMS THRESHOLDS
TTL Type: ________                  Far End Test No: ________________
  TTL Vendor: ____________              TTL Contact: ____________________
Trunk Vendor: ___________             Trunk Contact: ________________
Trunk Length: _____
                         MARGINAL          UNACCEPTABLE
                         Min   Max           Min   Max
           1004 Hz Loss: __    __            __    __

                         -Dev  +Dev          -Dev  +Dev
            404 Hz Loss:  _     _             _     _
           2804 Hz Loss:  _     _             _     _

Maximum C Message Noise:     __                  __
Maximum C Notched Noise:     __                  __
         Minimum SRL-HI:     __                  __
         Minimum SRL-LO:     __                  __
            Minimum ERL:     __                  __

     Allow ATMS Busyout, Error Logging and Alarming? _
  Maximum Percentage of Trunks Which Can Be Removed from Service by ATMS: __
```

CO Trunk Group ATMS Thresholds

! **CAUTION:**
Customers: Do not change fields on this page without assistance from Avaya or your network service provider.

The total number of pages, and the first page of Group Member Assignments, vary depending on whether the Administrable Timers and ATMS Thresholds pages display.

```
add trunk-group next
                                    TRUNK GROUP
                                      Administered Members(min/max): xxx/yyy
                                         Total Administered Members: xxx
GROUP MEMBER ASSIGNMENTS
       Port      Code  Sfx  Name         Night          Mode     Type    Ans Delay
    1: _______              __________   _____          _______  _______ ___
    2: _______              __________   _____          _______  _______ ___
    3: _______              __________   _____          _______  _______ ___
    4: _______              __________   _____          _______  _______ ___
    5: _______              __________   _____          _______  _______ ___
    6: _______              __________   _____          _______  _______ ___
    7: _______              __________   _____          _______  _______ ___
    8: _______              __________   _____          _______  _______ ___
    9: _______              __________   _____          _______  _______ ___
   10: _______              __________   _____          _______  _______ ___
   11: _______              __________   _____          _______  _______ ___
   12: _______              __________   _____          _______  _______ ___
   13: _______              __________   _____          _______  _______ ___
   14: _______              __________   _____          _______  _______ ___
   15: _______              __________   _____          _______  _______ ___
```

Group Member Assignments

Administered Members (min/max)

This display-only field shows the minimum and maximum member numbers that have been administered for this trunk group.

Port

If this trunk is registered as an endpoint in an IP application, this field will display T00000.

Enter the port number of each member. The member number of the trunk is the number displayed to the left of the Port field.

Code

This display-only field shows the type of circuit pack physically installed or logically administered at the location to which this member is assigned. If no circuit pack is installed or administered at the port address you enter, the field is blank.

Sfx

This display-only field shows the model suffix for the type of circuit pack physically installed at the location to which this member is assigned. If no circuit pack is installed at the port address you enter, the field is blank.

Name

Your vendor, sometimes need to identify specific trunks to work with your system. Therefore, the name you give to a trunk should identify the trunk unambiguously.

Examples of good names:

- The phone number assigned to incoming trunks
- The Trunk Circuit Identification number assigned by your service provider

Night

Use this field only if you want to assign a night service destination to individual trunks that is different from the group destination entered in the Night Service field on page 1. Incoming calls are routed to this destination when the system is placed in night service mode.

Mode

This field specifies the signaling mode used on tie trunks with TN722A or later, TN760B or later, TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs. This entry must correspond to associated dip switch settings on the circuit pack.

! **CAUTION:**
Customers should not attempt to administer this field. Please contact Avaya for assistance.

Enter **e&m** for 6-wire connections that pair 2 signaling wires with 4 voice wires. You'll use **e&m** in the vast majority of systems in the U.S.

Enter **simplex** for 4-wire connections that do not use an additional signaling pair. This configuration is very rare in the U.S.

Type

This field specifies the signaling type to be used with TN760B (or later release), TN722 (with any suffix), TN767, TN2140 (when the Trunk Signaling Type field is **cont**), TN437, TN439, TN464 with any suffix, or TN458 circuit packs.
(t1 stan, t1 comp, t5rev, type 5)

Ans Delay

Specifies the length of time (in ms) your switch will wait before it sends answer supervision for incoming calls on tie trunks using the TN722A or later, TN760 (B, C, or D), TN767, TN464 (any suffix), TN437, TN439, TN458, or TN2140 circuit packs.

CDR Reports

Yes - All outgoing calls on this trunk group will generate call detail records. If the Record Outgoing Calls Only field on the CDR System Parameters screen is n, then incoming calls on this trunk group will also generate call detail records.

No - Calls over this trunk group will not generate call detail records.

Managing vectors and VDNs

A vector is a series of commands that you design to tell the system how to handle incoming calls. A vector can contain up to 32 steps and allows customized and personalized call routing and treatment. Use call vectoring to:

- play multiple announcements
- route calls to internal and external destinations
- collect and respond to dialed information

Tip:
The vector follows the commands in each step in order. The vector "reads" the step and follows the command if the conditions are correct. If the command cannot be followed, the vector skips the step and reads the next step.
Your system can handle calls based on a number of conditions, including the number of calls in a queue, how long a call has been waiting, the time of day, dayof the week, and changes in call traffic or staffing conditions.

Vector directory numbers

A vector directory number (VDN) is an extension that directs an incoming call to a specific vector. This number is a "soft" extension number not assigned to an equipment location. VDNs must follow your dial plan.

Understanding Automatic Call Distribution (ACD)

Automatic Call Distribution (ACD) is a DEFINITY ECS feature used in many call centers. ACD gives you greater flexibility to control call flow and to measure the performance of agents.

ACD systems operate differently from non-ACD systems, and they can be much more complex. ACD systems can also be more powerful because they allow you to use features and products that are not available in non-ACD systems. Refer to *DEFINITY ECS Guide to ACD Call Centers* for more information on ACD call centers.

Enhancing an ACD system

First, all call center management systems (such as Avaya's Basic Call Management System (BCMS), BCMSVu, and the sophisticated CentreVu Call Management System) require ACD. These management systems give you the ability to measure more aspects of your center's operation, and in more detail, than is possible with standard DEFINITY ECS reports.

Call vectoring greatly enhances the flexibility of a call center, and most vectoring functions require ACD. Vectoring is a simple programming language that allows you to custom design every aspect of call processing.

Together, ACD and vectoring allow you to use Expert Agent Selection (EAS). For a variety of reasons, you may want certain agents to handle specific types of calls. For example, you may want only your most experienced agents to handle your most important customers. You may have multilingual agents who can serve callers in a variety of languages.

EAS allows you to classify agents according to their specific skills and then to rank them by ability or experience within each skill. DEFINITY ECS uses these classifications to match each call with the best available agent. Refer to *DEFINITY ECS Call Vectoring/EAS Guide* or the *DEFINITY BCS and Guestworks Call Vectoring/EAS Guide* for more information on call vectoring and EAS.

ARS/AAR

World class routing

Your system uses Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS) to direct outgoing calls.

- AAR routes calls within your company over your own private network.
- ARS routes calls that go outside your company over public networks. ARS also routes calls to remote company locations if you do not have a private network.

Automatic routing begins when a user dials a Feature Access Code (FAC) followed by the number the user wants to call. The switch analyzes the digits dialed, selects the route for the call, deletes and inserts digits if necessary, and routes the call over the trunks you specify in your routing tables. ARS and AAR can access the same trunk groups and share the same route patterns and other routing information. ARS calls can be converted to AAR calls and vice-versa.

The FAC for AAR is usually the digit 8. The FAC for ARS is usually the digit 9 in the US and 0 outside of the US. Your Avaya technician sets up AAR on your switch and usually assigns the AAR FAC at the same time. You can administer your own ARS FAC.

Managing calling privileges

Each time you set up a phone, you use the station screen to assign a COR. You can create different CORs for different groups of users. For example, you may want executives in your company to have different calling privileges than receptionists.

When you set up a COR, you specify a Facility Restriction Level (FRL) on the Class of Restriction screen. The FRL determines the calling privileges of the user. Facility Restriction Levels are ranked from 0–7, where 7 has the highest level of privileges.

You also assign an FRL to each route pattern preference in the route pattern screen. When a user makes a call, the system checks the user's COR. The call is allowed if the caller's FRL is higher than or equal to the route pattern preference's FRL.

Displaying ARS analysis information

Instructions

You'll want to become familiar with how your system currently routes outgoing calls. To display the ARS Digit Analysis Table that controls how the system routes calls that begin with 1:

Type **display ars analysis 1** and press RETURN.

The ARS Digit Analysis Table for dialed strings that begin with 1 appears. Note that the switch displays only as many dialed strings as can fit on one screen at a time.

To see all the dialed strings that are defined for your system, run an ARS Digit Analysis report:

Type **list ars analysis** and press RETURN.

The ARS Digit Analysis Report appears. You may want to print this report to keep in your paper records.

Understanding ARS analysis

With ARS, the switch checks the digits in the number called against the ARS Digit Analysis Table to determine how to handle the dialed digits. Your switch also uses Class of Restriction (COR) and Facility Restriction Level (FRL) to determine the calling privileges.

Let's look at a very simple AAR and ARS Digit Analysis Table. Your system likely has more defined dialed strings than our example.

```
change ars analysis                                          Page 1 of X

                         ARS DIGIT ANALYSIS TABLE
                                Location: ___      Percent Full: ___

     Dialed        Total    Route    Call    Node   ANI
     String       Min Max   Pattern  Type    Num    Reqd
_______________   __  __   _____    ___     ___     n
_______________   __  __   _____    ___     ___     n
_______________   __  __   _____    ___     ___     n
_______________   __  __   _____    ___     ___     n
_______________   __  __   _____    ___     ___     n
_______________   __  __   _____    ___     ___     n
_______________   __  __   _____    ___     ___     n
_______________   __  __   _____    ___     ___     n
_______________   __  __   _____    ___     ___     n
_______________   __  __   _____    ___     ___     n
```

ARS Digit Analysis Table

The far-left column of the ARS Digit Analysis Table lists the first digits in the dialed string. When a user makes an outgoing call, the system analyzes the digits, looks for a match in the table, and uses the information in the matching row to determine how to route the call.

Let's say a caller places a call to 1-303-233-1000. The switch matches the dialed digits with those in the first column of the table. In this example, the dialed string matches the '1'. Then the systems matches the length of the entire dialed string (11 digits) to the minimum and maximum length columns. In our example, the 11-digit call that started with 1 follows route pattern 30 as an fnpa call.

Tip:
The first dialed digit for an external call is often an access code. If '9' is defined as the ARS access code, the switch drops this digit and analyzes the remaining digits with the ARS Analysis Table.

The Route Pattern points to the route that handles the calls that match this dial string.

Call Type tells what kind of call is made with this dial string. Call type helps the switch decide how to handle the dialed string.

Route Pattern

The Route Pattern screen defines the route patterns used by your switch. Each route pattern contains a list of trunk groups that can be used to route the call. The maximum number of route patterns and trunk groups allowed depends on the configuration and memory available in your system.

Use this screen to insert or delete digits so AAR or ARS calls route over different trunk groups. You can convert an AAR number into an international number, and insert an area code in an AAR number to convert an on-network number to a public network number. Also, when a call directly accesses a local central office (CO), if the long-distance carrier provided by your CO is not available, your switch can insert the dial access code for an alternative carrier into the digit string.

```
change route-pattern 1                                          Page 1 of X
                                Pattern Number: 1_

                                 No.                                      DCS/
   Grp.  FRL NPA Pfx Hop Toll Del   Inserted                            QSIG  IXC
   No.           Mrk Lmt List Dgts  Digits                              Intw
 1: __   _ __  _  __  __   __   ______________________________________  n    user
 2: __   _ __  _  __  __   __   ______________________________________  n    user
 3: __   _ __  _  __  __   __   ______________________________________  n    user
 4: __   _ __  _  __  __   __   ______________________________________  n    user
 5: __   _ __  _  __  __   __   ______________________________________  n    user
 6: __   _ __  _  __  __   __   ______________________________________  n    user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature BAND  No.   Numbering LAR
     0 1 2 3 4 W     Request                              Dgts   Format
                                                        Subaddress
 1: y y y y y n  y  none  ___ both ept outwats-bnd____ ____  _    _______  none
 2: y y y y y n  y            rest     _______________       _    _______  next
 3: y y y y y n  y            rest     _______________       _    _______  rehu
 4: y y y y y n  y            rest     _______________       _    _______  none
 5: y y y y y n  y            rest     _______________       _    _______  none
 6: y y y y y n  y            rest     _______________       _    _______  none
```

Route Pattern

Pattern Number

Displays the route pattern number (**1** to **640**).

Grp No

Enter the trunk group number associated with this row (preference).

FRL

Enter the Facility Restriction Level (FRL) associated with the entries on this row (preference). 0 is the least restrictive and 7 is the most restrictive. The calling party's FRL must be greater than or equal to this FRL to access the associated trunk-group.

Network Specific Facility

Identifies the services and features used to complete a call.

NPA

This entry is not required for AAR.

Enter the 3-digit Numbering Plan Area (NPA) (or area code) for the terminating endpoint of the trunk group. Call your local telephone company to verify this number if you need help.

For WATS trunks, the terminating NPA is the same as the home NPA unless the Local Exchange Carrier requires 10 digits for local NPA calls.

No. Del. Digits

Use this field to modify the dialed number so an AAR or ARS call routes over different trunk groups that terminate in switches with different dial plans

Enter the total number of digits you want the system to delete before it sends the number out on the trunk. Use for calls that route:

Inserted Digits

Enter the digits you want inserted for routing. The switch can send up to 52 digits. This includes up to 36 digits you may enter here plus up to 18-digits originally dialed. Special symbols count as two digits each.

DEFINITY AUDIX

Using the Administration Terminal and Administrative Forms

DEFINITY AUDIX system administration is performed at a data terminal or PC connected to the DEFINITY AUDIX administration port. The following terminals are supported for the DEFINITY AUDIX system:

- 513
- 4410
- 4425
- 5420
- PC (using the ADAP package)
- 715 (The 715 terminal provides the ability to toggle back and forth between the DEFINITY AUDIX system and switch administration screens.

Logging into the DEFINITY AUDIX

To administer your DEFINITY AUDIX system, you must first log in at your administration terminal.

To log in:

1. Enter **cust** (the administrative login ID) at the login prompt.
2. Enter **custpw** (the default login password) at the password prompt.
3. Press RETURN at the system password prompt.
4. Enter **513** or the appropriate terminal type at the terminal type prompt.

After you are logged in, change the default login password and the system password to passwords of your choosing.

At this point, it is important for you to provide your subscribers with the appropriate documentation to help them use the DEFINITY AUDIX system properly. Select the appropriate template and make the necessary changes. The following DEFINITY AUDIX documentation is recommended also:

- *A Portable Guide to Voice Messaging*, 585-300-701
- *Voice Messaging Quick Reference*, 585-300-702
- *Multiple Personal Greetings Quick Reference*, 585-300-705
- *Voice Messaging Wallet Card*, 585-300-704
- *Voice Messaging Business Card Stickers*, 585-304-705
- *Outcalling Quick Reference*, 585-300-706

Feature Administration

Automated Attendant

The Automated Attendant feature allows you to set up the DEFINITY AUDIX system to answer extensions and prompt callers to press appropriate keys on their touch-tone telephones to transfer to desired extensions or leave messages for individual subscribers. You can set up any number of automated attendants, and you can nest them so that an option selected from one attendant menu dials another automated attendant to provide a completely new menu of options. Callers can be transferred directly to DEFINITY AUDIX mailboxes from automated attendant menu options without going to the switch, allowing you to efficiently handle DEFINITY AUDIX coverage for shared extensions and non-resident subscribers with an automated attendant.

The first stage of an automated attendant in a multilingual environment might ask the user to select a language. Subsequent stages can implement the auto-attendant function in the language chosen.

Define the automated attendant in the same manner that you would set up a new local subscriber on the Subscriber screen. Be sure to identify the attendant as such in the PERMISSIONS, Type field.

NOTE:
If you plan to use a number of automated attendants, you might want to set up a class of service with the PERMISSIONS, Type field already set to "auto-attendant" for use with automated attendants. If so, be sure that existing subscribers are not already assigned to that class of service. Also, you must run "audit subscriber-data" before class of service changes will be effective for automated attendants.

Be prepared to supply the following information on the Subscriber screen.

For the attendant:

- The attendant's name
- The attendant's extension
- The number of seconds to wait for time-out when a caller does not respond to a prompt

For each button (0 through 9) and the timeout that is active for the attendant:

- The extension to transfer the call to or the DEFINITY AUDIX mailbox extension when the button is pressed.

If this button is the first digit of an extension on your switch *and* you want callers to be able to directly enter an extension, put **"e"** in this field. If you want callers to be able to dial an extension directly from the automated attendant without using * T, you cannot use any button whose number is the first digit of an extension that could be called directly. For example, if internal extensions begin with 5, you cannot use button 5 for a separate selection like "To reach accounting, press 5." If you did, callers who attempted to dial specific extensions starting with 5 would instead be transferred to accounting in this example. Instead, assign this button as extension *e*. This allows the caller direct access to any extension that starts with the associated button. More than one button can be assigned as *e* if necessary. For this feature to work properly, addressing must be by extension on the automated attendant Subscriber screen.

NOTE:
Pay particular attention to the switch dial plan when assigning the *e* option. Consider that some extensions within the group may not exist, may not be assigned, may be assigned to special features like hunt groups or vectors, may be assigned to the attendant console, may be assigned to a Distributed Communication System (DCS) node, or may be assigned to the DEFINITY AUDIX voice ports. Any of these situations may cause problems when a caller attempts to dial one of them.

- A transfer treatment code to specify whether the call is to transfer to the extension's telephone **("transfer")** or directly to the extension's mailbox to play the subscriber's call answer greeting **('call-answer")** or the system guest greeting ("guest-greeting"). If the treatment is "call-answer" and no personal greeting is recorded or active, the system call answer greeting is played.
- A brief comment describing the button's function. This is your only record of the attendant's button functions and associated extension numbers, so spelling it out now in the comment field may save you some time later if you have to modify the attendant's functions or rerecord the attendant menu.

Recording the Attendant Menu

Use your touch-tone telephone to record the automated attendant menu that will be spoken to callers. This menu is actually the personal greeting for the attendant's extension. If setting up a TTY automated attendant, use the TTY keyboard to type the menu where the following instructions say speak or record and the touchtone telephone keypad when the instructions say press.

In the menu, you may want to include the following:

- A greeting followed by the menu choices available to the caller
- An instruction to wait if a time-out extension is administered

NOTE:
You also can set up a one-button press to repeat the menu by putting the same attendant's extension in the Extension field and "call-answer" in the treatment field.

Setting up a Business Schedule

Auto-Attend-Routing Business-Schedule

Type : **" ch auto bus 1"** (change auto-attendant-routing business-schedule 1)

Set up the business schedules by filling in these fields:

- Business Schedule — If the default name of the schedule does not seem descriptive enough, name the schedule in this field.

- Day Service Hours — Hours outside of this range are considered to be night service hours. (Use 24-hour notation: AM starts at 00:00, midnight: PM times are 12:00-23:59.)
  - — Start Time — Enter the time at which daytime operation should begin.
  - — End time — Enter the time at which daytime operation should end.
- Alternate Service Hours — This is time that may be considered an exception to normal day service (lunch time, for example). *An alternate service period must either fall entirely inside of day service hours or entirely outside of day service hours.* (Use 24-hour notation: AM starts at 00:00, midnight: PM times are 12:00-23:59.)
  - — Start Time — Enter the time at which alternate service is to begin.
  - — End Time — Enter the time at which alternate service is to end.

Setting up a Holiday Schedule

The second step of setting up auto-attendant routing is to access a change auto-attend-routing holiday-schedule administration screen. The screen you access will be one of four such screens you can use for separate schedules. Consequently, you will have to give the administration program the name or number of a particular schedule to access this screen (numbers 1 through 4 will access a screen anytime):

Auto-Attend-Routing Holiday-Schedule

Type: **"ch auto hol 1"** (change auto-attendant-routing holiday-schedule 1)

NOTE:
Before you start to fill in the schedule, make sure that the auto-attendants exist. They must have been created before you add them under the *Mailbox* column.

Set up the holiday schedules by filling in these fields:

- Holiday Schedule — If the default name of the schedule does not seem descriptive enough, name the schedule in this field. Use capital letters here if you intend to use them in the name of this schedule in the future.
- Holiday Name — Enter the name of the holiday here. (This field is for your reference. An entry in this field is optional.)

- Date — Enter the date on which the affected incoming call will be forwarded to mailbox.
- Mailbox — Enter the number to which the affected incoming call will be forwarded.

Filling in the Routing Table

Auto-Attendant-Routing Routing-Table

Type: **"ch auto rout"** (change auto-attendant-routing routing-table)

The routing function redirects calls to specified numbers. It redirects them to specified voice mailbox numbers according to the instructions given in the holiday schedules and the routing table. Fill in the routing table as follows:

- Incoming Called Number — Enter the numbers to be redirected. These can be any incoming numbers reported to AUDIX by the switch. If a number appears twice in this column, the first instance prevails. 802, for example, appears before the range 802-806, and will be treated separately per its first appearance. 805 appears after the range of numbers, however, so it will be treated as set out on the line associated with the range.
- Business Schedule — Enter the name or number of the business schedule that is to determine how the incoming number is to be treated. The name login is reserved. It indicates that a direct, external call to the associated incoming number is to be allowed AUDIX Login services. That is, if you call this extension. AUDIX will ask you to log in.
- Holiday Schedule — Enter the name or number of the holiday schedule (if any) that is to determine which auto-attendant mailbox the incoming number is to access on holidays.

Name Addressing

The automated attendant can be administered to use name addressing. For example, the automated attendant could greet callers with the following:

> "Thank you for calling ABC Company." "To leave a message for one of our sales agents, please enter his or her name starting with the last name."

To administer name addressing, enter "name" in the Addressing Format field on page 2 of the Subscriber screen. On page 3 of the Subscriber screen, enter “e” in the extension column for buttons 2 through 9. It is recommended that a Timeout Treatment also be administered on page 3 of the Subscriber screen. If the Length of Timeout on Initial Entry field is set to “1” or “0”, callers to the automated attendant will be transferred immediately for name addressing.

Broadcast Messages

The Broadcast Messages feature allows selected subscribers to send broadcast messages to all local subscribers and selected remote subscribers. Permission to send broadcast messages can be assigned on a per-subscriber basis or by class-of-service. You should limit permission to send broadcast messages to yourself or to a few selected subscribers to avoid overuse of this feature. Any user with broadcast permission can interact with the system in the language of her or his choice. The broadcaster should consider the appropriateness of broadcasting a uni-lingual message versus a multilingual message to a multilingual community.

The broadcast message is seen as the first message in the subscriber’s mailbox regardless of subsequent message activity. The subscriber can retrieve, listen, save, and — if the message is not private — forward the message. Broadcast messages are not actually distributed. Instead, subscribers hear the message from a mailbox that is administered as the broadcast mailbox. Since the message is not actually sent, the sender can specify a date when the message should no longer be played.

Class of Service

The Class of Service feature provides 12 unique service classes containing different combinations of features or mailbox parameters for use by subscribers with varying service requirements. Service classes are predefined on Class Of Service screens and can be individually modified to meet your specific requirements. Each subscriber is assigned a class of service by associating his/her Subscriber screen with a specific Class Of Service screen, or individual subscriber service options can be customized by entering service information directly on the individual’s Subscriber screen.

End of Message Warning

The End of Message Warning feature is enabled by you and causes message recording to be interrupted a predefined amount of time before the maximum recording length is reached. The DEFINITY AUDIX system announces that "n" seconds of recording time remain and prompts the user to resume recording. You can define the End of Message Warning feature on a system-wide basis. You also can define individual warning times for DEFINITY AUDIX subscribers who want to override the system-wide warning time.

Mailing List

The Mailing List feature allows DEFINITY AUDIX subscribers to create lists of names or extensions to send voice mail to instead of addressing each recipient individually. This is convenient for sending voice mail to groups who regularly receive mail, such as members of a department or project. Lists can be composed of individual subscriber names, extensions, and other lists. The creator of a list also can designate it as public or private, and public lists are available for use by other subscribers. Even if a list is public, only the owner of the list can modify it.

Multiple Personal Greetings

The Multiple Personal Greetings feature allows subscribers to record and store up to nine personal greetings and to activate as many as three of them at once for different call types (internal/external, busy/no-answer, out-of-hours). For example, one greeting can be activated for internal calls, a second for external calls, and a third for out-of-hours calls. Alternatively, one greeting can be activated for calls when the subscriber's phone is busy, a second for when there is no answer, and a third for after-hours calls. The Multiple Personal Greetings feature is not available to subscribers that use the Multilingual feature and have the Call Answer Language Choice field set to "y" on the Subscriber or Class of Service screen.

Internal/external and busy/no answer call types are mutually exclusive; the subscriber can specify separate greetings for internal and external calls *or* for busy and no answer calls, but not for both call types or a combination of both call types. Note that out-of-hours calls are answered with the out-of-hours greeting regardless of whether the call is internal/external or busy/no-answer. Even if subscribers choose not to provide separate greetings for different call types, the multiple personal greetings feature is useful for setting up a library of different personal greetings to activate for different occasions, such as during lunch or while on vacation.

Outcalling

The Outcalling feature allows the DEFINITY AUDIX system to call a subscriber on the phone or a pager for message notification instead of, or in addition to, notification by a MWI or stutter dial tone. This is useful for receiving DEFINITY AUDIX messages at a home phone or at a phone in another business-related location, or to program the DEFINITY AUDIX system to outcall to a paging service and enter a callback number.

This feature is not recommended for use as an emergency pager since the DEFINITY AUDIX system does not treat outcalling as its highest priority during busy periods and may not be fast enough for emergency applications. If the Multilingual feature is active on your system, the DEFINITY AUDIX system uses the Login Announcement Set of the subscriber when it outcalls.

This feature, which is initially off by default, is activated or deactivated on the System-Parameters Outcalling screen. Permission to use outcalling then can be assigned on a per-subscriber basis or by class-of-service. You control the hours during which outcalling is permitted (up to three periods each day), the maximum number of ports that can be used simultaneously for outcalling, and the maximum number of digits (up to 60) that subscribers can specify for the outcalled number.

Subscribers administer outcalling numbers and permissible hours (the same as or a subset of the system's permissible hours) using the DEFINITY AUDIX voice menu.

Adding New Subscribers

After the initial group of subscribers has been added, you must still regularly add subscribers as new employees join your company or existing employees without DEFINITY AUDIX service are added. Adding a subscriber involves assigning the individual a login and (optionally) a password, defining a set of permissions called *class of service* options that control the individual's DEFINITY AUDIX service, and (optionally) recording the subscriber's name as it will be spoken by the DEFINITY AUDIX system.

Use your administration terminal to add subscriber information (name, extension, password, and service options) to the DEFINITY AUDIX system.

Perform the following steps to add new subscribers:

1. Inform the switch administrator that new subscribers are being added and that call coverage paths need to be established for them.
2. Add subscriber information (name, extension, and service options) to the DEFINITY AUDIX data filesystem.

3. Assign subscribers an initial password that is less than the minimum password length defined on the System- Parameters Features screen. When the subscriber logs on to the DEFINITY AUDIX system for the first time, he or to enter a new password that is at least as long as the required minimum. This is a good way to ensure that subscribers do not continue using the default password you assign them initially. Also if the Password Aging feature is active, subscribers will be required to change their password according to the Password expiration interval defined on the System-Parameters Features Screen

4. If none of the predefined classes of service meet a particular subscriber's needs, you can customize the subscriber's service options by assigning a class of service and then changing information via the Subscriber screen. The subscriber's service options then are independent of the assigned class of service and will not be updated by changes made to any Class of Service screen.
5. Either you or the subscriber must make a voice recording of the subscriber's name for use as a voice fragment that the DEFINITY AUDIX system speaks during call answering. If the Name Record By Subscriber feature is used, the DEFINITY AUDIX system will prompt subscribers to speak their own names the first time they log in. It is recommended that you activate this feature since it will significantly decrease your workload. Otherwise, you must record all of the subscriber names yourself.
6. If you enter a large number of subscribers, you may want to manually back up the names filesystem when you are finished. This data is normally backed up automatically each Sunday, but you may want to protect your work in the interim if you have recorded name fragments for a large number of users. To perform a demand backup of the subscriber names data, execute the **save weekly** command. You also may want to back up system data, though your exposure to data loss is not as great because the system data filesystem is backed up automatically each night. However, to perform a demand backup of system data, execute the **save nightly** command. Refer to "Backing Up Filesystems and Subdirectories" in Chapter 6, "Ongoing Data Administration", for more information about manually backing up data.

Unlocking Locked Subscriber Logins

For security purposes, the DEFINITY AUDIX system accepts only three unsuccessful login attempts per subscriber session (call to the DEFINITY AUDIX system) before disconnecting. The DEFINITY AUDIX system also monitors the number of unsuccessful consecutive login attempts per subscriber. If this number exceeds the number defined on the System-Parameters Features screen, the DEFINITY AUDIX system *locks out* that subscriber's login ID, thus preventing further system access. The subscriber cannot access the DEFINITY AUDIX system until you unlock the subscriber's login.

Intuity™ AUDIX®

How to Log In to the System

To log in to the system:

1. Turn on your monitor.
2. Enter your administrator's login at the login: prompt.

   If you are the system administrator, enter **sa**

   If you are the voice messaging administrator, enter **vm**
3. Enter your administrator's password at the password: prompt.
4. Enter **at386** at the terminal type: prompt.

At the Main Menu, pick "Audix Administration" and press return.

Function Keys

Standard Function Keys

Most function keys perform standard actions regardless of the screen you are viewing.

This table describes the purpose of each standard function key. There is also a description of each function key at the bottom of each screen.

	Standard Function Keys
Keypad Command	**Function**
F1 (Cancel)	This stops the current activity and returns the cursor to the command line. When the cursor is in the command line, F1(Cancel) erases the entire contents of the command line. On a help screen, F1(Cancel) returns to the screen on which the help was requested.
F2 (Refresh)	This redraws or updates the screen.
F3 (Enter)	This submits the information entered on a screen for the action specified on the command line. When the cursor is on the command line, F3 (Enter) requests execution of the command.
	Note: RETURN has the same effect as F3 (Enter) when the cursor is on the command line. On a screen, RETURN moves the cursor forward from one field to the next.
F4 (Clearfld)	This clears an entire field on a screen or a single keyword from the command line. For example, if the command line contains the command list cos and you press F4(Clearfld), the command line changes to list.
F5 (Help)	When the cursor is on the command line, pressing this key is identical to typing the help command. That is, it displays a screen explaining all the types of help available in the Intuity AUDIX system. When the cursor is on a screen, this key requests help for the entire screen.
F6 (Choices)	When the cursor is on the command line, this key requests a menu of valid entries for command line keywords. Once this menu is displayed, use the UP ARROW or DOWN ARROW key to select an item from the menu. Pressing F6 (Choices) or RETURN selects the highlighted item from the menu. When the cursor is on a screen, this key requests help for the particular field where the cursor appears. The field help menu provides an explanation of the field and a list of valid values or actions for the field. When a field menu is displayed, pressing F6 (Choices) or RETURN selects the highlighted item from the menu.
F7 (Nextpage)	This moves forward through multiple-page administration screens, reports, or help screens.
F8 (Prevpage)	This moves backward through multiple-page administration screens, reports, or help screens.

How to Add a User

To add a user:

1. Start from the Avaya Intuity main menu (Figure1-1), and select:

   AUDIX Administration

2. At the enter command: prompt, enter **ad su *name/extension*** where *name* is the name of the user and *extension* is the telephone extension of the user you want to add to the system.

3. Press **return.**

4. Enter name and password as needed.

Note: If you don't enter a password, the **#** sign will become the default password.

5. Press F3 (Enter) to save the information.

References

Basic DEFINITY ECS documents

These documents are issued for all new and upgrade DEFINITY ECS Release 8.2 systems.

Administration

The primary audience for these documents consists of customer administrators.

DEFINITY ECS Release 8.2 — Administrator's Guide, 555-233-506, Issue 1

A task-based document that provides step-by-step procedures for administering the switch. This book contains information previously found in *DEFINITY ECS Administration and Feature Description*, 555-230-522, and *DEFINITY ECS Release 8 — Administrator's Guide*, 555-233-502, as well as new information for this release.

DEFINITY ECS Release 8.2 — Administration for Network Connectivity, 555-233-504, Issue 1

Describes how to administer connections between DEFINITY ECS switches (csi, si, and r models) for DCS messaging. The main focus is on TCP/IP connectivity introduced with DEFINITY Releases 7 and 8, including voice over IP (VOIP).

DEFINITY ECS Release 8.2 — Change Description, 555-233-411, Issue 1

Provides a high-level overview of what is new in DEFINITY ECS R8.2. Describes the hardware and software enhancements and lists the problem corrections for this release. It also includes any last-minute changes that come in after the remaining books have gone to production.

DEFINITY ECS Release 8.2 — System Description, 555-233-200, Issue 1

Provides hardware descriptions, system parameters, listing of hardware required to use features, system configurations, and environmental requirements.

DEFINITY System's Little Instruction Book for basic administration, 555-233-756, Issue 1

Provides step-by-step procedures for performing basic switch administration tasks. Includes managing phones, managing features, routing outgoing calls, and enhancing system security.

DEFINITY System's Little Instruction Book for advanced administration, 555-233-757, Issue 1

Provides step-by-step procedures for managing trunks, managing hunt groups, setting up night service, writing vectors, recording announcements, using reports, and understanding call centers.

DEFINITY System's Little Instruction Book for basic diagnostics, 555-233-758, Issue 1

Provides step-by-step procedures for baselining your system, solving common problems, reading alarms and errors, using features to troubleshoot your system, and contacting Avaya.

DEFINITY ECS Release 8.2 — Reports, 555-233-505, Issue 1

Provides detailed descriptions of the measurement, status, security, and recent change history reports available in the system and is intended for administrators who validate traffic reports and evaluation system performance. Includes corrective actions for potential problems. Previously known as *DEFINITY ECS System Monitoring and Reporting*.

BCS Products Security Handbook, 555-025-600, Issue 7

Provides information about the risks of telecommunications fraud and measures for addressing those risks and preventing unauthorized use of BCS products. This document is intended for telecommunications managers, console operators, and security organizations within companies.

DEFINITY Terminals and Adjuncts Reference, 555-015-201, Issue 10

Provides drawings and full descriptions for all phones, phone adjuncts, and data terminals that can be used with System 75, System 85, DEFINITY Communications System, and DEFINITY ECS. This document is intended for customers and Lucent Technologies account teams for selecting the correct equipment.

Guide Builder™ Software for DEFINITY® Telephones, 555-230-755, Issue 5

Provides the ability to produce laser-printed documentation for specific telephones. A comprehensive user's guide and on-line help support the software. This information applies to Release 8.2 as well as earlier DEFINITY systems. All customers receive this software.

Installation and maintenance

DEFINITY ECS Release 8.2 — Installation and Test for Single-Carrier Cabinets, 555-233-120, Issue 1

Provides procedures and information for hardware installation and initial testing of single-carrier cabinets. This document is availabe in languages other than English and can be ordered from the BCS Publications Catalog web site.

DEFINITY ECS Release 8.2 — Installation and Test for Multi-Carrier Cabinets, 555-233-114, Issue 1

Provides procedures and information for hardware installation and initial testing of multi-carrier cabinets.

DEFINITY ECS Release 8.2 — Installation for Adjuncts and Peripherals, 555-233-116, Issue 1

Provides procedures and information for hardware installation and initial testing of ECS adjunct and peripheral systems and equipment.

DEFINITY ECS Release 8.2 — Installation and Test for Compact Modular Cabinets, 555-233-118, Issue 1

Provides procedures and information for hardware installation and initial testing of compact modular cabinets.

DEFINITY ECS Release 8.2 — ATM Installation, Upgrades, and Administration, 555-233-124, Issue 1

Provides step-by-step instructions for how to install, upgrade, and administer ATM switches.

DEFINITY ECS Release 8.2 — Installation and Maintenance for Survivable Remote EPN, 555-233-121, Issue 1

Describes how to install, cable, test, and perform maintenance on a Survivable Remote Expansion Port Network (SREPN). Provides power, ground, and fiber connections.

DEFINITY ECS Release 8.2 — Upgrades and Additions for R8.2r, 555-233-115, Issue 1

Provides procedures for an installation technician to upgrade an existing DEFINITY Communications System or DEFINITY ECS to DEFINITY ECS Release 8.2.

Includes upgrade considerations, lists of required hardware, and step-by-step upgrade procedures. Also includes procedures to add control carriers, switch node carriers, port carriers, circuit packs, auxiliary cabinets, and other equipment.

DEFINITY ECS Release 8.2 — Upgrades and Additions for R8.2si, 555-233-122, Issue 1

Provides procedures for an installation technician to upgrade an existing DEFINITY Communications System or DEFINITY ECS to DEFINITY ECS Release 8.2.

Includes upgrade considerations, lists of required hardware, and step-by-step upgrade procedures. Also includes procedures to add control carriers, switch node carriers, port carriers, circuit packs, auxiliary cabinets, and other equipment.

DEFINITY ECS Release 8.2 — Maintenance for R8.2r, 555-233-117, Issue 1

Provides detailed descriptions of the procedures for monitoring, testing, troubleshooting, and maintaining the R8.2r ECS. Included are maintenance architecture, craft commands, step-by-step trouble-clearing procedures, the procedures for using all tests, and explanations of the system's error codes.

DEFINITY ECS Release 8.2 — Maintenance for R8.2si, 555-233-123, Issue 1

Provides detailed descriptions of the procedures for monitoring, testing, troubleshooting, and maintaining the R8.2si ECS. Included are maintenance architecture, craft commands, step-by-step trouble-clearing procedures, the procedures for using all tests, and explanations of the system's error codes.

DEFINITY ECS Release 8.2 — Maintenance for R8.2csi, 555-233-119, Issue 1

Provides detailed descriptions of the procedures for monitoring, testing, troubleshooting, and maintaining the R8.2csi (Compact Modular Cabinet) ECS. Included are maintenance architecture, craft commands, step-by-step trouble-clearing procedures, the procedures for using all tests, and explanations of the system's error codes.

Call center documents

These documents are issued for DEFINITY ECS Call Center applications. The intended audience is DEFINITY ECS administrators.

DEFINITY ECS Release 8 — Guide to ACD Call Centers, 555-233-503, Issue 2

This module contains information about the call center-specific features of the DEFINITY ECS.

DEFINITY ECS Release 8 — Call Vectoring/EAS Guide, 555-230-521, Issue 4

Provides information on how to write, use, and troubleshoot vectors, which are command sequences that process telephone calls in an Automatic Call Distribution (ACD) environment.It is provided in two parts: tutorial and reference. The tutorial provides step-by-step procedures for writing and implementing basic vectors. The reference includes detailed descriptions of the call vectoring features, vector management, vector administration, adjunct routing, troubleshooting, and interactions with management information systems (including the Call Management System).

DEFINITY ECS Release 8.2 — Maintenance for R8.2r, 555-233-117, Issue 1

Provides detailed descriptions of the procedures for monitoring, testing, troubleshooting, and maintaining the R8.2r ECS. Included are maintenance architecture, craft commands, step-by-step trouble-clearing procedures, the procedures for using all tests, and explanations of the system's error codes.

DEFINITY ECS Release 8.2 — Maintenance for R8.2si, 555-233-123, Issue 1

Provides detailed descriptions of the procedures for monitoring, testing, troubleshooting, and maintaining the R8.2si ECS. Included are maintenance architecture, craft commands, step-by-step trouble-clearing procedures, the procedures for using all tests, and explanations of the system's error codes.

DEFINITY ECS Release 8.2 — Maintenance for R8.2csi, 555-233-119, Issue 1

Provides detailed descriptions of the procedures for monitoring, testing, troubleshooting, and maintaining the R8.2csi (Compact Modular Cabinet) ECS. Included are maintenance architecture, craft commands, step-by-step trouble-clearing procedures, the procedures for using all tests, and explanations of the system's error codes.

Application-specific documents

These documents support specific DEFINITY applications.

ACD

DEFINITY Communications Systems G3 — Automatic Call Distribution (ACD) Agent Instructions, 555-230-722, Issue 5

Provides information for use by agents after they have completed ACD training. Includes descriptions of ACD features and the procedures for using them.

DEFINITY Communications Systems G3 — Automatic Call Distribution (ACD) Supervisor Instructions, 555-230-724, Issue 4

Provides information for use by supervisors after they have completed ACD training. Includes descriptions of ACD features and the procedures for using them.

Console operations

The primary audience for these documents consists of attendant console users.

DEFINITY ECS Release 7 — Console Operations, 555-230-700, Issue 4

Provides operating instructions for the attendant console. Included are descriptions of the console control keys and functions, call-handling procedures, basic system troubleshooting information, and routine maintenance procedures.

DEFINITY ECS Release 7 — Console Operations Quick Reference, 555-230-890, Issue 3

Provides operating instructions for the attendant console. Included are descriptions of the console control keys and functions, call-handling procedures, basic system troubleshooting information, and routine maintenance procedures. This document is available in languages other than English and can be ordered from the BCS Publications Catalog web site

Hospitality

DEFINITY ECS and Guestworks Release 8.2 — Hospitality Operations, 555-233-755, Issue 1

Provides step-by-step procedures for using the features available for the lodging and health industries to improve their property management and to provide assistance to their employees and clients. Includes detailed descriptions of reports.

Glossary and Abbreviations

Numerics

800 service
A service in the United States that allows incoming calls from certain areas to an assigned number for a flat-rate charge based on usage.

A

AA
Archangel. See angel.

AAC
ATM access concentrator

AAR
See Automatic Alternate Routing (AAR).

abandoned call
An incoming call in which the caller hangs up before the call is answered.

Abbreviated Dialing (AD)
A feature that allows callers to place calls by dialing just one or two digits.

AC
1. Alternating current.
2. See Administered Connection (AC).

AAR
Automatic Alternate Routing

ACA
See Automatic Circuit Assurance (ACA).

ACB
See Automatic Callback (ACB).

ACD
See Automatic Call Distribution (ACD).

ACD agent
See agent.

ACU
See Automatic calling unit (ACU)

ACW
See after-call work (ACW) mode.

access code
A 1-, 2-, or 3-digit dial code used to activate or cancel a feature, or access an outgoing trunk.

access endpoint
Either a nonsignaling channel on a DS1 interface or a nonsignaling port on an analog tie-trunk circuit pack that is assigned a unique extension.

access tie trunk
A trunk that connects a main communications system with a tandem communications system in an electronic tandem network (ETN). An access tie trunk can also be used to connect a system or tandem to a serving office or service node. Also called access trunk.

access trunk
See access tie trunk.

ACCUNET
A trademarked name for a family of digital services offered by AT&T in the United States.

ACD
See Automatic Call Distribution (ACD). ACD also refers to a work state in which an agent is on an ACD call.

ACD work mode
See work mode.

active-notification association
A link that is initiated by an adjunct, allowing it to receive event reports for a specific switch entity, such as an outgoing call.

active-notification call
A call for which event reports are sent over an active-notification association (communication channel) to the adjunct. Sometimes referred to as a monitored call.

active notification domain
VDN or ACD split extension for which event notification has been requested.

ACU
See Automatic calling unit (ACU).

AD
See Abbreviated Dialing (AD).

ADAP
AUDIX Data Acquisition Package

ADC
See analog-to-digital converter (ADC).

adjunct
A processor that does one or more tasks for another processor and that is optional in the configuration of the other processor. See also application.

adjunct-control association
A relationship initiated by an application via *Third Party Make Call*, the *Third Party Take Control,* or *Domain (Station) Control* capabilities to set up calls and control calls already in progress.

adjunct-controlled call
Call that can be controlled using an adjunct-control association. Call must have been originated via *Third Party Make Call* or *Domain (Station) Control* capabilities or must have been taken control of via *Third Party Take Control* or *Domain (Station) Control* capabilities.

adjunct-controlled split
An ACD split that is administered to be under adjunct control. Agents logged into such splits must do all telephony work, ACD login/ logout, and changes of work mode through the adjunct (except for auto-available adjunct-controlled splits, whose agents may not log in/out or change work mode).

adjunct-monitored call
An adjunct-controlled call, active-notification call, or call that provides event reporting over a domain-control association.

Adjunct-Switch Application Interface (ASAI)
A recommendation for interfacing adjuncts and communications systems, based on the CCITT Q.932 specification for layer 3.

ADM
Asynchronous data module

administer
To access and change parameters associated with the services or features of a system.

Administered Connection (AC)
A feature that allows the switch to automatically establish and maintain end-to-end connections between access endpoints (trunks) and/or data endpoints (data modules).

administration group
See capability group.

administration terminal
A terminal that is used to administer and maintain a system. See also terminal.

Administration Without Hardware (AWOH)
A feature that allows administration of ports without associated terminals or other hardware.

ADU
See asynchronous data unit (ADU).

AE
See access endpoint.

after-call work (ACW) mode
A mode in which agents are unavailable to receive ACD calls. Agents enter the ACW mode to perform ACD-related activities such as filling out a form after an ACD call.

AG
ASAI Gateway

agent
A person who receives calls directed to a split. A member of an ACD hunt group or ACD split. Also called an ACD agent.

agent report
A report that provides historical traffic information for internally measured agents.

AIM
Asynchronous interface module

AIOD
Automatic Identification of Outward Dialing

ALBO
Automatic Line Build Out

All trunks busy (ATB)
The state in which no trunks are available for call handling.

ALM-ACK
Alarm acknowledge

American Standard Code for Information Interchange
See ASCII (American Standard Code for Information Interchange).

AMW
Automatic Message Waiting

AN
Analog

analog
The representation of information by continuously variable physical quantities such as amplitude, frequency, and phase. See also digital.

analog data
Data that is transmitted over a digital facility in analog (PCM) form. The data must pass through a modem either at both ends or at a modem pool at the distant end.

analog telephone
A telephone that receives acoustic voice signals and sends analog electrical signals along the telephone line. Analog telephones are usually served by a single wire pair (tip and ring). The model-2500 telephone set is a typical example of an analog telephone.

analog-to-digital converter (ADC)
A device that converts an analog signal to digital form. See also digital-to-analog converter (DAC).

angel
A microprocessor located on each port card in a processor port network (PPN). The angel uses the control-channel message set (CCMS) to manage communications between the port card and the archangel on the controlling switch-processing element (SPE). The angel also monitors the status of other microprocessors on a port card and maintains error counters and thresholds.

ANI
See Automatic Number Identification (ANI).

ANSI
American National Standards Institute. A United States professional/technical association supporting a variety of standards.

answerback code
A number used to respond to a page from a code-calling or loudspeaker-paging system, or to retrieve a parked call.

AOL
Attendant-offered load

AP
Applications processor

APLT
Advanced Private-Line Termination

appearance
A software process that is associated with an extension and whose purpose is to supervise a call. An extension can have multiple appearances. Also called call appearance, line appearance, and occurrence. See also call appearance.

application
An adjunct that requests and receives ASAI services or capabilities. One or more applications can reside on a single adjunct. However, the switch cannot distinguish among several applications residing on the same adjunct and treats the adjunct, and all resident applications, as a single application. The terms application and adjunct are used interchangeably throughout this document.

applications processor
A micro-computer based, program controlled computer providing application services for the DEFINITY switch. The processor is used with several user-controlled applications such as traffic analysis and electronic documentation.

application service element
See capability group.

architecture
The organizational structure of a system, including hardware and software.

ARS
See Automatic Route Selection (ARS).

ASAI
See Adjunct-Switch Application Interface (ASAI)

ASCII (American Standard Code for Information Interchange)
The standard code for representing characters in digital form. Each character is represented by an 8-bit code (including parity bit).

association
A communication channel between adjunct and switch for messaging purposes. An active association is one that applies to an existing call on the switch or to an extension on the call.

asynchronous data transmission
A method of transmitting data in which each character is preceded by a start bit and followed by a stop bit, thus permitting data characters to be transmitted at irregular intervals. This type transmission is advantageous when transmission is not regular (characters typed at a keyboard). Also called asynchronous transmission. See also synchronous data transmission.

asynchronous data unit (ADU)
A device that allows direct connection between RS-232C equipment and a digital switch.

Asynchronous Transfer Mode (ATM)
A packet-like switching technology in which data is transmitted in fixed-size (53-byte) cells. ATM provides high-speed access for data communication in LAN, campus, and WAN environments.

ATB
See All trunks busy (ATB).

ATD
See Attention dial (ATD).

attendant
A person at a console who provides personalized service for incoming callers and voice-services users by performing switching and signaling operations. See also attendant console.

ATM
See Asynchronous Transfer Mode (ATM).

attendant console
The workstation used by an attendant. The attendant console allows the attendant to originate a call, answer an incoming call, transfer a call to another extension or trunk, put a call on hold, and remove a call from hold. Attendants using the console can also manage and monitor some system operations. Also called console. See also attendant.

Attention dial (ATD)
A command in the Hayes modem command set for asynchronous modems.

Audio Information Exchange (AUDIX)
A fully integrated voice-mail system. Can be used with a variety of communications systems to provide call-history data, such as subscriber identification and reason for redirection.

AUDIX
See Audio Information Exchange (AUDIX).

auto-in trunk group
Trunk group for which the CO processes all of the digits for an incoming call. When a CO seizes a trunk from an auto-in trunk group, the switch automatically connects the trunk to the destination — typically an ACD split where, if no agents are available, the call goes into a queue in which callers are answered in the order in which they arrive.

Auto-In Work mode
One of four agent work modes: the mode in which an agent is ready to process another call as soon as the current call is completed.

Automatic Alternate Routing (AAR)
A feature that routes calls to other than the first-choice route when facilities are unavailable.

Automatic Callback (ACB)
A feature that enables internal callers, upon reaching a busy extension, to have the system automatically connect and ring both parties when the called party becomes available.

Automatic Call Distribution (ACD)
A feature that answers calls, and then, depending on administered instructions, delivers messages appropriate for the caller and routes the call to an agent when one becomes available.

Automatic Call Distribution (ACD) split
A method of routing calls of a similar type among agents in a call center. Also, a group of extensions that are staffed by agents trained to handle a certain type of incoming call.

Automatic calling unit (ACU)
A device that places a telephone call.

Automatic Circuit Assurance (ACA)
A feature that tracks calls of unusual duration to facilitate troubleshooting. A high number of very short calls or a low number of very long calls may signify a faulty trunk

Automatic Number Identification (ANI)
Representation of the calling number, for display or for further use to access information about the caller.

automatic restoration
A service that restores disrupted connections between access endpoints (nonsignaling trunks) and data endpoints (devices that connect the switch to data terminal and/or communications equipment). Restoration is done within seconds of a service disruption so that critical data applications can remain operational.

Automatic Route Selection (ARS)
A feature that allows the system to automatically choose the least-cost way to send a toll call.

automatic trunk
A trunk that does not require addressing information because 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 communications-system attendant group. Also called automatic incoming trunk and automatic tie trunk.

AUX
Auxiliary

auxiliary equipment
Equipment used for optional system features, such as Loudspeaker Paging and Music-on-Hold.

auxiliary trunk
A trunk used to connect auxiliary equipment, such as radio-paging equipment, to a communications system.

Aux-Work mode
A work mode in which agents are unavailable to receive ACD calls. Agents enter Aux-Work mode when involved in non-ACD activities such as taking a break, going to lunch, or placing an outgoing call.

AVD
Alternate voice/data

AWOH
See Administration Without Hardware (AWOH).

AWG
American Wire Gauge

AWT
Average work time

B

B8ZS
Bipolar Eight Zero Substitution.

bandwidth
The difference, expressed in hertz, between the defined highest and lowest frequencies in a range.

barrier code
A security code used with the Remote Access feature to prevent unauthorized access to the system.

Basic Rate Interface (BRI)
A standard ISDN frame format that specifies the protocol used between two or more communications systems. BRI runs at 192 Mbps and provides two 64-kbps B-channels (voice and data) and one 16-kbps D-channel (signaling). The D-channel connects, monitors, and disconnects all calls. It also can carry low-speed packet data at 9.6 kbps.

baud
A unit of transmission rate equal to the number of signal events per second. See also bit rate and bits per second (bps).

BCC
See Bearer capability class (BCC).

BCMS
Basic Call Management System

BCT
See business communications terminal (BCT).

Bearer capability class (BCC)
Code that identifies the type of a call (for example, voice and different types of data). Determination of BCC is based on the caller's characteristics for non-ISDN endpoints and on the Bearer Capability and Low-Layer Compatibility Information Elements of an ISDN endpoint. Current BCCs are 0 (voice-grade data and voice), 1 (DMI mode 1, 56 kbps data transmission), 2 (DMI mode 2, synchronous/asynchronous data transmission up to 19.2 kbps) 3 (DMI mode 3, 64 kbps circuit/packet data transmission), 4 (DMI mode 0, 64 kbps synchronous data), 5 (temporary signaling connection, and 6 (wideband call, 128–1984 kbps synchronous data).

BER
Bit error rate

BHCC
Busy-hour call completions

bit (binary digit)
One unit of information in binary notation, having two possible values: 0 or 1.

bits per second (bps)
The number of binary units of information that are transmitted or received per second. See also baud and **bit rate**.

bit rate
The speed at which bits are transmitted, usually expressed in bits per second. Also called data rate. See also baud and bits per second (bps).

BLF
Busy Lamp Field

BN
Billing number

BOS
Bit-oriented signaling

BPN
Billed-party number

bps
See bits per second (bps).

bridge (bridging)
The appearance of a voice terminal's extension at one or more other voice terminals.

BRI
The ISDN Basic Rate Interface specification.

bridged appearance
A call appearance on a voice terminal that matches a call appearance on another voice terminal for the duration of a call.

BTU
British Thermal Unit

buffer
1. In hardware, 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.
2. In software, an area of memory that is used for temporary storage.

bus
A multiconductor electrical path used to transfer information over a common connection from any of several sources to any of several destinations.

business communications terminal (BCT)
A digital data terminal used for business applications. A BCT can function via a data module as a special-purpose terminal for services provided by a processor or as a terminal for data entry and retrieval.

BX.25
A version of the CCITT X.25 protocol for data communications. BX.25 adds a fourth level to the standard X.25 interface. This uppermost level combines levels 4, 5, and 6 of the ISO reference model.

bypass tie trunks
A 1-way, outgoing tie trunk from a tandem switch to a main switch in an ETN. Bypass tie 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 (usually eight) bits processed together.

C

cabinet
Housing for racks, shelves, or carriers that hold electronic equipment.

cable
Physical connection between two pieces of equipment (for example, data terminal and modem) or between a piece of equipment and a termination field.

cable connector
A jack (female) or plug (male) on the end of a cable. A cable connector connects wires on a cable to specific leads on telephone or data equipment.

CACR
Cancellation of Authorization Code Request

CAG
Coverage answer group

call appearance
1. For the attendant console, six buttons, labeled a–f, used to originate, receive, and hold calls. Two lights next to the button show the status of the call appearance.
2. For the voice terminal, a button labeled with an extension and used to place outgoing calls, receive incoming calls, or hold calls. Two lights next to the button show the status of the call appearance.

call-control capabilities
Capabilities (*Third Party Selective Hold, Third Party Reconnect, Third Party Merge*) that can be used in either of the Third Party Call Control ASE (cluster) subsets (Call Control and Domain Control).

Call Detail Recording (CDR)
A feature that uses software and hardware to record call data (same as CDRU).

Call Detail Recording utility (CDRU)
Software that collects, stores, optionally filters, and outputs call-detail records.

Call Management System (CMS)
An application, running on an adjunct processor, that collects information from an ACD unit. CMS enables customers to monitor and manage telemarketing centers by generating reports on the status of agents, splits, trunks, trunk groups, vectors, and VDNs, and enables customers to partially administer the ACD feature for a communications system.

call-reference value (CRV)
An identifier present in ISDN messages that associates a related sequence of messages. In ASAI, CRVs distinguish between associations.

call vector
A set of up to 15 vector commands to be performed for an incoming or internal call.

callback call
A call that automatically returns to a voice-terminal user who activated the Automatic Callback or Ringback Queuing feature.

call-waiting ringback tone
A low-pitched tone identical to ringback tone except that the tone decreases in the last 0.2 seconds (in the United States). Call-waiting ringback tone notifies the attendant that the Attendant Call Waiting feature is activated and that the called party is aware of the waiting call. Tones in international countries may sound different.

call work code
A number, up to 16 digits, entered by ACD agents to record the occurrence of customer-defined events (such as account codes, social security numbers, or phone numbers) on ACD calls.

CAMA
Centralized Automatic Message Accounting

carrier
An enclosed shelf containing vertical slots that hold circuit packs.

carried load
The amount of traffic served by traffic-sensitive facilities during a given interval.

CARR-POW
Carrier Port and Power Unit for AC Powered Systems

CAS
Centralized Attendant Service or Call Accounting System

capability
A request or indication of an operation. For example, *Third Party Make Call* is a request for setting up a call; *event report* is an indication that an event has occurred.

capability group
Set of capabilities, determined by switch administration, that can be requested by an application. Capability groups denote association types. For example, *Call Control* is a type of association that allows certain functions (the ones in the capability group) to be performed over this type of association. Also referred to as administration groups or application service elements (ASEs).

CA-TSC
Call-Associated Temporary Signaling Connection

cause value
A value is returned in response to requests or in event reports when a denial or unexpected condition occurs. ASAI cause values fall into two coding standards: Coding Standard 0 includes any cause values that are part of AT&T and CCITT ISDN specifications; Coding standard 3 includes any other ASAI cause values. This document uses a notation for cause value where the coding standard for the cause is given first, then a slash, then the cause value. Example: CS0/100 is coding standard 0, cause value 100.

CBC
Call-by-call or coupled bonding conductor

CC
Country code

CCIS
Common-Channel Interoffice Signaling

CCITT
CCITT (Comitte Consultatif International Telephonique et Telegraphique), now called *International Telecommunications Union* (ITU). See International Telecommunications Union (ITU).

CCMS
Control-Channel Message Set

CCS
See capability.

CCS or hundred call seconds
A unit of call traffic. Call traffic for a facility is scanned every 100 seconds. If the facility is busy, it is assumed to have been busy for the entire scan interval. There are 3600 seconds per hour. The Roman numeral for 100 is the capital letter C. The abbreviation for call seconds is CS. Therefore, 100 call seconds is abbreviated CCS. If a facility is busy for an entire hour, then it is said to have been busy for 36 CCS. See also
Erlang.

CCSA
Common-Control Switching Arrangement

CDM
Channel-division multiplexing

CDOS
Customer-dialed and operator serviced

CDPD
Customer database-provided digits

CDR
See Call Detail Recording (CDR).

CDRP
Call Detail Record Poller

CDRR
Call Detail Recording and Reporting

CDRU
See Call Detail Recording utility (CDRU).

CED
Caller entered digits

CEM
Channel-expansion multiplexing

center-stage switch (CSS)
The central interface between the processor port network and expansion port networks in a CSS-connected system.

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

central office (CO) codes
The first three digits of a 7-digit public-network telephone number in the United States.

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

CEPT1
European Conference of Postal and Telecommunications Rate 1

CESID
Caller's Emergency Service Identification

channel
1. A circuit-switched call.
2. A communications path for transmitting voice and data.
3. In wideband, all of the time slots (contiguous or noncontiguous) necessary to support a call. Example: an H0-channel uses six 64-kbps time slots.
4. A DS0 on a T1 or E1 facility not specifically associated with a logical circuit-switched call; analogous to a single trunk.

channel negotiation
The process by which the channel offered in the Channel Identification Information Element (CIIE) in the SETUP message is negotiated to be another channel acceptable to the switch that receives the SETUP message and ultimately to the switch that sent the SETUP. Negotiation is attempted only if the CIIE is encoded as *Preferred.* Channel negotiation is not attempted for wideband calls.

CI
Clock input

circuit
1. An arrangement of electrical elements through which electric current flows. 2. A channel or transmission path between two or more points.

circuit pack
A card on which electrical circuits are printed, and IC chips and electrical components are installed. A circuit pack is installed in a switch carrier.

CISPR
International Special Committee on Radio Interference

Class of Restriction (COR)
A feature that allows up to 96 classes of call-origination and call-termination restrictions for voice terminals, voice-terminal groups, data modules, and trunk groups. See also Class of Service (COS).

Class of Service (COS)
A feature that uses a number to specify if voice-terminal users can activate the Automatic Callback, Call Forwarding All Calls, Data Privacy, or Priority Calling features. See also Class of Restriction (COR).

cm
Centimeter

CM
Connection Manager

CMC
Compact Modular Cabinet

CMDR
Centralized Message Detail Recording

CMS
Call Management System

CO
See central office (CO).

common-control switching arrangement (CCSA)
A private telecommunications network using dedicated trunks and a shared switching center for interconnecting company locations.

communications system
The software-controlled processor complex that interprets dialing pulses, tones, and keyboard characters and makes the proper connections both within the system and external to the system. The communications system itself consists of a digital computer, software, storage device, and carriers with special hardware to perform the connections. A communications system provides voice and data communications services, including access to public and private networks, for telephones and data terminals on a customer's premises. See also switch.

confirmation tone
A tone confirming that feature activation, deactivation, or cancellation has been accepted.

connectivity
The connection of disparate devices within a single system.

console
See attendant console.

contiguous
Adjacent DS0s within one T1 or E1 facility or adjacent TDM or fiber time slots. The first and last TDM bus, DS0, or fiber time slots are not considered contiguous (no wraparound). For an E1 facility with a D-channel, DS0s 15 and 17 are considered contiguous.

control cabinet
See control carrier.

control carrier
A carrier in a multi-carrier cabinet that contains the SPE circuit packs and, unlike an G3r control carrier, port circuit packs. Also called control cabinet in a single-carrier cabinet. See also switch-processing element (SPE).

controlled station
A station that is monitored and controlled via a domain-control association.

COR
See Class of Restriction (COR).

COS
See Class of Service (COS).

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 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
An extension or attendant group, VDN, or ACD split designated as an alternate answering position in a coverage path.

covering user
A person at a coverage point who answers a redirected call.

CP
Circuit pack

CPE
Customer-premises equipment

CPN
Called-party number

CPN/BN
Calling-party number/billing number

CPTR
Call-progress-tone receiver

CRC
Cyclical Redundancy Checking

critical-reliability system
A system that has the following duplicated items: control carriers, tone clocks, EI circuit packs, and cabling between port networks and center-stage switch in a CSS-connected system. See also duplicated common control, and duplication.

CSA
Canadian Safety Association or Customer Software Administrator

CSCC
Compact single-carrier cabinet

CSCN
Center-stage control network

CSD
Customer-service document

CSM
Centralized System Management

CSS
See center-stage switch (CSS).

CSSO
Customer Services Support Organization

CSU
Channel service unit

CTS
Clear to Send

CWC
See call work code.

D

DAC
1. Dial access code or Direct Agent Calling
2. 2. See digital-to-analog converter (DAC).

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

data-communications equipment (DCE)
The equipment (usually a modem, data module, or packet assembler/disassembler) on the network side of a communications link that makes the binary serial data from the source or transmitter compatible with the communications channel.

data link
The configuration of physical facilities enabling end terminals to communicate directly with each other.

data module
An interconnection device between a BRI or DCP interface of the switch and data terminal equipment or data communications equipment.

data path
The end-to-end connection used for a data communications link. A data path is the combination of all elements of an interprocessor communication in a DCS.

data port
A point of access to a computer that uses trunks or lines for transmitting or receiving data.

data rate
See bit rate.

data service unit (DSU)
A device that transmits digital data on transmission facilities.

data terminal
An input/output (I/O) device that has either switched or direct access to a host computer or to a processor interface.

data terminal equipment (DTE)
Equipment consisting of the endpoints in a connection over a data circuit. In a connection between a data terminal and host, the terminal, the host, and their associated modems or data modules make up the DTE.

dB
Decibel

dBA
Decibels in reference to amperes.

dBrnC
Decibels above reference noise with C filter.

DC
Direct current

DCE
Data-communications equipment

D-channel backup
Type of backup used with Non-Facility Associated Signaling (NFAS). A primary D-channel provides signaling for an NFAS D-channel group (two or more PRI facilities). A second D-channel, on a separate PRI facility of the NFAS D-channel group, is designated as backup for the D-channel. Failure of the primary D-channel causes automatic transfer of call-control signaling to the backup D-channel. The backup becomes the primary D-channel. When the failed channel returns to service, it becomes the backup D-channel.

DCO
Digital central office

DCP
Digital Communications Protocol

DCS
Distributed Communications System

DDC
Direct Department Calling

DDD
Direct Distance Dialing

delay-dial trunk
A trunk that allows dialing directly into a communications system (digits are received as they are dialed).

denying a request
Sending a negative acknowledgment (NAK), done by sending an FIE with a *return error* component (and a cause value). It should not be confused with the denial event report that applies to calls.

designated voice terminal
The specific voice terminal to which calls, originally directed to a certain extension, are redirected. Commonly used to mean the forwarded-to terminal when Call Forwarding All Calls is active.

dial-repeating trunks
A PBX tie trunk that is capable of handling PBX station-signaling information without attendant assistance.

dial-repeating tie trunk
A tie trunk that transmits called-party addressing information between two communications systems.

DID
Direct Inward Dialing

digit conversion
A process used to convert specific dialed numbers into other dialed numbers.

digital
The representation of information by discrete steps. See also analog.

digital communications protocol (DCP)
A proprietary protocol used to transmit both digitized voice and digitized data over the same communications link. A DCP link is made up of two 64-kbps information (I-) channels and one 8-kbps signaling (S-) channel. The DCP protocol supports two information-bearing channels, and thus two telephones/data modules. The I1 channel is the DCP channel assigned on the first page of the 8411 station form. The I2 channel is the DCP channel assigned on the analog adjunct page of the 8411 station form or on the data module page.

Digital Communications Protocol. The DCP protocol supports two information-bearing channels, and thus two telephones/data modules. The I1 channel is the DCP channel assigned on the first page of the 8411 station form. The I2 channel is the DCP channel assigned on the analog adjunct page of the 8411 station form or on the data module page.

digital data endpoints
In DEFINITY ECS, devices such as the 510D terminal or the 515-type business communications terminal (BCT).

digital multiplexed interface (DMI)
An interface that provides connectivity between a communications system and a host computer or between two communications systems using DS1 24th-channel signaling. DMI provides 23 64-kbps data channels and 1 common-signaling channel over a twisted-pair connection. DMI is offered through two capabilities: bit-oriented signaling (DMI-BOS) and message-oriented signaling (DMI-MOS).

digital signal level 0 (DS0)
A single 64-kbps voice channel. A DS0 is a single 64-kbps channel in a T1 or E1 facility and consists of eight bits in a T1 or E1 frame every 125 microseconds.

digital signal level 1 (DS1)
A single 1.544-Mbps (United States) or 2.048-Mbps (outside the United States) digital signal carried on a T1 transmission facility. A DS1 converter complex consists of a pair, one at each end, of DS1 converter circuit packs and the associated T1/E1 facilities.

digital terminal data module (DTDM)
An integrated or adjunct data module that shares with a digital telephone the same physical port for connection to a communications system. The function of a DTDM is similar to that of a PDM and MPDM in that it converts RS-232C signals to DCP signals.

digital-to-analog converter (DAC)
A device that converts data in digital form to the corresponding analog signals. See also analog-to-digital converter (ADC).

digital transmission
A mode of transmission in which information to be transmitted is first converted to digital form and then transmitted as a serial stream of pulses.

digital trunk
A circuit that carries digital voice and/or digital data in a telecommunications channel.

DIOD
Direct Inward and Outward Dialing

direct agent
A feature, accessed only via ASAI, that allows a call to be placed in a split queue but routed only to a specific agent in that split. The call receives normal ACD call treatment (for example, announcements) and is measured as an ACD call while ensuring that a particular agent answers.

Direct Extension Selection (DXS)
A feature on an attendant console that allows an attendant direct access to voice terminals by pressing a group-select button and a DXS button.

Direct Inward Dialing (DID)
A feature that allows an incoming call from the public network (not FX or WATS) to reach a specific telephone without attendant assistance.

Direct Inward Dialing (DID) trunk
An incoming trunk used for dialing directly from the public network into a communications system without help from the attendant.

disk drive
An electromechanical device that stores data on and retrieves data from one or more disks.

distributed communications system (DCS)
A network configuration linking two or more communications systems in such a way that selected features appear to operate as if the network were one system.

DIVA
Data In/Voice Answer

DLC
Data line circuit

DLDM
Data-line data module

DMI
Digital-multiplexed interface

DND
Do not disturb

DNIS
Dialed-Number Identification Service

DOD
Direct Outward Dialing

domain
VDNs, ACD splits, and stations. The VDN domain is used for active-notification associations. The ACD-split domain is for active-notification associations and domain-control associations. The station domain is used for the domain-control associations.

domain-control association
A *Third Party Domain Control Request* capability initiates a unique CRV/link number combination, which is referred to as a domain-control association.

domain-controlled split
A split for which *Third Party Domain Control* request has been accepted. A domain-controlled split provides an event report for logout.

domain-controlled station
A station for which a *Third Party Domain Control* request has been accepted. A domain-controlled station provides event reports for calls that are alerting, connected, or held at the station.

domain-controlled station on a call
A station that is active on a call, and which provides event reports over one or two domain-control associations.

DOSS
Delivery Operations Support System

DOT
Duplication Option Terminal

DPM
Dial Plan Manager

DPR
Dual-port RAM

DS1
Digital Signal Level 1

DS1C
Digital Signal Level-1 protocol C

DS1 CONV
Digital Signal Level-1 converter

DSI
Digital signal interface

DSU
Data service unit

DTDM
Digital-terminal data module

DTE
Data-terminal equipment

DTGS
Direct Trunk Group Select

DTMF
Dual-tone multifrequency

DTS
Disk-tape system

duplicated common control
Two processors ensuring continuous operation of a communications system. While one processor is online, the other functions as a backup. The backup processor goes online periodically or when a problem occurs.

duplication
The use of redundant components to improve availability. When a duplicated subsystem fails, its backup redundant system automatically takes over.

duplication option
A system option that duplicates the following: control carrier containing the SPE, EI circuit packs in carriers, fiber-optic cabling between port networks, and center-stage switch in a CSS-connected system.

DWBS
DEFINITY Wireless Business System

DXS
Direct extension selection

E

E1
A digital transmission standard that carries traffic at 2.048 Mbps. The E1 facility is divided into 32 channels (DS0s) of 64 kbps information. Channel 0 is reserved for framing and synchronization information. A D-channel occupies channel 16.

E & M
Ear and mouth (receive and transmit)

EA
Expansion archangel

EAL
Expansion archangel link

ear and mouth (E & M) signaling
Trunk supervisory signaling, used between two communications systems, whereby signaling information is transferred through 2-state voltage conditions (on the E and M leads) for analog applications and through a single bit for digital applications.

EAS
See "Expert Agent Selection".

ECC
Error Correct Code

ECMA
European Computer Manufacturers Association

EEBCDIC
Extended Binary-Coded Decimal Interexchange Code

EFP
Electronic power feed

EI
Expansion interface

EIA
Electronic Industries Association

EIA-232
A physical interface specified by the EIA. EIA-232 transmits and receives asynchronous data at speeds of up to 19.2 kbps over cable distances of up to 50 feet. EIA-232 replaces RS-232 protocol in some DEFINITY applications.

electronic tandem network (ETN)
A tandem tie-trunk network that has automatic call-routing capabilities based on the number dialed and the most preferred route available. Each switch in the network is assigned a unique private network office code (RNX), and each voice terminal is assigned a unique extension.

Electronics Industries Association (EIA)
A trade association of the electronics industry that establishes electrical and functional standards.

emergency transfer
If a major system failure occurs, automatic transfer is initiated to a group of telephones capable of making outgoing calls. The system operates in this mode until the failure is repaired and the system automatically returns to normal operation. Also called power-failure transfer.

EMI
Electromagnetic interference

end-to-end signaling
The transmission of touch-tone signals generated by dialing from a voice terminal to remote computer equipment. These digits are sent over the trunk as DTMF digits whether the trunk signaling type is marked as tone or rotary and whether the originating station is tone or rotary. Example: a call to a voice-mail machine or automated-attendant service. A connection is first established over an outgoing trunk. Then additional digits are dialed to transmit information to be processed by the computer equipment.

enhanced private-switched communications service (EPSCS)
An analog private telecommunications network based on the No. 5 crossbar and 1A ESS that provides advanced voice and data telecommunications services to companies with many locations.

EPN
Expansion-port network

EPROM
Erasable programmable read-only memory

EPSCS
Enhanced Private Switched Communications Services

ERL
Echo return loss

Erlang
A unit of traffic intensity, or load, used to express the amount of traffic needed to keep one facility busy for one hour. One Erlang is equal to 36 CCS. See also capability.

ESF
Extended superframe format

ESPA
European Standard Paging Access

ETA
Extended Trunk Access; also Enhanced Terminal Administration

ETN
Electronic tandem network

ETSI
European Telecommunications Standards Institute

expansion archangel (EAA)
A network-control microprocessor located on an expansion interface (EI) port circuit pack in an expansion port network. The EA provides an interface between the EPN and its controlling switch-processing element.

expansion-archangel link (EAL)
A link-access function on the D-channel (LAPD) logical link that exists between a switch-processing element and an expansion archangel (EA). The EAL carries control messages from the SPE to the EA and to port circuit packs in an expansion port network.

expansion control cabinet
See expansion control carrier.

expansion control carrier
A carrier in a multicarrier cabinet that contains extra port circuit packs and a maintenance interface. Also called expansion control cabinet in a single-carrier cabinet.

expansion interface (EI)
A port circuit pack in a port network that provides the interface between a PN's TDM bus/ packet bus and a fiber-optic link. The EI carries circuit-switched data, packet-switched data, network control, timing control, and DS1 control. In addition, an EI in an expansion port network communicates with the master maintenance circuit pack to provide the EPN's environmental and alarm status to the switch-processing element.

expansion port network (EPN)
A port network (PN) that is connected to the TDM bus and packet bus of a processor port network (PPN). Control is achieved by indirect connection of the EPN to the PPN via a port-network link (PNL). See also port network (PN).

Expert Agent Selection
A feature allowing incoming calls to be routed to specialized groups of agents within a larger pool of agents.

extension-in
Extension-In (ExtIn) is the work state agents go into when they answer (receive) a non-ACD call. If the agent is in Manual-In or Auto-In and receives an extension-in call, it is recorded by CMS as an AUX-In call.

extension-out
The work state that agents go into when they place (originate) a non-ACD call.

external measurements
Those ACD measurements that are made by the External CMS adjunct.

extension
A 1- to 5-digit number by which calls are routed through a communications system or, with a Uniform Dial Plan (UDP) or main-satellite dialing plan, through a private network.

external call
A connection between a communications system user and a party on the public network or on another communications system in a private network.

F

FAC
Feature Access Code

facility
A telecommunications transmission pathway and associated equipment.

facility-associated signaling (FAS)
Signaling for which a D-channel carries signaling only for those channels on the same physical interface.

FAS
Facility-associated signaling

FAT
Facility access trunk

FAX
Facsimile

FCC
Federal Communications Commission

FEAC
Forced Entry of Account Codes

feature
A specifically defined function or service provided by the system.

feature button
A labeled button on a telephone or attendant console used to access a specific feature.

FEP
Front-end processor

fiber optics
A technology using materials that transmit ultrawideband electromagnetic light-frequency ranges for high-capacity carrier systems.

FIC
Facility interface codes

fixed
A trunk allocation term. In the fixed allocation scheme, the time slots necessary to support a wideband call are contiguous, and the first time slot is constrained to certain starting points.

flexible
A trunk allocation term. In the flexible allocation scheme, the time slots of a wideband call can occupy noncontiguous positions within a single T1 or E1 facility.

floating
A trunk allocation term. In the floating allocation scheme, the time slots of a wideband call are contiguous, but the position of the first time slot is not fixed.

FNPA
Foreign Numbering-Plan Area

foreign-exchange (FX)
A CO 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 CO other than its local CO.

foreign numbering-plan area code (FNPAC)
An area code other than the local area code, that must be dialed to call outside the local geographical area.

FRL
Facilities Restriction Level

FX
Foreign exchange

G

G3-MA
Generic 3 Management Applications

G3-MT
Generic 3 Management Terminal

G3r
Generic 3, RISC (Reduced Instruction Set Computer)

generalized route selection (GRS)
An enhancement to Automatic Alternate Routing/Automatic Route Selection (AAR/ARS) that performs routing based on call attributes, such as Bearer Capability Classes (BCCs), in addition to the address and facilities restriction level (FRL), thus facilitating a Uniform Dial Plan (UDP) that is independent of the type of call being placed.

glare
The simultaneous seizure of a 2-way trunk by two communications systems, resulting in a standoff.

GM
Group manager

GPTR
General-purpose tone receiver

grade of service
The number of call attempts that fail to receive service immediately. Grade of service is also expressed as the quantity of all calls that are blocked or delayed.

ground-start trunk
A trunk on which, for outgoing calls, the system transmits a request for services to a distant switching system by grounding the trunk ring lead. To receive the digits of the called number, that system grounds the trunk tip lead. When the system detects this ground, the digits are sent.

GRS
Generalized Route Selection

H

H0
An ISDN information transfer rate for 384-kbps data defined by CCITT and ANSI standards.

H11
An ISDN information transfer rate for 1536-kbps data defined by CCITT and ANSI standards.

H12
An ISDN information transfer rate for 1920-kbps data defined by CCITT and ANSI standards.

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

hertz (Hz)
A unit of frequency equal to one cycle per second.

high-reliability system
A system having the following: two control carriers, duplicate expansion interface (EI) circuit packs in the PPN (in G3r with CSS), and duplicate switch node clock circuit packs in the switch node (SN) carriers. See also duplicated common control, duplication, duplication option, and critical-reliability system.

HNPA
See home numbering-plan area code (HNPA).

holding time
The total length of time in minutes and seconds that a facility is used during a call.

home numbering-plan area code (HNPA)
The local area code. The area code does not have to be dialed to call numbers within the local geographical area.

hop
Nondirect communication between two switch communications interfaces (SCI) where the SCI message passes automatically without intermediate processing through one or more intermediate SCIs.

host computer
A computer, connected to a network, that processes data from data-entry devices.

hunt group
A group of extensions that are assigned the Station Hunting feature so that a call to a busy extension reroutes to an idle extension in the group. See also ACD work mode.

Hz
See hertz (Hz).

I

I1
The first information channel of DCP.

I2
The second information channel of DCP.

I2 Interface
A proprietary interface used for the DEFINITY Wireless Business System for the radio-controller circuit packs. Each interface provides communication between the radio-controller circuit pack and up to two wireless fixed bases.

I3 Interface
A proprietary interface used for the DEFINITY Wireless Business System for the cell antenna units. Each wireless fixed base can communicate to up to four cell antenna units.

IAS
Inter-PBX Attendant Service

ICC
Intercabinet cable or intercarrier cable

ICD
Inbound Call Director

ICDOS
International Customer-Dialed Operator Service

ICHT
Incoming call-handling table

ICI
Incoming call identifier

ICM
Inbound Call Management

IDDD
International Direct Distance Dialing

IDF
Intermediate distribution frame

IE
Information element

immediate-start tie trunk
A trunk on which, after making a connection with a distant switching system for an outgoing call, the system waits a nominal 65 ms before sending the digits of the called number. This allows time for the distant system to prepare to receive digits. On an incoming call, the system has less than 65 ms to prepare to receive the digits.

IMT
Intermachine trunk

in
Inch

INADS
Initialization and Administration System

ICLID
Incoming Caller ID

incoming gateway
A PBX that routes an incoming call on a trunk *not* administered for Supplementary Services Protocol B to a trunk *not* administered for Supplementary Services Protocol B.

information exchange
The exchange of data between users of two different systems, such as the switch and a host computer, over a LAN.

Information Systems Network (ISN)
A WAN and LAN with an open architecture combining host computers, minicomputers, word processors, storage devices, PCs, high-speed printers, and nonintelligent terminals into a single packet-switching system.

INS
ISDN Network Service

inside call
A call placed from one telephone to another within the local communications system.

Integrated Services Digital Network (ISDN)
A public or private network that provides end-to-end digital communications for all services to which users have access by a limited set of standard multipurpose user-network interfaces defined by the CCITT. Through internationally accepted standard interfaces, ISDN provides digital circuit-switched or packet-switched communications within the network and links to other ISDNs to provide national and international digital communications. See also Integrated Services Digital Network Basic Rate Interface (ISDN-BRI) and Integrated Services Digital Network Primary Rate Interface (ISDN-PRI).

Integrated Services Digital Network Basic Rate Interface (ISDN-BRI)
The interface between a communications system and terminal that includes two 64-kbps B-channels for transmitting voice or data and one 16-kbps D-channel for transmitting associated B-channel call control and out-of-band signaling information. ISDN-BRI also includes 48 kbps for transmitting framing and D-channel contention information, for a total interface speed of 192 kbps. ISDN-BRI serves ISDN terminals and digital terminals fitted with ISDN terminal adapters. See also Integrated Services Digital Network (ISDN) andInte-grated Services Digital Network Primary Rate Interface (ISDN-PRI).

Integrated Services Digital Network Primary Rate Interface (ISDN-PRI)
The interface between multiple communications systems that in North America includes 24 64-kbps channels, corresponding to the North American digital signal level-1 (DS1) standard rate of 1.544 Mbps. The most common arrangement of channels in ISDN-PRI is 23 64-kbps B-channels for transmitting voice and data and 1 64-kbps D-channel for transmitting associated B-channel call control and out-of-band signaling information. With nonfacility-associated signaling (NFAS), ISDN-PRI can include 24 B-channels and no D-channel. See also Integrated Services Digital Network (ISDN) and Integrated Services Digital Network Basic Rate Interface (ISDN-BRI).

intercept tone
A tone that 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 a system.

International Telecommunications Union (ITU)
Formerly known as International Telegraph and Telephone Consultative Committee (CCITT), ITU is an international organization that sets universal standards for data communications, including ISDN. ITU members are from telecommunications companies and organizations around the world. See alsoBX.25.

International Telegraph and Telephone Consultative Committee
See International Telecommunications Union (ITU).

interflow
The ability for calls to forward to other splits on the same PBX or a different PBX using the Call Forward All Calls feature.

intraflow
The ability for calls to redirect to other splits on the same PBX on a conditional or unconditional basis using call coverage busy, don't answer, or all criteria.

internal measurements
BCMS measurements that are made by the system. ACD measurements that are made external to the system (via External CMS) are referred to as external measurements.

in-use lamp
A red light on a multiappearance 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.

INWATS
Inward Wide Area Telephone Service

IO
Information outlet

ISDN
See Integrated Services Digital Network (ISDN).

ISDN Gateway (IG)
A feature allowing integration of the switch and a host-based telemarketing application via a link to a gateway adjunct. The gateway adjunct is a 3B-based product that notifies the host-based telemarketing application of call events.

ISDN trunk
A trunk administered for use with ISDN-PRI. Also called ISDN facility.

ISDN-PRI terminal adapter
An interface between endpoint applications and an ISDN PRI facility. ISDN-PRI terminal adapters are currently available from other vendors and are primarily designed for video conferencing applications. Accordingly, currently available terminal adapters adapt the two pairs of video codec data (V.35) and dialing (RS-366) ports to an ISDN PRI facility.

IS/DTT
Integrated Services/digital tie trunk

ISN
Information Systems Network

ISO
International Standards Organization

ISV
Independent software vendor

ITP
Installation test procedure

ITU
International Telecommunications Union

IXC
Interexchange carrier code

K

kHz
Kilohertz

kbps
Kilobits per second

kbyte
Kilobyte

kg
Kilogram

L

LAN
Local area network

LAP-D
Link Access Procedure on the D-channel

LAPD
Link Access Procedure data

LATA
Local access and transport area

lb
Pound

LBO
Line buildout

LDN
Listed directory number

LDS
Long-distance service

LEC
Local exchange carrier

LED
See light-emitting diode (LED).

light-emitting diode (LED)
A semiconductor device that produces light when voltage is applied. LEDs provide a visual indication of the operational status of hardware components, the results of maintenance tests, the alarm status of circuit packs, and the activation of telephone features.

lightwave transceiver
Hardware that provides an interface to fiber-optic cable from port circuit packs and DS1 converter circuit packs. Lightwave transceivers convert electrical signals to light signals and vice versa.

line
A transmission path between a communications system or CO switching system and a voice terminal or other terminal.

line appearance
See appearance.

line buildout
A selectable output attenuation is generally required of DTE equipment because T1 circuits require the last span to lose 15–22.5 dB.

line port
Hardware that provides the access point to a communications system for each circuit associated with a telephone or data terminal.

link
A transmitter-receiver channel that connects two systems.

link-access procedure on the D-channel (LAPD)
A link-layer protocol on the ISDN-BRI and ISDN-PRI data-link layer (level 2). LAPD provides data transfer between two devices, and error and flow control on multiple logical links. LAPD is used for signaling and low-speed packet data (X.25 and mode 3) on the signaling (D-) channel and for mode-3 data communications on a bearer (B-) channel.

LINL
Local indirect neighbor link

local area network (LAN)
A networking arrangement designed for a limited geographical area. Generally, a LAN is limited in range to a maximum of 6.2 miles and provides high-speed carrier service with low error rates. Common configurations include daisy chain, star (including circuit-switched), ring, and bus.

logical link
The communications path between a processor and a BRI terminal.

loop-start trunk
A trunk on which, after establishing a connection with a distant switching system for an outgoing call, the system waits for a signal on the loop formed by the trunk leads before sending the digits of the called number.

loss plan
The overall plan, used in network design and management, for creating and maintaining consistent signal strength across the network. The term also applies to local management of signal strength to achieve appropriate levels for specific applications.

LSU
Local storage unit

LWC
Leave Word Calling

M

MAC
Medium access

MADU
Modular asynchronous data unit

main distribution frame (MDF)
A device that mounts to the wall inside the system equipment room. The MDF provides a connection point from outside telephone lines to the PBX switch and to the inside telephone stations.

main-satellite-tributary
A private network configuration that can either stand alone or access an ETN. 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 one switch, with one listed directory number (LDN). A tributary switch is connected to the main switch via tie trunks, but has its own attendant positions and LDN.

maintenance
Activities involved in keeping a telecommunications system in proper working condition: the detection and isolation of software and hardware faults, and automatic and manual recovery from these faults.

management terminal
The terminal that is used by the system administrator to administer the switch. The terminal may also be used to access the BCMS feature.

major alarm
An indication of a failure that has caused critical degradation of service and requires immediate attention. Major alarms are automatically displayed on LEDs on the attendant console and maintenance or alarming circuit pack, logged to the alarm log, and reported to a remote maintenance facility, if applicable.

Manual-In work mode
One of four agent work modes: the mode in which an agent is ready to process another call manually. *See* Auto-In Work mode for a contrast.

MAP
Maintenance action process

MAPD
Multiapplication platform for DEFINITY

MA-UUI
Message-Associated User-to-User Signaling

Mbps
Megabits per second

M-Bus
Memory bus

Mbyte
Megabyte

MCC
Multicarrier cabinet

MCS
Message Center Service

MCT
Malicious Call Trace

MCU
Multipoint control unit

MDF
Main distribution frame

MDM
Modular data module

MDR
Message detail record

MEM
Memory

memory
A device into which information can be copied and held, and from which information can later be obtained.

memory shadowing link
An operating-system condition that provides a method for memory-resident programs to be more quickly accessed, allowing a system to reboot faster.

message center
An answering service that supplies agents to and stores messages for later retrieval.

message center agent
A member of a message-center hunt group who takes and retrieves messages for voice-terminal users.

MET
Multibutton electronic telephone

MF
Multifrequency

MFB
Multifunction board

MFC
Multifrequency code

MHz
Megahertz

MIM
Management information message

minor alarm
An indication of a failure that could affect customer service. Minor alarms are automatically displayed on LEDs on the attendant console and maintenance or alarming circuit pack, sent to the alarm log, and reported to a remote maintenance facility, if applicable.

MIPS
Million instructions per second

MIS
Management information system

MISCID
Miscellaneous identification

MMCS
Multimedia Call Server

MMCH
Multimedia call handling

MMI
Multimedia interface

MMS
Material Management Services

MO
Maintenance object

modem
A device that converts digital data signals to analog signals for transmission over telephone circuits. The analog signals are converted back to the original digital data signals by another modem at the other end of the circuit.

modem pooling
A capability that provides shared conversion resources (modems and data modules) for cost-effective access to analog facilities by data terminals. When needed, modem pooling inserts a conversion resource into the path of a data call. Modem pooling serves both outgoing and incoming calls.

modular processor data module (MPDM)
A processor data module (PDM) that can be configured to provide several kinds of interfaces (RS-232C, RS-449, and V.35) to customer-provided data terminal equipment (DTE). See also processor data module (PDM).

modular trunk data module (MTDM)
A trunk data module that can be configured to provide several kinds of interfaces (RS-232, RS-449, and V.35) to customer-provided data terminal equipment.

modulator-demodulator
See modem.

monitored call
See active-notification call.

MOS
Message-oriented signaling

MPDM
Modular processor data module

MS
Message server

ms
Millisecond

MS/T
Main satellite/tributary

MSA
Message servicing adjunct

MSG
Message service

MSL
Material stocking location

MSM
Modular System Management

MSS
Mass storage system

MSSNET
Mass storage/network control

MT
Management terminal

MTDM
Modular trunk data module

MTP
Maintenance tape processor

MTT
Multitasking terminal

multiappearance voice terminal
A terminal equipped with several call-appearance buttons for the same extension, allowing the user to handle more than one call on that same extension at the same time.

Multicarrier cabinet
A structure that holds one to five carriers. See also single-carrier cabinet.

Multifrequency Compelled (MFC) Release 2 (R2) signaling
A signal consisting of two frequency components, such that when a signal is transmitted from a switch, another signal acknowledging the transmitted signal is received by the switch. R2 designates signaling used in the United States and in countries outside the United States.

multiplexer
A device used to combine a number of individual channels into a single common bit stream for transmission.

multiplexing
A process whereby a transmission facility is divided into two or more channels, either by splitting the frequency band into a number of narrower bands or by dividing the transmission channel into successive time slots. See also time-division multiplexing (TDM).

multirate
The new N x DS0 service (see N x DS0).

MWL
Message-waiting lamp

N

N+1
Method of determining redundant backup requirements. Example: if four rectifier modules are required for a DC-powered single-carrier cabinet, a fifth rectifier module is installed for backup.

N x DS0
N x DS0, equivalently referred to as N x 64 kbps, is an emerging standard for wideband calls separate from H0, H11, and H12 ISDN channels. The emerging N x DS0 ISDN multirate circuit mode bearer service will provide circuit-switched calls with data-rate multiples of 64 kbps up to 1536 kbps on a T1 facility or up to 1920 kbps on an E1 facility. In the switch, N x DS0 channels will range up to 1984 kbps using NFAS E1 interfaces.

NANP
North American Numbering Plan

narrowband
A circuit-switched call at a data rate up to and including 64 kbps. All nonwideband switch calls are considered narrowband.

native terminal support
A predefined terminal type exists in switch software, eliminating the need to alias the terminal (that is, manually map call appearances and feature buttons onto some other natively supported terminal type).

NAU
Network access unit

NCA/TSC
Noncall-associated/temporary-signaling connection

NCOSS
Network Control Operations Support Center

NCSO
National Customer Support Organization

NEC
National Engineering Center

NEMA
National Electrical Manufacturer's Association

NETCON
Network-control circuit pack

network
A series of points, nodes, or stations connected by communications channels.

network-specific facility (NSF)
An information element in an ISDN-PRI message that specifies which public-network service is used. NSF applies only when Call-by-Call Service Selection is used to access a public-network service.

network interface
A common boundary between two systems in an interconnected group of systems.

NFAS
See Nonfacility-associated signaling (NFAS).

NI
Network interface

NID
Network Inward Dialing

NM
Network management

NN
National number

node
A switching or control point for a network. Nodes are either tandem (they receive signals and pass them on) or terminal (they originate or terminate a transmission path).

Nonfacility-associated signaling (NFAS)
A method that allows multiple T1 and/or E1 facilities to share a single D-channel to form an ISDN-PRI. If D-channel backup is not used, one facility is configured with a D-channel, and the other facilities that share the D-channel are configured without D-channels. If D-channel backup is used, two facilities are configured to have D-channels (one D-channel on each facility), and the other facilities that share the D-channels are configured without D-channels.

NPA
Numbering-plan area

NPE
Network processing element

NQC
Number of queued calls

NSE
Night-service extension

NSU
Network sharing unit

null modem cable
Special wiring of an RS-232-C cable such that a computer can talk to another computer (or to a printer) without a modem.

NXX
Public-network office code

O

OA
Operator assisted

occurrence
See appearance.

OCM
Outbound Call Management

offered load
The traffic that would be generated by all the requests for service occurring within a monitored interval, usually one hour.

ONS
On-premises station

OPS
Off-premises station

OPX
Off-premises extension

OQT
Oldest queued time

OSHA
Occupational Safety and Health Act

OSI
Open Systems Interconnect

OSS
Operations Support System

OSSI
Operational Support System Interface

OTDR
Optical time-domain reflectometer

othersplit
The work state that indicates that an agent is currently active on another split's call, or in ACW for another split.

OTL
Originating Test Line

OTQ
Outgoing trunk queuing

outgoing gateway
A PBX that routes an incoming call on a trunk administered for Supplementary Services Protocol B to a trunk *not* administered for Supplementary Services Protocol B.

P

PACCON
Packet control

packet
A group of bits (including a message element, which is the data, and a control information element (IE), which is the header) used in packet switching and transmitted as a discrete unit. In each packet, the message element and control IE are arranged in a specified format. See also **packet bus** and **packet switching**.

packet bus
A wide-bandwidth bus that transmits packets.

packet switching
A data-transmission technique whereby user information is segmented and routed in discrete data envelopes called packets, each with its own appended control information, for routing, sequencing, and error checking. Packet switching allows a channel to be occupied only during the transmission of a packet. On completion of the transmission, the channel is made available for the transfer of other packets. See also BX.25and **packet**.

PAD
Packet assembly/disassembly

paging trunk
A telecommunications channel used to access an amplifier for loudspeaker paging.

party/extension active on call
A party is on the call if he or she is actually connected to the call (in active talk or in held state). An originator of a call is always a party on the call. Alerting parties, busy parties, and tones are not parties on the call.

PBX
Private branch exchange

PC
See personal computer (PC).

PCM
See pulse-code modulation (PCM).

PCOL
Personal central-office line

PCOLG
Personal central-office line group

PCS
Permanent switched calls

PDM
See processor data module (PDM).

PDS
Premises Distribution System

PE
Processing element

PEC
Price element code

PEI
Processor element interchange

personal computer (PC)
A personally controllable microcomputer.

PGATE
Packet gateway

PGN
Partitioned group number

PI
Processor interface

PIB
Processor interface board

pickup group
A group of individuals authorized to answer any call directed to an extension within the group.

PIDB
Product image database

PKTINT
Packet interface

PL
Private line

PLS
Premises Lightwave System

PMS
Property Management System

PN
Port network

PNA
Private network access

POE
Processor occupancy evaluation

POP
Point of presence

port
A data- or voice-transmission access point on a device that is used for communicating with other devices.

port carrier
A carrier in a multicarrier cabinet or a single-carrier cabinet containing port circuit packs, power units, and service circuits. Also called a port cabinet in a single-carrier cabinet.

port network (PN)
A cabinet containing a TDM bus and packet bus to which the following components are connected: port circuit packs, one or two tone-clock circuit packs, a maintenance circuit pack, service circuit packs, and (optionally) up to four expansion interface (EI) circuit packs in DEFINITY ECS. Each PN is controlled either locally or remotely by a switch processing element (SPE). See also expansion port network (EPN) and processor port network (PPN).

port-network connectivity
The interconnection of port networks (PNs), regardless of whether the configuration uses direct or switched connectivity.

PPM
1. Parts per million
2. Periodic pulse metering

PPN
See processor port network (PPN).

PRI
See Primary Rate Interface (PRI).

primary extension
The main extension associated with the physical voice or data terminal.

Primary Rate Interface (PRI)
A standard ISDN frame format that specifies the protocol used between two or more communications systems. PRI runs at 1.544 Mbps and, as used in North America, provides 23 64-kbps B-channels (voice or data) and one 64-kbps D-channel (signaling). The D-channel is the 24th channel of the interface and contains multiplexed signaling information for the other 23 channels.

PRI endpoint (PE)
The wideband switching capability introduces PRI endpoints on switch line-side interfaces. A PRI endpoint consists of one or more contiguous B-channels on a line-side T1 or E1 ISDN PRI facility and has an extension. Endpoint applications have call-control capabilities over PRI endpoints.

principal
A terminal that has its primary extension bridged on one or more other terminals.

principal (user)
A person to whom a telephone is assigned and who has message-center coverage.

private network
A network used exclusively for the telecommunications needs of a particular customer.

private network office code (RNX)
The first three digits of a 7-digit private network number.

processor carrier
See control carrier.

processor data module (PDM)
A device that provides an RS-232C DCE interface for connecting to data terminals, applications processors (APs), and host computers, and provides a DCP interface for connection to a communications system. See also modular processor data module (MPDM).

processor port network (PPN)
A port network controlled by a switch-processing element that is directly connected to that PN's TDM bus and LAN bus. See also port network (PN).

processor port network (PPN) control carrier
A carrier containing the maintenance circuit pack, tone/clock circuit pack, and SPE circuit packs for a processor port network (PPN) and, optionally, port circuit packs.

PROCR
Processor

Property Management System (PMS)
A stand-alone computer used by lodging and health-services organizations for services such as reservations, housekeeping, and billing.

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

PSC
Premises service consultant

PSDN
Packet-switch public data network

PT
Personal terminal

PTC
Positive temperature coefficient

PTT
Postal Telephone and Telegraph

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

pulse-code modulation (PCM)
An extension of pulse-amplitude modulation (PAM) in which carrier-signal pulses modulated by an analog signal, such as speech, are quantized and encoded to a digital, usually binary, format.

Q

QPPCN
Quality Protection Plan Change Notice

quadrant
A group of six contiguous DS0s in fixed locations on an ISDN-PRI facility. Note that this term comes from T1 terminology (one-fourth of a T1), but there are five quadrants on an E1 ISDN-PRI facility (30B + D).

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.

R

RAM
See random-access memory (RAM).

random-access memory (RAM)
A storage arrangement whereby information can be retrieved at a speed independent of the location of the stored information.

RBS
Robbed-bit signaling

RC
Radio controller

RCL
Restricted call list

read-only memory (ROM)
A storage arrangement primarily for information-retrieval applications.

recall dial tone
Tones signalling that the system has completed a function (such as holding a call) and is ready to accept dialing.

redirection criteria
Information administered for each voice terminal's coverage path that determines when an incoming call is redirected to coverage.

Redirection on No Answer
An optional feature that redirects an unanswered ringing ACD call after an administered number of rings. The call is then redirected back to the agent.

release
To release a call is to initiate its disconnection.

release signal
The signal one switch sends to another to disconnect a call. If the calling switch ends the call, it sends a "forward" release signal. If the receiving switch ends the call, it sends a "backward" release signal.

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 (ARS) feature. Calls can be allowed or denied based on the area code and the dialed CO code rather than just the area code. If the call is allowed, the ARS pattern used for the call is determined by these six digits.

Remote Operations Service Element (ROSE)
A CCITT and ISO standard that defines a notation and services that support interactions between the various entities that make up a distributed application.

REN
Ringer equivalency number

reorder tone
A tone to signal that at least one of the facilities, such as a trunk or a digit transmitter, needed for the call was not available.

report scheduler
Software that is used in conjunction with the system printer to schedule the days of the week and time of day that the desired reports are to be printed.

RFP
Request for proposal

RHNPA
See remote home numbering-plan area code (RHNPA).

RINL
Remote indirect neighbor link

RISC
Reduced-instruction-set computer

RLT
Release-link trunk

RMATS
Remote Maintenance, Administration, and Traffic System

RNX
Route-number index (private network office code)

ROM
See read-only memory (ROM).

ROSE
See Remote Operations Service Element (ROSE).

RPN
Routing-plan number

RS-232C
A physical interface specified by the Electronic Industries Association (EIA). RS-232C transmits and receives asynchronous data at speeds of up to 19.2 kbps over cable distances of up to 50 feet.

RS-449
Recommended Standard 449

RSC
Regional Support Center

S

S1
The first logical signalling channel of DCP. The channel is used to provide signaling information for DCP's I1 channel.

S2
The second logical signaling channel of DCP. The channel is used to provide signaling information for DCP's I2 channel.

SABM
Set Asynchronous Balance Mode

SAC
Send All Calls

SAKI
See sanity and control interface (SAKI).

sanity and control interface (SAKI)
A custom VLSI microchip located on each port circuit pack. The SAKI provides address recognition, buffering, and synchronization between the angel and the five control time slots that make up the control channel. The SAKI also scans and collects status information for the angel on its port circuit pack and, when polled, transmits this information to the archangel.

SAT
System access terminal

SBA
Simulated bridged appearance

SCC
1. See single-carrier cabinet.
2. Serial communications controller

SCD
Switch-control driver

SCI
Switch communications interface

SCO
System control office

SCOTCH
Switch Conferencing for TDM Bus in Concentration Highway

SCSI
See small computer system interface (SCSI).

SDDN
Software-Defined Data Network

SDI
Switched Digital International

SDLC
Synchronous data-link control

SDN
Software-defined network

SFRL
Single-frequency return loss

SID
Station-identification number

simplex system
A system that has no redundant hardware.

simulated bridged appearance
The same as a temporary bridged appearance; allows the terminal user (usually the principal) to bridge onto a call that had been answered by another party on his or her behalf.

single-carrier cabinet
A combined cabinet and carrier unit that contains one carrier. See also Multicarrier cabinet.

single-line voice terminal
A voice terminal served by a single-line tip and ring circuit (models 500, 2500, 7101A, 7103A).

SIT
Special-information tones

SLS
Service Level Supervisor

small computer system interface (SCSI)
An ANSI bus standard that provides a high-level command interface between host computers and peripheral devices.

SMDR
Station Message Detail Recording

SN
Switch Node

SNA
Systems Network Architecture

SNC
Switch Node Clock

SNI
Switch Node Interface

SNMP
Simple Network Management Protocol

software
A set of computer programs that perform one or more tasks.

SPE
Switch Processing Element

SPID
Service Profile Identifier

split
See ACD work mode.

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

split number
The split's identity to the switch and BCMS.

split report
A report that provides historical traffic information for internally measured splits.

split (agent) status report
A report that provides real-time status and measurement data for internally measured agents and the split to which they are assigned.

SSI
Standard serial interface

SSM
Single-site management

SSV
Station service

ST3
Stratum 3 clock board

staffed
Indicates that an agent position is logged in. A staffed agent functions in one of four work modes: Auto-In, Manual-In, ACW, or AUX-Work.

STARLAN
Star-Based Local Area Network

Station Message Detail Recording (SMDR)
An obsolete term now called CDR — a switch feature that uses software and hardware to record call data. See Call Detail Recording (CDR).

standard serial interface (SSI)
A communications protocol developed for use with 500-type business communications terminals (BCTs) and 400-series printers.

status lamp
A green light that shows the status of a call appearance or a feature button by the state of the light (lit, flashing, fluttering, broken flutter, or unlit).

stroke counts
A method used by ACD agents to record up to nine customer-defined events per call when CMS is active.

SVN
Security-violation notification

switch
Any kind of telephone switching system. See also communications system.

switchhook
The buttons located under the receiver on a voice terminal.

switch-node (SN) carrier
A carrier containing a single switch node, power units, and, optionally, one or two DS1 converter circuit packs. An SN carrier is located in a center-stage switch.

switch-node (SN) clock
The circuit pack in an SN carrier that provides clock and maintenance alarm functions and environmental monitors.

switch-node interface (SNI)
The basic building block of a switch node. An SNI circuit pack controls the routing of circuit, packet, and control messages.

switch-node link (SNL)
The hardware that provides a bridge between two or more switch nodes. The SNL consists of the two SNI circuit packs residing on the switch nodes and the hardware connecting the SNIs. This hardware can include lightwave transceivers that convert the SNI's electrical signals to light signals, the copper wire that connects the SNIs to the lightwave transceivers, a full-duplex fiber-optic cable, DS1 converter circuit cards and DS1 facilities if a company does not have rights to lay cable, and appropriate connectors.

switch-processing element (SPE)
A complex of circuit packs (processor, memory, disk controller, and bus-interface cards) mounted in a PPN control carrier. The SPE serves as the control element for that PPN and, optionally, for one or more EPNs.

SXS
Step-by-step

synchronous data transmission
A method of sending data in which discrete signal elements are sent at a fixed and continuous rate and specified times. See also association.

SYSAM
System Access and Administration

system administrator
The person who maintains overall customer responsibility for system administration. Generally, all administration functions are performed from the Management Terminal. The switch requires a special login, referred to as the system administrator login, to gain access to system-administration capabilities.

system printer
An optional printer that may be used to print scheduled reports via the report scheduler.

system report
A report that provides historical traffic information for internally measured splits.

system-status report
A report that provides real-time status information for internally measured splits.

system manager
A person responsible for specifying and administering features and services for a system.

system reload
A process that allows stored data to be written from a tape into the system memory (normally after a power outage).

T

T1
A digital transmission standard that in North America carries traffic at the DS1 rate of 1.544 Mbps. A T1 facility is divided into 24 channels (DS0s) of 64 kbps. These 24 channels, with an overall digital rate of 1.536 Mbps, and an 8-kbps framing and synchronization channel make up the 1.544-Mbps transmission. When a D-channel is present, it occupies channel 24. T1 facilities are also used in Japan and some Middle-Eastern countries.

TAAS
Trunk Answer from Any Station

TABS
Telemetry asynchronous block serial

TAC
Trunk-access code

tandem switch
A switch within an electronic tandem network (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 (TTTN)
A private network that interconnects several customer switching systems.

TC
Technical consultant

TCM
Traveling class mark

TDM
See time-division multiplexing (TDM).

TDR
Time-of-day routing

TEG
Terminating extension group

terminal
A device that sends and receives data within a system. See also administration terminal.

tie trunk
A telecommunications channel that directly connects two private switching systems.

time-division multiplex (TDM) bus
A bus that is time-shared regularly by preallocating short time slots to each transmitter. In a PBX, all port circuits are connected to the TDM bus, permitting any port to send a signal to any other port.

time-division multiplexing (TDM)
Multiplexing that divides a transmission channel into successive time slots. See also multiplexing.

time interval
The period of time, either one hour or one-half hour, that BCMS measurements are collected for a reports.

time slice
See **time interval**.

time slot
64 kbps of digital information structured as eight bits every 125 microseconds. In the switch, a time slot refers to either a DS0 on a T1 or E1 facility or a 64-kbps unit on the TDM bus or fiber connection between port networks.

time slot sequence integrity
The situation whereby the N octets of a wideband call that are transmitted in one T1 or E1 frame arrive at the output in the same order that they were introduced.

to control
An application can invoke *Third Party Call Control* capabilities using either an adjunct-control or domain-control association.

to monitor
An application can receive *event reports* on an active-notification, adjunct-control, or domain-control association.

TOD
Time of day

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

TOP
Task-oriented protocol

trunk
A dedicated telecommunications channel between two communications systems or COs.

trunk allocation

The manner in which trunks are selected to form wideband channels.

trunk-data module
A device that connects off-premises private-line trunk facilities and DEFINITY ECS. The trunk-data module converts between the RS-232C and the DCP, and can connect to DDD modems as the DCP member of a modem pool.

trunk group
Telecommunications channels assigned as a group for certain functions that can be used interchangeably between two communications systems or COs.

TSC
Technical Service Center

TTI
Terminal translation initialization

TTR
Touch-tone receiver

TTT
Terminating trunk transmission

TTTN
See tandem tie-trunk network (TTTN).

TTY
Teletypewriter

U

UAP
Usage-allocation plan

UART
Universal asynchronous transmitter

UCD
Uniform call distribution

UCL
Unrestricted call list

UDP
See Uniform Dial Plan (UDP).

UL
Underwriter Laboratories

UM
User manager

Uniform Dial Plan (UDP)
A feature that allows a unique 4- or 5-digit number assignment for each terminal in a multiswitch configuration such as a DCS or main-satellite-tributary system.

UNMA
Unified Network Management Architecture

UNP
Uniform numbering plan

UPS
Uninterruptible power supply

USOP
User service-order profile

UUCP
UNIX-to-UNIX Communications Protocol

UUI
User-to-user information

V

VAR
Value-added reseller

VDN
See vector directory number (VDN).

vector directory number (VDN)
An extension that provides access to the Vectoring feature on the switch. Vectoring allows a customer to specify the treatment of incoming calls based on the dialed number.

vector-controlled split
A hunt group or ACD split administered with the vector field enabled. Access to such a split is possible only by dialing a VDN extension.

VIS
Voice Information System

VLSI
Very-large-scale integration

VM
Voltmeter

VNI
Virtual nodepoint identifier

VOA
VDN of origin announcement

voice terminal
A single-line or multiappearance telephone.

W

WATS
See Wide Area Telecommunications Service (WATS).

WCC
World-Class Core

WCR
World-Class Routing

WCTD
World-Class Tone Detection

WFB
Wireless fixed base

Wide Area Telecommunications Service (WATS)
A service in the United States that allows calls to certain areas for a flat-rate charge based on expected usage.

wideband
A circuit-switched call at a data rate greater than 64 kbps. A circuit-switched call on a single T1 or E1 facility with a bandwidth between 128 and 1536 (T1) or 1984 (E1) kbps in multiples of 64 kbps. H0, H11, H12, and N x DS0 calls are wideband.

wideband access endpoint
Access endpoints, extended with wideband switching to include wideband access endpoints. A wideband access endpoint consists of one or more contiguous DS0s on a line-side T1 or E1 facility and has an extension. The Administered Connections feature provides call control for calls originating from wideband access endpoints.

wink-start tie trunk
A trunk with which, after making 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.

work mode
One of four states (Auto-In, Manual-In, ACW, AUX-Work) that an ACD agent can be in. Upon logging in, an agent enters AUX-Work mode. To become available to receive ACD calls, the agent enters Auto-In or Manual-In mode. To do work associated with a completed ACD call, an agent enters ACW mode.

work state
An ACD agent may be a member of up to three different splits. Each ACD agent continuously exhibits a work state for every split of which it is a member. Valid work states are Avail, Unstaffed, AUX-Work, ACW, ACD (answering an ACD call), ExtIn, ExtOut, and OtherSpl. An agent's work state for a particular split may change for a variety of reasons (example: when a call is answered or abandoned, or the agent changes work modes). The BCMS feature monitors work states and uses this information to provide BCMS reports.

write operation
The process of putting information onto a storage medium, such as a hard disk.

WSA
Waiting session accept

WSS
Wireless Subscriber System

Z

ZCS
Zero Code Suppression