

Lucent Technologies
Bell Labs Innovations



What's New in DEFINITY[®]
Enterprise Communications Server
Release 8.2

Job Aid

555-233-754
Comcode 108678806
Issue 1
April 2000

Notice

Every effort was made to ensure that the information in this book was complete and accurate at the time of printing. However, information is subject to change.

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Part 68: Network Registration Number. This equipment is registered with the FCC in accordance with Part 68 of the FCC Rules. It is identified by FCC registration number AS593M-13283-MF-E.

Part 68: Answer-Supervision Signaling. Allowing this equipment to be operated in a manner that does not provide proper answer-supervision signaling is in violation of Part 68 Rules. This equipment returns answer-supervision signals to the public switched network when:

- Answered by the called station
- Answered by the attendant
- Routed to a recorded announcement that can be administered by the CPE user

This equipment returns answer-supervision signals on all DID calls forwarded back to the public switched telephone network. Permissible exceptions are:

- A call is unanswered
- A busy tone is received
- A reorder tone is received

Canadian Department of Communications (DOC) Interference Information

This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the radio interference regulations of the Canadian Department of Communications.

Le Présent Appareil Numérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils numériques de la class A prescrites dans le reglement sur le brouillage radioélectrique édicté par le ministère des Communications du Canada.

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European Union Declaration of Conformity

The "CE" mark affixed to the DEFINITY® equipment described in this book indicates that the equipment conforms to the following European Union (EU) Directives:

- Electromagnetic Compatibility (89/336/EEC)
- Low Voltage (73/23/EEC)
- Telecommunications Terminal Equipment (TTE) i-CTR3 BRI and i-CTR4 PRI

For more information on standards compliance, contact your local distributor.

Comments

To comment on this document, return the comment card at the front of the document.

Acknowledgment

This document was prepared by Product Documentation Development, Lucent Technologies, Denver, CO.

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New Switch and Adjunct Support

1

About this Document

This document introduces new and enhanced features and capabilities of DEFINITY Enterprise Communications Server (ECS) Release 8. It does not contain comprehensive instructions for switch administration or complete feature descriptions, nor does it contain information about how to install, maintain, repair, or troubleshoot the switch.

It is intended for DEFINITY system administrators and managers, users interested in information about specific features, and Lucent Technologies personnel responsible for planning, designing, configuring, selling, and supporting the system.

This document assumes that you are familiar with DEFINITY ECS in a recent release of DEFINITY ECS software.

Overview

DEFINITY[®] Enterprise Communications Server Release 8

- DEFINITY[®] ECS Release 8 is the next step in the evolution of the DEFINITY[®] ECS.
- It is a multi-purpose platform developed to support a number of offers.
- It builds upon existing capabilities of the DEFINITY[®] ECS standard and special development releases to implement cost reductions for Lucent Technologies, and a new set of advanced capabilities for the global communications marketplace.
- Release 8 introduces these major features:
 - Completion of C-LAN
 - QSIG CAS enhancements
 - ATM enhancements
 - ASAI enhancements
 - Call center enhancements
 - IP Solutions
 - Centralized Voice Mail via Mode Code
 - Attendant Vectoring

Upgrades

Paths

R8 is an upgrade path for existing DEFINITY[®] system customers.

- From pre-R5 to R8
- From R5, R6, and R7 to R8 (through R5.5 for si customers with critical reliability configurations)
- Between specific R8 models



NOTE:

There are no upgrade paths for:

- Non-csi models to csi models
- Vs models to R8

Release 8.1 Feature List

CCRON Renewal
De-admin Login
Abort Xfer
13 Digit Auth Codes
X-station Mobility
DECT Support
Message Waiting Enhancement
6400 Tip/Ring Data Module
QSIG CISC Enhancements
■ Status Signaling Group
QSIG VALU Enhancements
■ Call Coverage
■ Distinctive Alerting
■ Call Coverage and CAS
Feature + non DID via UDP
BCS/Guestworks Enhancement
■ Auto digit rotation for DID
■ Crisis Alert to Pager
■ Suite Check in Via Hunt To Feature
ISDN Bellcore call-ID
E&M Signaling Enhancements
■ Brazil Contin. E&M Signal
■ Brazil Pulsed E&M Signal
■ Hungary Signal
ATM Network Duplication
6200 Analog Native Support
GTS Restricted Presentation
Australian CIN Board Support
Italy Multi Pub Net Calling – Connect Number/System
Pass AOC to BRI Endpoints
China Special Dial Tone

China Outgoing Call No Answer
China Time Super. and Force Release
Auto Exclusion by COS
INS I500-AOC
Fast Analog Modem Support Testing
Enhanced Terminal Parameters
G3R EMC Compliance
TN746B Buffer FW Changes
Advocate Enhancements <ul style="list-style-type: none">■ Logged-in Agent Counting■ Standard Reports
ASAI/CTI Enhancements <ul style="list-style-type: none">■ CTI Regression Testing■ Pending Work Mode Changes■ Switch SW Version Query■ Trunk Group ID in Event Reports■ Cap Increase - Active Route Requirements■ Cap Increase - Split Skill Dom Ctrl■ UUI>32byte
Interworking Bandconstr (ATM)
ACD/ATM Capacity Increases <ul style="list-style-type: none">■ Hunt Group Members■ Hunt Groups■ Measure VDNs■ QueueS1ots■ Vectors■ 1000 Announcements
CMS ATM Trk Measurements
Site Stats Remote EPN
Call Master V Native Support
CMS High Avail. Support
Call Center Release Control
TN802B (Medpro)

TN799B
<ul style="list-style-type: none"> ■ Variable Length Pin
<ul style="list-style-type: none"> ■ TraceRoute
<ul style="list-style-type: none"> ■ Packet Error History
<ul style="list-style-type: none"> ■ SNMP Re-arch
<ul style="list-style-type: none"> ■ Host Route Admin Chan Cs
TN2211 Optical Drive
<ul style="list-style-type: none"> ■ UN332C MSSNET

Release 8.2 Feature List

IP Solutions
<ul style="list-style-type: none"> ■ IP SoftPhones
<ul style="list-style-type: none"> ■ H.323 Trunks
ATM PNC Reliability
QSIG Integration: Transfer to Audix
Attendant Vectoring (formerly CAS)
QSIG CAS & VALU Coverage Interaction Support
QSIG CAS Enhancement
<ul style="list-style-type: none"> ■ Attendant Display of COR
<ul style="list-style-type: none"> ■ Attendant Return of Call
<ul style="list-style-type: none"> ■ Display Enhancements
<ul style="list-style-type: none"> ■ Priority Queue
<ul style="list-style-type: none"> ■ RLT Emulation via PRI
Support for Japan National Private Networking
<ul style="list-style-type: none"> ■ 2MB Trunk — Ph 2: Codeset 5 TCMs
<ul style="list-style-type: none"> ■ 2MB Trunk — Ph 2: Connect Ack
<ul style="list-style-type: none"> ■ 2MB Trunk — Ph 2: Different Protocol Discriminator
<ul style="list-style-type: none"> ■ D Channel — Private ISDN Networks — Q.931.a, Q.932.a
<ul style="list-style-type: none"> ■ D Channel — Private ISDN Networks — 951.a
Administrable Loss Plan
24 Port Analog Line with Caller ID (TN793/TN2793B)
U.S. Analog Trunk & Line Board (TN797) without Busy Tone Detection

Centralized Voice Mail via Interswitch Mode Code
Trouble Isolation: List Trace Command
TN2313
T1 401A Sync Splitter
E1 402A Sync Splitter
E1 403A Sync Splitter

Platform - Enhancements/Renewal

DADMIN LOGIN

- Administration of the dadmin login uses the standard DEFINITY® login and permission form interfaces.
- Dadmin Login replaces Inads Login for Distributors.
- The dadmin login ID, restricts access to commands that are limited to Lucent Services.
- The dadmin login service level is located between the inads and craft logins service levels in the system administration/maintenance login permission hierarchy.

Restart Strategy for Dadmin Login

- Login administration parameters are retained for the dadmin login on recovery level 1 (warm) and recovery level 2 (cold) restarts and require retrieval from a translation storage device for all other restart levels.

Number of Logins

Platform	Maximum Number of Logins *	Login Distribution
R8(vs/si/csi)	16	<ul style="list-style-type: none">■ 4 Lucent Services Logins■ 1 Dadmin Login■ 11 Customer Logins
R8(r)	25	<ul style="list-style-type: none">■ 4 Lucent Services Logins■ 1 Dadmin Login■ 20 Customer Logins

*The maximum number of logins does not include the "mis" login which is used internally by the system to establish connections to call management systems, CMS.

Feature to Feature Interactions

Access Security Gateway (ASG)

- ASG authentication not required by default for the dadmin login.
- ASG field on the dadmin login form defaults to n(o) and all fields that appear on the Access Security Gateway page of the login form are unassigned or set to system defaults.
- Secret Key field is not defaulted for the dadmin login.

Constraints and Limitations

- The number of logins supported for various system configurations is increased by one to accommodate the addition of the "dadmin" login.

Customer Options

- Administration of Version and Location fields remain under "init" control on the System Parameters Customer Options forms.

Offer Categorization

- Administration remains under the control of the init login.

Enable/Disable Login

- The system supports execution of the enable and disable login commands for the dadmin login.
- Permission to execute these commands for the dadmin login is restricted to the init and dadmin logins, and to the inads login, if it has been administered to have "Administer Permissions" permissions.
- It is possible for the dadmin login to disable itself by executing the disable login dadmin command.
- Execution of the command will not terminate the current login session, however, subsequent attempts to log in to the system using the dadmin login will be denied.
- Re-enabling the dadmin login must be performed by the init login or the inads login, if it has been administered to have "Administer Permissions".

Optical Drive (TN2211)

- The TN2211 Optical Drive is a direct replacement for the TN1656 Tape Drive, and offers the following:
- The TN2211 provides removable storage for software upgrades, translation backups, announcement file backups, and core dumps

- The TN2211 Optical Drive resolves several problems related to the tape drive such as availability, cost, performance, reliability, and storage capacity.
- The TN2211 is designed to be less expensive and more reliable than the TN1656.
- Installation and Upgrade procedures for the TN2211 optical drive are nearly the same as for the TN1656 tape drive.
- The tape commands are replaced by removable-media commands.
 - test tape becomes test removable-media, or test rem for the shorthand version.
- An optical drive is faster than a tape drive, but it is not related to call flow.
 - A full backup to an optical disk cartridge takes approximately 20 minutes, compared to about 95 minutes for a tape.

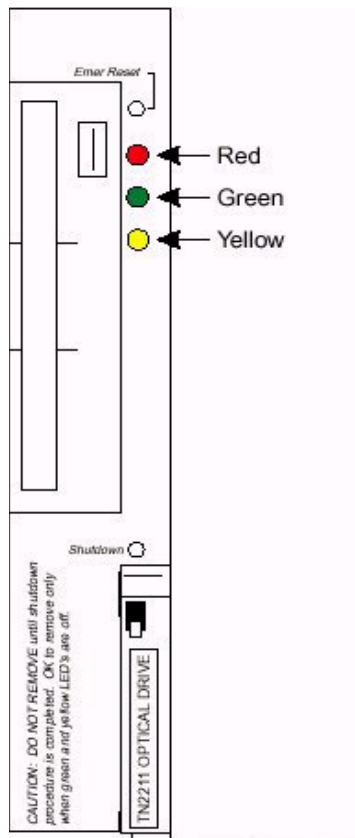


Figure 1. Optical Drive (TN2211)

Configuration

- The configuration is a G3r switch (simplex or duplex) with a TN2211 Optical Drive and a UN332C MSSNET OR a TN1656 Tape Drive and a UN332B MSSNET.
- All commands using tape are replaced by removable-media.
- Commands such as test tape, busy tape, release tape, etc. are replaced by removable-media.
 - The commands are test removable-media, busy removable-media, etc. removable-media can be shortened to rem (test rem, busy rem).
- If a user enters tape instead of removable-media, the error/help message "tape is no longer valid; use removable-media" is displayed on the SAT.
- All instances of tape on output displays are replaced with removable-media or R-MEDIA as appropriate.
- Both the tape drive and optical drive are supported by R8 software.
- G3r switches running R8 software are shipped with the TN2211 Optical Drive and the UN332C MSSNET
 - New switches are shipped with the TN1656 Tape Drive and hardware
- The UN332C MSSNET must be used with the TN2211 Optical Drive.



NOTE:

UN332C MSSNET is backward-compatible with the UN332B. UN332C will work with a tape drive.

Serviceability

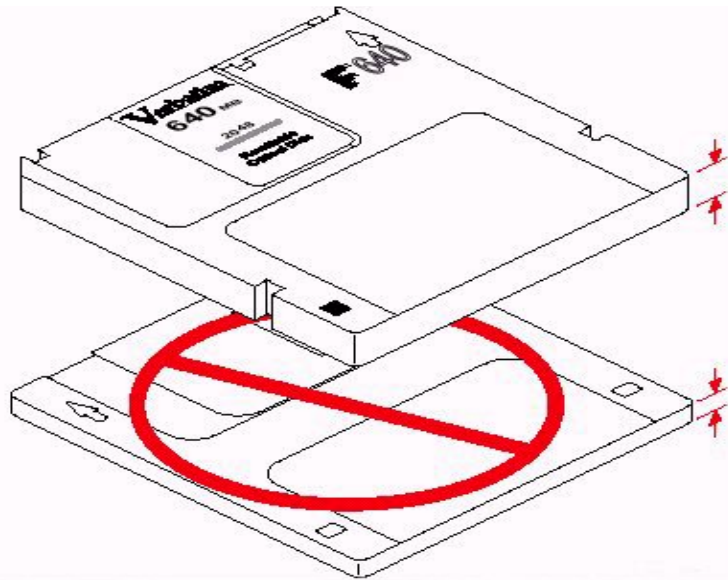
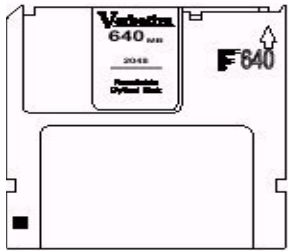
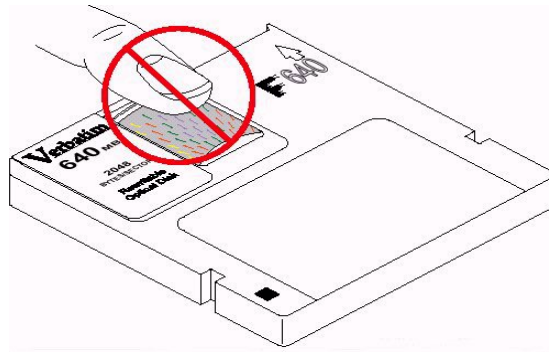
- A (remote) maintenance user can identify between the two drive types by using the list config command to determine the TN code.
- The tape commands are replaced with removable-media commands (rem for short).
- A user can verify the correct suffix of the UN332C by reading the circuit pack label or reading the output from the list configuration control command.

Upgrades

- Typically, a defective tape drive will be replaced with another tape drive for existing systems.
- Customers are not required to upgrade the tape drive to an optical drive for bugfix.
- For "purchased" upgrades to R8, the customer will be required to upgrade to the TN2211 optical drive and UN332C MSSNET.

- The procedure to both upgrade from pre-R8 to R8 and to change from a TN1656 to a TN2211 at the same time is not supported.
- Services must upgrade the hardware first (TN1656 to a TN2211 and a UN332B to a UN332C), then upgrade to the R8 software.
 - The hardware must be upgraded prior to the software.
 - This will take advantage of the faster optical drive technology for some of the upgrade steps.
 - This does not require that the new software be shipped on two different media forms (tape and optical disk cartridge).
- Replacement of the tape drive with the optical drive does not impact the call preserving nature of the upgrade.

Optical Drive Cartridge



Maintenance



NOTE:

Refer to Definity ECS Release 8.2 Upgrades and Additions for R8r, 555-233-115, Issue 1.

- A TN2211 is maintained in software as if it were a TN1656.
- After inserting an optical disk cartridge, the access lamp of the drive should be on for a few seconds, then should turn off.
 - Access lamp blinking afterwards typically means a cartridge should be reinserted.

Maintenance Object appears as "R-MEDIA"

- The Maintenance Object appears as R-MEDIA for either the TN1656 or the TN2211.
- This Maintenance Object is used in the output of commands such as display alarms, display errors, test removable-media, etc., and functions as follows:
 - If an error or alarm log contains an entry for TAPE, the description changes to R-MEDIA after the software has been upgraded to R8.
- The list config command output describes the TN1656 and the TN2211 as R-MEDIA DRIVE.
- Both the tape and optical drives are now generically described as R-MEDIA DRIVE.
 - The only way to differentiate between the two drives in R8 is to look at the TN code via the list config command.

No limit on the number of accesses

- The software is not limited the number of accesses of an optical disk cartridge.



NOTE:

Important! The TN2211 Optical Drive does NOT require cleaning.

Any attempt to clean the optical drive will damage it.

- The LMM will display TAPE on boot-up on the SAT, even if an optical drive and optical disk cartridge are present.
- In duplicated systems, either carrier may have an optical drive or a tape drive, since software supports both.
- The drives can be mixed between carriers (one carrier has a TN1656, the other has a TN2211).

U.S. Analog Trunk & Line Circuit Pack TN797

Description

This circuit pack is the combination 8-port Analog Trunk and Line Circuit Pack, TN797, for the US, Canada, and like countries. This provides the user with the capability to administer any of the 8 ports of this analog circuit pack as a:

- Central Office trunk, either loop start or ground start,
- CAMA E911 trunk,
- Direct Inward Dialing trunk, either wink start or immediate start; or
- An analog line, on or off-premises, with or without LED message waiting indication.

One way of visualizing the capabilities of this circuit pack is to think that when a port is administered as a CO trunk, its capabilities will be similar to those of a port of the analog CO trunk TN747B (or the TN429D used for CAMA/E911); if the port is administered as a DID trunk, its capabilities will be similar to those of the DID trunk TN753B, and if administered as a line circuit, its capabilities will be similar to those of the analog line TN746B.

Note that the TN797 circuit pack does not support ICLID on the analog trunk to the CO, nor Caller ID on the line side to a terminal. Additional capabilities, including additional international features, are being planned for a future release of this circuit pack.

One interesting application of this circuit pack is to provide Emergency Transfer capabilities to a system which has only DS1 connectivity to the central office, or to one that has only digital terminals. This circuit pack will permit analog connectivity to the central office from analog terminals using only one port slot.

There are no changes required in Call Processing for this circuit pack.

The TN797 circuit pack supports all analog terminals supported by the similar analog circuit packs mentioned above. However, neon lamp message waiting indication is not supported by this circuit pack.

TN2313 DS1 Interface (US (24-Channel) or International (32-Channel))

The TN2313 DS1 port board interfaces a DS1 trunk to the switch backplane via port slots that are standard for DEFINITY products. The TN2313 is compatible with previous DS1 circuit packs, including the TN464F (V19 and below), the TN2464 (V19 and below), and the TN767E DS1, except that it does not provide for packet adjunct capabilities. The TN2313 supports a variety of applications, including networking of DEFINITY switches, international trunk types, video teleconferencing, and wideband data transmission.

The TN2313 DS1 interface can be configured for domestic (24-channel, 1.544 Mbps) or international (32-channel, 2.048 Mbps) use. The TN2313 can supply two 8-Khz reference signals to the switch backplane for optional use by the tone/clock board in synchronizing the system clock to the received line clock.

24 Port Analog Line with Caller ID: TN793B & TN2793B

The TN793/TN793B is a dual coded, analog line 24-port, that performs all the functions of the TN746B, 16-port analog line circuit pack. Each port supports 1 voice terminal, such as 500 (rotary dial) and 2500 terminals (DTMF dial).

The TN793/TN793B supports on-premises (in-building) wiring with either touch-tone or rotary dialing and with or without the LED and neon Message Waiting Indicators. The TN793/TN793B supports off-premises wiring (out-of-building only with certified protection equipment) with either DTMF or rotary dialing, but LED or neon message waiting indicators are not supported off -premises.

The TN793/TN793B, along with a TN755B neon power unit per carrier or per single-carrier cabinet, supports voice terminals equipped with neon message waiting indicators (on-premises use only). The TN793/TN793B supports 3 ringer loads, only 1 voice terminal can have an LED or neon message waiting indicator. The TN793/TN793B allows a maximum of 12 simultaneous ports ringing.

The TN793/TN793B supports A-Law and μ -law companding and administrable timers. The TN793 supports queue warning level lights associated with the DDC and UCD features, recorded announcements associated with the Intercept Treatment feature, and PagePac paging system for the Loudspeaker Paging feature. Additional support is provided for external alerting devices associated with the TAAS feature, neon message waiting indicators, and modems The TN793/TN793B also supports secondary lightning protection.

The TN793B is compatible with the terminal configurations shown in the table below.

Table 1. Terminals and wiring configurations supported by the TN793

Terminal	Wire Size	Maximum Range
500-Type	24 AWG (0.2 mm ² /0.5 mm)	20,000 ft (6,096 m)
2500-Type	24 AWG (0.2 mm ² /0.5 mm)	20,000 ft (6,096 m)
6200-Type	24 AWG (0.2 mm ² /0.5 mm)	12,000 ft. (3,657m)
7100-Series	24 AWG (0.2 mm ² /0.5 mm)	20,000 ft. (6,096 m)
8100-Series	24 AWG (0.2 mm ² /0.5 mm)	12,000 ft. (3,657m)
9100-Series	24 AWG (0.2 mm ² /0.5 mm)	12,000 ft. (3,657m)

General Telephony Enhancements - Global

Coverage of Calls Redirected Off-Net (CCRON)

Release 8 Enhancements

- Remote SAC Activation/Deactivation Call Processing
- Threshold Activated CF Timer Call Processing
- Coverage After Forward (per station) Call Processing
- ISDN Renewal(long term)
- Maintain SBA at Principal Call Processing

Remote Send All Calls(SAC) Activation/ Deactivation

- The Remote Send All Calls Activation/Deactivation feature allows a remote user to activate or deactivate send all calls.

Remote SAC Activation/Deactivation

- New Feature Access Codes
- Added to the Feature Access Codes (FAC) form.
 - These new feature access codes, followed by the user's extension, "#," associated Station Security Code (SSC) and "#," are used to activate/deactivate Send All Calls (SAC) for the specified extension from any on-site extension or remotely (offnet).
 - Activating/Deactivating SAC Via The Remote SAC Feature Access Code Or Existing Method Activating or deactivating SAC via either the Remote SAC feature access codes or existing SAC button or feature access codes have the same effect.
 - SAC may be activated by using a local SAC FAC, SAC feature button, or by activating Remote SAC.
 - SAC may be inactivated by using a local SAC FAC, SAC feature button, or by activating Remote SAC.

Remote SAC Activation/Deactivation Form

- Added New "Remote Send All Calls Activation:" and "Deactivation:" Fields on the FEATURE ACCESS CODE (FAC) Form
 - FACs allow a remote user to activate or deactivate the Send All Calls feature.
 - Feature access codes may be 1 to 4 digits; * and # may be used as the first digit only. The defaults are blank.

Feature Access Code (FAC) Form (page 3 of x)

```
change feature-access-codes                                     Page 3 of X
FEATURE ACCESS CODE (FAC)
    Program Access Code: ____
    Refresh Terminal Parameters Access Code: ____
    Remote Send All Calls Activation: ____ Deactivation: ____
    Send All Calls Activation: ____ Deactivation: ____
    Station Security Code Change Access Code: ____
    Terminal Dial-up Test Access Code: ____
    Terminal Translation Initialization Merge Code: ____ Separation Code: ____
    Transfer to AUDIX Access Code: ____
    Trunk Answer Any Station Access Code: ____
    User Control Restrict Activation: ____ Deactivation: ____
    Voice Coverage Message Retrieval Access Code: ____
    Voice Principal Message Retrieval Access Code: ____
    Whisper Page Activation Access Code: ____
```

'#' as Abbreviation for Extension of Currently Active Appearance

- For either of the new feature access codes, a '#' dialed directly after the feature access code will signify that the extension of the appearance being dialed from is the extension being administered, and the extension need not be dialed; i.e., the initial '#' is followed by the Station Security Code, '#'.

Remote User

- Access this feature by dialing the telecommuting access extension telephone number directly or by being transferred to it.
 - Alternatively, the remote user can dial the remote access extension.
- On receipt of system dial tone the user would enter the appropriate FAC (either activate or deactivate SAC).

- The user then dials the station extension number, followed by a “#” and station security code (SSC), followed by a “#.”

Local User

- The Remote SAC Activation/Deactivation feature can also be activated by a local user from any local extension.
- The local user would access this feature on receipt of dial tone by entering the appropriate FAC (either activate or deactivate SAC) and receive recall dial tone.
- The user then dials the station extension number, followed by a “#” and station security code, followed by a “#.”
- If the user is dialing from the extension which is to be administered, the user may enter just the ‘#’ followed by the SSC, followed by “#,” this will signify to the system that the extension number of the appearance being dialed from is the extension number being administered.
- Then either confirmation or intercept tone is returned if any of the following conditions exist:
 - The extension number entered is invalid (logged as an invalid attempt).
 - The extension number is not accessible from the user’s partition.
 - The SSC is invalid for the extension entered (logged as an invalid attempt).
 - Interdigit time out has occurred.
- If activation/deactivation is successful, then confirmation tone is returned and SAC is either activated or deactivated at this point.

ISDN Renewal (Interworking Assumption)

When CCRON calls are extended over ISDN facilities, two scenarios are possible.

1. ISDN end-to-end facilities are established

Or

2. ISDN end-to-end facilities are not available; a portion of the connection is not ISDN and ISDN facilities interworking is required for some link between endpoints.
 - ISDN end-to-end was assumed for a period of time, after which interworking was assumed unless a positive indication from the ISDN network signaled otherwise.

Off-Net Coverage Points

- Any point in a principal's coverage path can be administered as an off-net destination.



NOTE:

Time-of-Day Coverage feature increases the number of coverage points from 3 to 6.

Increase Number of Remote Coverage Points

- Number of Remote Coverage Points available (on all platforms) increased from 225 to 999.

Increase Number of Coverage Paths

- Number of coverage paths increased from 600 to 999 on the small mips and from 7500 to 9999 on the large mips.

New System Parameters--Call Coverage / Call Forwarding Form, Page 1

```
change system-parameters coverage-forwarding Page 1 of 2
      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 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? _
```

New System Parameters--Call Coverage / Call Forwarding Form, Page 2 CCRON Enabled

```
change system-parameters coverage-forwarding Page 2 of 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
```

New System Parameters--Call Coverage / Call Forwarding Form, Page 2 CCRON Disabled

```
change system-parameters coverage-forwarding Page 2 of 2
SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)
Coverage Of Calls Redirected Off-Net Enabled? N
```

Threshold Activated Call Forward Timer

- Activates "Threshold for Blocking Off-Net Redirection of Incoming Trunk Calls" field for Release 8.
- Field has a range of 1-7 and "n".
 - "n" will mean that call processing never activates the Call Forward timer
 - An infinite number of calls to a principal may be redirected off-net.
 - The default is 1, which provides the current operation.
 - When a customer is upgraded from a release prior to R8, this field is set to 1.
- All incoming trunk calls to a particular station are blocked from redirecting offnet during the call forward timer interval.
- The R8.1 user can change this value.

Coverage After Forward (Station Option)

- Adds a new field entitled "Coverage After Forwarding (station specific)?" to page 2 of the Station form for all stations.
- The field may contain any of three different values:
 - y - indicates, that for this station, coverage treatment should be applied after forwarding, regardless of the value of the "Coverage After Forwarding?" field on the SYSTEM PARAMETERS - CALL COVERAGE / CALL FORWARDING form
 - n - indicates, that for this station, coverage treatment should not be applied after forwarding., regardless of the value of the "Coverage After Forwarding?" field on the SYSTEM PARAMETERS - CALL COVERAGE / CALL FORWARDING form.
 - default - indicates, that for this station, use the value of the "Coverage After Forwarding?" field on the SYSTEM PARAMETERS - CALL COVERAGE / CALL FORWARDING form.



NOTE:

Application of coverage treatment does not guarantee that the call will definitely redirect to a coverage point, since the coverage criteria as applied at the forwarded-to destination may not be satisfied.

- When a DEFINITY® system with CCRON activated is upgraded to R8, the value “s” (for ‘system’) set in the “Coverage After Forwarding ?” field on the Station form for all stations
- When a customer is upgraded from a release prior to R8, this field is set to “s” for all stations, indicating that call processing uses the value on the Coverage After Forwarding field on the SYSTEM PARAMETERS--CALL COVERAGE / CALL FORWARDING form.
- To override the system-wide parameter for a given station, the system administrator must set the field on the Station form.

Station Form (Page 2 of X)

add station 1014	STATION	Page 2 of X
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? _	
	Special Character for Restricted Number? n	
MWI Served User Type: _____	Display Client Redirection? n	
AUDIX Name: _____	Select Last Used Appearance? n	
Messaging Server Name: _____	Coverage After Forwarding? _	
Recall Rotary Digit? n	Multimedia Early Answer? n	

Maintain SBA at Principal

- Allows customers to prevent the appropriation of a call appearance button on the principal's phone by the simulated bridge appearance by a coverage call.
 - The ability to take phone calls may be more important than the capability of the principal to retrieve calls that have already redirected to coverage.

Option to Dispense with Call Classifier On Coverage Call

- Established a new field, "Maintain SBA At Principal?"
- This is a y/n field.
- The default is y, which maintains current operation.
- When a customer is upgraded from a release prior to R8, this field is set to y.

13-Digit Authorization Codes

- The system now supports authorization codes from a minimum of 4 digits to a maximum of 13 digits in length.
 - The number of authorization codes allowed (90,000 in a G3r, 5,000 in a G3i) (not changed).
- The user may have to dial up to 13 digits to enter an authorization code.
- The use of the authorization code information for ARS, AAR, incoming trunk authorization, remote access, or any other feature that may require the collection of an authorization code has not changed.
- If a standard CDR format is used and there are more than 7 digits in an authorization code, the authorization code is truncated to the high 7 digits to fit the format.
- The customized format supports 13-digit authorization codes, and is encouraged for those with larger authorization codes.

Feature Administration

- Feature Administration allows the length of the authorization code assigned on the system - parameters features form to be from 4 to 13 digits.
- The security violations status form has changed to allow 13-digit authorization codes.
- The Authorization Code form shows 4 columns of 16 authorization codes instead of 6 columns.

OLD SCREEN

```
monitor security-violations authorization-code
```

```
SECURITY VIOLATIONS STATUS
```

```
Date: 17:51 MON SEP 28 1999
```

```
AUTHORIZATION CODE VIOLATIONS
```

Date	Time	Originator	Auth-Cd	TG No	Mbr	Bar-Cd	Ext	CLI/ANI
09/28	17:19	Trunk	9639632	1				1
09/28	17:10	Trunk	8998989	1				1

NEW SCREEN

```
SECURITY VIOLATIONS STATUS
```

```
Date: 17:51 MON SEP 28 1999
```

```
AUTHORIZATION CODE VIOLATIONS
```

Date	Time	Originator	Auth-Cd	TG/Mbr	Bar-Cd	Ext	CLI/ANI
09/28	17:19	Trunk	9639632	123456	1/		1
09/28	17:10	Trunk	8998989	123456	1/		1

```
list authorization-code  
LIST AUTHORIZATION CODES REPORT
```

Authorization Code	Class of Restriction(COR)
0000000000000	0
1111111111111	1
2222222222222	2
3333333333333	3

Call Detail Recording (CDR) Formatting



NOTE:

CDR formats are "standards" therefore no attempt was made to modify them to fit larger authorization codes.

A customized CDR format handles 13-digit authorization codes and allows any format to be created.

- The form for translating authorization codes has changed to fit the new maximum length.

- The new form displays a maximum of 48 codes (3 columns), leaving only 1 column for adding more codes.
- To enter more codes, execute change authorization - code 9999999.... to show a blank screen, allowing 64 codes to be submitted at one time rather than the original 96.

Original screen

```
change authorization-code 1111111 Page 1 of 1
Authorization Code - COR Mapping

      NOTE: 96 codes administered. Use 'list' to display all codes

      AC  COR  AC  COR  AC  COR  AC  COR  AC  COR  AC  COR
1111111 1 2222222 2 3333333 3 4444444 4 5555555 5 6666666 6
1111112 1 2222223 2 3333334 3 4444445 4 5555556 5 6666667 6
1111113 1 2222224 2 3333335 3 4444446 4 5555557 5 6666668 6
. (12 more rows)
.
.
1111125 1 2222236 2 3333347 3 4444458 4 5555569 5 6666680 6
```

New screen

```
change authorization-code 1111111111111 Page 1 of 1
Authorization Code - COR Mapping

      NOTE: 64 codes administered. Use 'list' to display all codes

      AC  COR  AC  COR  AC  COR  AC  COR
1111111111111 1 2222222222222 2 3333333333333 3 4444444444444 4
1111111111112 1 2222222222223 2 3333333333334 3 4444444444445 4
1111111111113 1 2222222222224 2 3333333333335 3 4444444444446 4
. (12 more rows)
.
.
1111111111125 1 2222222222236 2 3333333333347 3 4444444444458 4
```

- If using both authorization codes and account codes, be aware that there is a problem with phone calls that involve more than 36 digits.
 - The length of the account code and authorization code should not have more than 36 digits.

As an example:

- ars access code = 1 digit
- phone number = 15 digits
- account code access code = 3 digits
- This leaves a total of 17 digits for the total of authorization and account code lengths.

- 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**.

Feature-Related System Parameters screen - Page 10

```
change system-parameters features                               Page 10 of 10
                        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
```

- **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 switch.

Valid entries	Usage
---------------	-------

y	Enables automatic exclusion by a class of service.
----------	--

n	Exclusion operates normally.
----------	------------------------------

- **Automatic Exclusion Coverage/Hold**

Appears when 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 the 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 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.
----------	--

- Recall Rotary Digit

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

Valid entries	Usage
---------------	-------

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

64 Bridged Call Appearances

- DEFINITY[®] system bridged appearance limit of 26 expanded to 64.
 - Bridged appearance feature button for a given extension increased to 63, reserving one appearance for the primary set.
- There are limits on the number of sets per system that can be administered for more than 26 bridged appearances per set.
 - R8 csi and si can have up to 50 such sets, while R8r can have up to 250 such sets.

Circular Station Hunting

- Provides a new method of administering a hunt group.
- A new group type, circ (for circular) added to the list of available group types entered on the Hunt Group form.
 - When administering a circular station hunt group, the order in which the extensions for those stations participating in the hunt group are administered is the order in which calls will be directed.
 - System must keep track of the last extension in the hunt group, to which a call was connected, such that when the next incoming call arrives, the system can determine the next idle extension in the circular hunt group.
 - Extensions in the hunt group that are busy are skipped in the algorithm, and the next idle extension within the hunt group is selected regardless of past call history.
- The caller hears a busy tone when all extensions in the hunt group are busy.
 - The capability exists to have a call go to coverage or to the attendant if there are no idle stations available in the hunt group.

Group Call Pick-up

- Work groups in Japan use this feature.
- Allows users to dial a FAC and then a pickup group number, in order to answer a call pickup call from a different call pickup group.

Long Hold Recall – Warning

- When a call stays on hold past an administered time, this feature will give visual or audible warnings to the set that put the call on hold, similar to the warning given to attendants when calls stay in queue too long.

Reset Shift Call

- When a call is made to a busy station, special dial tone is heard (called second dial tone by Japanese PBX manufacturers).
- After special dial tone, a single digit may be entered which replaces the last digit of the originally dialed extension, and the call is offered to the new station.
- If call coverage is supplied for the dialed extension, the call goes to coverage as normal.
- If the coverage extension is busy, the special dial tone is heard and a single digit may be entered.
- Active for station to station calls, but not for incoming calls nor outgoing calls.
- Operation for transfer and conference is similar: if the third party is busy, special dial tone is heard and an opportunity is provided for a last digit re-entry as on a normal call.

Station Self Display

- When a station user goes offhook, the station set will display the extension number of the set itself.
- Once the user starts to dial, the dialed number is displayed.
- People in small work groups in Japan use this feature as they move from desk to desk while working.

Special Dial Tone [China]

- Adds the ability to play a special dial tone whenever an analog set would normally receive dial tone, but there is a feature or condition active at the set that the user needs to be notified.

- Examples of such features or conditions include:
 - Features that can be activated and deactivated via a FAC from the station, and cause some re-routing of calls to that station
 - Features that can be activated and deactivated via a FAC from the station and invoke some restriction of calls from the station
- The special dial tone of 450 Hz, on 400ms, off 50ms, repeating, at -10 +/- 3 dBm0.
- Tone is administered via the country code, and via the current tone administration.

6200 Analog Native Support

Description

- The 6200 Analog Native Support feature provides the user with the capability to administer the 6200 family of analog terminals directly in DEFINITY[®] ECS rather than aliasing them as some other existing terminal.
- The terminal type, CallrID, added to permit the aliasing of various terminals and adjuncts for Caller ID purposes.
- When CallrID (one word limited to seven characters) is designated, this permits the aliasing of analog terminals with Caller ID capabilities, or analog terminals with Caller ID boxes, or adjuncts, capable of displaying calling party information via Bellcore (US, Bahrain V.23) or NTT (Japan) standards.
- Analog terminals are 6210, 6218, and 6220.

Upgrades

- To function with the Caller ID feature, TN793B or TN2793B is required with terminals administered or aliased as CallrID.

Administration

Screen for 6200 Terminals

- The administration screen shows the Page 1 of the STATION administration form where, as an example, station 1000 has been administered as a 6210 terminal.
 - This screen shows the default values that will appear when 6210, 6218 or 6220 value is entered as a terminal type.

Station Form

```
add station next                                     Page 1 of 3
STATION

  Extension: 1000                                     Lock Messages? n
    Type: 6210                                       Security Code:      TN: 1
    Port:                                           Coverage Path 1:   COR: 1
    Name:                                           Coverage Path 2:   COS: 1
                                                    Hunt-to Station:   Tests? y

STATION OPTIONS
  Off Premises Station? n                           Message Waiting Indicator: led
                                                    Message Lamp Ext: 1000
```

- HELP values for the "Type" field will include 6210, 6218 and 6220.

Caller ID Terminals

- A Caller ID terminal is required, in addition to the TN793B or TN2793B circuit pack and Software Release 8 (or later), to implement the Caller ID feature.
- Provides an expedient manner for administering non-Lucent Caller ID terminals when a customer requires the Caller ID feature.
- New fields on the STATION form:
 - Display Caller ID?
 - Caller ID Message Waiting Indication?

STATION Form

```
add station next                                     Page 1 of 3
                                                    STATION

  Extension: 1000                                     Lock Messages? n
    Type: CallrID                                    Security Code:      TN: 1
    Port:                                           Coverage Path 1:   COR: 1
    Name:                                           Coverage Path 2:   COS: 1
                                                    Hunt-to Station:   Tests? y

STATION OPTIONS
  Off Premises Station? n                           Message Waiting Indicator:
                                                    Display Caller ID? y
  Caller ID Message Waiting Indication? n
```

- HELP values for the Type field include CallrID.



NOTE:

CallrID is one word, limited to seven characters by Type field constraints.

- The fields Display Caller ID? y and Caller ID Message Waiting Indication? n appear in this form only when CallrID is chosen as the terminal Type.
- When Display Caller ID field is set to y (yes), calling party information is transmitted to the Caller ID terminal via Bellcore (US, Bahrain V.23) or NTT (Japan) protocol.
 - When not desired, field should be set to n.
- Caller ID Message Waiting Indication? n: When set to n (no), FSK message waiting indication is not transmitted to the Caller ID terminal.
 - If FSK message waiting indication is transmitted to a Caller ID terminal, this field should be set to y.
 - Sending FSK message waiting indication to a Caller ID terminal is independent of the administration of the Message Waiting Indicator field.
 - The choices for these two fields should be selected according to the capabilities of the Caller ID terminal used.

Description of the terminals

6210

- 6210 is a Lucent set with handset volume control, ringer volume control, timed switch-hook disconnect, tone dialing (only), Flash, Set Redial, Set Hold, Data Jack (same T&R pair), and DEFINITY[®] system LED MWI.

6218

- 6218 is a Lucent set that is the same as the 6220, but without Speakerphone and without Mute.

6220

- 6220 is a Lucent set with 6210 features plus Speakerphone, Mute, Repertory Dialing, Repertory Keylock, Set Personalized Ring, System Hold CallrID, and Caller ID analog terminals (analog terminals with Caller ID boxes, or adjuncts) that support Caller ID according to Bellcore, Bellcore V.23, or NTT standards.
- MWI is supported via FSK (Bellcore standard). MWI is also optionally supported via DEFINITY[®] system LED voltage method or via neon lamp.
- TeleMatrix and Teledex Caller ID terminals use a neon lamp MWI in hospitality environments.

⇒ NOTE:
Caller ID feature requires the TN793B or TN2793B circuit pack.



Figure 2. 6210 Analog Telephone



Figure 3. 6220 Analog Telephone

Description of the terminals(continued)

- The 6200 family of analog terminals requires DEFINITY® ECS R8 software.
- If R8.1 is used as a bugfix, access to the CallrID terminal type is blocked.
 - Access is not blocked for the 6200 family of terminals.
- 6200 Analog Native Support is available and supported in both Offer Category A and Offer Category B.
- 6200 family is compatible with existing DEFINITY® system analog line circuit packs.
 - Can be used where the 8110 terminal is used.

Caller ID terminals

- TN793B/TN2793B circuit pack supports the following Caller ID terminals:
 - TeleMatrix 2802 CID Guest Room Telephone
 - Teledex Millennium 2505C and 2510C
 - Commercial sets with Caller ID capabilities that comply with the protocols supported by the Caller ID
 - Caller ID display boxes, or adjuncts, that comply with the protocols supported by the Caller ID



NOTE:

Many Caller ID protocols specify FSK signaling for MWI.

6400 Tip/Ring (Data Module Features)

- The 6400 Tip/Ring feature supports the operation of an analog adjunct operating independently on an I2 channel with its own extension for 64xx DCP telephones.
- Allows the operation of an analog adjunct to be independent of the digital telephone's extension for the use of fax machines or modems without compromising the user's voice extension.
 - 6416D+
 - 6424D+ DCP



NOTE:

When set for I2 operation, the module transmits an S2-channel button press message for button 01h of group 00h whenever it detects an analog switch hook flash.

The DEFINITY® system ignores any of these button depression messages. This does not cause any trouble report. While set for I2 operation, the button press message cannot be suppressed.



Figure 4. 6416D+ DCP Terminal



Figure 5. 6424D+ DCP Terminal

Administration

The 6416D+ and the 6424D+ terminals are administered with a data option field for an analog adjunct.

- Administer as a data module field on add station form.
- An option on the add station form allows users to administer their own particular extension.
 - Previously, what is now the data option field, was the data module field for all of the 6400 terminals.
 - User can choose from analog, data module, or none.

Station Form page 1 used for 6416D+ and 6424D+ terminals only

STATION		Page 1 of X
Extension: 1003	Lock Messages? n	BCC: 0
Type: 6424D+	Security Code: _____	TN: 1_
Port: _____	Coverage Path 1: _____	COR: 1_
Name: _____	Coverage Path 2: _____	COS: 1_
Hunt-to-Station: _____		
STATION OPTIONS		
Loss Group:	Personalized Ringing Pattern: 1	
Data Option: analog	Message Lamp Ext: 1000	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english	Expansion Module? n	
	MM Complex Data Ext: n	

- DEFINITY[®] system 8.1 provides the following form only if the set is a 6416D+ or 6424D+ and Expansion Module = y on Page 1.

Station form used only with 6416d+ or 6424d+ equipped with XM24 expansion module

STATION		Page X of X
EXPANSION MODULE BUTTON ASSIGNMENT		
1: _____	13: _____	
2: _____	14: _____	
3: _____	15: _____	
4: _____	16: _____	
5: _____	17: _____	
6: _____	18: _____	
7: _____	19: _____	
8: _____	20: _____	
9: _____	21: _____	
10: _____	22: _____	
11: _____	23: _____	
12: _____	24: _____	

- DEFINITY[®] system 8.1 provides the following form if the set is a 6416D+ or 6424D+ and data option is equal to analog

Station Form used only with the 6416D+ or the 6424D+ when Data Option equal to analog

		Page 6 of 6	
STATION			
Analog Adjunct			
Data Extension: 1003	Name:	BCC: 0	
	COS: 1		
	COR: 1		
ITC: restricted	TN: 1		
ABBREVIATED DIALING			
List1:			
SPECIAL DIALING OPTION:			
ASSIGNED MEMBER (Station with a data extension button for this analog adjunct)			
Ext Name			
1:			



NOTE:

DSA is impacted when implementing these new features.

Station Form Page 3 Used For All 6400 Voice Terminals Except the 6402/D

		Page 3 of X	
STATION			
SITE DATA			
Room: _____	Headset? n		
Jack: _____	Speaker? y		
Cable: _____	Mounting: d		
Floor: _____	Cord Length: 0		
Building: _____	Set Color: _____		
ABBREVIATED DIALING			
List 1: _____	List 2: _____	List 3: _____	
BUTTON ASSIGNMENTS			
1: _____	5: _____		
2: _____	6: _____		
3: _____	7: _____		
4: _____	8: _____		

Station Form Page 3 Used For 6402/D Voice Terminals Only

		Page 3 of X
STATION		
SITE DATA		
Room: _____	Headset? n	
Jack: _____	Speaker? y	
Cable: _____	Mounting: d	
Floor: _____	Cord Length: 0	
Building: _____	Set Color: _____	
ABBREVIATED DIALING		
List 1: _____	List 2: _____	List 3: _____
BUTTON ASSIGNMENTS		
1: call-appr _____		
2: call-appr _____		

Station Form Page 4 Used For 6402/D Voice Terminals Only

		Page 4 of X
STATION		
FEATURE BUTTON ASSIGNMENTS		
1: _____		
2: _____		
3: _____		
4: _____		
5: _____		
6: _____		
7: _____		
8: _____		
9: _____		
*: _____		
0: _____		
#: _____		

Station Form Page 4 Used For 6416D+ Only

STATION	Page 4 of X
FEATURE BUTTON ASSIGNMENTS	
1: _____	
2: _____	
3: _____	
4: _____	
5: _____	
6: _____	
7: _____	
8: _____	

Station Form Page 4 Used For 6424D+ Only

STATION	Page 4 of X
FEATURE BUTTON ASSIGNMENTS	
1: _____	9: _____
2: _____	10: _____
3: _____	11: _____
4: _____	12: _____
5: _____	13: _____
6: _____	14: _____
7: _____	15: _____
8: _____	16: _____

Station Form Page 5 Used for all 6400 Display Sets Other Than the 6402D

Page 5 of X

STATION

SOFTKEY BUTTON ASSIGNMENTS

- 1: _____
- 2: _____
- 3: _____
- 4: _____
- 5: _____
- 6: _____
- 7: _____
- 8: _____
- 9: _____
- 10: _____
- 11: _____
- 12: _____

Station Form used Only With 6416D+ or 6424D+ Equipped With XM24 Expansion Module

Page X of X

STATION

EXPANSION MODULE BUTTON ASSIGNMENT

- | | |
|-----------|-----------|
| 1: _____ | 13: _____ |
| 2: _____ | 14: _____ |
| 3: _____ | 15: _____ |
| 4: _____ | 16: _____ |
| 5: _____ | 17: _____ |
| 6: _____ | 18: _____ |
| 7: _____ | 19: _____ |
| 8: _____ | 20: _____ |
| 9: _____ | 21: _____ |
| 10: _____ | 22: _____ |
| 11: _____ | 23: _____ |
| 12: _____ | 24: _____ |

6400 Terminal Parameters Form

		Page X of X
		STATION
6400-TYPE TERMINAL PARAMETERS		
Default Parameter Set: 1		Customize Parameters? y
OPTIONS		
Display Mode:		Handset Expander Enabled?
Volume:		
PRIMARY LEVELS		
Voice Transmit (dB):		Voice Sidetone (dB):
Voice Receive (dB):		Touch Tone SideTone (dB):
Touch Tone Transmit (dB):		
2-WAY BUILT-IN SPEAKER LEVELS:		
Voice Transmit (dB):		Voice Receive (dB):
Touch Tone Transmit (dB):		
1-WAY BUILT-IN SPEAKER LEVELS		
Voice Receive (dB):		Touch Tone SideTone (dB):

Tip Fields

Fields, field placements, and valid keywords

6400	8400/ 9400	Differences
6402/D	8403B	6402 has 1 less call appearance.
6408	8410B	6408 has 2 less call appearances and no 2 way-speaker-phone.
6408+	8405B	6408+ has 3 more call appearances
6408D	8405D	6408D has 3 more call appearances and no 2-way speaker-phone.
6408D+	8405D+	6408D+ has 3 more call appearances.
6416D+	8434D	6416D+ has 18 less call appearances and 3 less soft-keys (12 v 15). The 8434 has a 2 x 40 display instead of a 2 x 24 display
6424D+	8434D	6416D+ has 18 less call appearances and 3 less soft-keys (12 v 15). The 8434 has a 2 x 40 display instead of a 2 X 24 display

Support for 24-button expansion modules

- The XM24 expansion module is fully functional when administered together with the 6416D+ and 6424D+ voice terminals.

Abort Transfer

- The Abort Transfer provides the user with the optional capability to abort a transfer attempt when pressing a non-idle line appearance or going on-hook, putting the previously soft-held call on hard hold.
- In a Pre Release 8.1 operation, the user might have received an urgent call that they needed to pick up and then later transfer the urgent call to yet another party.
 - The user forgets that the first call is still on soft-hold awaiting transfer and the user instead presses the transfer button to transfer the urgent call to a third party.
 - As soon as the user presses the transfer button, the first and second calls are transferred together which is not what the user intended.
 - If the user of a DCP, Hybrid, Wireless, or ISDN-BRI terminal presses a non-idle call appearance (or goes on-hook) in the middle of performing a Transfer operation, the Transfer operation is not aborted but is left in an active state.
- Abort Transfer allows abort the transfer operation whenever he/she presses a non-idle call appearance in the middle of the transfer operation or when the user hangs-up (assuming that the Transfer Upon Hang-Up feature is not enabled).
- The original call that was on soft-hold in the transfer-pending mode will be changed to be on hard-hold so that the original call is not lost.
- When the user then presses the transfer button, the user will be initiating a new transfer for the call appearance he/she is currently on.

Abort Conference

- The Abort Conference enhancement provides the user with the optional capability to abort a conference attempt when going on-hook, putting the previously soft-held call on hard hold.
- This allows the user of a DCP, Hybrid, Wireless, or ISDN-BRI terminal to abort the conference operation whenever they hang-up.

NOTE:

The original call that was on soft-hold in the conference-pending mode will be changed to be on hard-hold so that the original call is not lost.

- When the user later presses the conference button, the user will be initiating a new conference for the call appearance they are currently on.



NOTE:

The Abort Transfer / Abort Conference features are not applicable to Attendant Consoles.

Administration

Enable Abort Transfer Operation and Enable Abort Conference Operation

- Enabled/disabled on a per-system basis by the use of 2 new fields on the system-parameters features form.
- Automatically disabled on upgrades.

New on the system-parameters features form

- "Abort Transfer?" and "Abort Conference Upon Hang-Up?" are added to Page 5 of the form as shown

```
change system-parameters features                                     Page 5 of 10
                           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): 160
Reset Shift Timer (seconds): 0
Station Call Transfer Recall Timer (seconds): 0
DID Busy Treatment: tone
Allow AAR/ARS Access from DID/DIOD? _
Allow ANI Restriction on AAR/ARS? _ Transfer Upon Hang-Up? n
Abort Conference Upon Hang-Up? n Abort Transfer? n
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: _
```

- Abort Transfer? Fixed field identifying whether the Abort Transfer feature is enabled or disabled.
 - Help: n(o), y(es)
 - Default: n
 - Error: invalid entry, press help.
- Abort Conference Upon Hang-Up? Fixed field identifying whether the Abort Conference feature is enabled or disabled.
 - Help: n(o), y(es)

- Default: n
- Error: invalid entry, press help.

Attendant Vectoring

Attendant Call Vectoring is a stand alone feature that is supported on a single switch environment with or without CAS. Attendant Call Vectoring permits unique treatment for each of the attendant-seeking calls according to a number of factors, as defined by the Vector Directory Numbers for each functionality, so as to provide a highly flexible approach for managing incoming calls to an attendant on a DEFINITY switch.

Terminology and Acronyms

Table 2. Terms and Acronyms

Term	Meaning
BSR	Best Service Routing
CAS	Centralized Attendant Service
IAS	Inter-PBX Attendant Service
RLT	Release Link Trunk
TAAS	Trunk Answer Any Station
TN	Tenant Number. Tenant numbers are assigned to all endpoints in the system and indicates which tenant partition the endpoint belongs to. If tenant partitioning is not administered the system has only one "universal" tenant partition and all endpoint TN numbers are defaulted to 1.
VDN	Vector Directory Number

Attendant Group Calls

Attendant group calls are calls directed to an attendant group; calls to an individual attendant (via the assigned extension) are not considered attendant group calls. Calls begin as or become attendant group under the following conditions:¹

- Call to Listed Directory Number (LDN)
- Attendant group in coverage path
- Attendant control of trunk group access
- Attendant return calls
- Attendant serial calls
- Calls forwarded to attendant group
- Controlled Restriction
- Dialed attendant access code
- Emergency access (dialed FAC or off hook timeout)
- VIP Wakeup
- DID/Tie/ISDN intercept treatment
- DID time-out due to Unanswered DID Call Timer expiry
- DID busy treatment
- Call Park time-out
- Call Transfer time-out
- Security Violation Notification (SVN)
- Multifrequency signaling with attendant group as terminating destination
- CDR buffer full with attendant group as Call Record Handling Option
- Trunk incoming destination is attendant group
- Trunk group night service destination is attendant group
- Hunt group night service destination is attendant group
- Automatic Circuit Assurance (ACA) referral calls
- VDN routes to the attendant access code

Attendant group calls terminate to the attendant group associated with a particular party's tenant number (TN). When tenant partitioning is not administered, the system can have only one "universal" tenant partition and attendant group in which attendants can be members. Any attempt to assign an attendant to an

1. Multimedia data calls can also be sent to an attendant group, but only voice is being supported for R8 attendant vectoring. Data calls were not originally supported for basic vectoring and support is still incomplete. Since there is no need to provide this functionality, MMCH calls will be blocked from attendant vectoring.

attendant group other than 1 is denied. All endpoints in the system are defaulted to TN 1 which by default belongs to attendant group 1. Any attempt to assign TN numbers other than 1 is also denied. So, attendant group calls will always terminate to attendant group 1. For example, station 2000 is assigned TN 1 so when the attendant access code is dialed from station 2000, the call terminates to attendant group 1.

When tenant partitioning is administered, attendants can be assigned to multiple attendant groups and endpoints can be assigned TN numbers for any of the tenant partitions. The TN is associated with an attendant group via the tenant form. So, tenant partitions 1 and 2 can be associated with separate attendant groups or the same attendant group. Attendant group calls still terminate to the attendant group assigned to a party's TN. For example, station 2000 is assigned TN 1 which is associated with attendant group 1. Station 3000 is assigned TN 2 which is associated with attendant group 2. When the attendant access code is dialed from station 2000, the call terminates to attendant group 1 and when dialed from station 3000, terminates to attendant group 2. This is a flexible method of assigning multiple tenant partitions to be controlled by one attendant group or assigning each tenant partition a separate attendant group.

Call Vectoring

Call vectoring provides a highly flexible approach to managing calls. It consists of three basic components:

1. A Vector Directory Number (VDN)
2. Vectors
3. Vector commands

The VDN is an extension which can be directly dialed or administered as a destination (e.g., last point in a coverage path, night service destination, etc.). Each VDN is assigned a vector in which call handling procedures are programmed using vector commands. Once the call goes to a vector, the call's routing and treatment are determined by the commands in the vector.

Attendant Vectoring

A new customer option for attendant vectoring will be added and will not be available if RLT CAS is turned on. So, this feature will be available in non-distributed attendant environments and distributed attendant environments for IAS and QSIG CAS. When the new option is on, attendant vectoring will be available for customers to program how they want their attendant group calls processed. This feature will be based upon and use existing call vectoring and attendant administration.

Just as TN assignment currently determines which attendant group calls are terminated to, the TN assignment will also determine which VDN the calls are redirected to (i.e., a VDN field will be added for each tenant partition). If a VDN is administered, attendant group calls will be redirected to the VDN rather than the attendant group. If a VDN is not assigned calls will terminated to the associated

attendant group as they currently do. How Definity determines which party's TN to use in call scenarios will remain unchanged by attendant vectoring. The VDN for the selected TN will apply. For example, the selected TN for calls covered to an attendant group, is the called user's, not the calling user's, TN. When tenant partitioning is not administered, the system can have only one partition and attendant group. All attendant group calls are directed to attendant group 1. The form to administer TN associations is not accessible so system-wide console assignments apply. To follow the existing principals of this administration, the attendant vectoring VDN assignment will appear on the Console Parameters form when partitioning is turned off. When it is turned on, the field will be removed from the console form and the contents will be automatically copied to TN 1.

To associate VDNs and vectors for attendant vectoring, a new field will be added to both the VDN and the call vectoring forms to indicate attendant vectoring. When attendant vectoring is indicated for VDNs and vectors, all call center associated fields will be removed (Skills, BSR, etc.). There will be no restrictions applied to attendant and non-attendant vectoring. For example, an attendant VDN can point to a non-attendant vector and vice versa. The same is true for vector commands. For example, an attendant VDN which points to an attendant vector can have a vector step which routes to another VDN which is a non-attendant VDN. In this case, the call will be removed from queue and treated as though it had just entering vector processing rather than a continuation from one VDN to another. The reverse will also be true if non-attendant VDNs get subsequently routed to attendant VDNs. The following commands will be provided for attendant vectoring:

- Announcement <extension>
- busy
- disconnect after announcement <extension>
- goto step <step #> if time-of-day is <day> <hour>:<minute> to <day> <hour>:<minute>
- goto step <step #> if <unconditionally, queue-fail>
- goto vector <vector #> if time-of-day is <day> <hour>:<minute> to <day> <hour>:<minute>
- goto vector <vector #> if <unconditionally, queue-fail>
- queue-to attd-group
- queue-to attendant <extension>
- queue-to hunt-group <group number> pri <priority level>
- route-to number <number> with cov <y|n> if <unconditionally>¹
- wait-time <seconds> secs hearing <silence, ringback, music>

1. In this case, unconditional is not a choice, it is the only option

- wait-time <seconds> secs hearing <extension> then <silence, ringback, music, continue>
- stop

If attendant vectoring results in putting a call in the attendant queue, it will be placed in queue with the priority as administered on the console parameter form. There will be no changes made to the attendant priority queue for attendant vectoring. Note that even when partitioning is turned on and multiple attendant groups exist, all queues have the same priority assignments. Priority queue administration also applies for calls to an individual attendant (via the assigned extension).

If attendant vectoring results in putting a call in the hunt group queue, it will be placed in queue with the indicated priority. To use this command, the hunt group must be vector controlled.

Until some mechanism is provided for customers to apply vector commands or specialized administration to specific types of attendant group calls, the following will not be redirected to the attendant VDN:

- Emergency Access - these calls will still be sent directly to the attendant group. However, an attendant vectoring VDN can be assigned as the emergency access redirection extension.
- Attendant return calls - these calls will still be sent to the original attendant if available or placed into the attendant group queue if no attendants are available.
- Serial calls - as with return calls, serial calls will still be returned to the original attendant if available and placed in queue if no attendants are available.
- VIP Wakeup calls - these reminder calls will still be sent directly to the attendant group.
- Call Park time-out - these calls result in a conference (caller, principal, and attendant) and call vectoring does not allow conferenced calls to be vectored.
- Call Transfer time-out - these calls are controlled by the attendant return call timer and are processed as though they had been attendant extended calls (i.e., actual attendant return calls). It is unclear when or why this functionality was introduced, but the design will not be changed in order to allow the calls to be vectored.

One of the main reasons for providing attendant vectoring is to allow attendant group calls to be routed anywhere when the system is in night service. For R8, there will be no additional night service functionality provided for attendant vectoring. The desired night service routing can be provided using the existing night station service in conjunction with attendant vectoring. All existing night service rules will remain in place (e.g., night console service supersedes night station service which supersedes TAAS). Attendant group calls will not be

redirected to attendant vectoring when the system is in night service unless a night console is available. Otherwise, they will continue to be redirected to the applicable night service processing. In order to achieve attendant vectoring for calls when the system is in night service without a night console, the night station service extensions must be attendant vectoring VDN extensions.

Attendant VDN

The fact that VDN extensions can be dialed directly or calls can be transferred to VDN extensions will remain unchanged for attendant VDNs.

Currently, VDN extensions can be assigned to:

- Hunt Group night destination - An attendant vectoring VDN can be assigned as a hunt group's night destination. Calls to that hunt group when it is in night service will be redirected to the VDN and attendant vectoring will apply. Note that hunt group night service does not apply if the hunt group is vector controlled. When *vector?* is *y*, the night service destination field is removed from the form. In order for a hunt group to be available in vectoring for the *queue-to hunt-group* command, the hunt group must be vector controlled. So, the hunt group in the *route-to* command could be in night service and the call would then terminate to the indicated night service destination.
- LDN and trunk night destination - One or all trunk groups can be placed into night service and an attendant vectoring VDN can be assigned as the group's night service destination. If a night destination is assigned for LDN calls, it will over-ride (for LDN calls) the trunk group's night destination. Either of these destinations can be an attendant vectoring VDN. However, if tenant partitioning is administered and the trunk group night service destination is the attendant group, the call will be redirected to the VDN associated with the trunk group's TN. If, instead, the night service destination is explicitly assigned to a particular attendant vectoring VDN, it may or may not be the VDN that would have resulted had the night destination been the attendant group.
- Tenant night destination - For tenant partitioning, each partition can be assigned a night destination. When tenant partitioning is turned off, local attendant group calls are sent to the LDN night destination. When partitioning is turned on, local attendant seeking calls are sent to the partition's night destination.
- Trunk group incoming destination - The incoming destination can be an attendant vectoring VDN except for RLT trunk groups. As in trunk group night service, an assigned incoming destination to an attendant vector could result in the call being sent to a different VDN than if the destination had been assigned to the attendant group.
- Last coverage point in a coverage path - Attendant VDNs can be assigned as a coverage point.
- Abbreviated dialing lists - Attendant VDNs can be assigned to abbreviated dialing lists.

- Emergency access redirection - An attendant VDN can be assigned to emergency access redirection. When the attendant's emergency queue over-flows or when the attendant group is in night service, all emergency calls will be redirected to this VDN. Careful thought should be given to routing these calls off-switch.
- QSIG CAS number for attendant group calls - An attendant VDN can be assigned to this number which determines where attendant group calls at a QSIG Branch are processed. This will allow local vectoring at a Branch prior to routing the calls to the Main or elsewhere.
- Auxiliary data for the following button assignments - In keeping with existing procedures, attendant VDNs will not be denied as auxiliary button data for:
 - Facility busy indication - visual indication of busy or idle status for the associated extension
 - Manual message waiting indication - light a message waiting lamp on the station associated with the button
 - Manual signaling - rings the station associated with the button
 - Remote message waiting indicator - message waiting status lamp automatically lights when a LWC message has been stored in the system for the associated extension

Attendant Routing Examples

The following examples outline how the Attendant Vectoring commands can be used to route calls in an attendant environment. For these examples, consider the following vectors and vector administration.

Assume the following attendant vectors (tenant partitioning is turned on):

Table 3. Attendant Vectoring Vectors

VDN 1999 vector 1	VDN 2999 vector 2	VDN 3999 vector 3
1. goto step 6 if time-of-day is all 12:00 to 13:00 2. queue-to attd-group 3. goto step 7 if queue-fail 4. wait 999 secs hearing music 5. busy 6. route-to number 4000 with cov y if unconditionally 7. route-to number 93035381000 with cov y if unconditionally	1. queue-to attd-group 2. goto step 6 if queue-fail 3. announcement 9000 4. wait 999 seconds hearing music 5. disconnect after announcement 9001 6. queue-to hunt-group 1 7. goto step10 if queue-fail 8. wait 999 secs hearing ringback 9. busy 10. route-to number 93035381000 with cov y if unconditionally	1. goto step 7 if time-of-day is all 12:00 to 13:00 2. queue-to attd-group 3. goto step 7 if queue-fail 4. announcement 9000 5. wait 15 seconds hearing music 6. goto step 4 if unconditionally 7. queue-to attendant 6000 8. goto step 10 if queue-fail 9. wait 999 secs hearing ringback 10. route-to number 93035381000 with cov y if unconditionally

Administration is as follows:

- a. All stations are assigned TN 1 which is associated with attendant group 1, VDN 1999, and music source 1
- b. All trunk groups are assigned TN 2 which is associated with attendant group 1, VDN 2999, and music source 2
- c. All VDNs are assigned TN 3 which is associated with attendant group 2, VDN 3999, and music source 3
- d. Extension 4000 is assigned to a hunt group 1
- e. Extension 6000 is assigned to an attendant console for direct access

Local Attendant Group Access Code

When a station dials the attendant access code, the call is redirected to vector 1. If it is lunch time, the call is sent to a hunt group and vector processing terminates. If it is not lunch time, the call is sent to attendant group 1. If an attendant is available, the call is terminated to the attendant and vector processing terminates.

Otherwise, the call will be queued to the attendant group and the caller hears music from the music source assigned to TN 1 until an attendant answers the call. If the call cannot be queued, it is routed to a remote location with coverage and vector processing terminates. If the call is unanswered after 999 seconds in the attendant queue, the caller hears busy and vector processing terminates.

NOTE:

The *route-to* command leaves vector processing as soon as the call is successfully routed. So, in the example above, if it is lunch time the call will route to the hunt group and all hunt group processing will then apply. If the group is assigned a queue and the call can be queued, it will be. If not, and coverage criteria is met, the call will follow the hunt group's coverage path. If the hunt group is in night service, the call will go to the hunt group's night service destination. If the *route-to* command had indicated coverage "n", the hunt groups coverage path would not have been followed and vector step 7 would apply.

Incoming Trunk Calls to Attendant Group

When a call is received on a trunk which has the attendant group assigned as the incoming destination or the call is addressed to the attendant group, the call is redirected to vector 2. The call is then sent to attendant group 1. If an attendant is available, the call is terminated to the attendant and vector processing terminates. Otherwise, the call will be queued to the attendant group and the caller will hear the announcement followed by music from the music source assigned to TN 2. If the call is unanswered after 999 seconds in the attendant queue, the caller is dropped after hearing an announcement and vector processing terminates. If queueing to the attendant fails, the call is queued to hunt group 1. If a member is available to take the call, it will be terminated to the member and vector processing terminates. If a member is not available and the call can be queued, it will be and the caller will hear ringback until a member answers. If the call is

unanswered after 999 seconds in the hunt group queue, the caller hears busy and vector processing terminates. If the call cannot be queued, the call will be routed to the remote location and vector processing will terminate.

 **NOTE:**

The main difference from scenario 1 is queueing the call to the hunt group rather than routing the call there. In this scenario, the call will not follow the hunt group's coverage path or night service destination.

Incoming LDN Calls

When a call is received for an LDN, the call is redirected to vector 3. If it is lunch time, the call is sent to attendant 6000. If the attendant is available, the call is answered and vector processing terminates. If the attendant is not available the call is placed into queue and the caller hears ringback until the attendant answers the call. If the call is unanswered after 999 seconds in the attendant's queue, the call is sent to the remote location and vector processing terminates. If the call cannot be placed in attendant 6000's queue, the call is routed to a remote location and vector processing terminates. If it is not lunch time, the call is sent to attendant group 2. If an attendant is available, the call is terminated to the attendant and vector processing terminates. Otherwise, the call will be queued to the attendant group and the caller will hear an announcement followed by music (from the music source assigned to TN 3) every 15 seconds. If the call cannot be queued, it is sent to attendant 6000.

 **NOTE:**

Vector 3 attempts to queue the call to attendant 6000. A *route-to* command could have been used also, but care should be taken since an attendant cannot be assigned a coverage path.

QSIG CAS

This example shows how you can use Attendant Vectoring with CAS.

CAS BRANCH

Suppose the customer would like to always play an announcement at a QSIG CAS branch before routing the call to the QSIG CAS main. An attendant VDN (assume 1000) would need to be administered in the QSIG CAS Number field at the branch instead of the number to the QSIG CAS main's attendant access code (assume 303-538-0 with AAR access code 9). The following vector would play an announcement and then route the call to the QSIG CAS main.

Table 4. QSIG CAS Vector

VDN 1000 vector 1
1. announcement 9000 2. route-to number 93035380 with cov y if unconditionally

CAS MAIN

Calls from a QSIG branch are sent to the main with the main's attendant access code as the destination address so these calls are automatically attendant group calls. The VDN to which these calls get redirected will depend on the TN of the incoming trunk.

Night Station Service

This example shows how you can make use of the Attendant Vectoring features for night service.

Administration is as follows:

Table 5. Night Station Service Vectors

VDN 4999 vector 4	VDN 5999 vector 5
1. route-to number 93035381000 with cov y if unconditionally	1. route to number 6000 with cov n if unconditionally 2. route-to number 93035381000 with cov y if unconditionally

- a. Trunk group 1 is assigned TN 2 which is associated with attendant group 1, and night destination 4999
- b. Trunk group 2 is assigned TN 1 which is associated with attendant group 2, and night destination 5999
- c. Extension 6000 is assigned to a station
- d. System night service is on

When a non-DID call comes in on trunk group 1, the call is redirected to VDN 4999 which routes it to a remote location.

When a non-DID call comes in on trunk group 2, the call is redirected to VDN 5999 which routes it to station 6000. If station 6000 is unavailable, the call will not

cover on station 6000's coverage path. Vector processing will continue and route the call to a remote location.

 **NOTE:**

When station night service is active, calls are processed according to the administered night destination for the trunk group, not the night destination for the associated TN (i.e., these are not attendant group calls). If the night destination had been assigned as *attd* or left unassigned, the calls would have become attendant group calls and would be processed according to the partitions night destination.

Trouble Isolation: List Trace Command

Two new options for the **list trace** command allow customers and Tier 1 technicians to troubleshoot misdirected calls, trunking/routing problems, and call denials.

- **list trace station nnnnn** (where nnnnn is the number of the station you want to troubleshoot)
- **list trace tac xxxx** (where xxxx is the identifier of the trunk you want to troubleshoot)

Execute the **list trace** command with the relevant option word.

- To see all the **list trace** options, execute **list trace** at the command line.
- To redisplay the most-recently done trace, execute **list trace previous** at the command line.
- To cancel a trace in progress, press **F7**.

 **WARNING:**

DEFINITY release 8.2 does not record all denial events. Always check translations before you escalate a problem.

For example, COR to COR restrictions are not recorded in the event log. If the event log does not have entries for the calling/called parties, check the COR translations.

Diagnostic Procedure

1. Execute **list trace station nnnnn** or **list trace tac xxxx** from the command line.

For tone traces (denial events) convert the call ID (cid) from decimal to hexadecimal, and continue with the following steps.

2. Execute the **display events** command from the command line. Search for **CATEGORYCATEGORY denial**.
3. Search the **EVENTS REPORT** for the *hexidecimal* cid to troubleshoot the problem.

Diagnostic Example: list trace to Diagnose Intercept Tone (Denial)

In this example, the customer heard an intercept tone (denial of service).

Execute **list trace station nnnn** and make the test call from that station:

```
list trace station 51001

                        LIST TRACE

time                data

11:58:57    active station    51001 cid 528
11:59:02    dial 9538
11:59:02    denial station    51001 cid 528
11:59:02    TRACE COMPLETE station 51001 cid 528
```

Convert 528 to hexadecimal (0x210) and execute **display events CATEGORY denial**.

Search for event data 2 for hexadecimal cid 210:

```
                        EVENTS REPORT

Event Event          Event Event  First      Last      Evtnt
Type  Description      Data 1 Data 2  Occur      Occur      Cnt

1343 Need higher FRL to route 6      210      02/11/11:59 02/11/11:59 1
```

Diagnostic Example: list trace to Diagnose Fast Busy (Reorder Tone)

In this example, no trunks were available in trunk group 1.

Execute **list trace station nnnnn** and make the test call from that station:

```
list trace station 51001
```

LIST TRACE

```
time          data
12:04:30      active station    51001 cid 531
12:04:32      dial 56001
12:04:32      reorder trunk-group 1  cid 531
12:04:032     TRACE COMPLETE station 51001  cid 531
```

Convert 531 to hexadecimal (0x213) and execute **display events CATEGORY denial**.

Search for event data 2 for hexadecimal cid 213:

EVENTS REPORT

Event Type	Event Description	Event Data 1	Event Data 2	First Occur	Last Occur	Evnt Cnt
1012	Destination Unavailable	6	213	02/11/12:04	02/11/12:04	1

Diagnostic Example: list trace to Diagnose DID not Ringing Correctly

In this example, the trace verifies the inbound digits from the network and traces DID trunk to VDN 52099, to vector 1, terminated to agent 51001.

Execute **list trace tac xxxx** and place the test call from the network:

```
list trace tac 7

                                LIST TRACE

time          data
11:38:30     active trunk-group 79 member 1 cid 516
11:38:33     dial 52099
11:38:33     ring vector 1 cid 516
11:38:41     active station 51001 cid 516
11:38:41     TRACE COMPLETE station 51001 cid 516
```

Diagnostic Example: list trace to Diagnose Outpulsing Error

In this example, the trace verifies the outpulsing of incorrect digits to the central office (area code 303 is required). Trunk-group 81 member 1 outpulsed **5381788**. The network rejected the call because it required a 10-digit number. The call did not complete, and the caller went on-hook.

Execute **list trace station nnnnn** and place the test call from the station:

```
list trace station 51001

                                LIST TRACE

time          data
11:17:17     active station 51001 cid 522
11:17:20     dial 9538
11:17:20     term trunk-group 81 cid 522
11:17:21     dial 953817
11:17:21     seize trunk-group 81 member 1 cid 522
11:17:24     dial 95381788
11:17:24     outpulse done 5381788
11:17:32     active trunk-group 81 member 1 cid 522
11:17:34     idle station 51001 cid 522
11:17:34     TRACE COMPLETE station 51001 cid 522
```


Diagnostic Example: list trace to Diagnose Station not Ringing

In this example, the trace verifies the inbound DID call to x52001 is forwarded off-net and the trunk hung up.

Execute **list trace station nnnnn**:

```
list trace tac 7

                                LIST TRACE

time          data

10:42:15     active trunk-group 79 member 1 cid 504
10:42:18     term station 52001 cid 504
10:42:18     call-forwarding 9538
10:42:18     term trunk-group 81 cid 504
10:42:18     call-forwarding 95381788
10:42:18     seize trunk-group 81 member 1 cid 504
10:42:21     call-forwarding 95381788
10:42:21     outpulse done 3035381788
10:42:30     idle trunk-group 81 member 1 cid 504
10:42:30     TRACE COMPLETE trunk-group 79 cid 504
```

Diagnostic Example: list trace to Analyze Call Coverage

In this example, the trace verifies the inbound DID call to x52002, no answer; rings x51002, no answer; 2010 no answer, 2010 (final point of coverage) continues to ring and then answers.

Execute **list trace station nnnnn**:

```
list trace tac 7

                                LIST TRACE

time          data

10:21:03     active trunk-group 79 member 1 cid 512
10:21:06     dial 52002
10:21:06     ring station 52002 cid 512
10:21:16     no answer station 52002 cid 512
10:21:16     coverage station 51002 cid 512
10:21:32     no answer station 51002 cid 512
10:21:32     coverage station 2010 cid 512
10:21:47     no answer station 2010 cid 512
10:21:47     coverage cid 512
10:21:55     active station 2010 cid 512
10:21:57     idle station cid 512
10:21:57     TRACE COMPLETE trunk-group 79 cid 512
```

Diagnostic Example: list trace to Analyze Trunk Signaling Failure

In this example, the trunk wink was not received from the network.

Notice there is no trunk seizure message and digits are not outpulsed. Check the hardware error log for trunk group 2 errors indicating seizure failure.

```
list trace station 51001
time          data
10:35:13     active station    51001 cid 1347
10:35:18     dial 9255
10:35:18     term trunk-group 2   cid 1347
10:35:23     dial 92556001
10:35:23     term trunk-group 2   cid 1347
10:35:38     idle station        51001 cid 1347
10:35:38     TRACE COMPLETE station 51001 cid 1347
```

Diagnostic Example: list trace to Analyze ISDN Call Rejection

In this example, the outbound ISDN call has been rejected by the network or far-end.

Note the ISDN SETUP message has been sent prior to the trunk seizure indicated by the CALL PROCEEDING message. However there is no CALL ALERTING message, indicating that the call was denied by the far-end and the trunk was dropped by the far-end.

```
list trace station 51001
time          data
11:00:37     active station    51001 cid 1518
11:00:41     dial *06255
11:00:41     term trunk-group 11   cid 1518
11:00:44     dial *062556999
11:00:44     seize trunk-group 11 member 2   cid 1518
11:00:44     proceed trunk-group 11 member 2   cid 1518
11:00:44     idle trunk-group 11 member 2   cid 1518
11:00:46     idle station        51001 cid 1518
11:00:46     TRACE COMPLETE station 51001 cid 1518
9454
```

Diagnostic Example: list trace to Display ISDN Call Completion

In this example, the outbound ISDN call is completed.

Note the ISDN ALERTING message was received followed by an active trunk message indicating the called party answered.

```
list trace station 51001

time          data

10:30:02      active station    51001 cid 1305
10:30:07      dial 9255
10:30:07      term trunk-group 11    cid 1305
10:30:09      dial 92556001
10:30:09      seize trunk-group 11 member 4  cid 1305
10:30:09      proceed trunk-group 11 member 4  cid 1305
10:30:09      alert trunk-group 11 member 4  cid 1305
10:30:18      active trunk-group 11 member 4  cid 1305
10:30:20      idle trunk-group 11 member 4  cid 1305
10:30:20      TRACE COMPLETE station 51001  cid 1305
```

NewHospitality-Related Features

Auto Selection Direct Inward Dial (DID) Numbers

- Provides the capability to have the system automatically choose a number among a list of available DID numbers to be assigned to a guest's room upon check-in.
- Useful for any hotel wishing to have direct dialing to rooms without attendant intervention. It is unlikely to get a central office to assign DID numbers directly associated with a hotel's room numbering plan.
- Useful for hotels that wish to give guests a phone number that is not related to the room number.
- Invoked when a check-in is done from the switch (via a button on the console) or remotely via a PMS system.
- The switch assigns a DID number to the checked-in room from a list that is administered as DID stations at the switch.
- All calls made to the DID number are directed to the room's extension as if the room was called directly.
- The Auto Digit Rotation for Direct Inward Dial feature uses x-ports to create the XDIDs; thus, the effective station capacity of the system is reduced by the number of DIDs administered.



NOTE:

xdid assignments also count against the port count for the "port sensitive pricing."

- The Auto Digit Rotation for Direct Inward Dial feature increases hotel guests' security as they can publish their "phone number" without giving away their room number.

Process

Example

When guests check in, they receive the 7- or 10-digit number associated with their DID number assignment.

- Hotel personnel know the area code and office code to precede the DID number to create the 7- or 10-digit public network number.
- All outside calls to the guest use the 7- or 10-digit number associated with that DID number.
- The guests phone will ring and cover as if the room was dialed directly.

- From inside the hotel, either the room/extension number or the 2- to 5-digit DID number may be used.



NOTE:

XDIDs that are unused will follow the coverage path.

All xdids cover to voice mail so that they can get a recording that says nobody is checked in.

Administration

Coverage

- XDID ports perform the Hunt-to function before coverage.
- After hunting, coverage criteria for these calls is based upon the DID, but the coverage points are based upon the hunted-to station (room).
- The Hunt-to feature is modified so that calls placed properly to the DID number route properly to the mailbox associated with the room's extension (hunted-to number).

New field to activate DID assignment feature

- A new field is added to the system-parameters hospitality form to activate the Automatic DID assignment feature Automatic DID Assignment on Check-in?
 - The field accepts only valid values of y or n, which indicates whether the feature is active.
 - The field defaults to n (no).
- Whether the feature is administered from the switch-activated check-in or from a PMS check-in depends on which check-in mode is active on the switch.

Enhance list station command to show hunt-to assignments

- The list station command has a new field to display hunt-to assignments.
- This field always shows regardless of whether the DID feature is assigned. It is directly under the name field on line 2.

New button type xdid-view

- New button type xdid-view is similar to the check-in button.
- The system allows this button's use whether the switch or a PMS administers check-ins, to allow DID assignments to be looked up and changed at any time.
- The button is allowed only on display sets.
- When changed, the old DID, if it existed, is placed back into the queue.

- Use will only be allowed on a station with console permissions.
- The DID can continue to be changed as long as y is pressed.

New Screens

```
add station 1014                                     Page 1 of X
                                                    STATION
Extension: 1014                                     Lock Messages? n      BCC: 0
Type: 6048D+                                       Security Code: _____ TN: 1
Port: _____                                   Coverage Path 1: ____ COR: 1
Name: _____                                   Coverage Path 2: ____ COS: 1
                                                    Hunt-to-Station: ____

STATION OPTIONS
Loss Group: _
Data Module? n
Speakerphone: 2-way
Display Language? English
Personalized Ringing Pattern: 1
Message Lamp Ext: 1014
Mute button enabled? y
```

```
                                                    Page 2 of 3
                                                    HOSPITALITY
Dual Wakeup? y Daily Wakeup? y VIP Wakeup? y
VIP Wakeup Per 5 Minutes: _____
Room Activated Wakeup With Tones? _____
Time of Scheduled Wakeup Activity Report: _____
Time of Scheduled Wakeup Summary Report: _____
Time of Scheduled Emergency Access Summary Report: _____
Announcement Type: _____

Length of Time To Remain Connected To Announcement: 30____
Extension To Receive Failed Wakeup LWC Messages: _____
Routing Extension On Unavailable Voice Synthesis: _____
Display Room Information in Call Display? n
Automatic DID Assignment on Check-in? n
Number of Digits from PMS: 1
PMS Sends Prefix? y
Number of Digits in PMS Coverage Path: 3
Digit to Insert/Delete:
```

Call Detail Recording (CDR)

- With Call Detail Recording (CDR), incoming calls to the DID port are recorded as coming to the DID rather than the room's extension number.
- Outgoing calls are recorded as coming from the room's extension number.

Call Processing

- PMS can request a DID assignment from the switch in the check-in message.
- When the Automatic DID assignment feature is active and check-ins are done from the PMS, the system assigns all check-ins received from the PMS a DID number from the list of XDID numbers.
 - The number is chosen from the list in a FIFO manner.
 - The DID chosen is always the oldest number.
 - This number is sent back to the PMS in a new check-in message.
- The new message is always sent if the DID feature is active, regardless of whether the PMS requested a DID assignment.
- The XDID port has the Hunt-to field filled in with the room extension.
- Checkouts clear the Hunt-to field and return xdid, if there is one, to the free list.

Switch check-in allows optional DID assignment

- When the check is normally complete, then the number is displayed on the set that is doing the check-in.
- The display is changed to "CHECK-IN COMPLETE, DID=dddd" where dddd is the DID assignment.
- The XDID port has the Hunt-to field filled in with the room extension.
- When the DID feature is active, hunt-to stations are treated as the called station.
- When the DID station number is called, the call hunts before coverage and the hunted-to station is treated as though it was dialed directly.
- Coverage applies as if from the hunted-to station.
- Incoming calls use the COS and COR of the DID.
- Outgoing calls use the room extension's COS and COR.

Crisis Alert to Pager

The primary use of this feature is in addition to the Crisis Alert to an Attendant or Digital station.

- It is not recommended to notify only a pager of the Crisis Alert due the possibility of the paging message not getting immediate response.

- If a crisis alert call is originated in an emergency situation, a message will be sent to a pager to notify the page of the extension and location where the emergency call was originated.
 - The message begins with a customer administrable crisis code, then the room number or extension and then the location, which can be the main telephone number or any type of location code.
- Depending on the type of paging service, the Crisis Alert message may be from 7 to 22 digits.



NOTE:

Important! Be aware that a crisis alert call will use from 2 to 4 trunks. This is because one trunk is used for the actual emergency call and 1 to 3 trunks are used to notify the pager(s) depending on the how many pagers are administered.

- Not limited to a particular digital numeric pager and service but must adjust the length of the main number field to fit the pager display.
- The feature should already be administered in the ARS digit analysis table, if not a digit pattern must be selected, i.e. 911, and the call type must be set to "alrt".
- If a Crisis Alert situation occurs the pager will be alerted and display the crisis alert code, the extension/room number and the main number, if one is entered, so that the pagee will know the location where the emergency call was originated.

Administration

Administration of the Crisis Alert to Pager feature consists of the following:

- Adds new fields on the Feature-Related SYSTEM PARAMETERS form.
- New fields added to the CRISIS ALERT form.
 - An administrable originating extension field that is the originating extension for the alert call to a pager.
 - A Crisis Alert Code field that is administrable by the customer for identifying a crisis alert pager message.
 - Code is the first part of the alert message that is sent to a pager, for example in the U.S. the code may be 911.
- When the alrt digit pattern is dialed a visual and audible alert is sent to all attendants and digital stations that have a crss-alert button assigned.
- The Crisis Alert to a Digital Pager feature must be turned on using the Alert Pager field on the CRISIS ALERT SYSTEM PARAMETERS form.

- When the Alert Pager field is turned on new fields appear, Originating Extension, Crisis Alert Code, Retries, Retry Interval, Main Number, three pager numbers and three pin numbers.
- The Originating Extension field requires a valid extension to be entered for use as the extension originating the call to send a crisis alert message to a pager.
- A crisis alert message sent to a pager needs to be identified, so the customer can administer a code in the Crisis Alert Code field.
- The Retries field allows the user to choose the number of times the system tries to send out the alert message in case of an unsuccessful attempt.
- The Retry Interval field allows an administrable time period between retries.
- The Main Number field is for entering a main phone number to the location or a location code, up to 15 digits.
- Three pager number fields are available for inputting up to a 15-digit number.
 - Required that one of the three pager number fields has a number entered or the form cannot be submitted.
 - The last three fields are for a pin number associated with each of the pager number fields.
 - The user has 15 spaces for entering the letter p, *, #, or digits in any combination or the field can be left blank.
 - The p is for pause for timing purposes, so the system will wait to send out the pin number and then wait after the pin number to send the message.
 - A pager number or the main number may be entered using dashes to separate the digits, such as 123-456-7890.
- Two fields for DTMF duration, Tone (msec) and Pause (msec), allow the tone and pause time to be administered.
- The tone time is the actual length of the DTMF tone out pulsed and the pause time is the time between tones.
 - These fields have a range of 20 to 2550 in milliseconds.
- For CO type trunks, on the trunk form either the trunk Answer Supervision Timeout field be set to 0 (zero) and the Receive Answer Supervision field set to y or make appropriate adjustment to the Answer Supervision Timeout with the Receive Answer Supervision field set to n.

FEATURE-RELATED SYSTEM PARAMETERS

```
FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
  System Printer Extension:                Lines Per Page: 60
  EIA Device Bit Rate: 9600

SYSTEM-WIDE PARAMETERS
  Switch Name:

MALICIOUS CALL TRACE PARAMETERS
  Apply MCT Warning Tone? n              MCT Voice Recorder Trunk Group:

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:

CRISIS ALERT:Every User Responds? n
```

Alert Pager field turned off

```
change system-parameters crisis-alert
  CRISIS ALERT SYSTEM PARAMETERS

ALERT STATION
  Every User Responds? n
ALERT PAGER
  Alert Pager? Y
```

Alert Pager field turned on

```
change system-parameters crisis-alert
                                CRISIS ALERT SYSTEM PARAMETERS

ALERT STATION
  Every User Responds? n

ALERT PAGER
  Alert Pager? y
  Originating Extension: 7768
  Crisis Alert Code: 911
  Retries: 5
  Retry Interval (sec): 30
  Main Number: 303-555-0800

                                Pager Number      Pin Number
                                1: 3035559001      1: 7614567890
                                2: 123456789012345  2: ppp1234567890pp
                                3: 123456789012345  3: ppp1234567890pp

                                DTMF Duration - Tone (msec): 100  Pause (msec): 100
```

Attendant receives alert

- To receive the Crisis Alert message there must be at least one attendant with a crss-ahrt button administered. With the Alert Pager field turned on, the attendant will receive the correct alert.

Crisis Alert to Digital Station

Digital Station(s) receives alert

- To receive the Crisis Alert message there must be at least one digital display set with a crss-ahrt button administered.
- With the Alert Pager field turned on, any station with a crss-ahrt button will receive the correct alert.

Emergency call goes out on Cama trunk

- When an emergency call is originated it will connect over a Cama trunk.

Multiple messages

- If there are multiple crisis alert messages, each message is sent out to the pager service in the order that they are initiated.
- Emergency 911 call is not affected by message to pager.
- Entry logged on the emergency log.

- Result of attempt to pager sent to journal printer.
 - The journal printer prints out the result, successful or unsuccessful, of the final attempt of sending a message to the pager.
 - Each pager number has status sent to the journal printer.

Hardware

- Digital Numeric Pager
- TN2182 - Tone Board/Call Classifier or TN744B, C.
- Applies to all DEFINITY® platforms.

Check In via the Hunt-To Feature

- Check In via the Hunt-To Feature provides the user with the capability to have the system automatically check in several related extensions with one check-in command. The check-in command can come from the user's PMS, or can be an on-switch check-in from the attendant.
- This feature allows hotels that offer suite rooms with several phones to check in all the phones associated with that suite at once.

Process

- When a guest checks into a room, the PMS sends a check-in message to the DEFINITY® system.
- The DEFINITY® system turns on the phone in the room by removing controlled outward restriction, adds the guest's name to the station record, sets the coverage path, removes any LWC messages, and marks the room as occupied.
- With the suite check-in feature active, when a room phone is checked in, the system will look if there is a hunt-to extension associated with that station. If so, the system will also check in the station found in the hunt-to field.
- The system will continue checking in stations until it encounters a station in the chain with a blank hunt-to field.

Administration

- System optionThe Station Hunt Before Coverage option allows the customer to choose that a call to a busy station will perform station hunting before going to coverage. This feature allows customers to have calls hunt to the other stations associated with their room (suite) before going to coverage, if the primary number called is busy.
- To activate the Suite Check-in feature, the Suite Check-in? field on the system-parameter hospitality form must be set to y.

- For a group of stations to appear as a suite, the primary extension (the extension that the PMS will be passing to the switch in the check-in message) must have the secondary extension administered in its hunt-to field on the station form.
- To have additional extensions associated with this suite, the user should enter the next extension into the secondary extension's hunt-to field. The final station in the hunt-to chain will have a blank entry in its hunt-to field, thus denoting the end of this suite.
- If the Suite Check-in field is set to y, and a check-in message is received, the extension sent in the check-in message will be checked in. If the extension to be checked in has an extension in its hunt-to field on the station form, that extension will be checked in as well.
- To activate the Station Hunt Before Coverage feature, the "Station Hunt Before Coverage?" field on the system-parameters coverage form must be set to y. The system will check to see if there is a hunt-to station assigned to the busy station. If there is, the system will try to terminate to the hunt-to station before going to coverage. If the hunt-to station is busy as well, the system will continue hunting down the hunt-to chain. If all stations in the hunt-to chain are busy, the call will go to the principal's coverage.
- When the DEFINITY® receives a check-out message, the extension sent in the message as well as any associated extensions (those found in the hunt-to chain) will be checked out. If the primary extension sent on a check-in, or any stations on the hunt-to chain are already in the checked in state, the room will not be checked in, and the message "Room already Occupied" will be given. If the primary extension sent on a check-out, or any stations on the hunt-to chain are already in the checked-out state, the rooms will be checked out, and the message "Room already Vacant" is given.

Changes to the hospitality form

```
change system-parameters hospitality page 3 of 3
HOSPITALITY
ROOM STATES:
Definition for Rooms in State 1: Rooms in State 1
Definition for Rooms in State 2: Rooms in State 2
Definition for Rooms in State 3: Rooms in State 3
Definition for Rooms in State 4: Rooms in State 4
Definition for Rooms in State 5: Rooms in State 5
Definition for Rooms in State 6: Rooms in State 6

HOSPITALITY FEATURES
Suite Check-in ? n
```

Changes to the system-parameters coverage-forward form

change system-parameters coverage-forwarding page 1 of 2

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 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? _

New Mobility-Related Features

X-Station Mobility (DECT)

- X-Station Mobility Feature allows phones remoted over a trunk interface to be controlled by DEFINITY® as if they were directly connected to it.
- Accomplished by administering a station record for each station to be so controlled as of type X-mobile.
- X-mobile stations have additional administration information further describing the capabilities of the remote station.
- Each X-mobile station has a Trunk Group associated with it.
- If the call or other signal needs to be sent to it, DEFINITY® uses the associated D-Channel to communicate with the remote terminal.
- When a call arrives over a D-Channel that carries X-mobile traffic, that call is analyzed to check if an X-mobile remote station originated it.
 - If it did, then the call is associated in call processing as an internally originated call from the local X-mobile station.
 - Allows a large number of DEFINITY® system features to work for the remotely controlled station, including bridging.
- Used to provide DEFINITY® system feature integration with wireless handsets.
- DEFINITY® Wireless DECT Release 2.0 uses this capability to field a competitive wireless offering.

Administration

Enabling X-Station Mobility

- New field Maximum XMOBILE Stations is created on the Customer Options form and defaults to 0.
- When this field is set to 0, the feature is disabled.
- It is only possible to set the field to $N > 0$ if the system is administered as R8 or later.
- This customer option is disabled by default for new R8 systems and for older systems that upgrade to R8.

Change system-parameters customer-options

```
change system-parameters customer-options Page 1 of 5

OPTIONAL FEATURES

G3 Version: V8 Maximum Ports: 0
Location: 1 Maximum XMOBILE Stations: 0
Abbreviated Dialing Enhanced List? N CAS Branch? n
Access Security Gateway (ASG)? n CAS Main? n
Analog Trunk Incoming Call ID? n Cvg Of Calls Redirected Off-net? n
A/D Grp/Sys List Dialing Start at 01? n DCS (Basic)? n
Answer Supervision by Call Classifier? N DCS Call Coverage? N
```

Administering X-mobile Stations

- The station administration form changes to allow the station type of XMOBILE.
- An X-mobile station has the following new fields:

Field	Range	Default	Description
Trunk Group	1 to Max Trunk Groups	1	The trunk group to be signaled
MW Type	NONE, DISP[LAY]	NONE	Update X-mobiles
Display for MWI	Length of Display	1, 2	1 Number of lines in the display

- If the station type is changed to be non-xmobile, then all other fields are the same as for 2500 series stations.

Change station

```
change station 3000 Page 1 of 3
STATION

Extension: 3000 Lock Messages? n BCC:0
Type: XMOBILE Security Code: _____ TN: 1
Coverage Path 1: ___ COR: 1
Name: Jeff's Mobile Coverage Path 2: ___ COS: 1

Mobility Trunk Group: 1
Message Lamp Ext: 3000
Message Waiting Type: NONE

Length of Display: 1
```


Terminal Type

- Setting the type in the station screen to XMOBILE will identify an X-mobile station.

Signaling Group administration

- New field X-Mobility/Wireless Type is added to the signaling group form to indicate the type of X-mobile endpoints allowed for it (DECT or NONE).
- The default for the field is NONE.

Add signaling-group

```
add signaling-group 1 Page 1 of 5
SIGNALING GROUP

          Group Number: 1
    Associated Signaling? y Max number of NCA TSC: 0
          Primary D-Channel: 01b1216 Max number of CA TSC: 0

Trunk Group for NCA TSC:
Trunk Group for Channel Selection: 1           X-Mobility/Wireless Type: DECT
Supplementary Service Protocol: a
```

- The X-Station capability is turned off unless specifically enabled in the special applications form.

Signaling

- ISDN PRI is modified to allow INFO messages with Display Information Elements to be sent at any time.
- ISDN PRI is also modified to allow INFO messages with Keypad Facility Information Elements to generate DTMF when the special application is enabled.
- The X-mobile station is treated as an analog station in terms of feature interactions except as noted in previous requirements.

Description

- When a call comes into a station administered to be X-mobile, the software checks for a trunk group associated to the station.
 - It then signals the call to that station using that trunk group.
- Very similar to how incoming calls are handled for BRI stations.

- When a call comes in from a trunk group marked as having X-mobile endpoints, the SETUP message is checked for a CALLING PARTY NUMBER Information Element.
 - If present, and if it matches the extension of a station marked as X-mobile, that call is re-stimulated as an internal call originating from the X-mobile station.

Calling Capabilities

- Administration of an X-mobile station allows an administrator to customize call options, based on the parameters for an analog station.

Call Detail Recording

- A Call origination from an X-mobile WT is present in the CDR as a call origination from the X-mobile station.
- A Call Termination to an X-mobile WT presents as a call termination to the X-mobile station in the CDR.

Xmobile to Trunk mapping

- No limit to the number of X-mobile stations that can be supported per Trunk Group.

Call Coverage

- The X-mobile station is allowed to cover calls not answered by the X-mobile, either Ring No Answer or Subscriber Absent, when coverage is administered for the X-mobile station.



NOTE:

If remote coverage is administered for an X-mobile station, it then must be the last point of coverage.

Capacity Constraints and Feature Limitations

- The X-Station Mobility works only when stations are remoted over ISDN trunks.
- An X-mobile station counts as wired DEFINITY® station in terms of station limits and configuration consideration.

Memory Impact of Capacity Changes

- The X-mobile capability is administered on the station admin screen
- It is implemented in such a way as to *not* increase the memory required on a per station basis.

Scenarios

- All feature operations are intended to work exactly as if the X-mobile station were directly connected to the DEFINITY® System.
- Analog type feature operation were chosen, since most wireless terminals support only flash operations and are not multi-appearance or context sensitive.
- In almost all cases, the operation expected for an X-mobile station is identical to a similarly administered analog station.

XStation Mobility (MWL)

- Allows the DEFINITY® system to control and manage handsets not directly connected to the DEFINITY® system as if they were directly connected wired stations.
- The handset may have bridged appearances on wired terminals and thus be able to use DEFINITY® system features like call forwarding and call pickup.
- Icon-based message waiting update is also supported.
- Achieved by using the DEFINITY® system's Administration Without Hardware (AWOH) capability and extending it to actual signal remote points.

Description

- When a call comes into a station administered to be X-Mobile, the software checks for a trunk group associated to the station.
- It then signals the call to that station using that trunk group.
 - Similar to how incoming calls are handled for BRI stations.
- MWI Activation and Deactivation messages are exchanged between the DEFINITY® system and the DAS to update MW status on the X-Mobile WTs.
- Upon successful location registration of an X-Mobile WT, the DAS sends a location registration indication message to the DEFINITY® system.
 - Sends back an MWI Activation or Deactivation message back to the DAS to refresh the X-Mobile WT's MW status.
- Depending on the level of support on the X-Mobile WT, a detach indication may be sent to the DEFINITY® system indicating that the WT has left the system.
 - The DEFINITY® system ignores this indication until further definition of this capability is available.

⇒ NOTE:

If the X-Mobile trunk does not return alerting, and instead returns a disconnect or release, the DEFINITY[®] system software advances to the next coverage point, if administered.

If the next coverage point is not administered, DEFINITY[®] system software disconnects ringback tone and present busy tone.

On an ISDN interface, after XX seconds of busy tone, a disconnect with cause SUBSCRIBER ABSENT (17) is passed if the originator was ISDN.

- Allows phones remote over a trunk interface to be controlled by the DEFINITY[®] system as if they were directly connected to it.
- Accomplished by administering a station record for each station to be controlled as type X-Mobile.
- These X-Mobile stations have additional administration information further describing the capabilities of the remote station.
- Each X-Mobile station has an associated Trunk Group.
- If the call or other signal needs to be sent to it, the DEFINITY[®] system uses the associated D-Channel to communicate with the remote terminal.
- When a call arrives over a D-Channel which carries X-Mobile traffic, that call is analyzed to check if an X-Mobile remote station originated it.
 - If it was, then the call is re-associated in call processing as an internally originated call from the local X-Mobile station.
 - Allows a large number of DEFINITY[®] system features to work for the remotely controlled station, including bridging.
- Icon-based message waiting enables X-Mobile WT users to check for messages while idle instead of the display-based message waiting feature that only informs status while on a call.
 - When message is waiting at an idle X-Mobile WT, its switch-hook icon flashes.
- Both types of message waiting indication can co-exist in a system.

Administration

Enabling X-Station Mobility

- New field Maximum XMOBILE Stations is created on the Customer Options form and defaults to 0.
- When this field is set to 0, the feature is disabled.
- It is only possible to set the field to $N > 0$ if the system is administered as R8 or later.

- This customer option is disabled by default for new R8 systems and for older systems that upgrade to R8.

Change system-parameters customer-options

```
change system-parameters customer-options                               Page 1 of 5

OPTIONAL FEATURES

                G3 Version: V8                                Maximum Ports: 0
                Location: 1                                  Maximum XMOBILE Stations: 0
Abbreviated Dialing Enhanced List? N                                CAS Branch? n
    Access Security Gateway (ASG)? n                                CAS Main? n
    Analog Trunk Incoming Call ID? n                                Cvg Of Calls Redirected Off-net? n
A/D Grp/Sys List Dialing Start at 01? n                            DCS (Basic)? n
Answer Supervision by Call Classifier? N                            DCS Call Coverage? N
```

- The station administration form is changed to allow the station type of XMOBILE.
- An X-Mobile station has the following new fields:

Field	Range	Default	Description
Trunk Group	1 to Max Trunk Groups	1	The trunk group to be signaled
Message Waiting Type	NONE, DISP[LAY], ICON	ICON	Update X-Mobile Display for MWI
Length of Display	1, 2	1	Number of lines in the display

- All other fields are the same as for 2500 series stations.

Change station

```
change station 3000                                                    Page 1 of 3

                STATION

Extension: 3000                Lock Messages? n                BCC: 0
Type: XMOBILE                Security Code: _____        TN: 1
                Coverage Path 1: ____        COR: 1
Name: Vorry's Mobile        Coverage Path 2: ____        COS: 1

Mobility Trunk Group: 1
Message Lamp Ext: 3000
Message Waiting Type: ICON

Length of Display: 1
```

Station form indicates MWI update type

- The Message Waiting Type field on the station form indicates the mode of MWI update to the WT.
- The MWI activation/deactivation message is used when the field is set to ICON.
- The MW is attached to the end of the display line during an incoming or outgoing call if the field is set to DISP.
- No MWI update is sent if the field is set to NONE.

Mobility Trunk Group field on station form changes

- This Mobility Trunk Group field is updated dynamically.
- If the field is changed through the change command instead of the add command, the following warning message appears on the SAT:

Overriding this field may cause failure in delivering incoming calls to this station.

- In a multiple DAS configuration, the Mobility Trunk Group field may change without user intervention after the last form submittal.
- As an X-Mobile WT roams among the DASs, it sends a location registration indication message to the DEFINITY[®] system through the signalling link of the trunk group associated with the particular DAS.
- This informs the DEFINITY[®] system to deliver incoming calls to the station through the updated trunk group's signalling link.
 - If this is overridden, incoming calls may be sent to a DAS that the X-Mobile WT is not currently registered to.

Signaling Group administration

- New field has been added to the signaling group form to indicate the type of X-Mobile endpoints allowed for it (DECT or NONE).
- The default for the field is NONE.

add signaling-group

```
add signaling-group 1
SIGNALING GROUP
```

Page 1 of 5

```
Group Number: 1
  Associated Signaling? y
  Primary D-Channel: 01b1216
Trunk Group for Channel Selection: 1
Supplementary Service Protocol: a
```

```
Max number of NCA TSC: 0
Max number of CA TSC: 0
Trunk Group for NCA TSC: _
X-Mobility/Wireless Type: DECT
```

- If the administered software version is V5 or earlier this feature cannot be enabled.
- The only form that includes new fields is the signaling group form. The new field is defaulted to none on upgrades.
- The feature must be enabled from the customer options screen. If number of X-Mobile Stations is set to $N > 0$, then X-Station Mobility is enabled for this system.
- X-Mobility functionality requires that ISDN PRI is also enabled.

Capacity Constraints and Feature Limitations

- The X-Station Mobility only works when stations are remote over ISDN trunks.
- An X-Mobile station counts as a wired DEFINITY® system station in terms of station limits and configuration consideration.

Call Processing

Basic Call Termination

- A call for an X-Mobile station looks up the current trunk group associated with the X-Mobile station.
- It then extends the call over the trunk group to the remote station.

NOTE:

For PRI trunks, this involves sending a setup message with a calling party number and a called party number indicating the extension number of the X-Mobile WT.

Basic Call Origination

- A call from an X-Mobile station appears to DEFINITY® system Call Processing to have been locally originated by that station.
- DEFINITY® associates call origination to X-Mobile stations by analyzing the CALLING PARTY NUMBER Information Element in the SETUP message received over an ISDN trunk marked as having X-Mobile endpoints and matching the number with the extension of an X-Mobile station.

NOTE:

The X-Mobile station is modeled as an analog station as far as a DEFINITY® system's internal call structures are concerned. The calling party number is the X-Mobile station's extension number. Digit modification is not required.

Flash Operation

- If an INFO message containing a KeyPad Facility Information Element of the value 0x15 or values 0x1B, 0x48, the message is interpreted as a Flash-hook operation from the remote station.



NOTE:

These values are defined by the ETSI DECT GAP standard and the RCR-28 PHS standard.

Bridged Appearances

- An X-Mobile station can have bridged appearances on wired stations, or be a bridged appearance of a station.
 - Those bridge appearance status LEDs reflect the status of the X-Mobile station.
- The X-Mobile station is able to bridge on or off calls at will, just like any bridge of a directly connected wired station.



NOTE:

When a wired phone has a bridged appearance of an X-Mobile, it is an analog bridge.

When the X-Mobile is a bridge of another phone, it is a digital button bridge.

Call Progress Tones

- When an X-Mobile station is on a call, the DEFINITY[®] system provides local call progress tones for that call
 - Dial Tone
 - Alerting
 - Coverage Tone
 - Busy
 - Re-order
 - Custom Alert
 - Call Waiting Tone
 - Call Prompting Tone
 - Confirmation Tone
 - Stutter Dial Tone

Display

- When an X-Mobile station is ringing or active, an ISDN Info message is used to update its display whenever appropriate.
- The display for an X-Mobile station is limited compared with standard DEFINITY[®] system displays.
- The length is limited to 15 characters.
- Other than the optional message waiting indication, display information is limited to calling/called party names or numbers.
- There are no redirection indications nor conference displays.

Called Party Name

- When an X-Mobile station calls another party, the DEFINITY[®] system provides to the X-Mobile WT the name administered for the called party via an ISDN info message.

Connected Party

- When call processing would normally update a station's display with text information (such as the connected party), it sends that information to the X-Mobile station in an INFO message contained in a DISPLAY Information Element.

Calling Party Number

- Calling Party Number is sent to the X-Mobile station when available.

The calling party number is not displayed when the number is restricted from display as defined by DEFINITY[®] system translation.

X-Mobile to Trunk mapping

- There is no limit to the number of X-Mobile stations that can be supported per Trunk Group.

NOTE:

The number of simultaneous calls supported by the interface is limited to the number of trunks assigned to an X-Mobile trunk group.

Call Coverage

- The X-Mobile station is allowed to cover calls not answered by the X-Mobile, either Ring No Answer (RNA) or Subscriber Absent, when coverage is administered for the X-Mobile station. An X-Mobile station is also allowed to be a coverage point.

Calling Party Name

- The X-Mobile station sends the calling party name information in a DISPLAY Information Element whenever normal call processing for that station would update the display of the station to indicate the calling party.



NOTE:

After connect, an INFO message overrides the handset operation of the display in order to keep the handset display current.

Call Transfer

- When an X-Mobile station indicates a flash during a call, it is treated as initiating a transfer operation.
- The operation is defined by DEFINITY® system's standard operation for transfer from an analog phone.



NOTE:

Transfer is completed with an onhook button press.

Call Forward

- An X-Mobile station is allowed to activate and deactivate call forwarding (all, busy, don't answer) just as if it were a directly connected station.
- The remote station state is used to determine if call-forwarding criteria are met.
- If the remote station is busy, then busy criteria is met.
- If the remote station does not answer the call, RNA criteria is met.

Terminal Type

- Setting the type in the station screen to XMOBILE will identify an X-Mobile station.

Calling Capabilities

- Administration of an X-Mobile station allows an administrator to customize call options, based on the parameters for an analog station.

DTMF Generation

- When an X-Mobile station is on a call, and INFO messages are received indicating KeyPad numeric button pushes, DTMF is generated on the call.



NOTE:

Digits are stimulated as CP_DIGITS to the Callp, which eventually are translated to ESIG_EXDs and invoke gm_esig to generate DTMF tone.

Call Progress Tones to Remote Party

- When a party dials to an X-Mobile station, the DEFINITY® system provides ring-back, busy, or re-order tone as appropriate.
- If the remote party is put on hold by the X-Mobile station, hold tone or music on hold is provided as administered for the system.

Call Waiting

- When a call arrives for an X-Mobile station, and Call Waiting is administered, Call Waiting Tone is applied, as defined by the DEFINITY® system's operation for call waiting for analog phones.

Conference

- If an X-Mobile station has initiated a second call by a previous flash operation and the third party has answered the second call, then a flash operation will conference all three parties.
- Another flash operation by the X-Mobile station will DROP the last added party.
- A Disconnect indication from the X-Mobile station while on a conference will leave the remaining parties connected and idle the X-Mobile station.
- The operation is defined by DEFINITY®'s treatment for conference by an analog station.

⇒ NOTE:

Applies only to second (inquiry) calls originated by the X-Mobile, it does not apply to call waiting calls.

Call Park/Answer Back

- An X-Mobile station is able to park its current call after a flash operation in the normal method by feature access code.
- An X-Mobile station is able to answer a parked call by using the Answer Back feature access code.

Call Shuttle

- If administered, an X-Mobile station is alerted via call waiting tone when another incoming call arrives for it while the X-Mobile station is on a call.
- The X-Mobile station can accept the second call via a flash operation.
 - This will put the first party on soft hold.
- Subsequent flash operations will shuttle between the second and first parties.
- A disconnect operation in this case, where a party is being soft held, results in clearing the current call, and recalling the X-Mobile station to attempt to reconnect the soft held party.

Display-based Message Waiting Indication

- All text information sent to an X-Mobile station contains information indicating a message is waiting for it, if the station is marked as having a message waiting.
- Noted by an MW at the end of the display line.
- Content of the string is administerable.



NOTE:

The display IE is used to indicate message waiting indication and is sent anytime the display updates.

Distinctive Ringing

- All calls presented to the X-Mobile station as internal calls use a numbering plan code of 0x30 (network specific) in the calling party IE.
- For external calls, the numbering plan code is one of the standard public numbering plan values.
- As an example, 0x21 is national ISDN.

Send All Calls

- Able to activate send all calls for itself by feature access code.

Terminating Extension Groups

- Allowed to be a member of a terminating extension group.

Hunt Group

- An X-Mobile station is allowed to be a member of a hunt group.

Coverage Answer Group

- An X-Mobile station is allowed to be a member of a coverage answer group.

Abbreviated Dial

- An X-Mobile station is allowed to use its station abbreviated dial lists and system abbreviated dial lists.

Call Pickup

- An X-Mobile station is allowed to be in a Call Pickup Group.

Process

Idle X-Mobile WT's switch-hook icon flashes when message is waiting

- When a message is left for an X-Mobile Station either through a messaging adjunct (for example, INTUITY™ AUDIX®) or one of the DEFINITY® system-based features (for example, Leave Word Calling), the X-Mobile WT's message status indication is updated immediately.
- If the X-Mobile WT is idle, the switch-hook icon flashes.
- If the X-Mobile WT is offhook on a call, then the switch-hook icon starts flashing immediately after it goes onhook.
- The DEFINITY® system sends an MWI Activation message to the corresponding DAS whenever there is a change in the number of messages waiting at the X-mobile station resulting in one or more messages waiting for the user.
- Depending on the type of message server, the message may not be sent with every change in status unless it results in a change in the indicator to an on-state.
- The number of messages field in the MWI activation message is always set to 1 regardless of the number of messages.
- The MW status is stored locally on the X-Mobile WT.
- It starts flashing the switch-hook icon immediately after the station goes onhook.
- The MWI Activation message is sent to the control link that the DEFINITY® system received the last location registration indication from the X-Mobile WT.
- This should handle the configuration where more than one DAS are connected to the DEFINITY®.
- Idle X-Mobile WT's switch-hook icon remains off when no message is waiting.
- When a message is deleted from an X-Mobile Station resulting in no more message waiting for the station, the flashing switch-hook icon on the idle X-Mobile WT is turned off immediately. If the X-Mobile WT is offhook on a call, the switch-hook icon will not flash after the station goes onhook.
- DEFINITY® sends an MWI Deactivation message to the corresponding DAS whenever there is a change in the number of messages waiting at the X-mobile station resulting in no more messages waiting for the user.
- Depending on the type of message server, the message may not be sent with every change in status unless it results in a change in the indicator to an off-state.
- The number of messages field in the MWI Deactivation message is always set to 0.
- The MW status is stored locally on the X-Mobile WT.

- It will not flash the switch-hook icon after the station goes onhook.
- MWI operation follows ASN.1 specification in ETS 300 745-1.
- MWI Activation/Deactivation messages are conveyed through FACILITY messages.
- The DEFINITY[®] system sends the MWI Activation/Deactivation messages in a connectionless mode through the ISDN FACILITY messages.
 - This type of message does not require an associated call.
 - Call reference value is always set to 0x00.

Interface supports immediate mode only

- The DEFINITY[®] system supports the immediate mode MWI only.
- The messages are sent immediately upon a change in the number of message waiting at the X-mobile station.
- Depending on the type of message server, the message may not be sent with every change in status unless it results in a change in the indicator state.
- Does not retransmit the MWI status during a call origination.
- The deferred or combined modes are not supported.
- The DEFINITY[®] system refreshes MWI status during location registration.
- Upon successful location registration, the X-Mobile WT's MW status is refreshed.
- While idle, its switch-hook icon starts flashing if one or more messages are waiting at the station.
- After successful location registration with the X-Mobile WT, the DAS sends a location registration indication message to the DEFINITY[®] system.
- Upon receipt of the message, the DEFINITY[®] system sends an MWI Activation or Deactivation message back to the DAS depending if message is waiting at the WT.

Memory Impact of Capacity Changes

- The X-Mobile capability is administered on the station admin screen.
- There is a moderate user manager memory impact.

Maintenance

Status Station

- An X-Mobile station will display switch-hook state and connected port information.

- The display is the same as a standard analog station with one exception.
- When the station is onhook, the port field is blank.
- When the station is offhook the port field contains the identity of the connected ISDN trunk port.

DEFINITY[®] refreshes MWI hourly

- The DEFINITY[®] system runs an hourly MWI refresh (also known as audit) on a time available basis.
- It limits the refresh rate to no more than 6 MWI messages per second.
- The foreground traffic is assumed to be negligible as the refresh is run at a time available basis.
- At this rate, the DEFINITY[®] system does not overload the DAS's worst case MWI message handling capacity of 6 MWI messages per second.

Signaling

- ISDN PRI is modified to allow INFO messages with Display Information Elements to be sent at any time.
- ISDN PRI is also modified to allow INFO messages with Keypad Facility Information Elements to generate DTMF when the feature is enabled.
- The X-Mobile station is treated as an analog station in terms of feature interactions except as noted in previous requirements.

Call Detail Recording

- A Call origination from an X-Mobile WT appears in the CDR as a call origination from the X-Mobile station.
- A Call Termination to an X-Mobile WT appears as a call termination to the X-Mobile

Performance Impact

- There is a minimum impact on performance (similar to a BRI call).
- When a new call is originated from a trunk group marked as allowing X-mobility, it is analyzed to check if the origination is from an X-Mobile station.
 - If it is, the origination is treated as a local station origination.
 - The check yields the UID of the X-Mobile station if successful, the additional processing is minimal.

Scenarios

- All feature operations are intended to work exactly as if the X-Mobile station were directly connected to the DEFINITY[®] system.
- Analog-type feature operation was chosen, since most wireless terminals support only flash operations and are not multiappearance or context sensitive.
- In almost all cases, the operation expected for an X-Mobile station is identical to a similarly administered analog station.
- All flash (recall) operations behave identically to an analog station administered for flash answer back, including those that result in re-origination.
- In case of far-end (forward) disconnect, an X-Mobile station is forced onhook; therefore, there is no possibility of re-origination or hearing of tone (busy or intercept) after a far-end drop, as happens when a wired analog station remains offhook.

Telephony Applications - International

Reset Shift Call

When a station call is made to a busy station, reset shift dial tone, called second dial tone by the Japanese PBX manufacturers, is heard. During reset shift dial tone, a single digit may be entered, which replaces the last digit of the originally dialed extension, and the call is offered to the new station.

The length of the reset shift dial tone is administrable on a system basis and should have a range of 0-255. It is normally set to 10 seconds. Japanese PBXs can give this tone for up to 1 or 2 minutes before reverting to busy tone. If call coverage is supplied for the incoming number, the call goes to coverage as normal. If the coverage path is busy, the reset shift dial tone is heard and a single digit may be entered as described above. This feature is only active for station to station calls, not for incoming or outgoing calls. More specifically, the originator and destination of the call must both be stations connected to the DEFINITY[®] system. In DCS or QSIG environments, both stations need not be connected to the same DEFINITY[®] system, but must be on the same DCS or QSIG networks. Operation for transfer and conference is similar. If, while one is setting up a conference or transfer, the dialed party is busy, reset shift dial tone is heard and an opportunity is provided for a last digit reentry as on a normal call.



NOTE:

Multi-function analog terminal (MFAT) sets do not receive reset shift dial tone.

Performance Impact

This feature uses a port on a tone detector the entire time that a station is offhook and listening to reset shift dial tone. This means that whenever this feature is assigned, additional tone detector circuit packs may be needed. The number of additional tone detector circuit packs will vary depending on the following things:

- The percentage of calls that encounter busy.
- The average length of time that station set users will wait, listening to reset shift dial tone, before they dial a digit.
- The administered maximum duration of reset shift dial tone.

Requirements

- TN2182 tone/clock circuit pack, any suffix or vintage, or
- TN780 tone/clock circuit pack, vintage 2 or later.

This feature applies to all hardware platforms.

 NOTE:

Since Japan does not have regulations on the characteristics for reset shift dial tone, users require training to properly identify the tone that DEFINITY® ECS uses for this feature. If the PBX is being used by people with limited dexterity, the timer associated with this feature is set to a large value, to give such users enough time to press the extra digit.

Call Processing

Reset shift feature operation

Play reset shift dial tone on busy calls if:

- A station or attendant originates a call, and
- The dialed endpoint is in the dial plan and is one of the following types:
 1. Station extension (extension)
 2. Prefixed extension (pextension)
 3. AAR (aar)
- The destination uses out-of-band signaling — either CCMS or ISDN trunk signaling — to signal “busy” back to the caller, then the calling set will receive reset shift dial tone.

Out of band signaling operation

Play reset shift dial tone on busy calls for non-ISDN trunks that use out of band busy signaling. This includes DCS trunks using out of band signaling such as X.25 or TCP/IP. Subsequent reset shift dial attempts if a user receives reset shift dial tone, enters a single digit, and the resulting destination is busy, the calling set will receive reset shift dial tone again and restart the reset shift timer.

Turn off special dial

The PBX turns off reset shift dial tone when it receives a dialed digit from the station user. If a station user is listening to reset shift dial tone and the Reset Shift Timer expires, the station user receives busy tone.

 NOTE:

The reset shift timer can be set to 0, in which case busy tone is sent instead of reset shift dial tone.

Maintain previous operation

When Reset Shift Timer field on systems parameter form is set to 0, operation will revert as if the Reset Shift Call Feature was not implemented.

Administration

Initial Administration

To administer the reset shift call feature on a DEFINITY[®] system where it was not previously used, the following procedure should be utilized:

1. In order to fully use the reset shift call feature, station extensions should be assigned with contiguous numbers for groups of users. Thus, users with similar job functions should have extensions that differ by only the last digit.
2. Set the Reset Shift Timer (seconds): field on the system parameters form to a value other than 0. The value entered in the Reset Shift Timer (seconds): field is the number of seconds that reset shift dial tone will be audible before busy tone is heard.

Removing Administration

To disable the reset shift call feature on a DEFINITY[®] system where it is active, set the Reset Shift Timer (seconds): field in the system parameters form to 0.

Timer administration

The system parameters features form has a new field for setting the reset shift call timer. The new field is called Reset Shift Timer (seconds). The timer will have a range between 0-255.

On an upgrade from a previous release to R8, the Reset Shift Timer field defaults to 0, disabling the feature.

The Reset Shift Timer (seconds): field is on the feature related system parameters form. It specifies the number of seconds that a station user can listen to reset shift dial tone and have the opportunity to launch a new call by pressing a single digit.

```
change system-parameters features Page 5 of 10
      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): 160
      Reset Shift Timer (seconds): 0
      Station Call Transfer Recall Timer (seconds): 0
      DID Busy Treatment: tone
      Allow AAR/ARS Access from DID/DIOD? _
      Allow ANI Restriction on AAR/ARS? _ Transfer Upon Hang-Up? n
      Abort Conference Upon Hang-Up? n Abort Transfer? n
      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: _
```



NOTE:

Japan does not have national regulations covering the cadence, frequency, or level of the reset shift dial tone. However, the de-facto standard is to mimic Japanese PBX operations.

Reset shift dial tone frequency

The reset shift dial tone has a frequency of 400 Hz $\pm \pm \pm 20$ Hz. This requirement is based on the Fujitsu switch documentation for center frequency, and the acceptable range is based on the NTT ranges 3 specified for the public network.

Reset shift dial tone cadence

The reset shift dial tone has a repetitive cadence of 100 ($\pm \pm \pm 20\%$) milliseconds of silence, followed by 150 ($\pm \pm \pm 20\%$) milliseconds of tone, followed by 100 ($\pm \pm \pm 20\%$) milliseconds of silence, followed by 650 ($\pm \pm \pm 20\%$) milliseconds of tone.

This requirement is based on the Fujitsu switch documentation for cadence, and the acceptable range is based on NTT specification for the public network.

Reset shift dial tone level

The reset shift dial tone has the same level as the currently implemented Japanese dial tone $\pm \pm \pm 2$ dBm. The range for this requirement is also obtained from the NTT Tone board support.

The reset shift dial tone is provided by systems equipped with either the TN780 or the TN2182B circuit packs.

Terminals

This feature applies to all sets with the following exception: multi-function analog terminal (MFAT) sets can not receive reset shift dial tone, although they can be the targets of a reset shift call from another terminal type. 7303S, 7305S, 7303H01A, 7305H01A, 7305H02A, 7309H, 7313H, 7314H, 7315H, 7316H, 7317H, MDC9000, and MDW9000 are MFAT sets.

Call Detail Recording (CDR)

If the following CDR SYSTEM PARAMETERS fields, Record Outgoing Calls Only? and Suppress CDR for Ineffective Call Attempts?, are both set to n, each call attempt made by the reset shift call feature will be recorded in CDR as a separate record containing the full number — not just the last digit.

Call Center

Overview

- ASAI/CTI Enhancements
 - Capacity Increases for ASAI Multiple CTI Application Control Support
 - Active Route Requests – from 126 to 4,000
 - Split/Skill Third Party Domain Controls – from 300 to 2,000
 - ASAI Pending Work Mode Changes
 - ASAI/CTI Regression Testing
 - ASAI DEFINITY[®] Switch Software Version Query
 - ASAI Trunk Group Identification Information in Event Reports
 - CentreVu Computer Telephony on MAPD - NT (co-resident with DEFINITY[®] LAN Gateway)
- Advocate Related Enhancements
 - Logged-In Advocate Agent Counting
 - Standard Advocate Reports
- ASCII Agent Administration Support
- ATM Support
 - Increased Call Center capacities
 - Measured VDNs – from 8,000 to 20,000
 - Hunt Groups – from 600 to 999
 - Members/group – from 999 to 1,500
 - Queue Slots – from 15,000 to 25,000
 - Vectors – from 512 to 999
 - Recorded Announcements/Audio sources – from 256 to 1,000
 - Site Stats for ATM Connected Remote EPNs
 - CMS Measurement of ATM Trunks
- Call Center Release Control
- CALLMASTER V (CC 6416D+) Native Support
- DEFINITY[®] DoLAN IP Trunking Testing
- PASTE Update
- Support for CMS High Availability (Dual CMS Links)

- CMS High Availability
 - Dual CMS systems
 - Manual alarm clears
 - Historical log
 - Non-disruptive cmsadm backups
 - Reasonable restores
 - Non-disruptive maintenance backups
 - External Call History alarms
- R3V8 CMS not a bug fix for R3V6
- R3V8 CMS does not support G2.2 or G3V1
- V8 Visual Vectors not a bug-fix for V1
- DEFINITY® ACD becomes “general PBX” call preserving
- Migration from R2 CMS to R3V8 supported
- ASCII agent admin interface restored in R3V6
- Windows 98 supported
- Simplified Chinese translations will be available after other languages
- Solaris version 2.7 nomenclature becomes “Solaris 7”
- MAPD hardware for NT Computer Telephony will be TN801C

Call Center - ASAI DEFINITY Software Version Query

- Provides ASAI applications with the capability to obtain the DEFINITY software version running on the system and the administered field version.
- Applications previously had no way of obtaining this information.



NOTE:

There is no new administration required. For R8, Administered Field Version and DEFINITY Software Version will always be present.

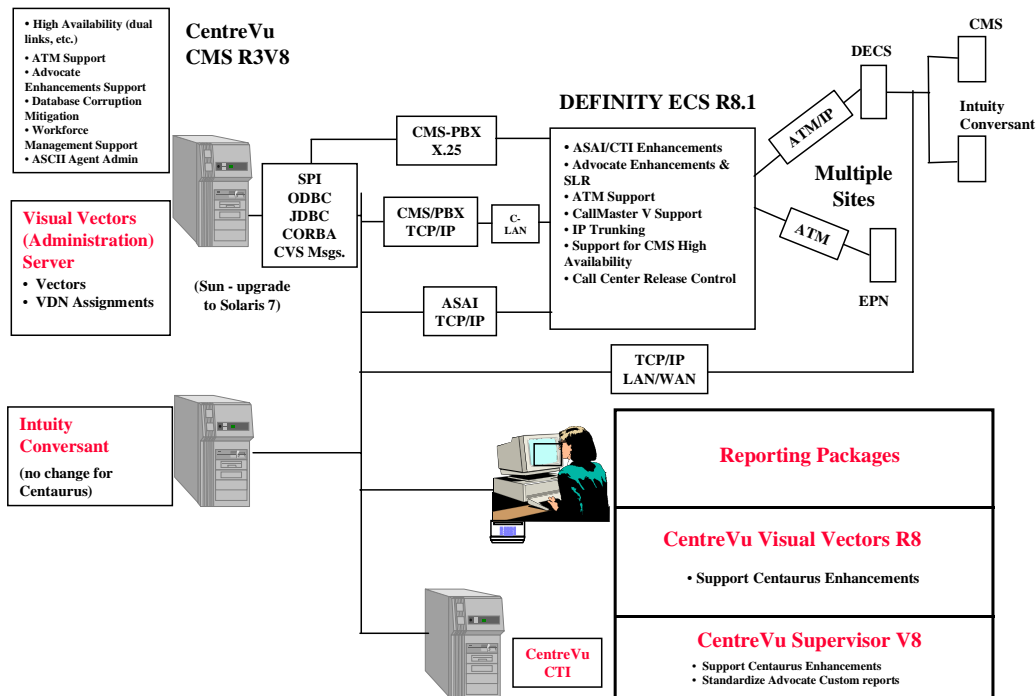
CallVisor

- CallVisor PC API supports the switch version query so that the G3PD can access it.

CentreVu

- CentreVu Telephony Server API supports the software version query and the G3PD uses it to obtain the software version.

Call Center Design Graphic



Call Center Release Control

- This is a field provided on the Customer Options form to set the Call Center release purchased by the customer.
- The release is indicated in this field using DEFINITY® release nomenclature down to the dot release setting (R8.1).



NOTE:

The field can not be set to a release greater than the software installed.

- Release field is used by new call center enhancements and feature software (except ASAI) to control whether the enhancement/feature is to be made available to the customer.
- The requirements for the enhancement/feature specifies which release settings the enhancement/feature will be available with.
- Some enhancements/features also require a Customer Option field to be active.

Call Center - ASAI Trunk Group Identification

- Provides the application with the capability to obtain Trunk Group information regardless of whether Calling Party Number is reported.
- Prior to this release, trunk group information (which includes trunk group number and trunk member number) was provided in ASAI event reports for inbound calls, but only if calling party number was not known.
 - If the network did not provide ANI, then ASAI reported trunk group information.



NOTE:

Refer to DEFINITY ECS CallVisor ASAI Technical Reference, 555-230-220 for a complete list of terms and acronyms relating to ASAI.

Usability

- Useful to recording applications that record at the trunk side of the connection rather than at the station side.
- These applications will be able to associate a particular call with a specific trunk member of a trunk group to trigger the recording device.
- Examples of such applications are those provided by NICE Systems and Dictaphone.

Applicable Messages

- Trunk group information passed in the following messages: Call Offered, Alerting, Connected, Conference and Transfer events.
- Trunk group information also passed in the Adjunct Route Request.
- If there are trunks involved with the call, then Conference and Transfer events will contain trunk_ids for the trunks.
- The events will contain as many trunk_id parameters as there are trunks on the call. There will be a one-to-one correspondence between a Connected Number IE containing ##### and a Trunk ID IE.

PCOL Trunk

- The trunk group information provided for a PCOL trunk corresponds to a PCOL group administered on the system.
- The PCOL group number could be the same as another, existing trunk group on the model.
- To be able to distinguish between the two groups, a new number (other than the numbers already administered for the Trunk Groups on the switch) is used for the PCOL trunk group.

Call Center – ASAI/CTI Pending Work Mode Changes

- This feature provides ASAI applications with the capability to change the current work mode of an agent while the agent is busy on a call by making it a pending change, which takes effect as soon as all current calls are cleared.
- Previously, ASAI did not allow an application to change the work mode of an agent.
 - Applications got a negative acknowledgment indicating that the agent was busy.
 - If the request was to change the work mode of the agent from any other mode into a different work mode, including AI and MI, the DEFINITY® NAKs the Request Feature message with cause value “user busy”.
- Feature addresses the needs of customers/applications that need to allow changing the agents’ work mode while they are busy on calls.
- The type of applications that benefit from this ASAI capability are those that function in an ICM-OCM call blending environment.
 - Call Blending Applications by Davox and EasyPhone, are examples of this type of applications.
- Feature will allow call blending applications to “reserve” an inbound agent by requesting to change the work mode of that agent from the AI work mode to the AUX or ACW work modes while the agent is on a call.
 - Prevents additional calls to be delivered to the agent upon processing a current call.
 - As soon as the agent finishes processing a current call, the Agent Work Mode Feature Request message
- Ensures that existing customer applications will work correctly in the R8 software environment and, in turn, protects the customer’s application investment.
- Modified to allow the adjunct application to specify that it can support pending work mode changes.

CallVisor PC

- CallVisor PC, both Unix and NT versions support Pending Agent Work Mode Changes within the response to Agent Status Query, as well as an indication of a pending reason code, as appropriate.
- CV LAN interfaces (client and server) provide identical support.

Cross-Product Compatibility/Commonality

Terminals

- If there are feature buttons on an agent's phone for AUX, AI, MI and ACW work modes and an ASAI Change of Work Mode request is received by DEFINITY® for an agent with active calls, the lamp for the current work mode button will stay lit while the lamp for the pending work mode will flash.
- If there is a pending work mode change at an agent's phone previously triggered by ASAI, then a button press for a different work mode will override that particular pending change and the work mode pertaining to the button will be pending.
- If the agent decides to go to the AUX work mode and presses the AUX button, and then the switch will process the button press and have the agent's pending work mode change become AUX instead of ACW.
- Any Agent Status Value Query received by the switch after the AUX button was pushed will show the AUX work mode as a pending change.

CALLMASTER V (Call Center 6416D+) Native Support

- Adds the CALLMASTER V as a supported terminal type with a specific administration screen.
- This terminal is the Call Center edition of the global 6416D+ DCP terminal
 - Headset jacks replace the handset
 - Removes speakerphone microphone
 - Adds recorder interface
 - Meets European requirements.

The native support includes voice path enabling/disabling (muting while agent is waiting for a call) in the same manner as the previous CALLMASTER terminals.

Call Center - ASAI Capacity Increases for Multiple Application Support

- Capacities increased in DEFINITY® R8 for Third Party Domain Control of ACD Split/Skill domains and the number of Adjunct Route Requests that can be outstanding (active) at any point in time.
- Capacity increases allow more ASAI applications to better perform ACD Split/Skill monitoring and adjunct routing tasks with the DEFINITY® system.

- Increases for these capability groups are:
 - Third Party Domain Control of ACD Split/Skill extension domains
 - Number of associations - per extension domain from: 1 per domain in R7 to 4/8 per domain in R8csi/R8r
 - Number of associations - system-wide limit increase from: 99/600 in R7csi, R7si/R7r to 300/2000 in R8csi/R8r
 - Adjunct Route Requests
 - Number of active associations - system-wide limit: In R8csi, the system-wide limit will be set to 500 active associations at any one time. The R8r system-wide limit of 4000 active associations at any one time.
 - Number of active associations - per ASAI link: limit per link increases for R8csi up to 500 active associations per link while the R8r will support up to 4000 active associations per link.
 - As a result of the increase in the system limit for Adjunct Route Requests, the system limit for Adjunct Control Associations (Feature Requests, Value Queries, and Third Party Call Control) changes.
 - Active Adjunct Control Associations increases to 600/6000 in R8csi/ R8r.



NOTE:

These associations are shared with the Adjunct Route Request capability.



NOTE:

Customers are strongly encouraged to transition out of BRI and/or the DEFINITY LAN Gateway (DLG) running on the Multi Function Board (MFB) and into the DLG running on the MAPD board to avoid ASAI link congestion due to the system packet bus overflow.

- In order to reach the maximum number of outstanding Adjunct Route associations per link (4000); the CRV length must be administered for two bytes.

Administration

- The new capacities are available only if the system is administered as an R8 (V8).
- Adjunct Route new capabilities require the ASAI link version to be version 4.

Display Internal Data Hunt Group Command

- Updated for capacity increases for split/skill domain control.
- The User Ids for the controlling links added

Display internal-data hunt

```
display internal-data hunt-group 1 Page 1 of 3

HNT_GRP hnt_tbl[ 1] for 00820001
grp_typ:ucd-mia num_memb:0 mbr_min:1000 mbr_max:0 anncl:00000000 delay1:-1
annc2:00000000 delay2:-1 max_q_leng:1 ncalls_qwarn:-1 uid_ncalls:00000000
cid_ncalls:00000000 uid_time:00000000 cid_time:00000000 ns_ext:0
ns_uid[0]:00000000 ns_uid[1]:00000000 ns_uid[2]:00000000 noans_uid[0]:00000000
time_qwarn:-1 inthresh:999 supv:00000000
asai_linkst:n meas_by:none mch_type:none changed:n eanswer:n
data_group:n sls_option:n sls_state:0 isdn_dis:null
free_h:-1 free_t:-1 used_h:-1 used_t:-1 mbr_h:-1 mbr_t:-1
ctrl_uid_asai:00000000 adj_link:0 adj_idx:00 dom_asai_uid[0]:00000001
dom_asai_uid[1]:00000000 dom_asai_uid[2]:00000000 dom_asai_uid[3]:00000000
dom_asai_uid[4]:00000000 dom_asai_uid[5]:00000000 dom_asai_uid[6]:00000000
dom_asai_uid[7]:00000000
notf_asai_uid[0]:00000000 notf_asai_uid[1]:00000000 notf_asai_uid[2]:00000000
notf_asai_uid[3]:00000000 notf_asai_uid[4]:00000000 notf_asai_uid[5]:00000000
wat[low]:737280 wat[med]:737280 wat[high]:737280 wat[top]:737280
watinit:180 ewt_flags:00
asa:0 wrk_agnts:0 noans_intvl:0 noans_dest:00000000
rona_agent:00000000
cf_dest: srvc_lvl:0 tacw:0 sls_1_thresh:0 sls_2_thresh:0
```

List Hunt Group Command

- Updated to support the R8 capacity increases for split/skill domain control.

List Hunt Group

```
list hunt-group
Grp Grp
No. Name/      Grp  ACD/      Que No. Cov  Notif/  Dom  Message
Ext           Type MEAS Vec  MCH  Siz Mem Path Ctg Adj  Ctrl  Center
1   Sales New
    1500      ucd-mia y/N   y  none 1  0      C 1011  1000 n
                                   1001
                                   1002
                                   1003
2   Customer Service
    1501      ucd-mia y/N   y  none 1  0      n      1004 n
```

Feature Interactions

The increase of the system-wide limit of active Adjunct Route Requests does not impact other features

Call Center-Logged in Advocate Agent Counting

- Modified in the CentreVu Advocate R 8.1 DEFINITY.
- The “Logged-In Advocate Agents” customer option that was added in DEFINITY R7.1 modified to only count agents towards the Advocate agent limit if:
 - Percentage Allocation or a Reserved Skill is assigned to the agent's Login ID
 - Or
 - One of the agent's skills is assigned LOA or Service Level Supervisor. Checking whether the agent login ID is assigned as AAS (Auto Available Skill/Split) or AUDIX does not need to be done.

Feature Interactions

- Interactions with the Add and Remove Skill features.
 - If a skill is added or removed using the Add and Remove Skill features, and this change causes a CentreVu Advocate agent to no longer be a CentreVu Advocate agent, or vice versa, the count of Logged-In CentreVu Advocate agents is adjusted accordingly.
- CentreVu Advocate and Best Service Routing Reports have no Feature interactions.

Installation

- Requires the installer to set logged in Advocate Agent limit field to the value that the client has purchased.
- The CentreVu Advocate and Best Service Routing Reports of this feature do not impose any special requirements upon the CentreVu CMS or CentreVu Supervisor for installation.
- The new and modified reports are included in the installation software CD.

ATM_CMS

Background

- ATM trunks support the emulation of T1/E1 ISDN-PRI.
 - Can be used as access, tandem or tie trunks
 - Cannot be used to interface with the public network
- ATM trunking uses the ATM Trunk Interface which is supported by the TN2305 multi-mode fiber pack and the TN2306 single mode fiber pack.

- The advantage to a call center of using ATM trunks is that this type of trunk can be a cheaper alternative to the existing T1/E1 option.
- ATM WAN PNC provides the ability to interconnect port networks using an Asynchronous Transfer Mode (ATM) wide-area network (WAN).
- Each distinct DEFINITY® ACD can be converted to one or more EPNs and interconnected across the ATM WAN to make a single, large DEFINITY® ACD.
- EPNs can be dispersed across the country. A single DEFINITY® system can span different time zones and/or dial plans.
- Locations/location numbers are provided through an optional DEFINITY® Multiple Locations feature.
- DEFINITY® can support up to 44 locations/port networks (1 through 44).
- No enhancements are added to BCMS to support the added capacities and site specific reporting.



NOTE:

Changed capacities are bolded.

	DEFINITY® R7.1r Category A (ECS & ProLogix)	DEFINITY® R7.1r Category B (BCS & GuestWorks)	R3V6 CMS	DEFINITY® R8.1r Category A (ECS & ProLogix)	DEFINITY® R8.1r Category B (BCS & GuestWorks)	R3V8 CMS
Total Assignable VDNs on Switch	20,000	20	NA	20,000	20	NA
CMS Measured VDNs (external or both)	8,000	Not Applicable (CMS not available)	8,000	20,000	Not Applicable (CMS not available)	20,000
Announce- ments (extensions)	256	256	256	1000	1000	NA
Hunt Groups (splits/skills)	600	600	1000	999	999	1000
Max. Members per Hunt Group	999	999	NA	1,500	1,500	NA
Hunt Group Queue Slots	15,000	15,000	NA	25,000	25,000	NA
Max. Queue Slots for a Hunt Group	999	999	NA	999	999	NA
Vectors	512	20	4,096	999	20	7,992

- The changes to support ATM Trunk Measurements make R3V8 CMS incompatible with G2.2 and G3V1.
- R3V8 CMS is backward compatible with G3V2 onward.
- R3V8 can not be a bug-fix load for R3V6 due to the Solaris Operating System upgrade.

CMS measurement of ATM trunks

- Prior to R8, ATM trunks could not be measured externally by CMS due to ATM trunks' use of an expanded equipment location format (circuit number increased to 3 digits from 2 to support up to 256 circuits).
- DEFINITY[®] system
 - When an ATM trunk group is externally measured, the DEFINITY[®] system passes to CMS the expanded equipment location (pnn or cabinet, carrier, slot, circuit) for each ATM trunk in that group.
 - Fields that make up the equipment location number in XTRUNK6 and ARTRUNK6 messages between the DEFINITY[®] system and CMS change to support measurement of ATM trunks:
 - The circuit field expand to 9 bits to support circuit numbers of up to 256
 - The module field is removed
 - The cabinet field expand to 7 bits for future use
 - The location ID field is added
- CentreVu[®] CMS
 - ATM trunks use a revised equipment location format that was not previously supported by CMS.
 - R3V8 CentreVu[®] CMS converts all trunk equipment locations to an 8 character format.
 - Supports sending the revised trunk equipment location format in the Call Data Unit.
 - CentreVu[®] Explorer
 - Supports the revised trunk equipment location format in the call record.

- ATM trunks are supported through new circuit packs that use a different equipment location format than trunks currently supported by CMS
 - ATM trunks require a circuit number of 3 digits to be supported (up from 2 digits).
 - CMS supports a revised trunk equipment location number with the expanded circuit number.
 - This includes messaging, reporting and storing in the call record.
- The capability to measure ATM trunks on CMS is not controlled by the DEFINITY® Call Center Release field.
 - Available when R8.1 is used as a bug-fix for R7.1 (DEFINITY® version field set to V7) and CMS is upgraded to a R3V8.

ATM Trunking and ATM WAN Port Network Connectivity (PNC)

- ATM trunking provides call centers with an alternative to T1/E1 facilities for various private networking configurations.
- ATM WAN PNC enables merging together into a single DEFINITY® ACD what are separate, geographically-dispersed DEFINITY® ACDs.
- Interconnecting port networks over an ATM WAN merges together separate, geographically-dispersed DEFINITY® ACDs.



NOTE:

Traffic handling related capacities (maximum number of agents, trunks or BHCC capacity) are not enhanced. Permissions continue to be deactivated when more than 2,000 VDNs are measured.

CentreVu® CMS

- Capacity increases provided at one site do not require other ACDs connected to the same CMS to be upgraded to the same release.
- Supports the ability to measure 999 vectors for a single ACD and the expanded vector range for the goto vector ... commands.
- Permissions checking is supported for the increase in vectors.
 - CMS can support up to 8 ACDs.
 - The total multi-ACD capacity for vectors is 7,992.
 - The per CMS limit on measured hunt groups not increased.

Dictionary

- CentreVu® CMS provides dictionary support for the capacity increase.

CentreVu[®] Visual Vectors

- Provides support for 20,000 VDNs
- Provides support for 999 vectors
- Supports the changes to the vector commands affected by the increase in members per hunt group
- Supports the changes to the vector commands affected by the increase in hunt groups

CentreVu[®] Supervisor

- CentreVu[®] Supervisor supports capacity increases for VDNs, vectors and hunt groups.
- Dictionary support for the capacity increases is provided.

Agent Site Tracking

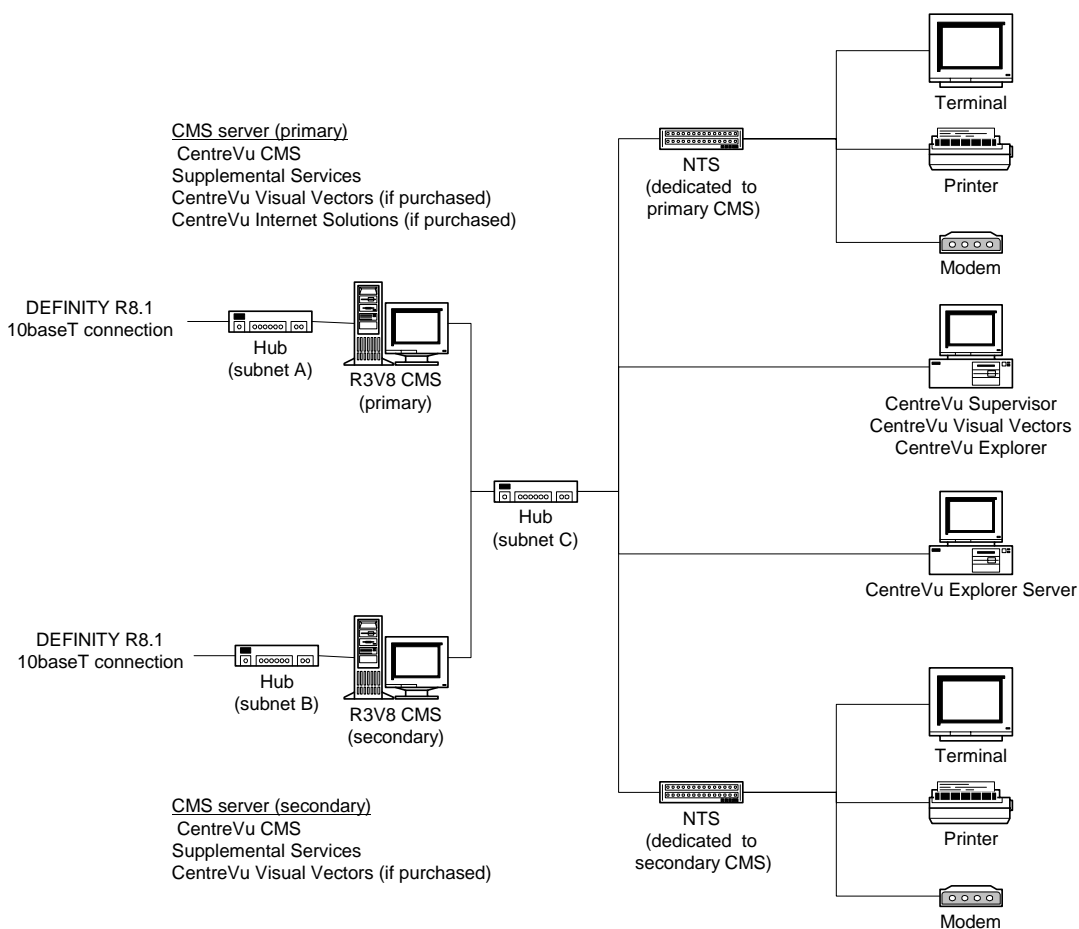
- This enables a customer who uses the Multiple Locations feature to associate each agent with a particular location ID upon login to the ACD and to be able to run reports that provide data on agent activity at a particular location.
- Support for Agent Site Tracking begins with DEFINITY[®] system.
 - The smaller DEFINITY[®] platforms, R8si and R8csi, only send location ID 0 to CMS.
 - Beginning in R8, the DEFINITY[®] system passes to CMS, for each agent that logs in to the ACD, the location ID associated with the equipment location of the agent's voice terminal.
 - With DEFINITY[®] Multiple Locations option enabled, the value of the location ID passed to CMS ranges between 1 and 44.
 - The sending of the location ID is supported for both EAS and non-EAS environments.
 - DEFINITY[®] passes to CMS the location ID associated with an agent when any of the following occurs:
 - Upon agent login to the ACD
 - At pump-up for each agent already logged in to the ACD
- The location ID of an agent can be changed at any time without the agent having to log out of the ACD.
 - This is done through the use of the change cabinet x form where x is the cabinet that contains the port network(s) to which the agent's voice terminal belongs.

CMS High Availability

CMS High Availability applies to all standard R3V8 CMS platforms (SPARC 5, SPARC 20, Ultra 5, Ultra Enterprise 3000, and Ultra Enterprise 3500), and provides for the following:

- Non-disruptive cmsadm backup.
 - Ability to perform a cmsadm backup with data collection turned on during the entire cmsadm backup process.
- Non-disruptive maintenance backup.
 - Ability to perform any of the maintenance backups with data collection turned on during the entire maintenance backup process.
 - Exists today, but the maintenance backups are slightly modified to support backing up data from one CMS server and restoring the data on another CMS server.
- Non-disruptive maintenance restores.
 - Ability to perform any of the maintenance restores with data collection turned on during the entire maintenance restore process.
- Non-disruptive R3 migration.
 - Ability to perform any of the R3 migrations with data collection turned on during the entire migration process.
 - R3 migration is critical to synchronizing data during the upgrade process (either R3V6 to R3V8 or R3V8.1 to R3V8.2 when database schema changes are required).
- Capability to turn ECH on and off.
 - The capability to turn ECH on and off allows the ability to minimize the amount of ECH data loss.
 - ECH can be authorized and installed on both CMS servers. ECH data can be turned on only one CMS server. If that CMS server fails, then ECH can be turned off on the failed CMS server and ECH can be turned on the other CMS server.
- Manual alarm clears.
 - Enhancement to the Alarm Origination Manager.
 - Manual alarm clears is the ability to list the current active alarms and to change the alarm status to resolved.
- New call history alarms.
 - Allows alarms to be sent to a remote system used by either the Technical Service Center (TSC) or the Centers of Excellence (COE).
 - The ECH alarms are sent when the call record data is nearing or exceeding the ECH buffer capacity.

CMS High Availability Configuration



Dual ACD links

- The dual ACD links feature allows the customer to eliminate data loss during an ACD link failure when combined with the CMS High Availability offer.
- The enhancements for the standard R3V8 CMS includes increasing CMS system availability and providing additional alarming support required for the High Availability configuration.
- The configuration significantly reduces the amount of data loss by eliminating single points of failure. This relies heavily on manual data synchronization and manual administration synchronization.

Reliability

- The CMS data collection process is improved by duplicating the ACD links.
- The customer only loses ACD data associated with an ACD link failure under these circumstances:
 - ACD software failure
 - Simultaneous C-LAN card failures
 - Simultaneous network failures across both ACD links
 - Combinations of failures on both ACD links.
- If one C-LAN card fails or a network failure occurs on one ACD link, then the customer would be vulnerable to data loss as only one CMS server would be receiving data.
- The CMSs are not operating in a high availability mode but no data loss occurs as the second CMS still collects data.

Capacity Constraints and Feature Limitations

Dual ACD Links

- The dual ACD links feature does not support X.25 communications between the DEFINITY[®] system and CMS.
- DEFINITY[®] R8.1 only supports two ACD links.

PC Application Software Translation Exchange (PASTE) Update

- This adds new button types and other new feature data implemented in R8 to the PASTE application.
- The PC Application Software Translation Exchange feature allows some system and terminal administration parameters to be passed to a PC application attached to a DCP display terminal.
- This development was required to provide continued support for CentreVu Agent.

1 New Switch and Adjunct Support
Call Center

112

Networking Enhancements

2

Networking - International

Continuous and Pulsed E&M Signaling (Brazil)

This feature provides the user with the capability to make and receive calls over Brazil pulsed or continuous E&M signaling trunks and to administer the signaling bit on all CAS DS1 trunk types.

Also addressed is Hungarian pulsed E&M. Hungarian Discontinuous signaling differs from Brazil Pulsed E&M only by the signaling bit and inversion used. This can be accomplished by administering a country code of 23 and the trunk signaling type to discont.

Capacity Constraints and Feature Limitations

- These signaling protocols do not support Brazil's BCC (Blocked Collect Call) feature.
- These signaling protocols are not available on Personal Central Office (PCOL) trunks.
- These signaling protocols are not available on TN767 boards. Although the TN767E vintage 19-board firmware can support the protocols, the administration software on the switch, which supports the TN767, can not handle the protocols. We understand that this is not a problem because the TN767 is not currently used in either Brazil or Hungary.

This feature requires the TN464F vintage 19, or TN2464 vintage 19, or a later suffix or vintage.

Feature Administration

To administer this feature for use in Brazil, one administers the forms for DS1 Circuit pack and Trunk Group. In particular, the fields shown in bold print in those sections should have one of the values listed.

This feature provides only trunk line signaling; and in Brazil the COs that use these protocols expect R2-MFC line signaling. In order to use the feature in Brazil, R2-MFC register signaling must also be administered for these trunks. In every case, the outgoing dial type and the incoming dial type must be set to tone or MF

Firmware on the DEFINITY[®] ECS digital circuit packs will not allow these protocols to be used with rotary signaling.

The Pulsed E&M signaling protocol has been tried at one site in Brazil, and at that site the following trunk parameter values were found to work. We recommend using these values in Brazil, although it is possible that slightly different values would also work:

- Incoming Disconnect(msec): 600
- Outgoing End of Dial(sec): 40
- Outgoing Disconnect(msec): 600
- Outgoing Seizure Response(sec): 255

To administer this feature for use in Hungary, set the Trunk Signaling Type field to **discont**.

Brazil Block Collect Call

In both continuous and pulsed E&M signaling, the Brazil's BCC (Blocked Collect Call) feature will not be included.

Personal Central Office Line (PCOL)

Personal Central Office Line (PCOL) trunks can not use continuous nor pulsed nor discontinuous E&M signaling.

When the Trunk Signaling Type field on the CO, FX, WATS, DID, and DIOD forms is administered to either cont-E&M or pulsed-E&M, that will cause Receive Release Ack, Send Release Ack and Send Answer Supervision fields to appear too.

Administer E&M signaling per trunk group

The line signaling protocols cont (for Brazil continuous E&M), pulsed (for Brazil pulsed E&M), discont (for Hungary discontinuous E&M), and blank (for standard R2) will be administrable per trunk group 4 for public network trunk groups. This requirement applies to trunk group type DIOD.

Seize when maintenance busy per network trunk group

This requirement is administrable per public network trunk group, and applies to trunk group type DIOD. The administration is via the existing field called Seize When Maintenance Busy being ported from the tie trunk screen to the above trunk group type screen.

Administration steps

1. Administer the basic system.
2. Administer trunks with the trunk group field values Trunk Group.
3. Administer R2-MFC signaling.
4. Administer the DS1 form as DS1 Circuit packs.

The following forms have to be administered when Brazil continuous E&M, Brazil pulsed E&M, or Hungary discontinuous E&M is necessary.

DS1 Circuit pack

add ds1 xxxxxx	DS1 CIRCUIT PACK	Page 1 of 2
Location: _____	Name: _____	
Bit Rate: _____	Line Coding: _____	
Signaling Mode: CAS_____		
Interconnect: CO__	Country Protocol: ____23_	
Interface Companding: _____		
Idle Code: _____		
Received Digital Metering Pulse Minimum (ms):		
Received Digital Metering Pulse Maximum (ms):		
Received Digital Metering Pulse Value:		
Slip Detection: __	Near-end CSU Type: _____	

Trunk Group DIOD

```
Page 1 of 10

TRUNK GROUP

Group Number:                Group Type: diod                CDR Reports: y
Group Name:                   COR: 1                TN: 1                TAC:
Direction: two-way           Outgoing Display? n           Trunk Signaling Type: cont

Dial Access? n                Busy Threshold: 99
Queue Length: 0                Country: 16

Prefix-1? y                    Auth Code? n                Digit Absorption List:
Trunk Flash? n                Toll Restricted? y

TRUNK PARAMETERS
Trunk Type: immed-start
Outgoing Dial Type: tone                Incoming Dial Type: tone
Trunk Termination: rc
Digit Treatment:                        Digits:
Expected Digits:                        Sig Bit Inversion: none
Terminal Balanced? n                RA Trunk Loss: 0db
Trunk Gain: high                Drop Treatment: silence
Disconnect Supervision - In? y Out? n
Answer Supervision Timeout: 10                Receive Answer Supervision? n
```

Trunk Signaling Type field

The Trunk Signaling Type field is new to the DIOD form, although it already exists on tie trunk forms. On the DIOD form:

- It is visible only if both the trunk types are set to immed-start and the country is set to 23.
- It defaults to blank.
- It takes on only 3 new values: pulsed, cont, and discont. It does not accept the values it takes on the tie trunk forms, except for blank. Administering blank will result in current DS1 board operation.
- It has new end validations: if it is set to one of the values pulsed, cont, or discont, then:
 - All ports in the trunk group must be on a TN464F vintage 19 or higher, or a TN2464 vintage 19 or higher.
 - The outgoing dial type and incoming dial type must be set to tone.or MF

TRUNK FEATURES

```

                                     TRUNK FEATURES
                                     Page 2 of X
ACA Assignment? n Measured: none
Maintenance tests? y
Data Restriction? n
Seize When Maintenance Busy: both-ends
Suppress # Outpulsing? n
Incoming Tone (DTMF) ANI? y
```

Seize When Maintenance Busy field

The Seize When Maintenance Busy field is new to trunk group type DIOD, but is the same field with the same definition, values, help and error messages as the Seize When Maintenance Busy field currently shown for tie trunk groups. On the DIOD form it displays only when the group type is diod and the Trunk Signaling Type field is set to one of the values pulsed, cont, or discont.

The Seize When Maintenance Busy field is now available for trunk group type DIOD when the country field is set to 23 and the trunk signaling type is one of pulsed, cont, or discont. This is in addition to the tie trunk types that used this field in earlier releases.

```

                                     ADMINISTRABLE TIMERS
                                     Page 3 of X
Outgoing Dial Guard (msec) : 1600
Incoming Glare Guard (msec) : 1500 Outgoing Glare Guard (msec) : 1500
Outgoing Seizure Response (sec) : 5
Programmed Dial Pause (msec) : 1500 Disconnect Signal Errors (sec) : 240
Flash Length(msec) : 540 Incoming Incomplete Dial Alarm (sec) : 255
Outgoing Last Digit(sec) : 10 Normal Outgoing Seize Send (msec) : 150
Release Ack Send(msec) : 600 Seize Ack Delay (msec) : 100
Answer Send(msec) : 150 Outgoing Disconnect Send (msec) : 600
Seize Ack Send(msec) : 150 Incoming Disconnect Send (msec) : 600
OUTPULSING INFORMATION
PPS: 10 Make (msec) : 40 Break (msec) : 60
```

Recommend values for Brazil

- Receive Release Ack? Y (y or n).
- Send Answer Supervision? Y (y or n).
- Send Release Ack? Y (y or n).
- Trunk Signaling Type: pulsed-E&M.
- Trunk Type (in/out): discont/discont (many options).
- Answer Send (ms): 150 (10 - 2550).
- Incoming Disconnect Send (ms): 600 (500 - 1200).

- Normal Outgoing Seize send (ms): 150 (10 - 990).
- Outgoing Disconnect Send (ms): 600 (100 - 9900).
- Outgoing Last Digit (sec): 40 (10 - 40).
- Outgoing Seizure Response (sec): 255 (1 - 255).
- Release Ack Send (ms): 600 (500 - 1200).
- Seize Ack Delay (ms): 120 (40 - 120).
- Seize Ack send (ms): 150 (10 - 990).

China Howler After Busy & Disconnect on No Answer

Disconnect on No Answer feature drops outgoing trunk calls that users leave unanswered too long.

Howler after Busy feature plays howler tone when users leave their set off hook too long.

When an outgoing trunk call or call attempt is not answered, the PBX will disconnect the trunk and play busy tone to the caller, next play howler tone, then go to high-and-dry state. Whether this feature applies to any particular call at all, and the amount of time each of these states last, depends on switch administration and the call type: local, toll, or international.

When any of the following timers expire during an outgoing call attempt, the PBX will apply busy tone for 40 seconds followed by howler tone for 60 seconds.

- Pre-dialing timer
- Inter-digit timer
- Outgoing seizure acknowledge timer
- 60-, 90-, and 120-second no answer disconnect timers, based on ARS call type
- 120-second timer used for calls without a call type, such as TAC calls

The tones will last for the duration set by these timers:

- Busy tone timer: 40 seconds
- Howler tone timer: 60 seconds

Capacity Constraints and Feature Limitations

This feature does not cover outgoing calls that are automatically generated by the switch and have no human involved in the call. Examples of such calls include outgoing alarm calls from the INADS port, and transmission test calls from the ATMS feature.

This feature does not apply to an attendant console, or an Outgoing Call Management (OCM) agent, or a BRI set.

The Disconnect on No Answer portion of this feature applies only to outgoing trunk calls, not to station-to-station calls. There is an exception to this rule for any station-to-station calls that are carried over a trunk, for example, CCRON calls.

This feature requires a TN2182 tone clock. If the howler tone feature is enabled on a system without a TN2182 tone clock, the station set users will hear silence instead of howler tone.

The Howler After Busy portion of this feature is most useful for customers who have a high percentage of analog sets. The main purpose of howler tone is to remind users to place sets back onhook, but digital sets can receive calls even while the handset is offhook.

Howler After Busy

If the Station Tone Forward Disconnect field on the Feature Related System Parameters form is set to anything other than busy, the user receives the treatment specified by that field instead of busy tone followed by howler tone.

Administration

- The system administrator is able to administer whether Howler after Busy and Disconnect on No Answer applies.

Steps to administer these features

For the Howler after Busy feature to work, the following must be administered:

1. The Station Tone Forward Disconnect field on the Feature Related System Parameters form is set to busy.
2. The Howler after Busy field on the System Parameters Country Options form is set to yes.

For the Disconnect on No Answer feature to work, the following must be administered:

1. The Disconnect on No Answer by Call Type field on the System Parameters Country Options form is set to yes.

Loop start trunks do not provide answer supervision.

For such trunks, many sites use the answer supervision timeout, which is not useful with the Disconnect on No Answer feature.

If an administrator were to simply disable the timeout, then the call will drop during the speech phase.

1. Administer answer supervision by classifier.

The field used for that is receive answer supervision on the trunk group form

1. Defaults to no.

FEATURE-RELATED SYSTEM PARAMETERS

```
change system-parameters features                                     Page 5 of 10
      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): 0
      Reset Shift Timer (seconds): 0
      Station Call Transfer Recall Timer (seconds): 0
      DID Busy Treatment: tone
      Allow AAR/ARS Access from DID/DIOD? n
      Allow ANI Restriction on AAR/ARS? n
      Transfer Upon Hang-Up? n                                Abort Transfer? n
      7405ND Numeric Terminal Display? n                      7434ND? n
DISTINCTIVE AUDIBLE ALERTING
      Internal: 1 External: 2 Priority: 3
      Attendant Originated Calls: external
```

System Parameters Country Options

```
change system-parameters country-options                           Page 1 of 7
      SYSTEM PARAMETERS COUNTRY-OPTIONS

      Companding Mode: Mu-Law                                Base Tone Generation Set: 1
      440Hz PBX-dial Tone? n                                440Hz Secondary-dial Tone? n
      Analog Ringing Cadence: 1 Set Layer 1 timer T1 to 30 seconds? n
      Analog Line Transmission: 1
      64/84xx Display Character Set? Roman
      Howler After Busy: y Disconnect on No Answer by Call Type: y

      TONE DETECTION PARAMETERS
      Tone Detection Mode: 5                                Dial Tone Validation Timer:
      Interdigit Pause: short
```

Time Supervision & Force Release (China)

Time Supervision & Force Release (China) provides three no-answer limits based on call type only for calls in which digits are analyzed by ARS. Outgoing trunk calls that are not analyzed by digit analysis have a fixed no-answer time limit of 120 seconds. Outgoing trunk calls that are analyzed by AAR digit analysis do not have a no-answer time limit.

When any of the following timers expire during an outgoing call attempt, the PBX will apply busy tone for 40 seconds followed by howler tone for 60 seconds.

- pre-dialing timer
- inter-digit timer
- outgoing seizure acknowledge timer
- 60-, 90-, and 120-second no answer disconnect timers, based on ARS call type
- 120-second timer, used for calls without a call type, such as TAC calls

The tones will last for the duration set by these timers:

- busy tone timer: 40 seconds
- howler tone timer: 60 seconds

Capacity Constraints and Feature Limitations

This feature does not cover outgoing calls that are automatically generated by the switch and have no human involved in the call. Examples of such calls include outgoing alarm calls from the INADS port and transmission test calls from the ATMS feature. This feature does not apply to an attendant console, or an Outgoing Call Management (OCM) agent, or a BRI set.

The Disconnect on No Answer portion of this feature applies only to outgoing trunk calls, not to station-to-station calls. There is an exception to this rule for any station-to-station calls that are carried over a trunk, for example CCRON calls.

This feature requires a TN2182 tone clock. If the howler tone feature is enabled on a system without a TN2182 tone clock, the station set users hear silence instead of howler tone.

Call Processing

Alternate treatment for seizure failure

If the switch is administered and the Outgoing Seizure Response timer expires, the switch will drop the trunk and treat the calling set. The Outgoing Seizure Response timer will typically be set to 2 seconds in China.

Alternate treatment for offhook after a call

If the switch is administered and any station set is the last party remaining offhook on a call, the switch will play tones to offhook sets if the calling set is not an analog set.

Alternate treatment for unanswered outgoing ARS calls

If the switch is administered and an outgoing ARS call from any station set has remained unanswered for:

- 60 seconds on local or service calls, for example, calls with call type: alrt, emer, hnpa, locl, lpvt, op, pubu, svcl, svfl, svct, and svft
- 90 seconds on toll calls, for example, calls with call type: fnpa, natl, npvt, and nsvc
- 120 seconds on international calls, for example, calls with call type: intl, iop

Timers for the above amounts of time start immediately after the last digit was sent or the TTR was removed on an outgoing call,.

Alternate treatment for unanswered outgoing non-ARS calls

If the switch is administered and an outgoing trunk call from any station set remains unanswered for 120 seconds, the switch will drop the trunk. This does not apply to ARS, or AAR calls.

Play tones to offhook sets

If the calling set is not an analog set and when any of the timers expire, the switch will play busy tone to the set for 40 seconds. After that, the switch will play howler tone to the set for 60 seconds. After that, the line appearance LED will turn off or the set will be placed in line lockout state.

Disconnect on No Answer by Call Type

A call is considered unanswered as long as none of the following events have happened:

- The network tells the PBX that the call has been answered, for example via an answer supervision line signal or an ISDN connect message.
- The call classifier classifies the call as answered.
- The Answer Supervision Timeout timer expires.
- The Outgoing End of Dial Timer expires.

Other ARS call types

Howler After Busy

If the Station Tone Forward Disconnect field on the Feature Related System Parameters form is set to anything other than busy, the user receives the treatment specified by that field instead of busy tone followed by howler tone.

Administration

Any entry in the version field is always usable no matter what the version field is set to on the system parameters customer options form.

Steps to administer these features

For the Howler After Busy feature to work, the following must be administered:

1. The Station Tone Forward Disconnect field on the Feature Related System Parameters form is set to busy.
2. The Howler After Busy field on the System Parameters Country Options form is set to yes.

For the Disconnect on No Answer feature to work, the following must be administered:

1. The Disconnect on No Answer by Call Type field on the System Parameters Country Options form is set to yes.

Although the Disconnect on No Answer feature will work when set as a non-zero value to the answer supervision timeout or the outgoing end of dial timer, it is better to set these timers to 0 to disable them. If the timers are set to more than the China values, then the trunk will always drop before the timers time out. Someone reading the administration screens later could be misled by the fact that they are set to something.

Loop start trunks do not provide answer supervision. For such trunks, many sites use the answer supervision timeout, but it is best to administer answer supervision by classifier. The field used for that is receive answer supervision on the trunk group form. It defaults to no.

TRUNK GROUP

```
Page Y of X
TRUNK GROUP
ADMINISTRABLE TIMERS
Incoming Disconnect(msec): 500 Outgoing Disconnect(msec): 500
Incoming Dial Guard(msec): 70 Outgoing Dial Guard(msec): 1600
Incoming Glare Guard(msec): 1500 Outgoing Glare Guard(msec): 1500
Incoming Partial Dial(sec): 18
Outgoing Rotary Dial Interdigit(msec): 800_
Outgoing End of Dial(sec): 0 Outgoing Seizure Response(sec): 5
Programmed Dial Pause(msec): 1500 Disconnect Signal Error(sec): 240
Incoming Incomplete Dial Alarm(sec): 255
END TO END SIGNALING
Tone (msec): 350 Pause (msec): 150
OUTPULSING TIMING
PPS: 10 Make (msec): 40 Break (msec): 60
```

```
Page 1 of X
TRUNK GROUP
Group Number: xxx Group Type: co CDR Reports: y
Group Name: OUTSIDE_CALL_____ COR: 1_ TN: 1__ TAC: ____
Direction: two-way_ Outgoing Display? n
Dial Access? n Busy Threshold: 99_ Night Service: ____
Queue Length: 0__ Country: 1_ Incoming Destination: ____
Comm Type: rbavd Auth Code? n Digit Absorption List: _
Prefix-1? y Trunk Flash? n Toll Restricted? y
BCC: _
TRUNK PARAMETERS
Trunk Type: _____
Outgoing Dial Type: tone_____ Cut-Through? n
Trunk Termination: rc_____ Disconnect Timing(msec): 500_
Auto Guard? n Call Still Held? n Sig Bit Inversion: none
Trunk Gain: high
Bit Rate: 1200 Synchronization: _____ Duplex: ____
Disconnect Supervision - In? y Out? n
Answer Supervision Timeout: 0 Receive Answer Supervision? Y
```

Feature Related System Parameters

```
Page 7 of 9
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? n
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: busy
Level Of Tone Detection: precise
Charge Display Update Frequency (seconds): 30
Date Format on 6400/8400 Terminals: mm/dd/yy
On-hook Dialing on 6400/8400 Terminals? n
RECALL TIMING
Flashhook Interval? y Upper Bound (msec): 1000
Lower Bound (msec): 200
Forward Disconnect Timer (msec): 600
ITALIAN DCS
Italian DCS Enabled? n
Apply Intercept Locally? _ Enforce PNT-to-PNT Restrictions? _
Page 1 of 21
SYSTEM PARAMETERS COUNTRY-OPTIONS
Companding Mode: A-Law Base Tone Generation Set: 18
440Hz PBX-dial Tone? n 440Hz Secondary-dial Tone? n
Digital Loss Plan: 18
Analog Ringing Cadence: 18 Set Layer 1 timer T1 to 30 seconds? n
Analog Line Transmission: 18 Enhanced 84xx Display Character Set?
Howler After Busy: y Disconnect on No Answer by Call Type: y
TONE DETECTION PARAMETERS
Tone Detection Mode: 4 Dial Tone Validation Timer: 600
Interdigit Pause: long
```

Administrable Loss Plan

Description

Capacity Constraints and Feature Limitations

This feature provides for dynamic administration of loss levels per station. If singing connections are the problem, use the feature to add more loss. If users complain about being unable to hear each other clearly, use the feature to add more gain. At any time, system administrators can reset the PBX to the default loss values by changing the "Customize?" fields back to "n", and resubmitting the system-parameters country-options form.

The following types of endpoints, although they do have extension numbers, are not true stations. This feature does not assign loss per these:

- Hunt group
- Listed Directory Number

- Terminating Extension Group
- Expert Agent Login ID
- Vector Directory Number
- Multimedia Applications Server Interface
- Attendant Common Shared Call Park Extension
- MASI stations
- Xmobile stations
- XDID stations

This feature does not administer loss between other types of endpoints and the following:

- Recorded announcement ports
- Attendant consoles
- Data modules
- MASI trunks
- Access endpoints
- Wideband access endpoints
- PRI endpoints

Configuration/Environment, Software Version, Configuration, Customer Option, Offer Category

The Administrable Loss Plan feature requires R8 software, and is available on all offer categories and all hardware platforms. But, it is available only if the new `Digital Loss Plan Modifications?` field on the **system parameters-customer options** form is set to y. This screen (shown below) allows access to features purchased by customers. An authorized Lucent Technologies representative sets these parameters for you.

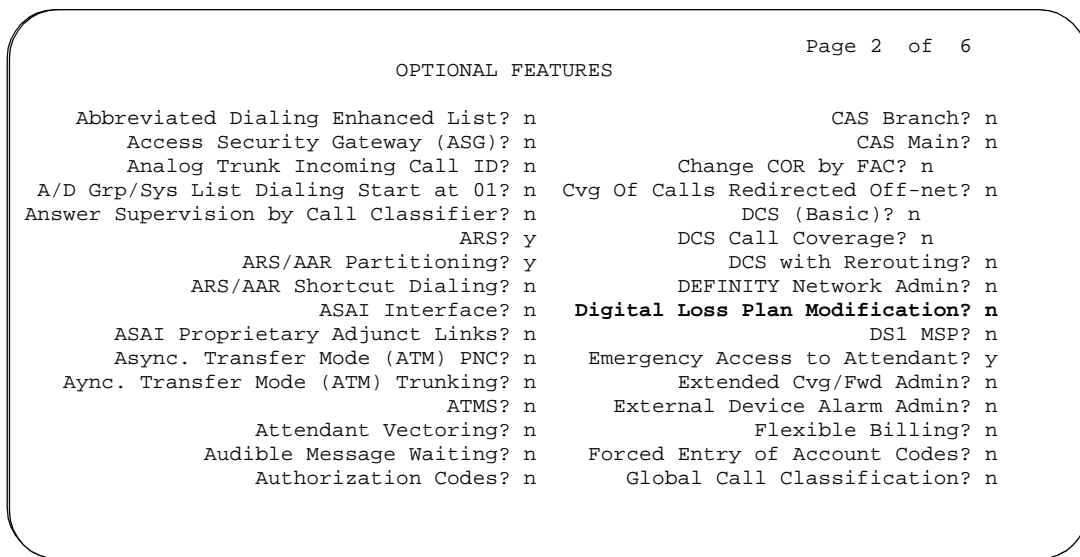


Figure 6. System-Parameters Customer-Options

Relevant Field description for page 2

Digital Loss Plan Modification

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

Valid entries	Usage
---------------	-------

y	Allows you to make specific changes.
n	Uses the default country code for the loss plan, tone plan, and conference loss plan. No modifications to the loss plan are permitted.

The screens below illustrate a sample 2-Party Loss Plan and a Tone and Conference Loss Plan, respectively. They are administered on the **system-parameters country-options** form. This form implements parameters associated with certain international (including North American) call characteristics.

Refer to the following table for country codes used in these screens.

Country code table

Code	Country
1	United States, Canada
2	Australia
3	Japan
4	Italy
5	Netherlands
6	Singapore
7	Mexico
8	Belgium, Luxembourg
9	Saudi Arabia
10	United Kingdom
11	Spain
12	France
13	Germany
14	Czech Republic, Slovakia
15	Russia (CIS)
16	Argentina
17	Greece
18	China
19	Hong Kong
20	Thailand
21	Macedonia
22	Poland
23	Brazil
24	Nordic
25	South Africa

display system-parameters country-otptions Page 2 of 7
 2 PARTY LOSS PLAN

Digital Loss Plan: 1

TO

Customize? n

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17
1:	-1	0	0	0	3	0	0	0	0	3	0	6	6	6	0	3	3
2:	0	0	0	0	0	3	3	2	2	3	0	6	6	6	2	3	3
3:	0	0	0	0	0	3	3	3	2	3	0	6	6	6	0	3	3
4:	15	0	0	0	6	0	0	0	0	3	0	6	6	6	0	3	3
5:	0	-3	-3	0	0	-3	-3	-3	-3	0	-3	3	0	0	-3	3	3
6:	0	3	3	0	0	6	8	6	5	5	5	9	9	9	5	3	3
F 7:	0	3	3	0	0	8	8	6	5	5	5	9	9	9	5	3	3
R 8:	0	3	3	0	0	6	6	6	3	5	3	9	6	6	3	3	3
O 9:	0	2	2	0	0	5	5	3	0	0	2	3	3	3	9	3	3
M 10:	3	3	3	3	3	5	5	5	0	0	3	3	3	3	3	3	3
11:	0	0	0	0	0	5	5	3	2	3	0	6	6	3	0	3	3
12:	0	0	0	0	0	3	3	3	-3	-3	0	0	0	0	0	3	3
13:	0	0	0	0	0	3	3	3	-3	-3	0	0	0	0	0	3	3
14:	0	0	0	0	0	3	3	3	-3	-3	-3	0	0	0	0	3	3
15:	0	2	0	0	0	5	5	3	0	3	0	6	6	6	0	3	3
16:	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3
17:	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3	3

Figure 7. System Parameters Country-Options

Field descriptions for page 2

Digital Loss Plan

Provides the default values for digital loss plan and n-party conference loss.

Valid entries	Usage
---------------	-------

1 - 25	Refer to the Country code table at the beginning of this screen description.
--------	--

Customize

Appears when the Digital Loss Plan Modification field is enabled on the System-Parameters Customer-Options screen.

Valid entries	Usage
---------------	-------

y/n	Enables customization on the 2-party loss table.
-----	--

FROM / TO

Identifies the variable digital loss values.

Valid entries	Usage
---------------	-------

-3 - 15	An unsigned number is a loss, while a number preceded with a minus sign is a gain.
---------	--

```
display system-parameters country-otptions                                     Page 3 of 7
                                     TONE & CONFERENCE LOSS PLANS

Digital Tone Plan: 1                                                         Customize? n

                                     TO
Dial:      1  2  3  4  5  6  7  8  9 10 11 12 13 14 15 16 17
Confirm:   0  3  3  0  0  6  6  6  5  0  6  5  5  5  5  0  0
Reorder:   0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
Busy:      0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
Ringing:   0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
Spec Ring: 0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
Intercept: 0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
Waiting:   0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
Verify:    0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
Intrude:   0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0  0
Zip:       3  3  3  -3 -3 -3 -3 -3 -3 -3 -3 -3 -3 -3 -3  0
Music:    0  3  3  0  0  6  6  6  3  0  6  3  3  3  3  0  0
```

End-to-End total loss (dB) in a n-party conference:
3:____ 4:____ 5:____ 6:____

Figure 8. System Parameters Country-Options

Field descriptions for page 3

Digital Tone Plan

Provides the default values for tone loss for the selected country.

Valid entries	Usage
---------------	-------

1 - 25	Refer to the Country code table at the beginning of this screen description.
--------	--

Customize

Appears when, on the System-Parameters Customer-Options screen, the Digital Loss Plan Modification field is **y**.

Valid entries	Usage
---------------	-------

y/n	Enables customization on the 2-party loss table.
-----	--

End-to-End total loss (dB) in a n-party conference

Provides total loss for a conference call with the designated number of parties.

Valid entries	Usage
---------------	-------

0 - 99	The higher the number listed for a call with a fixed display number of parties, the more loss the switch adds into a conference call with that number of parties; therefore, the conference call is quieter.
--------	--

FROM / TO

Identifies the variable digital tone values.

Valid entries	Usage
---------------	-------

-3 - 15	An unsigned number is a loss, while a number preceded with a minus sign is a gain.
---------	--

Hardware Restrictions

An installer using this feature should make sure that tie trunk groups do not contain a mix of TN439 ports and ports on other analog trunk circuit packs. While such a tie trunk group could be used, the transmission levels on calls placed over such a group would vary by a few dB from call to call, depending on which circuit pack type a particular call happened to use.

Other Gain/Loss Factors

The following table shows existing fields that also affect loss through a switch:

Form	Field	Loss or gain is applied:
system-parameters multifrequency-signaling	Received Signal Gain (-Loss) (dB)	Between the tone receiver and the trunk port during MFC tone reception.
system-parameters multifrequency-signaling	Transmitted Signal Gain (- Loss) (dB)	Between the tone receiver and the trunk port during MFC tone generation.
system parameters ocm call classification	Global Classifier Adjustment (dB)	Between the tone receiver and the trunk port during call classification.

Networking - ISDN Private

QSIG Call Independent Signaling Connection (CISC) Enhancements

- Provides the user with the capability to transport QSIG operation information independent of an active call, between two QSIG compliant nodes.
- QSIG CISCs are similar to the DEFINITY NCA TSCs (Non Call Associated Temporary Signaling Connections) except that QSIG CISCs is established on demand, not via a permanent administered connection interface.
 - QSIG PRI and BRI Trunk interfaces.
- QSIG Call Independent Signaling Connections (CISCs) are defined in ISO/IEC 11582
 - Call Completion
 - Message Waiting
- The per system maximum of NCA TSCs.
 - On a per Signaling Group basis, administrable maximum of NCA TSCs that can be established so no one signaling group may use up all of the NCA TSCs.

Administration

Feature-Related System Parameters Form

- To facilitate processing of a CISC, a global extension is applicable to all CISCs.
 - Unrestricted COR and the highest level of COS
 - Not a "real" extension that can be called or dialed.

NOTE:

The administration for this extension is not tied to "Basic Supplementary Services?" enabled on the system-parameters customer-options form. The extension may be administered, prior to any intended CISC transport.

- Support field "QSIG TSC Extension:" in the "ISDN PARAMETERS" section.
- Values according to the Dial Plan.
- The Unrestricted COR and COS are associated with the extension so a CISC can be routed at a Transit or Terminating PBX without requiring authorization.
- Users get Intercept treatment if this extension is called directly.

Feature-Related System Parameters form

```
change system-parameters features Page 5 of 8
      FEATURE-RELATED SYSTEM PARAMETERS

      Public Network Trunks on Conference Call: 5          Auto Start? n
      Conference Parties with Public Network Trunks: 6      Auto Hold? y
      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
      DID Busy Treatment: tone_____
      Allow AAR/ARS Access from DID/DIOD? n

      ISDN PARAMETERS
      Send Non-ISDN Trunk Group Name as Connected Name? n  QSIG TSC Extension: _____
      Display Connected Name/Number for ISDN DCS Calls? n

      DISTINCTIVE AUDIBLE ALERTING
      Internal: 1 External: 2 Priority: 3
      Attendant Originated Calls: external
```

Signaling Group Form

- All existing fields on page 1 apply to CISCs.
- The signaling group form page 1 has a field "Supplementary Service Protocol:" which may be set to "a" (national protocol, including AT&T) or "b" (QSIG).
 - Field determines whether AT&T NCA TSCs or QSIG CISCs are sent over the interface.
 - Because CISCs are on demand only, pages 2-5 of the form for administered connections do not apply to CISCs.
- BRI Trunk CISC administration via the BRI Trunk Board form.
 - AT&T TSCs not supported over BRI Trunk D-Channels.
- "Max NCA TSC" column controls the maximum number of CISCs that a particular BRI Trunk D-channel may carry.
- If a signaling group's SS Protocol field is changed from "a" to "b", the ISDN TSC Gateway Channel Assignment form must be checked first to make sure no NCA TSC on the signaling group form is administered as a Gateway Channel.
 - Users must remove the NCA TSC Gateway assignment first before the signaling group's SS Protocol field may be changed to "b".
- If a signaling group's "Supplementary Service Protocol:" is set to "b", it is an implementation option to either block administration for pages 2 through 5 or to not show pages 2 through 5.

- If pages 2 through 5 are administered and the "Supplementary Service Protocol:" is changed from "a" to "b", there is a warning message that those pages are blanked when the form is submitted.
- If a signaling group's "Supplementary Service Protocol:" is set to "b", the ISDN TSC Gateway form is checked.
- If the Gateway form uses an existing NCA TSC that would be removed by setting "Supplementary Service Protocol:" to "b", the signaling group change is blocked

Signaling Group form (page 1)

```
add signaling-group next Page 1 of 1
                        SIGNALING GROUP

Group Number: 1 Associated Signaling? n           Max number of NCA TSC: 0__
                Primary D-Channel: _____   Max number of CA TSC: 0__
                Secondary D-Channel: _____   Trunk Group for NCA TSC: __
Trunk Group for Channel Selection: __
Supplementary Service Protocol: b

Trunk Brd Interface ID Trunk Brd Interface ID
1: _____ 11: _____
2: _____ 12: _____
3: _____ 13: _____
4: _____ 14: _____
5: _____ 15: _____
6: _____ 16: _____
7: _____ 17: _____
8: _____ 18: _____
9: _____ 19: _____
10: _____ 20: _____
```

Route Pattern Form

- Existing "TSC" field is consulted to find a preference that can route a tandemed CISC (the field must be set to "y").

Trunk Group Form

- "NCA-TSC Signaling Group:" field points to the Signaling Group over which a tandem NCA TSC is routed on.
 - BRI Trunks are not administered with Signaling Groups, this field is changed to accommodate both PRI and BRI CISC routing.
 - There must be at least one member in the Trunk Group in order to use it to route tandem NCA TSC.
- The Trunk Group Form field "NCA-TSC Signaling Group" is changed to "NCA-TSC Trunk Member" so that the signaling interface associated with the administered Trunk Member is used to route a tandem NCA TSC.
 - DEFINITY verifies that this field is administered to a valid trunk group member.

Trunk Group form

```
change trunk-group 1 Page 2 of 12
TRUNK FEATURES
ACA Assignment? n Measured: none___ Wideband Support? n
Maintenance Tests? y
Data Restriction? n NCA-TSC Trunk Member: 7__
Send Name: y Send Calling Number: y
Used for DCS? n Send Connected Number: y
Suppress # Outpulsing? n Numbering Format: public_

Charge Conversion: 1___
Decimal Point: none__
Currency Symbol: ___
Charge Type: units__
```

System Capacity Form

- The title Fixed TSCs on page 6 of the System Capacity form is changed to Administered TSCs.

System Capacity form (page 6)

```
display capacity Page 6 of 7
SYSTEM CAPACITY

System
Used Available Limit
-----

Recorded Announcement Analog Queue Slots: * * 150

TEMPORARY SIGNALING CONNECTIONS (TSC)
Administered TSCs: * * 128
NCA-TSC Calls: * * 256

TRUNKS
DS1 Circuit Packs: * * 30
ICHT For ISDN PRI Trunks: * * 288
ISDN CBC Service Selection Trunks: * * 10
Trunk Groups: * * 99
Trunk Ports: 25 375 400
```

QSIG VALU Call Coverage

- QSIG-VALU is a term that describes DCS features that are ported to work in a QSIG network consisting of DEFINITY® ECS systems.
 - The messaging needed to implement the feature transparency is sent using the Manufacturer Specific Information (MSI) encoding provided by the QSIG standards.

- The encoding allows QSIG-VALU features to work between DEFINITY[®] systems in a QSIG network, even if the connection between the DEFINITY[®] systems includes non-DEFINITY[®] systems.
- QSIG-VALU Call Coverage feature extends the Call Coverage feature transparency when a coverage point is located on a remote DEFINITY[®] system connected via a QSIG network.
 - The Principal and coverage point users, located on DEFINITY[®] systems, see similar operational and display behavior for this feature as DCS Call Coverage.
- The signaling for this feature is built on top of the existing QSIG Call Diversion/Forwarding supplementary service.
- QSIG-VALU Distinctive Ringing feature extends the Distinctive Ringing feature transparency when a call is routed to a remote DEFINITY[®] system connected via a QSIG network.
- QSIG-VALU features require the administration of two fields:
 - QSIG-VALU on Customer Options form.
 - Trunk Group form.

Configuration/Environment

- To enable QSIG-VALU, the following customer options must also be enabled:
 - Offer Category "A"
 - ISDN-PRI or ISDN-BRI Trunks or Asynch. Transfer Mode (ATM) Trunking
 - QSIG Basic Supplementary Services
 - Software Version must be R8 or later.

Feature Description

Distinctive Ringing

- In DEFINITY[®] ECS R8 calls that are routed over QSIG network have the capability to provide distinctive ringing.
- It provides two kinds of ringing pattern to the terminal based on the administration and origination of the call before routed over the network.
 - Internal
 - External
- When there are gateway switches between the originating and terminating PBXs, functionality depends on the trunks connecting the PBXs.

Distinctive Ringing Administration

1. Option Value-Added Lucent (VALU) is Enabled on System-parameters customer-options form
2. The field QSIG Value-Added Lucent is selected as y and Supplementary Service Protocol Option shall be selected as B on a Trunk Group form to send QSIG-VALU Call Coverage MSI information to a remote switch in association with a call routing on that trunk group.

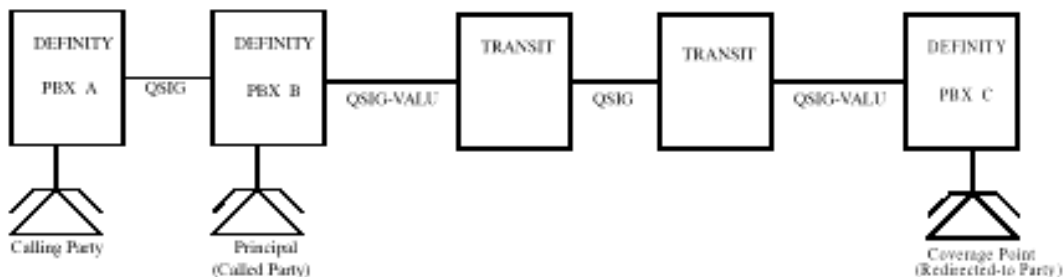


NOTE:

The administration to govern the ringing pattern at the called user's terminal is the same as the Distinctive Ringing feature.

QSIG-VALU Call Coverage

- QSIG-VALU Call Coverage enhances the operations for the users, similar to DCS Call Coverage, in the following ways:
 - Principal's coverage path and criteria are used to re-direct the call to QSIG-VALU coverage points.
 - If the QSIG-VALU coverage point is not available, and the call can be redirected to the subsequent coverage point based on the Principal's coverage path, the call to the coverage point is disconnected and the subsequent coverage point is alerted.
 - Coverage Point is alerted with distinctive ringing.
- Feature interactions for QSIG-VALU Coverage are in most cases the same as DCS Call Coverage.
- The DCS Call Coverage feature refers to Call Coverage feature for interactions.
- This encoding allows QSIG-VALU features to work between DEFINITY[®] systems in a QSIG network, even if the intervening connection includes non-DEFINITY[®] systems, as long as the intervening connection supports QSIG protocol (See the following diagram.).
- The signaling for this feature is built on top of the existing QSIG Call Diversion/ Forwarding supplementary service.



QSIG-VALU Call Coverage

- Display equipped Coverage point sees the display as:

a= <Calling Party's Name OR >	<to>	<Principal's Name OR >	<flag>
Calling Party's Number		Principal's Number OR	
		Incoming Trunk Group's Name	

OR

(When Calling Party's Name and Number not available)

a= <Incoming Trunk Group's Name>
<flag>

- Principal user is able to bridge on to the call while the call is routed to QSIG-VALU coverage points and after the call is answered by the coverage point.
- If QSIG-VALU is enabled, CCRON is enabled and the call is routed over QSIG-VALU trunk, the QSIG-VALU Coverage has a higher precedence than CCRON.
- After the QSIG-VALU coverage call is redirected to the coverage point, the Principal user's switch attempts to redirect the call to the subsequent coverage point.
 - If the subsequent coverage point is unavailable, the call stays at the previous coverage point.
 - If the QSIG-VALU coverage call fails to find any available coverage point, the call stays at the Principal user.

QSIG-VALU Coverage Administration

1. Option Value-Added Lucent (VALU) is Enabled on System-parameters customer-options form
2. The field QSIG Value-Added Lucent is selected as y and Supplementary Service Protocol Option is selected as B on a Trunk Group form to send QSIG-VALU Call Coverage MSI information to a remote switch in association with a call routing on that trunk group.
3. Field Immediate Redirection on receipt of PROGRESS inband information on the System Parameters -Call Coverage / Call Forwarding Form. The default value is n (no).
4. Field QSIG VALU Coverage Overrides QSIG Diversion with Rerouting on the System Parameters - Call Coverage / Call Forwarding is administered. The default value is n (no).
5. Field Maintain SBA at Principal on the System Parameters - Call Coverage / Call Forwarding form is administered. The default value is y (yes).

Remote Call Coverage Table

- The QSIG-VALU Call Coverage in R8 only works with covering users in the Remote Call Coverage Table.

New Administration for QSIG-VALU Coverage

1. Field Immediate Redirection on receipt of PROGRESS inband information on the System Parameters - Call Coverage / Call Forwarding form is administered.
The default value is n (no).
2. Field QSIG VALU Coverage Overrides QSIG Diversion with Rerouting on the System Parameters -Call Coverage / Call Forwarding has been added.
The default value is n.
This field is visible only if the customer purchased both of the following optional features and they are enabled:
 - a. Option Value-Added Lucent (VALU)" (enabled on System-parameters customer-options form).
 - b. Option Supplementary Services With Rerouting (enabled on System-parameters customer-options form).
3. The field Maintain SBA at Principal on the System Parameters - Call Coverage / Call Forwarding form is visible.

QSIG Integration: Transfer to Audix

This feature provides the user with the capability to transfer directly into the voice mail box of a subscriber on the voice mail system when DEFINITY and the voice mail are connected via a QSIG link. QSIG Transfer Into QSIG Voice Mail is not a feature defined by QSIG standards, so MSI is needed to accomplish the desired functionality.

QSIG Transfer Into QSIG Voice Mail uses the existing QSIG Call Transfer along with an MSI message specific to Transfer Into Voice Mail to provide the ability to transfer directly into a subscriber's mailbox.

QSIG Transfer Into QSIG Voice Mail is activated in the same manner as the current Transfer Into AUDIX. Transfer Into AUDIX uses the Transfer feature. Instead of dialing an extension to transfer to, the transferring party dials the FAC for Transfer Into Voice Mail. That is, for a multiline station, while on an existing call, the user presses the transfer button to put the call on hold, then dials the FAC for Transfer Into Voice Mail. The user then may press the transfer button again to complete the transfer and connect the calling party with the voice mail system. Following existing transfer procedures, until the transferring party presses the transfer button a second time, the original call is on hold and the transferring party is connected with the voice mail system.

NOTE:

For pre-R8 releases, the FAC is called the Transfer To AUDIX Access Code.

For a singleline station (without a transfer button), either pressing the switchhook or the recall button provides the same functionality as pressing the transfer button. For an attendant console, pressing the start button provides the same functionality as pressing transfer the first time, and pressing the release button provides the same functionality as pressing transfer the second time.

Alternatively, the user may assign an abbreviated dialing button to contain the Transfer Into Voice Mail FAC. Then, instead of dialing the FAC, the user can press the abbreviated dialing button.

The subscriber's mailbox to which the call is transferred is dependent upon the call scenario, as described in the following paragraphs.

- If the call to the user is either a direct call or the result of a transfer, and the called user activates Transfer Into Voice Mail, the calling party is transferred into the called party's mailbox. This is useful in the cases of fax calls. Suppose a subscriber has a mailbox which is fax enabled and the subscriber is receiving a fax call. If the subscriber answers the phone because they do not know it is a fax, they can transfer the call into their own mailbox which can record the fax.
- If the call is a result of forwarding or coverage and the terminating party activates Transfer Into Voice Mail, the calling party is transferred into the covered-from or forwarded-from party's mailbox. This feature is useful in

cases where a secretary is a coverage point for a boss. Suppose the secretary answers a call which has covered from the boss. If the calling party would like to leave a message for the boss, then the secretary can activate Transfer Into Voice Mail and the calling party will be transferred into the boss's mailbox.

Configuration/Environment

QSIG Transfer Into QSIG Voice Mail requires Definity R8 or higher software, and belongs to Offer Category B on the platforms G3r, G3si, and G3csi.

In order to enable QSIG Transfer Into QSIG Voice Mail, the following options must be enabled:

- ISDN-PRI or ISDN-BRI
- QSIG Basic Supplementary Services

NOTE:

There is nothing to prevent turning on QSIG Transfer Into QSIG Voice Mail when the voice mail system is non-Lucent, but this is not recommended because a non-Lucent voice mail system should not be able to recognize our MSI operation.

Administration

The QSIG Transfer Into QSIG Voice Mail requires the following administration of *existing* fields:

- The system-parameters customer-option form must be administered with option "V8" or later for field "G3 version".
- The system-parameters customer-option form must be administered with option "y(es)" for field "ISDN PRI" , or it must be administered with option "y(es)" for field "ISDN-BRI trunks".
- The system-parameters customer-options form must be administered with option "y(es)" for field "Basic Supplementary Service?"
- The trunk group form must be administered with option "b" for field "Supplementary Service Protocol".
- A Feature Access Code must be assigned in the "Transfer to Voice Mail Access Code" field of the Feature Access Code (FAC) form.

NOTE:

In pre-R8 systems, this field is called "Transfer to AUDIX Access Code".

- A hunt group must be in the coverage path of the user 's mailbox to be transferred into, as administered on the station and coverage path forms. For this hunt group, the hunt group form must have "qsig-mwi" for field "Message Center:" and number of the voice mail system must be administered in field "Voice Mail Number".

The QSIG Transfer Into QSIG Voice Mail procedures also require the following *new* administration, detailed below:

- The system-parameter customer-option form must be administered with option "y(es)" for field "Transfer Into Lucent QSIG Voice Mail?"

The new field can only be enabled if "Basic Supplementary Services?" is also set to "y(es)", and either "ISDN PRI" or "ISDN-BRI trunks" is set to "y(es)", and the G3 version is set to "V8" or later.

```
change system-parameters customer-options
```

```
Page 4 of 4
```

```
QSIG OPTIONAL FEATURES
```

```
Basic Call Setup? n  
Basic Supplementary Services? n  
Interworking with DCS? n  
Supplementary Services with Rerouting? n  
Transfer Into Lucent QSIG Voice Mail? n  
Value Added Lucent (VALU)? n
```

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

The current name of the access code for the existing Transfer Into AUDIX feature is "Transfer Into AUDIX Access Code". Since the QSIG Transfer Into QSIG Voice Mail feature opens up transfer into Audix functionality for voice mail systems that are not Audix, the name of the field has been changed to "Transfer Into Voice Mail

Access Code". There will be no changes to the existing use of the field and no additional admin checking will be done. The only change is to the name of the field.

```
change feature-access-codes                               Page 3 of 5
                FEATURE ACCESS CODE (FAC)
Station Security Code Change Access Code: ___
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: ___
Voice Principal Message Retrieval Access Code: ___
Whisper Page Activation Access Code: ___
```

QSIG CAS & VALU Coverage Interaction Support

QSIG-VALU is a general term that describes those DCS features that will be "ported" to work in a QSIG network consisting of DEFINITY ECS systems. The messaging needed to implement the DCS-like feature transparency will be sent using the Manufacturer Specific Information (MSI) encoding provided by the QSIG standards. This encoding should allow QSIG-VALU features to work between DEFINITY systems in a QSIG network, even if the intervening connection includes non-DEFINITY systems and as long as the intervening connection supports QSIG protocol.

The QSIG-VALU Call Coverage feature will extend the Call Coverage feature transparency when a coverage point is located on a remote DEFINITY system connected via a QSIG network. The calling party, principal and coverage point users, located on DEFINITY, will see the same operational and display behavior for this feature as DCS Call Coverage. The signaling for this feature will be built on top of the existing QSIG Call Diversion/Forwarding supplementary service. The following figures describes the basic scenario:

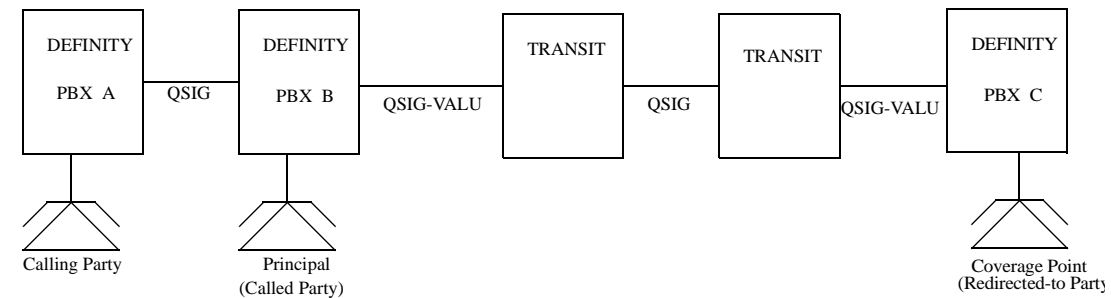


Figure 9. Basic QSIG-VALU Call Coverage Scenario

QSIG-VALU Call Coverage will enhance the operations for the users, similar to DCS Call Coverage, in the following ways:

1. Principal's coverage path and criteria will be used to re-direct the call to QSIG-VALU coverage points.
2. If the QSIG-VALU coverage point is not available, and the call can be redirected to the subsequent coverage point based on the Principal's coverage path, the call to the coverage point is disconnected and the subsequent coverage point is alerted.
3. Coverage Point will be alerted with distinctive ringing.
4. Display equipped Coverage point will see the display as:

```
a= <Calling Party's Name OR > <to> <Principal's Name OR > <flag>
   Calling Party's Number      Principal's Number OR
                               Incoming Trunk Group's Name
```

or, when Calling Party's name and number are not available,

```
a= <Incoming Trunk Group's Name> <flag>
```



NOTE:

<flag> values are: "b" (for Busy/Active), "c" (for Cover all), "d" (for Don't answer), and "s" (for Send All Calls).

5. Principal user will be able to bridge on to the call while the call is routed to QSIG-VALU coverage points and after the call is answered by the coverage point.
6. Calling party will hear a coverage redirection tone before the call is routed to the 1st coverage point if the caller is on the same DEFINITY PBX as the Principal, or if the incoming call received over a DCS or QSIG network provides an indication that the caller is provided with a ring pattern "internal" (DALERT or MSI_DALERT indicating type of originator as internal).
7. Calling party's display will see the reason code "Cover" (for Coverage).
8. After the call is answered by the covering user, QSIG Path Replacement will be initiated, if the Principal user's Simulated Bridge Appearance (SBA) is not maintained, to provide path optimization. If the Principal use's SBA is maintained, then a partial path replacement is possible between either the Calling user's PBX and the Principal user's PBX, or the Covering user's PBX and the Principal user's PBX.

Administration

The QSIG-VALU Call Coverage feature will require the following administration:

1. Option "Value-Added Lucent (VALU)" shall be Enabled on System-parameters customer-options form
2. The field `QSIG Value-Added Lucent` shall be selected as **y** and `Supplementary Service Protocol Option` shall be selected as **B** on a Trunk Group form to send QSIG-VALU Call Coverage MSI information to a remote switch in association with a call routing on that trunk group.

When a call is routed to a remote switch using QSIG-VALU Call Coverage, it will have higher precedence than the CCRON feature. Also, the signaling for this feature is built on top of QSIG Call Diversion/Forwarding supplementary service, so when a call is redirected to a remote switch using QSIG-VALU Call Coverage, the Principal User's switch will act as a Rerouting PBX (i.e. Call Diversion/Forwarding by forward switching).

QSIG CAS Enhancement

The following enhancements to the QSIG Centralized Attendant Service (CAS) have been added:

- Attendant Display of Class of Restrictions (COR)
- Attendant Return of Call
- Display Enhancements
- Priority Queue
- RLT Emulation via PRI

These enhancements require the administration of existing fields as follows:

- The **system-parameters customer-option** form must be administered with option "**V8**" or later for field "G3 version"
- The **system-parameters customer-option** form must be administered with option "**y(es)**" for field "ISDN PRI", or it must be administered with option "**y(es)**" for field "ISDN-BRI trunks".
- The **system-parameters customer-options** form must be administered with option "**y(es)**" for field "Basic Supplementary Service?".
- The **system-parameters customer-options** form must be administered with option "**y(es)**" for field "Supplementary Services with Rerouting?"

Attendant Display of COR

Currently, while on a call with a user on the main PBX, the attendant can press a "COR Display" button and obtain a display that tells the COR of the user. The attendant can then use this information to decide if it is acceptable to transfer this user to a desired destination. QSIG CAS Attendant Display of COR allows the same feature to function in the same manner if the other user is on the branch PBX and the attendant and the user are connected via QSIG ISDN trunks.

QSIG CAS Attendant Display of COR uses QSIG MSI messages. QSIG Manufacturer Specific Information (MSI) "envelopes" exchange QSIG private networking Information Elements (IE's) among nodes of a QSIG network. MSI "envelopes" are designed to transit (i.e., tandem) through other vendors' QSIG nodes, even though the data within is meaningless to them. This adds VALU to Lucent nodes within a multi-vendor network.

Attendant Return of Call

If an attendant transfers a call and that call goes unanswered after an administrable amount of time, the call is returned to the attendant. For a call that originated at the main, the call first attempts to go to the same attendant who originally handled it. If that attendant is unavailable, then the call goes to the attendant group, where it is extended to the next available attendant. For a call that originally came into the attendant from a branch over RLT, the return call would go to the attendant group. CAS Attendant Return Call allows attendant return call to function for calls that come into the attendant from a branch over ISDN trunks in the same manner as calls that come directly to or originate at the main. (There is also a display enhancement for return calls from a QSIG CAS branch, which is described in the CAS Display Enhancement section to follow.)

Display Enhancements

Release Link Trunks (RLT) are limited in the amount of information that can be passed and displayed to the attendant. CAS Display Enhancements are possible by replacing RLT with QSIG ISDN trunks. Display enhancements are achieved by using QSIG MSI messages. Display enhancements for calls coming into the attendant from a branch are:

- For an incoming LDN call, LDN name or number is shown
- For a '0' out of Audix call, a reason code and called number are shown
- For return calls, the called name is shown

Priority Queue

On the console-parameters form, customers can administer the priorities of incoming attendant calls that must wait in the Attendant Priority Queue because no attendant is immediately available. Previously, with RLT CAS, all calls incoming to the attendant from a branch over RLT trunks were assigned the priority administered to "Tie Trunk Calls." Now, using QSIG ISDN trunk between the branch and main, more information can be passed to the main PBX using QSIG MSI messages. This extra information enables calls coming in from a QSIG CAS branch to be placed in their appropriate place in the queue, as if the call came directly into the main.

RLT Emulation via PRI

Previously, Release Link Trunks were required in a Centralized Attendant Service. The purpose of the RLT is to route calls from the branch PBX to main PBX, and to provide trunk optimization in the case that the attendant subsequently transfers the call. Today, the same effects can be achieved by using only ISDN QSIG trunks. CAS RLT Emulation via PRI allows customers to set up a Centralized Attendant Service using solely ISDN QSIG trunks, and no RLT are needed. Using QSIG Path Replacement, trunk optimization of QSIG trunks is possible.

Centralized Voice Mail Via Mode Code

You can use a single voice mail system to support multiple DEFINITY and Merlin Legend/Magix systems in a network via mode code. This capability is available for:

- DEFINITY ECS R8
- Prologix R3
- BCS Issue 6
- Merlin Legend R6.1
- Merlin Magix 1.0



NOTE:

DEFINITY One and Guestworks are not supported.

Voice mail systems that support these connections are:

- Intuity AUDIX R4.4 or higher running on a MAP5, with up to 18 ports
- Octel 100, with up to 16 ports

Configuration requirements

Centralized voice mail via mode code requires the following:

- A DEFINITY ECS switch as the hub of the voice mail network, with the voice mail system directly connected to it.
- Direct ISDN PRI tandem trunk connections, using DS1 service between the DEFINITY ECS hub and the switches the voice mail supports. The system uses the D-channel to transmit mode code signals to light message waiting lights on remote extensions.
- A uniform dial plan for all switches in the network, with a 4-digit plan if Merlin Legend/Magix is part of the network.
- One and only one mailbox for each extension in the network.



NOTE:

DCS software, X.25 hardware, and CLAN hardware/software are not required for this type of network. Additionally, you cannot network switches simultaneously using both mode code and DCS.

Configuration example

[Figure 10](#) shows what a configuration of centralized voice mail via mode might look like.

In this configuration, DEFINITY system A is the hub. Voice mail system X is the centralized voice mail system. All other systems in the network are supported by voice mail system X *except* Legend system E and DEFINITY system D. These switches do not have a direct ISDN-PRI connection to the hub.

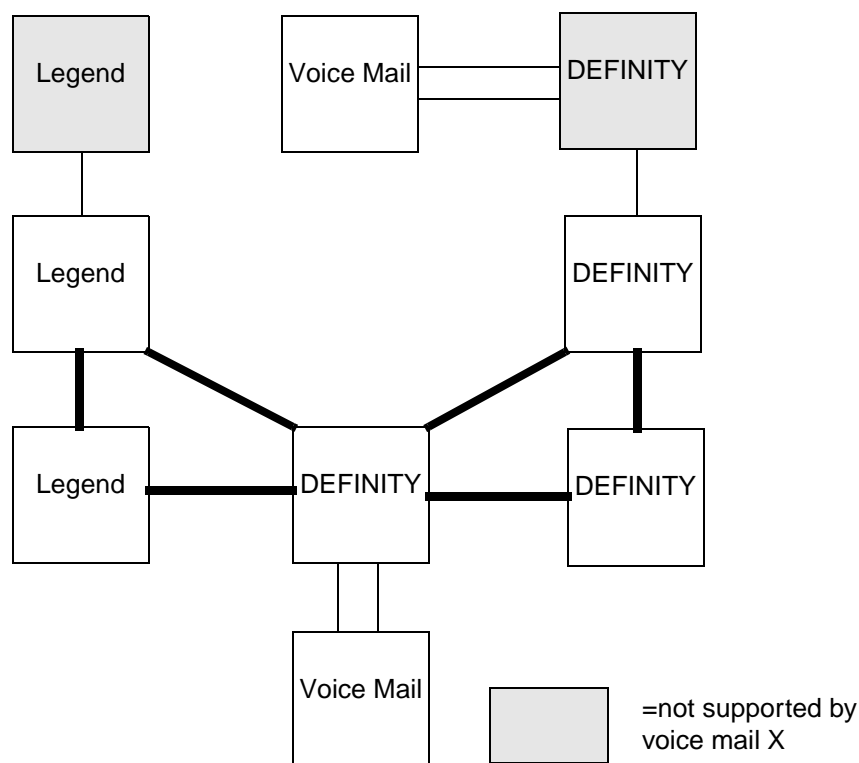


Figure 10. Centralized Voice Mail Via Mode Code Example Configuration

Feature Support

Features that are supported are:

- Calling party name/number sending/retrieval
- Message waiting light activation
- Remote coverage to voice mail
- Fax, as well as voice, mail

Features Not Supported

The following capabilities are not supported via mode code.

- Most DCS feature transparency.
- Centralized voice mail for a tandem switch (does not have a direct connection to the hub switch).
- Transfer into voice mail

Setting Up Centralized Voice Mail Via Mode Code

The following step describe how to set up DEFINITY switches to support centralized voice mail. For information on setting up Merlin Legend/Merlin Magix, see *Merlin Magix Integrated Systems Network Reference*, 555-715-114. For information on setting up INTUITY Messaging Solutions, see *INTUITY Messaging Solutions Integration with System 75, Generic 1 and 3, and R5/6*.

1. Enter **change system-parameters customer-options** on the command line of your system administration screen.
2. On page 3 ([Figure 11](#)), verify fields as follows:
 - **ISDN-PRI** field is **y**.
 - **Mode Code for Centralized Voice Mail?** field is **y**.
 - **Mode Code Interface?** field is **y**.
 - **UDP** field is **y**.

```

change system-parameters customer-options                               Page 2 of 4
                                OPTIONAL FEATURES

                                Hospitality (Basic)? y
Hospitality (G3V3 Enhancements)? n
                                H.323 Trunks? n           Processor and System MSP? n
                                IP Stations? n             Private Networking? n
                                SDN Feature Plus? y        Restrict Call Forward Off Net? y
                                ISDN-BRI Trunks? y         Secondary Data Module? y
                                Malicious Call Trace? n    Survivable Remote Processor? n
Mode Code for Centralized Voice Mail? y                 Tenant Partitioning? n
                                Mode Code Interface? y    Terminal Trans. Init. (TTI)? n
                                Multifrequency Signaling? y Time of Day Routing? n
Multimedia Appl. Server Interface (MASI)? n                Uniform Dialing Plan? y
                                Multimedia Call Handling (Basic)? n Usage Allocation Enhancements? y
Multimedia Call Handling (Enhanced)? n
                                Wideband Switching? n
                                Personal Station Access (PSA)? n Wireless? n

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

Figure 11. Modified “change system-parameters customer-option” form

3. On the hub switch, enter **add trunk group xxxx** on the command line of your system administration screen, where **xxxx** is the number of the ISDN-PRI trunk group connecting the hub with the remote switch.
4. On page 1, verify fields as follows:
 - **Group Type** field is **ISDN**.
 - **Service Type** field is **TIE**.
5. On page 2, verify fields as follows:

- **Send Name** field is **y**.
 - **Send Calling Number** field is **y**.
 - **Numbering Format** field is **Private**.
 - **Send Connected Number** field is **y**.
6. On each remote switch, repeat steps 3-5.
 7. On each DEFINITY switch in the network, enter **change dialplan** on the command line of your system administration screen.
 8. Administer the dial plan for each node in the network (see Chapter 4 in this document for examples). Usually the hub is considered Node 1.

If the network contains Merlin Legend/Magix systems:

- **Uniform Dialing Plan** field is **4** for each node.

If the network contains only DEFINITY systems:

- **Uniform Dialing Plan** field is **4** or **5**.

9. For each DEFINITY node, enter **change feature-access-codes** on the command line.
10. On page 2, verify fields as follows:
 - **Leave Word Calling Send a Message** field is **#90**.
 - **Leave Word Calling Cancel a Message** field is **#91**.

 **NOTE:**

All nodes in the system and the Voice Mail system must match this setting

11. For each DEFINITY node, enter **add ds1UUCSS** on the command line, where **UUCSS** is the address of the DS1 circuit pack.
12. On page 1, verify fields as follows:
 - **Line Code** field is **B8ZS**.
 - **Framing** field is **extended superframe**.
 - **Signaling Mode** field is **ISDN/PRI**.
 - **Connect** field is **PBX**.
 - **Interface** field is **network** (for the hub) and **user** (for the remote switch).

 **NOTE:**

Mode Codes *will not work* with D4 or SuperFrame

13. For each DEFINITY node, enter **change sign next** on the command line. Administer the signaling group (see example in Chapter 4 of this document).

14. For each DEFINITY node, enter **change isdn private-numbering**, and verify fields as follows:
15. **Set Network Level** field is **0**
This setting overrides the signaling on the D channel, allowing the Message Waiting lamp activation signal to be sent
16. On the DEFINITY node, enter **change sys params mode-code** on the command line.
17. On the hub switch, set the **VMS Hunt Group Extension** field to the voice mail hunt group extension ([Figure 10](#)).

```

MODE CODE RELATED SYSTEM PARAMETERS
Page 1

MODE CODES (FROM SWITCH TO VMS)
  Direct Inside Access: __
  Direct Dial Access - Trunk: __
  Internal Coverage: __
  External Coverage: __

  Refresh MW Lamp: __

  System In Day Service: __
  System In Night Service: __

OTHER RELATED PARAMETERS
DTMF Duration On (msec): __ Off (msec): __ Sending Delay (msec): __
VMS Hunt Group Extension : ____
Remote VMS Extensions - First:      Second:
```

Figure 12. Mode Code Related System Parameters

18. On the remote switches, repeat Step 15. Enter the voice mail hunt group extension in the **Remote VMS Extension - First** field.
19. For each DEFINITY node, enter **change aar an** on the command line.
20. Verify the following:
 - **Call Type** field is **lev0**
21. On the hub switch, enter **change station extension** for each port extension in the voice mail hunt group.
22. On Page 1, verify the following:
 - **Type** field is **vmi**.
23. On Page 2 of the Station screen, administer or verify the following:
 - **LWC Reception** field is **msa-spe** (Message Server Adjunct-System Processing Element).
 - **Leave Word Calling** field is **y**.
 - **Adjunct Supervision** field is **y**.

- **Distinctive Audible Alert** field is **n**.
 - **Switchhook Flash** field is **y**.
 - **LWC Activation** field is **y**.
24. For each remote DEFINITY node, enter **change coverage remote** on the command line.
 25. Administer or verify the following:
 - 01 contains the extension of the voice mail hunt group.

Support for Japan National Private Networking

In release R8, DEFINITY's implementation of ISDN PRI for Japan will be modified to support the private networking environment. The private protocol borrows heavily from the national specific public ISDN PRI specification. The differences are such that the Public Network protocol is insufficient to provide basic call support in their private networks. Existing supplementary services implementations of calling line and connected line information presentation and restriction will be tested for inclusion. There is no expectation to modify these implementations for operation in Japan. From a marketing point of view, this protocol is being implemented to penetrate the existing private networking market that is already entrenched. In the near future, QSIG will become a dominant private networking protocol as Japanese vendors implement it.

The Japanese standards organizations, TTC, has defined QSIG to be the same as the ISO standard which DEFINITY already supports. This private networking protocol will provide the basic level of networking in order to meet the immediate connectivity needs of our customers. QSIG will be positioned, going forward, as the private networking platform for Japan. A new administration option will be developed to distinguish this protocol from the Japanese public network.

- 2 MB Trunk - Ph2: Codeset 5 TCM's
- 2 MB Trunk - Ph2: Connect Ack
- 2 MB Trunk - Ph2: Different Protocol Discriminator
- D Channel - Private ISDN Ntwks - Q.931.a, Q.932.a (Basic Call Setup)
- D Channel - Private ISDN Networks - 951.a

Networking - ISDN Public

BellCore Calling Name ID

Bellcore Format with V.23 FSK

- The 24-Port Analog Line with CID feature supports the option to send Bellcore specification messages using V.23 modem protocol.
- Bellcore Format with V.23 FSK differs from the standard Bellcore format only in the frequencies used for the FSK signaling in the Physical Layer.
- Encompasses some Middle East countries such as Bahrain.

It is necessary to set the US NI Calling Name for outgoing Calls? field to **y/n** on page 6 of the **change system-parameters features form**, as shown below.

```
change system-parameters features                               Page 6 of 10
                        FEATURE-RELATED SYSTEM PARAMETERS

ISDN PARAMETERS

  Send Non-ISDN Trunk Goup Name as Connected Name? _
  Display Connected Name/Number for ISDN DCS Calls? _
    Send ISDN Trunk Group Name on Tandem Calls? n
      CPN Replacement for Restricted Calls: _____
      CPN Replacement for Unavailable Calls: _____
        QSIG TSC Extension: _____
          Path Replacement with Measurements? n
  MWI - Number of Digits Per AUDIX Subscriber: _
    Feature Plus Ext: _____
      National CPN Prefix: _____
      International CPN Prefix: _____
        Pass Prefixed CPN to ASAI? n
  Unknown Numbers Considered Internal for AUDIX ? n Maximum Length: 5
    US NI Calling Name for Outgoing Calls? n
```

Screen 1. Feature-Related System Parameters screen

US NI Calling Name for Outgoing Calls?

Valid entries	Usage
y/n	<p>Enter y to send a name on outgoing calls over NI PRI trunks.</p> <p>Important: Be sure you have validated that your service provider's central office is capable of accepting calling name information from DEFINITY in this way. For example, if the central office switch is a 5ESS, it must be a generic 5EXX or later. Failure to validate the central office capability may cause the central office to drop outgoing calls from your switch. In this case, change the value in this field to n.</p> <p>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.</p>

Bellcore National ISDN Calling Name Supplementary Service for PRI

- Provides the user with the capability on incoming calls, to receive Network provided Calling Name information, along with name presentation indication.
- Provides the user with the capability on outgoing calls, based on trunk group administration, for DEFINITY to provide the calling name and its presentation status to the network.
- Enables DEFINITY to provide users the same Calling Name capability provided by Central Office to residential customers.



NOTE:

DEFINITY only supports the Bellcore National ISDN Calling Name feature for PRI, and does not support the Bellcore National Calling Name feature for BRI.

There is no support for original called name information when a call has been forwarded or redirected.

Bellcore Calling Name is a public network feature and its feature operation is very much dependent on how the National ISDN trunk interfaces are provisioned at the CO switch (whether to accept CPE originated Calling name, how to deliver Calling Name with or without Privacy Override).

Capacity Constraints and Feature Limitations

- DEFINITY will be capable of receiving up to 50 characters of Bellcore National ISDN Calling Name information but display a maximum of 15 characters.
 - Truncation behaviors will truncate names longer than 15 characters to the first 15 characters to be displayed.



NOTE:

DEFINITY does not support optional capabilities to interpret/display the original called name information when a call has been forwarded or redirected.

- If the Bellcore Calling Name information is received during Call Setup time, this feature presents the same end user interface when displaying calling name information.
- Based on user administration for this feature, if Calling Name is received later in a FACILITY message, there may be a display update to include Calling Name in the “calling” part of an incoming call display.
 - This display update may erase any Call Center collected/prompted/ CINFO/II digits being displayed on agent CallMaster sets but agents can redisplay the digits via a feature button.

Feature Administration

- Available if “ISDN-PRI” is administered on the system-parameters customer option form.
- Following the DEFINITY administration scheme, the Bellcore Calling Name feature for PRI is selected for ISDN PRI or ATM trunk groups as follows:
 - The DS-1 interface for PRI or the signaling group for ATM Trunk is administered with country protocol “1b” for Bellcore.
 - The ISDN Trunk Group SS option (Supplementary Service option) is administered “A” for National Supplementary Service protocol .
 - The ISDN Trunk Group “Send Name:” field is administered to either “y” or “n” in order to originate Calling Name information for outgoing calls.

Calling Name Display Update

- Before DEFINITY release 8.1 calling number and/or calling name display to the called station was provided when the SETUP message was received, except in the case of Call-by-Call subscription of ANI delivery by the AT&T PRI network.

- The 5ESS network database query timer is set to one to six seconds (three seconds is the default).
 - The aim of DEFINITY support of the Bellcore Calling Name is to present as similar a user interface as the LEC Caller-ID service. Caller-ID usually delivers the Caller ID information on the 2nd ring cycle with each ring-silence cycle defined as 6 seconds.
- Given the Caller-ID information delivery timing, the LEC network switch should be delivering the Calling name information to DEFINITY within 2 rings as the call is being alerted and therefore there is no need to delay any call alerting on DEFINITY.
 - Customers can select whether to get a display update with name information if name was not initially displayed when the call was first alerting at the Called station.

Bellcore National ISDN Calling Name Update

- A new administration parameter added to the ISDN Trunk Group form if the "SS option" field is set to "A":
- The new field "US NI Delayed Calling Name Update?" is a "y/n" field and defaults to "n"(o).
 - If it is set to "n" and Calling Name information is received after the incoming call has been delivered to the Called station, there will be no display update to the terminating station with the Calling name information.
 - If the field is set to "y" and Calling Name information is received after the incoming call has been delivered to the terminating station, there will a display update with the Calling Name information in the "calling" portion of the display.
 - This field can only be set to "y" for ISDN PRI or ATM trunk group types with corresponding trunk interface that selected "1b" as the Country Protocol (on DS-1 form for PRI trunk or on signaling group form for ATM trunk).
 - If this field is set to "y" for other ISDN trunk group types, at form submission time the SAT cursor will placed on this field and the following error message displays "BRI trunks do not support this value".

Option A Change

- With the support of Bellcore Calling Name under SS Protocol "A" (National SS), the "r" choice is allowed.
- DEFINITY Call Processing code enforce a check if "Send Name: r" is administered for an SS Option A trunk group not associated with a Country Protocol "1b" PRI interface results in no Calling name being sent (as if the "Send Name:" field is set to "n").

SS Option A Choices

- The "Send Name:" field on page 2 of ISDN Trunk Group form allows the "r"(estricted) choice when the "Supplementary Service Protocol" is set to "A" (National SS protocol). Standard error and help messages apply.

Trunk Group Page 2 screen with the new field choice and new field.

```
change trunk-group 1                                     Page 2 of 10
TRUNK FEATURES

      ACA Assignment? n           Measured: none           Wideband Support: n
                                Internal Alert? n           Maintenance Tests? y
                                Data Restriction? n           NCA-TSC Trunk Member:
                                Send Name: r                 Send Calling Number: y

      Used for DCS? n
      Suppress # Outpulsing? n   Numbering Format: public
      Outgoing Channel ID Encoding: preferred UUI IE Treatment: service-provider

                                Replace Restricted Numbers? n
                                Replace Unavailable Numbers? n
                                Send Connected Number: y

      Send UCID? n
      Send Codeset 6/7 LAI IE? y

                                US NI Delayed Calling Name Update? n
```



NOTE:

For the Bellcore National ISDN Calling Name feature for PRI, DEFINITY supports the callingName, NAME, NamePresentationAllowed, NamePresentationRestricted, NameNotAvailable, NameData, NameSet, and CharacterSet.

Interworking

- DEFINITY supports Calling Name for the following types of ISDN networks:
 - QSIG private network (multi-vendor networks)
 - DCS/DCS+ private network (DEFINITY only networks)
 - AT&T Custom PRI private network (DEFINITY only networks)
 - AT&T Custom PRI public network (4ESS networks)
- Calling Name information interworking is supported between any two of these networks
 - QSIG
 - DCS/DCS+
 - AT&T Custom PRI private or public network.

- With the addition of Bellcore National ISDN Calling Name feature, interworking of Calling Name in the Bellcore National ISDN network will also be fully supported if the Bellcore Calling Name information is available in the SETUP message.
 - There is no interworking of the Bellcore Calling Name with other PRI networks if the Bellcore Calling Name is not in the SETUP message.

Feature Plus – Non-DID Calling via UDP

ISDN Feature Plus Phase 2

- Provides users without Direct Inward Dialing (DID) the capability of direct dialing (via the public network) users on a remote PBX.
- Available according to the following parameters:
 - ISDN Feature Plus is a set of features using non-standard ISDN signaling to give users feature transparency over public networks.
 - A non-blank subaddress entry on the route pattern form causes routing digits to be sent to the terminating end.
 - When these digits are received, the call will route based on the received routing digits.
 - Must only be used for route preferences destined for public LDNs.
 - For networks supporting less than the standard 23 octet Subaddress IE, either calling name or Pseudo DID information can be transported, but not both.
 - Recommended to turn off calling name since it will be truncated to only 6 characters.
 - For users without DID, remote extensions can only be reached over public networks by dialing the Listed Directory Number (LDN) of the remote location and having the remote attendant redirect the call to the appropriate extension.
 - Using the Feature Plus platform to provide Pseudo DID, a remote user's extension is transported across a public network in a Subaddress IE.
 - The remote user is accessed by their assigned extension and any other routing digits required, such as an AAR access code.
 - Utilizing existing DEFINITY® system digit manipulation, the remote user's extension is converted to the correct public LDN.

- When a LDN call is received, the incoming SETUP message is checked for a Calling Party Subaddress IE containing a DID extension.
- If the extension is present, the call will not be sent to the attendant, as is currently the case. The call will, instead, be routed to the indicated extension.
- Feature Plus Pseudo DID will only work over trunk groups assigned Supplementary Service Protocol Option f.
- Requires the originating and terminating nodes in a network to be DEFINITY® ECS release 8 or later.
- Tandem nodes in the network may be other vendor's so long as they support transmitting the Calling Party Subaddress IE in SETUP.
- A pre-release 8 DEFINITY® ECS can also be included in the network as a tandem node.
- Applicable offer category is B.

Administration

- The number of digits to send in the Calling Party Subaddress IE must be administered.
- In order to allow the most flexibility for routing alternatives, the administration was added to the route pattern form.
- New field only administered if the software is version 8 or above and ISDN Feature Plus is enabled.
- Sending Feature Plus information is only valid on trunk groups administered with Supplementary Service Protocol.
- A new numeric field No. Dgts Subaddress added to the Route Pattern form.
 - Default is blank.
 - Field can not be administered unless the switch is administered as version 8 or later.
 - Field can not be administered unless ISDN Feature Plus is enabled.
 - Field can not be administered unless the trunk group is ISDN.
 - Field is only valid for ISDN trunk groups.

Route Pattern Form

Page 1 of 3

change route-pattern 1
Pattern Number: 1 0

Grp. No.	FRL	NPA	Pfx	Hop	Toll	No. List	Del	Inserted	IXC
No.	Mrk	Lmt				Digits		Digits	
1:	---	---	---	---	---	-----			user
2:	---	---	---	---	---	-----			user
3:	---	---	---	---	---	-----			user
4:	---	---	---	---	---	-----			user
5:	---	---	---	---	---	-----			user
6:	---	---	---	---	---	-----			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	n		rest			-----	next
3:	Y	Y	Y	Y	Y	n	n		rest			-----	rehu
4:	Y	Y	Y	Y	Y	n	n		rest			-----	none
5:	Y	Y	Y	Y	Y	n	n		rest			-----	none
6:	Y	Y	Y	Y	Y	n	n		rest			-----	none

Example

- Leased line from DEFINITY® system A to DEFINITY® system B.
- The first routing preference can be assigned to the leased line so User B1 can be dialed directly without sending Pseudo DID subaddress information.
- If preference 1 is not available, preference 2 can be assigned to the LDN on DEFINITY® system B, and the dialed Pseudo DID extension can be sent as the subaddress.
- The 4000 block of extensions was not purchased from the public network and extension 4010 in the public network.
- With Feature Plus, User A can reach User B1 directly by dialing 94010. Digit manipulation must be administered (AAR digit analysis and route pattern) to delete the dialed extension (4010) and insert the LDN (303 XXX XXXX).
- The LDN is sent in the Called Party IE as the destination address and the DID extension is sent in the Calling Party Subaddress IE.
- Using the DID extension received in the Subaddress IE at DEFINITY® system B, the call terminates to User B1 rather than to the attendant. If User A wants to reach the attendant, the full ARS number must be dialed (for example, 8 303 XXX XXXX).

Alerting Ring Pattern

- If the incoming trunk group, which terminates to the LDN, is administered as tie, the ring pattern administered on the trunk for incoming calls apply.
 - If Internal Alert? is yes, the system ring pattern administered for internal calls apply.
 - If Internal Alert? is no, the system ring pattern for external calls apply.
 - If the trunk group is not tie, the system ring pattern administered for external calls applies.
- When calls are received for LDNs and the switch is in night service, the calls are redirected to the administered night service extension.
 - The night service extension are ignored for Pseudo-DID calls and routed to the received extension regardless of night service status.

Shared Calling Party Subaddress IE Contents

- When a Pseudo DID extension is assigned in conjunction with calling name (trunk group Send Name? y), both the calling user's name and the DID extension are transported in the Calling Party Subaddress IE.
- By default, for networks supporting less than the standard 23 octet IE, calling name information is discarded.

Terminating DID Subaddress

- When a call is received over an SSF interface for an LDN extension, the Calling Party Subaddress IE is checked for a DID extension.
- If present, the call is routed to the DID extension rather than the attendant.
- Once the call has been routed to the DID extension, all procedures for a direct incoming call apply.
- Since a DID extension received as a subaddress can be the result of AAR, ARS or UDP dialing on the originating side, ringing applies based on the administered ring pattern for external calls unless the trunk group is administered with Internal Alert? yes (applies only to tie trunks).

Multiple Pubnet Calling/Connect Numbers/ System (Italy)

This feature provides multiple public-network calling/connected numbers per system. It provides the user with the capability to send out different "main" numbers for trunks associated with different network providers.

- In Tenant Partitioning where each tenant is using specific trunks, this can ensure that proper calling/connected numbers are sent out based on the trunks used.

- Public networks may check the accuracy of the Calling Line Identification number information sent by a CPE.
- The number is checked to make sure it is within a provisioned range.
- If the number is not accurate based on the trunk group used, the DEFINITY[®]-provided Calling Number would be deleted and a network-provided "main billing" number would be used instead.
- This creates a situation in which the caller's extension information is lost, adversely impacting the caller identification in both emergency and normal situations.
- In outbound call management applications for third party companies, multiple trunks and numbers would be used and accurate Calling Number information is provided by DEFINITY[®] system on call originations.

It also passes network Advice of Charge (AOC) information to all protocol WCBRI endpoints provides the user with the capability to tandem transparently network provided AOC information to all WCBRI endpoints. The AOC information can be used by WCBRI stations for local station display purposes. Network AOC information to be passed include: ETSI, CEPT Italy, and NTT INS-NET 1500 Japan.

Capacity Constraints and Feature Limitations

- When a switch is administered for the multiple public network numbering enhancement, the number of administered entries in the ISDN public-unknown number form could increase by n fold with "n" being the number of distinct main numbers that the switch needs to originate for outgoing calls.
 - The maximum form entries (8 pages of 30 entries each) was considered sufficient, and therefore not increased.
- Maximum number of number digits displayed including any national/international prefix remains at 15 digits.
 - Numbers longer than 15 digits will be truncated to the first 15 digits of number information (this is consistent with the current DEFINITY[®] system number/name truncation behaviors).
- Maximum national or international prefix length is 5 digits.
- Network provided AOC information is tandemed transparently to WCBRI endpoints.
- Display of the AOC information at end user stations is dependent on the station capability.
- DEFINITY[®] system does not control WCBRI station display functions.
- There is no mapping or conversion of Network side (PRI) AOC protocol to Line side (WCBRI) AOC protocol.

Configuration/Environment

- These enhancements are available regardless of the administered software version on a DEFINITY[®] switch.
- The ISDN enhancements are considered as basic networking requirements for many international countries, and thus are available in both Offer A (ECS, including a full DEFINITY[®] with expensive circuit packs) and Offer B (BCS and GuestWorks[®]) categories.

Feature Interactions

Call Coverage Remote Off-Net (CCRON)

- When a remote coverage call is brought back into the DEFINITY[®] system to follow the called party's original coverage path, the call is subject to prefixing if the prefixing feature is administered.

Calling Line Identification (CLI) Prefix

- Can apply a prefixing treatment (digit deletion, digit insertion, new PI encoding) to outgoing tandem calls (an updated CPN IE is sent out) and to locally terminated calls (an updated CPN number information is displayed).
- If CLI Prefix is administered on a trunk group, any incoming Calling Party Number IE is modified first by the CLI Prefix feature, then any International/National prefix is applied to the modified CPN IE.
- If CLI Prefix is administered for a Trunk Group, any incoming Calling Party Number IE is first modified by the CLI Prefix feature.
- If the call is to be tandemmed out, the CLI Prefix modified CPN IE is sent in the outgoing SETUP message per existing DEFINITY[®] system behavior.
- If the call terminates locally, the CLI Prefix modified CPN IE is checked to see if any International/National prefix should be applied.

Display

- If the International/national prefix feature is administered, features that would display any number information (Calling, Connected, QSIG-VALU Called, QSIG-VALU Busy) display any resulting prefix-appended national/international number to user stations.

ISDN – Basic Rate Interface

- The Calling Party Number IE tandemmed to BRI interface remains the originally received CPN IE from PRI, without any prefixing applied.

ISDN – WCBRI

- The Calling Party Number IE tandemmed to WCBRI interface remains the originally received CPN IE from PRI, without any prefixing applied.

ISDN – DCS over ISDN PRI D-Channel (DCS+)

- DCS+ may be administered to display the ISDN Connected number information, which will be subject to the prefixing enhancement.

Adjunct Switch Applications Interface (ASAI)

- For many customers over the ASAI interface, the received Calling/ Connected Party Number IE will be sent, without any International/National prefix.
- In certain CTI applications, there is a need to receive prefixed Calling/ Connected Number information so the correct call-back applications may be launched using the prefixed number information.

Passing Prefix to ASAI

- If the Pass Prefixed CPN to ASAI? field on the system-parameters features form isdn page is set to y, and if a Calling Number or a Connected Number received a prefixing treatment, the prefixed Calling/Connected Number information is used in the CPN IE digit string in Octet 4 and beyond sent out over any ASAI BRI link.

Administration



NOTE:

The AOC enhancement does not require new administration.

Multiple Public Numbering Admin for Trunk Groups

- Add a new field, Trk Grp(s), for each CPN isdn public-unknown-numbering form.
- Field can be set to any valid administered “isdn” type trunk group's number, or to any range of valid administered “isdn” type trunk group numbers.
- Field defaults to blank and can have valid values of 1 to the maximum number of trunk groups supported on the system.
- If this field is blank, the identification numbers (Calling, Connected, QSIG-VALU Called, or QSIG-VALU Busy) originated will not be dependent on which trunk group the call is carried on.

National and International Prefix Administration

- The ISDN parameters section of the system-parameters features form (which only appears if the administered software version is V5 or later) new administrable number/text fields for the prefixing feature.
- The fields entitled National CPN Prefix: and International CPN Prefix: have a field length of 5.

- The default for these fields are blank, to be interpreted that no national or international prefix is appended to identification numbers displayed.
- Valid entries for these fields include any digit 0-9, *, and #.
- Any digit or * or # may be entered in any position within the prefix and may be repeated within the prefix.
- The ISDN parameters section of the system-parameters features form new field Pass Prefixed CPN to ASAI?.
- Valid values of n (no) and y (yes) and shall default to n (no).
- Can be administered regardless if ASAI has been selected on the system-parameters customer-options form.

Numbering Form Administration

Example

- User extensions range from 300 to 900 and these extensions can make outgoing calls and receive incoming calls.
- The customer currently has 5 total entries for these extensions in the isdn public-unknown numbering form administered to send a single main number "672987" followed by the extension number.
- ACD agents extensions range from 2000 to 2999 and these extensions can receive incoming calls only and are not allowed to make external calls.
- There are no isdn public-unknown numbering form entries for these extensions.
- There are 59 ISDN PRI trunk groups, 24 trunk groups are bidirectional, 29 trunk groups are used for incoming calls only, and 6 trunk groups have no members and are used for testing only.
 - 24 trunk groups for which the switch may send Calling/Connected/QSIG-VALU Called/QSIG-VALU Busy numbers, and 29 trunk groups for which the switch may send Connected/QSIG-VALU Called/QSIG-VALU Busy numbers.
- 29 physical E1 PRI trunk interfaces representing actually 8 different main numbers.
- If the customer decides to administer to send different main numbers based on trunk groups used, and assuming that:
 - Stations with 3 digits extensions can send Calling Number when using a single trunk group numbered 1; stations with 3 digits extensions can send Connected/QSIG-VALU Called/QSIG-Valu Busy Numbers when using trunk group 1.
 - If one numbering form entry is used for each set of extensions (300-900), a total of 7 entries would be entered.

- Stations 2000 to 2999 can send Calling Number information when using the 24 bidirectional trunk groups numbered 2-25; stations 2000 to 2999 can send Connected/QSIG-VALU Called/QSIG-VALU Busy Number information when using the 29 incoming-only trunk groups numbered 26-54.
 - If one numbering form entry is used for the entire group of 2000-2999 extensions, a maximum of 53 entries may be entered (if no trunk group number range is used, but each individual trunk group number is entered).
 - But if some trunk group ranges are used, the number of entries could be significantly less.

Pass Prefixed CPN to ASAI? field

- Default: n
- Help: n (no) y (yes)
- Error(s): x is an invalid entry; please press HELP

Administering the CPN Prefix feature

- Select an ISDN networking option on the system-parameters customer-options form either ISDN or ISDN-BRI Trunks.
- If a national prefix is desired, administer the National CPN Prefix field on the system-parameters features form.
 - If an international prefix is desired, administer the International CPN Prefix field.
 - The appropriate prefix is then applied to all incoming ISDN calls with the corresponding type of national or international Calling Number encoding.
- Administer the Pass Prefixed CPN to ASAI? field to y if desiring the non-default value, then any Calling Number for inbound calls passed to ASAI will include prefix if the call should get prefix treatment, or any Connected Number for outbound calls passed to ASAI will include prefix if the call should get prefix treatment.

Administration Procedures

Step by step

- Select an ISDN networking option on the system-parameters customer-options form: either ISDN or ISDN-BRI Trunks.
- Administer two or more ISDN type trunk group with trunk group number 11 and 22.

- Administer entries in the isdn public-unknown- numbering form
 - For each Ext Len/Ext Code combination desired, one entry may be entered with 11 as the Trk Grp(s) value, another entry may be entered with 22 as the Trk Grp(s) value.
 - Enter the appropriate CPN Prefix and Total CPN Length values for each entry.
 - For calls originated in Trunk Group 11, a CPN number is generated for outgoing calls using the Ext Code/CPN Prefix/Total CPN length values for the Trk Grp(s) 11 entry in the numbering table.
 - For calls originated in Trunk Group 22, a different CPN number is generated by using the Ext Code/CPN Prefix/ Total CPN Length values for the Trk Grp(s) 22 entry.
- If trunk groups numbered from 11 to 22 should all use the same CPN Prefix (for the same public trunk's main number), one entry in the isdn public-unknown numbering form can be administered with 11-22 in the Trk Grp(s) field.

Call Processing

- Originating Calling/Connected Number Based on Trunk Group Numbers
 - For ISDN calls, if a specific Identification Number (Calling/Connected) is to be originated by the DEFINITY[®] system (if the Trunk Group form Page 2's Numbering Format field is public), DEFINITY[®] use of the ISDN Number-Public/Unknown table to determine the number sent is enhanced to check the new Trk Grp(s) field.
 - If there is an Ext Code entry for the station in question and if the Trk Grp(s) field is administered to the Trunk Group number that the call is carried on, or if the Trk Grp(s) field is administered to a Trunk Group number range which contains the Trunk Group number that the call is carried on, the station's Identification number is generated using this entry's administration.
 - Otherwise, an entry with Ext Code that matches the station extension and with a blank Trk Grp(s) field is used to generate the station's Identification number.
 - DEFINITY[®] system implements QSIG-VALU Called and Busy Number procedures.
 - Any prefixing also applies to these types of identification numbers received.

- Originating QSIG-VALU Called/Busy Number based on Trunk Group Numbers
 - For ISDN calls, if a specific QSIG-VALU Number (Called/Busy) is originated by the DEFINITY® system using the ISDN Number-Public/Unknown table to determine the number sent is enhanced to check the Trk Grp(s) field.
 - If there is an Ext Code entry for the station in question, and if the Trk Grp(s) field is administered to the Trunk Group number that the call is carried on, or if the Trk Grp(s) field is administered to a Trunk Group number range which contains the Trunk Group number that the call is carried on, the station's Identification number is generated using this entry's administration.
 - Otherwise, an entry with Ext Code that matches the station extension and with a blank Trk Grp(s) field is used to generate the station's Identification number.

Example

- Any prefixed Calling/Connected/QSIG-VALU Called/QSIG- VALU Busy number that's longer than 15 digits is truncated from the least significant digits.
- The prefixing feature does not change any DEFINITY® system display format and any prefixed CPN/Called/Busy numbers continues to be displayed.

Pass network Advice of Charge (AOC) information to all protocol WCBRI endpoints

- Provides the user with the capability to tandem transparently network provided AOC information to WCBRI endpoints.
- The AOC may be used by WCBRI endpoints for local station display purpose.
- Network AOC information to be passed include: ETSI, CEPT Italy, and NTT INS-NET 1500 Japan.

Advice of Charge

- The WCBRI Platform supports these line-side country protocols
 - Bellcore National ISDN-1
 - Australia
 - ETSI
 - Japan
 - Singapore

- Network provided AOC is tandemed to WCBRI regardless of the administered WCBRI country protocol.



NOTE:

If the far end hangs up on a call, AOC may be passed to WCBR.

If the local WCBRI endpoint hangs up first, any AOC received from the network side will not be passed to the WCBRI endpoint if a WCBRI endpoint is administered for "1-step clearing" procedures.

Otherwise the network provided AOC will be passed to WCBRI in a RELEASE message.

Multiple Public Network Numbering

- The Identification Numbers selected are sent out in outgoing ISDN messages according to DEFINITY[®] ISDN signaling procedures.

National or International Prefix

- The administered national or international prefix is appended for display purpose mainly.
- These prefixes are not used when a call is transferred/forwarded/covered off switch.
- The original received Calling/Connected/Called/Busy number is sent on as is in DEFINITY[®] system behavior without any national/international prefixing.

Terminals

- Display of any prefixed Identification numbers applies to all types of display-capable terminal equipment
 - DCP
 - BRI
 - ATL
 - attendant.
- WCBRI station display is under local station control and is not controlled by the DEFINITY[®] system.

MCU Meet-Me Conference

- The Multi-point Control Unit for videoconference provides the Meet-Me Conference application.

- When there is a dial-out Meet-Me Conference, the DEFINITY® system generates an event report and needs to determine the Calling Number information for the call by using the isdn public-unknown form without knowing what trunk group the call is carried.
- For an individual dial out Meet-Me Conference, there may be several related calls carried on different trunks and they have the same Calling Number regardless of the Trunk Groups used for the calls.
- For switch supporting the Meet-Me Conference application, the isdn public-unknown form is properly administered with blank Trk Grp(s) entries for generating Calling Numbers for Meet-Me Conference calls.

Voice Mail adjuncts (AUDIX®, Intuity™, Octel®)

- The CPN to Intuity™ feature continues to pass unprefixed CPN information to Intuity™.
- The ISDN Calling Party Number information is not passed to voice messaging systems such as AUDIX® so the International/National Prefix feature does not impact voice mail adjuncts.

Pass Advice of Charge (AOC) to BRI Endpoints

- Provides the capability to tandem transparently the network provided AOC information to WCBRI endpoints.
- The AOC may be used by WCBRI endpoints for local station display purpose. Network AOC information to be passed include the ETSI, CEPT Italy, and NTT INS-NET 1500 Japan.

Description

- The AOC information can be used by WCBRI stations for local station display purpose.

Administration

CPN Prefix

- Select an ISDN networking option on the system-parameters customer-options form: either ISDN or ISDN-BRI Trunks.
- If a national prefix is desired, administer the National CPN Prefix field on the system-parameters features form.
 - If an international prefix is desired, administer the International CPN Prefix field.
 - The appropriate prefix is then applied to all incoming ISDN calls with the corresponding type of national or international Calling Number encoding.

Multiple Public Numbering

- Select an ISDN networking option on the system-parameters customer-options form: either ISDN or ISDN-BRI Trunks.
- Administer two or more ISDN-type trunk groups with trunk group number 11 and 22.
- Administer entries in the isdn public-unknown-numbering form: for each Ext Len/Ext Code combination desired, one entry may be entered with 11 as the Trk Grp(s) value, another entry may be entered with 22 as the Trk Grp(s) value.
 - Enter the appropriate CPN Prefix and Total CPN Length values for each entry.
 - For calls originated in Trunk Group 11, a CPN number will be generated for outgoing calls using the Ext Code/CPN Prefix/Total CPN length values for the Trk Grp(s) 11 entry in the numbering table.
 - For calls originated in Trunk Group 22, a different CPN number is generated by using the Ext Code/CPN Prefix/Total CPN Length values for the Trk Grp(s) 22 entry.
- If trunk groups numbered from 11 to 22 should all use the same CPN Prefix (for the same public trunk's main number), one entry in the isdn public-unknown numbering form can be administered with 11-22 in the Trk Grp(s) field.

Call Processing

Multiple Public Network Numbers

- For ISDN calls, if a specific Identification Number (Calling/ Connected) is originated by the DEFINITY[®] system (based on switch administration) and the Trunk Group form Page 2's Numbering Format field is administered to public.
- If there is an Ext Code entry for the station and if the Trk Grp(s) field is administered to the Trunk Group number that the call is carried on, or if the Trk Grp(s) field is administered to a Trunk Group number range, the station's Identification number is generated using this entry's administration.
- An entry with Ext Code that matches the station extension and with a blank Trk Grp(s) field is used to generate the station's Identification number.

Originating QSIG-VALU Called/Busy Number based on Trunk Group Numbers

- Any prefixed Calling/Connected/QSIG-VALU Called/QSIG-VALU Busy number that's longer than 15 digits is truncated from the least significant digits according to the current DEFINITY[®] system behavior.
- The prefixing feature does not change any DEFINITY[®] system display format and any prefixed CPN/Called/Busy numbers continues to be displayed.

Capacity Constraints and Feature Limitations

- When a switch is administered for the multiple public network numbering enhancement, the number of administered entries in the isdn public unknown number form can increase by n-fold.
- The n could be the number of distinct main numbers that the switch needs to originate for outgoing calls (number of public network trunks with distinct main numbers).
- Maximum form entries (8 pages of 30 entries each) is sufficient.
- Maximum number of number digits displayed including any national/international prefix remains at 15 digits.
- Numbers longer than 15 digits are truncated to the first 15 digits of number information.
- The maximum national or international prefix length is 5 digits.
- Network provided AOC information is tandemed transparently to WCBRI endpoints.
- Display of the AOC information at end user stations is dependent on the station capability.
- The DEFINITY[®] system does not control WCBRI station display functions.
- If the network and line protocols are different (for example, ETSI PRI - WCBRI Australia country protocol), there is no mapping or conversion of Network side (PRI) AOC protocol to Line side (WCBRI) AOC protocol.

Memory Impact of Capacity Changes

- This enhancement allows multiple attd entries to be entered into the table.
- This change has a moderate memory impact.

National or International Prefix

- These prefixes are not used when a call is transferred/forwarded/ covered off switch.
- The original received Calling/Connected/ Called/Busy number is sent on without any national/international prefixing.

Advice of Charge

- The WCBRI Platform supports the line-side country protocols of Bellcore National ISDN-1, Australia, ETSI, Japan, and Singapore.
- Network provided AOC is tandemed to WCBRI regardless of the administered WCBRI country protocol.
- Bellcore National ISDN and Singapore networks are not known to support AOC services.

Pass AOC to WCBRI

- If network provided Advice of Charge information is received for a call connected to a WCBRI endpoint, the AOC information is tandemed transparently to WCBRI.
- Tandemed AOC's
 - ETSI AOC.
 - For SS option C (etsi) trunk calls, the DEFINITY[®] system passes any FACILITY message containing Codeset 0 Facility IEs to WCBRI .
 - The DEFINITY[®] system passes any Codeset 0 Facility IE within the first call clearing message (DISCONNECT, RELEASE, or RELEASE COMPLETE) to WCBRI.
 - Italy CEPT AOC.
 - The DEFINITY[®] system passes the Shift to Codeset 6 IE and codeset 6 Charging Information IE received in CONNECT, INFORMATION, DISCONNECT, and RELEASE messages to WCBRI.
 - Japan AOC.
 - The DEFINITY[®] system passes the Shift to Codeset 6 IE and the INS-Net 1500 codeset 6 Advice of Charge IE received in DISCONNECT, RELEASE, and RELEASE COMPLETE messages to WCBRI .

Terminals

- Display of any prefixed Identification numbers applies to all types of display-capable terminal equipment (DCP, BRI, ATL, and attendant).
- WCBRI station display is under local station control and is not controlled by the DEFINITY[®] system.

Transmission Interface

- The ISDN enhancements only affect D-channel information signaling and end-user displays.
- No impact on the B-Channel connection used for each call.
- The IEs include Codeset 0 Facility IE (ETSI), Codeset 0 Display IE (Australia), Codeset 6 (CEPT Italy) Charging Information IE, and Codeset 6 AOC IE (Japan).

Call Coverage Remote Off-Net (CCRON)

- When a remote coverage call is brought back into the DEFINITY® system to follow the called party's original coverage path, the call is subject to prefixing if the prefixing feature is administered.

Calling Line Identification (CLI) Prefix

- Can apply a prefixing treatment (digit deletion, digit insertion, new PI encoding) to outgoing tandem calls (an updated CPN IE is sent out) and to locally terminated calls (an updated CPN number information is displayed).
- If CLI Prefix is administered on a trunk group, any incoming Calling Party Number IE is modified first by the CLI Prefix feature, then any International/National prefix is applied to the modified CPN IE.

Display

- If the International/national prefix feature is administered, features that display any number information (Calling, Connected, QSIG-VALU Called, QSIG-VALU Busy) display any resulting prefix-appended national/international number to user stations.

ISDN – Basic Rate Interface

- The Calling Party Number IE tandemmed to BRI interface remains the originally received CPN IE from PRI, without any prefixing applied.

ISDN – WCBRI

- The Calling Party Number IE tandemmed to WCBRI interface remains the originally received CPN IE from PRI, without any prefixing applied.

ISDN – DCS over ISDN PRI D-Channel (DCS+)

- DCS+ may be administered to display the ISDN Connected number information.

Malicious Call Trace (MCT)

- The MCT feature displays prefixed Calling Number information for consistent local displays.

Message Sequence Tracer (MST)

- The MST records the CPNs and other identification numbers as received on the link.
- The MST is enhanced to record new message types and IEs containing AOC information tandemed over WCBRI signaling links.

Multi-point Control Unit (MCU) Meet-Me Conference

- The Multi-point Control Unit for videoconference provides the Meet-Me Conference application.
- When there is a dial-out Meet-Me Conference, the DEFINITY® system generates an event report and determines the Calling Number information for the call by using the isdn public-unknown form.
- For an individual dial out Meet-Me Conference, there can be several related calls carried on different trunks and they will have the same Calling Number regardless of the Trunk Groups used for the calls.
- The isdn public-unknown form needs to be properly administered with blank Trk Grp(s) entries for generating Calling Numbers for Meet-Me Conference calls.

Adjunct Switch Applications Interface (ASAI)

- For many customers over the ASAI interface, the received Calling/Connected Party Number IE will be sent, without any International/National prefix.
- In certain CTI applications, there could be a need to receive prefixed Calling/Connected Number information so the correct call-back applications may be launched using the prefixed number information.

Restricted Presentation

- Provides new administration options to control whether a "Presentation Restricted" Calling Party Number is replaced by an administrable display string or the number will be displayed as received.
- If a Calling Party Number is received encoded as "Number Not Available Due to Interworking" or if a Calling Party Number IE is not received, a new administration option controls whether an administrable display string is displayed.
- Can be administered to append a special character to restricted Calling Numbers passed to ASAI adjuncts (to allow special handling by ASAI applications).
 - Will not pass Restricted numbers to AUDIX, CMS, and PMS adjuncts.

- The ISDN Presentation Restriction feature corrects the pre-R8 DEFINITY's specification non-conforming behavior of displaying all received Calling Party Numbers, regardless of the Presentation Indicator encoding as follows:
 - Presentation restricted
 - If the ISDN trunk group is administered to replace Restricted Numbers, the administered replacement string is displayed in place of the Calling/Connected, QSIG VALU Called/Busy number information.
 - Number not available due to interworking
 - If the ISDN trunk group is administered to replace Unavailable Numbers, the administered replacement string will be displayed in place of any Calling/Connected, QSIG VALU Called/Busy number information (some number digits may still be received).
 - Number not received
 - If the ISDN trunk group is administered to replace Unavailable Numbers, the administered replacement string displayed in the number field of user station's display.
 - ASAI Interface
 - ASAI Station form may be administered to identify restricted numbers by appending an "*" to the Calling Party Number string sent to the host adjunct via the ASAI BRI link.
 - Adjunct Interfaces
 - Restricted Calling Numbers are not forwarded to AUDIX, CMS, and PMS.
- Restricted Calling Numbers continue to be sent to CDR.
- Number display applies to Calling/Connected, QSIG VALU Called/Busy Numbers.
- DEFINITY Number Display procedures comply with FCC rulings and QSIG CLIP/CLIR specifications.
 - Allows Type Approvals and multi-vendor testings to proceed without exception.
- Does not affect Name display procedures.
- Any received QSIG Name information that is not "Presentation allowed" is treated as if no name is received, and there is no replacement string for Name display purposes.
- Allows a per trunk group control on whether to display Restricted (or Number not available), or to replace the number information with an administrable string.

- If no Calling/Connected, QSIG-VALU Called and/or QSIG-VALU Busy numbered Number IE is received at all, the same trunk group administration determines whether a replacement string is used as the number field in a display.
- Allows a special character to be appended to Restricted Calling numbers sent over the ASAI link.
- Recommended that the systems-parameters features form be administered first, then the ISDN Trunk Group forms, then any ASAI station form.

Display Enhancement

- Prior to R8.1, DEFINITY displayed maximum 15 digits for any number information (Calling/Connected, QSIG VALU Called/Busy), including any administered national or international prefix digits .
- Now increases the Calling/Connected, QSIG VALU Called/Busy numbers display to 20 digits plus a prepended prefix.
- The maximum 17-digit number display enhancement applies to Calling and Connected numbers only, there is no requirement to enhance the QSIG VALU number display to 17 digits.
- If Calling/Connected numbers plus any applicable prefix results in a number longer than 17 digits, DEFINITY will follow existing truncation behavior and will truncate from the rightmost digits and display the 17 leftmost digits.
- Display formats will be changed to display 17-digit Calling/Connected numbers only if there are sufficient blank spaces that can be shifted to the number field.
 - Category includes: all direct calls to stations and attendants, redirected calls to stations via transfer/coverage/forward, and redirected calls to attendants with one-character Reason Codes.
- On direct calls to stations or attendants, with 4-character display stations, the 17-digit Calling/Connected number field is position 19-35, the MISCID field is position 37-40.
- On 2x24 display stations, the calling number field is 1-17 on line two.



NOTE:

On redirected calls, the Calling number is displayed only if there is no Calling name information, otherwise the Calling Name is displayed in position 3-17.

On redirected calls to stations or attendants with 1-character Reason Codes, with 4-character display stations, the 17-digit Calling number field is position 3-19, and the "to" field at positions 21-22. On 2x24 display stations, the calling number field is 1-17 on line two.

Administration

System Parameters Special-applications Form

- If the release version on the Customer Option form is "R8" or later, the field "ISDN Presentation Restriction?" field is removed from the system-parameters special-applications form.

Trunk Group Form

- Because the ISDN Presentation Restriction feature available when an R8 release is used as a bug fix release, the new Trunk group form fields are not dependent on the software release version being R8 or later.
- Replaced Restricted Number Field on Trunk Group Form
 - Field on page two of the trunk group form, for isdn trunk types, added to indicate whether restricted Calling/ Connected, QSIG VALU Called/Busy numbers are replaced with administrable strings for incoming and outgoing calls assigned to the specified trunk group.
 - "Replace Restricted Numbers?"
 - Field valid values of "(y)es" and "(n)o"
 - default to "(n)o".
 - If this field is set to "y", the display is replaced regardless of the service type of the trunk.
 - Available for both PRI and BRI.
- Replaced Unavailable Number Field on Trunk Group Form
- New field on page two for isdn trunk types added to indicate whether unavailable Calling/Connected, QSIG VALU Called/Busy numbers are to be replaced with administrable strings for incoming and outgoing calls assigned to the specified trunk group.
- New field titled "Replace Unavailable Numbers?"
- default to "(n)o".
- Available for both PRI and BRI

ASAI Station Form

- Station Form ASAI Restricted Number which enables an ASAI station to identify restricted numbers.
 - "Special Character for Restriction Number?" on page 2 of ASAI station type form.
 - Valid values of "(y)es" and "(n)o"
 - Default to "(n)o"

System Parameters Features Form

- Two administrable text fields on page 6 under the ISDN parameters title.
- Fields titled "CPN Replacement for Restricted Calls" and "CPN Replacement for Unavailable Calls".
- Fields enable the user to administer a text string of up to 15 characters used to replace the restricted or unavailable Calling/Connected, QSIG VALU Called/Busy numbers upon display.
- Default is a blank string.
- Help message is the standard string translation help as displayed for the display message fields.
- An error message returned if either the "CPN Replacement for Restricted Calls:" or the "CPN Replacement for Unavailable Calls:" field is set to blank, when there is at least one trunk group form with either "Replace Restricted Numbers?" or "Replace Unavailable Numbers?" field is set to "y".

System Parameters Special-applications Form

```
change system-parameters special-applications
```

Page 2 of 2

```
SPECIAL APPLICATIONS
```

```
          (SA7666) - COS Conference Tone Check?n  
          (SA7880) - ASAI Internally Measured Data?n  
          (SA7779) - Enhanced DID Routing?n  
          (SA7777) - Night Service on DID Trunk Groups?n  
          (SA7778) - Display UUI Information?n  
          (SA7776) - Display Incoming Digits for ISDN Trunk Groups?n  
          (SA7852) - # and * in Vector Collect Step?none  
          (SA7933) - Busy Tone with SAC and No Available Cvg Points?n  
          (SA7844) - Station Used as Virtual Extension?n  
          (SA7963) - Dial By Name?n  
          (SA7900) - Service Observe Physical Set?n  
          (SA7991) - Variable Length* Account Code?n  
          (SA7710) - Enhanced Display for 8434 on Redirected Calls?n  
          (SA8052) - ISDN Redirecting Number?n  
          (SA8077) - Russian Power Industry Feature?n  
          (SA7161) - NORTEL SL1 PRI and DMS Name Display?n  
          (SA7578) - Integrated Directory Service over DCS?n  
          (SA7975) - Shortcut to AAR/ARS?n
```

- SA7992 removed

Trunk Group Form

```
change trunk-group 6                                     Page 2 of 10
TRUNK FEATURES
    ACA Assignment?9          Measured:none          Wideband Support?n
                                Maintenance Tests?y
    Data Restriction?n       NCA-TSC Signaling Group:1
    Send Name:n              Send Calling Number:y
    Used for DCS?n          Send Connected Number:y
    Suppress # Outpulsing:n  Numbering Format:public
    Outgoing Channel ID Encoding:preferred UUI IE Treatment:service-provider

                                Replace Restricted Numbers?n
                                Replace Unavailable Numbers?n
Numbers?n Restricted Replace
lace
    Send UCID:n
    Send Codeset 6/7 LAI IE?y
```

- New fields to support ISDN Presentation Restriction

ASAI Station Form

```
add station 3006                                         Page 2 of 4
STATION
FEATURE OPTIONS
    Event Minimization?

—

    Per Station CPN - Send Calling Number? ___
    Special Character for Restricted Numbers?n
```

- New field to support ISDN Presentation Restriction

System-parameters Feature Form

change system-parameters features
FEATURE-RELATED SYSTEM PARAMETERS

Page 6 of 9

ISDN PARAMETERS

Send Non-ISDN Trunk Group Name as Connected Name?n

Display Connected Name/Number for ISDN DCS Calls?n

Send ISDN Trunk Group Name on Tandem Calls?n

CPN Replacement for Restricted Calls:

CPN Replacement for Unavailable Calls:

QSIG TSC Extension:_____

Path Replacement with Measurements?y

- New fields to support ISDN Presentation Restriction

IP Solutions

DEFINITY ECS IP Solutions provide TCP/IP connectivity for two types of trunks and three types of IP Softphones. On DEFINITY ECS, IP Solutions use the TN802B IP Interface assembly, which includes a Windows NT server that resides on the TN802B circuit pack inside the DEFINITY ECS. The TN802B IP Interface, introduced in Release 8, operates in either the IP Trunk mode (for ITS-E connections) or in the MedPro mode (for H.323 trunk connections). The TN802 IP-Interface, introduced in Release 7, operates only in the IP Trunk mode. The MedPro mode requires the TN802B plus the new TN799B C-LAN circuit pack.

The following table summarizes the IP Solutions configurations and the circuit packs and software used with each.

IP Solutions		Circuit Pack Requirements		Software Requirements
		R7	R8	
Trunks	IP Trunk	TN802	TN802B in <i>ip trunk</i> mode with TN799B	–
	H.323 Trunk	–	TN802B in <i>medpro</i> mode with TN799B	–
IP Softphones	VOIP IP Softphone	–	TN802B in <i>medpro</i> mode with TN799B	DEFINITY IP Softphone, Microsoft NetMeeting ^a
	Dual-Connection IP Softphone	–	TN799B	DEFINITY IP Softphone
	CentreVu IP Agent	–	TN799B	DEFINITY IP Softphone

- a. The IP Softphones should work with other audio applications that are fully H.323 v2-compliant. DEFINITY IP Solutions was developed and tested with Microsoft NetMeeting.

TCP/IP-Connected Trunks

IP-connected trunks allow trunk groups to be defined as tie lines between switches over a virtual private network (VPN).

Trunks using IP connectivity provide cost-reduction and simplified management. Benefits include a reduction in long distance voice and fax expenses, facilitation of global communications, full-function networks with data and voice convergence, and network optimization by using the existing network resources.

H.323 Trunk

The TN802B IP Interface in Medpro mode enables H.323 trunk service using IP connectivity between two DEFINITY ECS systems. H.323 trunk groups can be configured as DEFINITY-specific tie trunks supporting ISDN trunk features such as DCS+ and QSIG, or as generic tie trunks permitting interconnection with other

vendors' H.323 v2-compliant switches, or as direct-inward-dial (DID) type of "public" trunk providing access to the switch for unregistered users. A variety of signaling options can be chosen by the system administrator.

IP Trunk

The IP Trunk mode will typically be chosen for interoperability with existing TN802 (as opposed to the TN802B) IP Interface circuit packs. IP Trunk can be used only between two DEFINITY switches. Each IP Interface circuit pack in IP Trunk mode provides a basic twelve-port package that can be expanded up to a total of 30 ports.

H.323 Trunk Administration

This section describes the administration steps needed to setup H.323 trunks. The first subsection covers the enabling administration that needs to be in place before the trunk administration can be done. The second subsection gives a summary of the H.323 trunk administration. Detailed procedural steps and the screens used for this administration are given in *DEFINITY ECS Administration for Network Connectivity*, 555-233-504.

Screen fields not mentioned here are administered as in previous releases.

Enabling Administration

Before you can administer an H.323 trunk, a few customer options and circuit pack parameters need to be properly set. In addition, there are some optional maintenance and IP parameters that can be administered. These enabling settings are summarized here.

Optional Features (System Parameters Customer Options)

The Optional Features screen must be administered by the *init* login. Open the screen with the change command (**ch sys cu**) and set the following fields:

page 1

- *G3 Version* = **V8**
- *Maximum H.323 Trunks* = number purchased; must be **greater than 0**.

page 3

- *H.323 Trunks?* = **y**
- *ISDN-PRI?* = **y**

Circuit Pack

The C-LAN and IP Interface assembly circuit packs must be administered on the Circuit Pack screen. Open the Circuit Pack screen with the change command (**ch ci**) and enter the board codes in available port slots:

- **C-LAN**
- *Code* = **TN799**
- *Sfx* = **B**
- *Name* = **C-LAN**

IP Interface assembly — enter in a slot with at least two empty slot before it

- *Code* = **TN802**
- *Sfx* = **B**
- *Name* = **MAPD Board** {entered automatically by system}

The two slots immediately before this slot are automatically populated as follows:

- *Code* = **DSMAPD** {displays automatically}

Change DSMAPD to **MEDPRO** {the IP Interface board defaults to the IP Trunk mode, which is specified by the DSMAPD in this field. Changing this field to MEDPRO changes the board's mode to MEDPRO}

- *Sfx* = {blank}
- *Name* = **Reserved - IP** {displays automatically}

Maintenance-Related System Parameters

Four parameters on the Maintenance-Related System Parameters screen set threshold values for network performance:

page 4

- Roundtrip Propagation Delay (sec) High: 800 Low: 400
- Packet Loss (%) High: 40 Low: 15
- Ping Test Interval (sec): 20
- Pings per Measurement Interval: 10

These parameters have no effect unless the bypass function is activated on the Signaling Group screen. If the bypass function is activated for a signaling group, ongoing measurements of network activity collected by the system are compared with these values. If the values of these parameters are exceeded by the current measurements, the bypass function terminates further use of the network path associated with the signaling group. The following actions are taken when thresholds are exceeded:

- existing calls are maintained
- incoming calls are allowed

- outgoing calls are blocked on this signaling group; if so administered, blocked calls are diverted to alternate routes (either IP or circuits) as determined by the administered routing patterns

You can use the default values set for these parameters, or you can change them to fit the needs of your network. The Maintenance-Related System Parameters screen can be administered by the *init*, *inads*, or *craft* logins. Open the screen with the change command (**ch sys ma**).

IP Parameters

The IP Parameters screen allows you to specify the type of codec used for voice compression/decompression. The difference between codecs is in the compression algorithm: some codecs compress the voice data more than others. A greater degree of compression results in lower bandwidth requirements on the network, but may also introduce transmission delays and lower voice quality. The default codec is set for G711. This can be changed to one of four others if the G711 does not meet your desired voice-quality/bandwidth tradeoff specification. Also, if the far-end switch is not a DEFINITY ECS, you may need to change the codec to match one that is supported by that switch.

The order in which the codecs are listed on this screen is the order of preference of usage. A trunk call between two DEFINITY switches will be set up to use the first common codec listed on the two IP Parameters screens.

You can also use the Audio IP port number range field to specify a block of port numbers to be used for audio connections. Then, if your network is equipped with the appropriate data filtering devices, audio data can be segregated from other data traffic to improve quality of service.

Open the screen with the change command (**ch ip-p**) and set the following fields:

- *Codec type* = {enter as many of the following types as supported by this switch, in the preferred order of usage — G711A, G711Mu, G723-6.3k, G723-5.3k, G.729A}. Refer to the following table of bandwidth requirements to decide which codecs to administer:

Codec	Bandwidth Requirement
G711 A-law @ 64Kbps	117.6 Kbps
G711 Mu-law @ 64Kbps	117.6 Kbps
G723 @ 6.3Kbps	31.1 Kbps
G723 @ 5.3Kbps	29.6 Kbps
G729 A-law @ 8Kbps	33.6 Kbps

- *Audio IP port number range* =

UDP Port Range
Min: 2048
Max: 65535

H.323 Trunk Administration — Task Summary

After the enabling administration is complete, you can administer the H.323 IP trunks. The screens and fields that need to be administered are listed below.



NOTE:

This is only a list of screens and the fields that need to be administered. The values that need to be entered in these fields are described in *DEFINITY ECS Administration for Network Connectivity*, 555-233-504.

Node names

A node name and IP address must be entered on the Node Names screen for each C-LAN and IP Interface board on the local switch. For each far-end node that the H.323 trunks on the local switch will connect to, a node name and IP address must be entered for the far-end gatekeeper, which is a C-LAN board if the far-end is a DEFINITY switch. Enter values in these fields:

- Node Name
- IP Address

IP Interface

Each DEFINITY switch in an IP network has one IP Interfaces screen. One line on this screen must be administered for each C-LAN and each IP Interface board. Enter values in these fields:

- Network regions are interconnected?
- Enable
- Type
- Slot
- Code
- Sfx
- Node Name
- Subnet Mask
- Default Gateway Address
- Region

Ethernet Data Module

A data module screen, type *ethernet*, must be administered for each C-LAN board on the switch. Enter values in these fields:

- Type
- Port
- Link

- Name
- Network uses 1's for broadcast address?

Signaling Group

Each H.323 trunk must be assigned to a signaling group. Enter values in these fields:

page 1

- Group Type
- Trunk Group for Channel Selection
- Near-end Node Name
- Near-end Listen Port
- Far-end Node Name (Optional)
- Far-end Listen Port (Optional)
- LRQ Required?
- Calls Share IP Signaling Connection?
- Bypass If IP Threshold Exceeded?

Page 2 — Administered NCA TSC Assignment

NCA TSCs need to be administered only if this signaling group is to be used for DCS, AUDIX, MASI, or as Gateway. Administration of TSCs is the same as in previous releases.

Trunk Group

Each H.323 trunk must be assigned to a trunk group, which is assigned to a signaling group. Enter values in these fields:

page 1

- Group Type
- Carrier Medium
- Service Type
- Codeset to Send Display
- TestCall ITC
- TestCall BCC

page 2

- Used for DCS?
- PBX ID
- DCS Signaling

page 4—trunk group members assignment

- Port
- Code
- Name
- Night
- Sig Grp

Network Regions

The Region field on the IP Interfaces screen allows you to set up segregated groups of C-LAN and IP Interface (Medpro) resources. This feature can be used in a variety of ways. For example, you could use regions to allocate specific C-LAN and IP Interface boards to H.323 trunks and others to stations; or you could use regions to avoid multimedia traffic over low bandwidth or high-latency network links.

DEFINITY Internet Protocol (IP) Softphones

DEFINITY IP[®] Softphone is a collection of computer telephony integration (CTI) applications that enables you to control telephone calls (both incoming and outgoing) directly from your personal computer (PC). DEFINITY IP Softphone enables you to log into your company's DEFINITY server remotely and make and receive telephone calls from the telephone extension. These applications run with Microsoft[®] Windows[®] NT Workstation 4.0 or later, Microsoft Windows NT[®] Server 4.0 or later, Microsoft Windows 98, and Microsoft Windows 95.

DEFINITY IP Softphones extend the level of DEFINITY services. They turn a PC or a laptop into an advanced telephone. DEFINITY IP Softphones allow you to send voice and fax from the DEFINITY ECS through an Internet Protocol (IP) network to other DEFINITY systems that have this feature. You can place calls, take calls, and handle multiple calls on your PC.

DEFINITY IP Softphones extend DEFINITY multifunction, multiline features support to IP-connected endpoints (typically user PCs). With certain exceptions, every feature available for wired-endpoint voice calling is available for IP-based calling; it supports full internetworking with conventional circuit-switched stations and trunks.

IP Softphone Configurations

DEFINITY IP Softphone supports the following three configurations:

- Road-Warrior Configuration (Voice over IP)

The road-warrior configuration enables travelers to use the full DEFINITY ECS feature set from temporary remote locations anywhere in the world (such as a hotel room). The road-warrior configuration consists of two

software applications running on a PC that is connected to DEFINITY ECS over an IP network. The single network connection between the PC and the DEFINITY ECS carries two channels, one for the signaling path and one for the voice path. DEFINITY IP Softphone handles the call signaling and an H.323 V2-compliant audio application, such as Microsoft NetMeeting 3.0x, handles the voice communications. You place and receive calls via DEFINITY IP Softphone and use the H.323 PC audio application to speak to and hear the other parties on calls via a headset connected to your PC or a microphone connected to (or integrated with) your PC. The DEFINITY server routes all of your incoming and outgoing calls over the connection between your PC and the DEFINITY server.

In the road-warrior configuration, Microsoft NetMeeting has been configured to run in the background and uses only the audio portion of the application. The data collaboration feature of NetMeeting can be used as a stand-alone application, but it is disabled during a DEFINITY IP Softphone session.

You can use an H.323 application other than Microsoft NetMeeting 3.0x in the road-warrior configuration. However, if you use an H.323 application other than Microsoft NetMeeting 3.0x, that H.323 application will not run in the background like Microsoft NetMeeting.

 **NOTE:**

DEFINITY IP Softphone works with any H.323 Version 2-compliant application (such as Microsoft NetMeeting 3.0x) that can register with an E.164 alias.

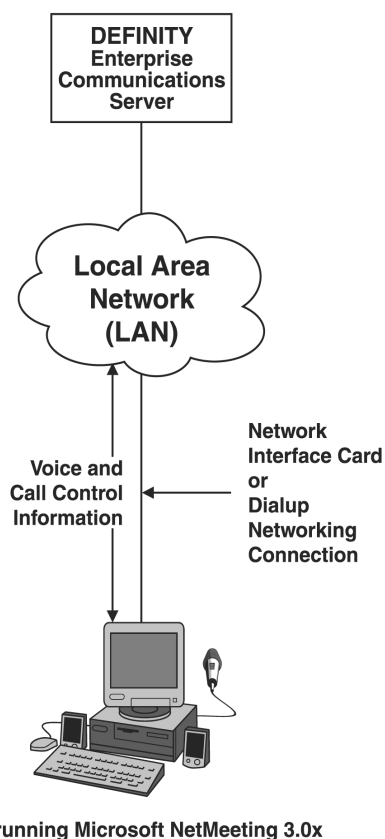


Figure 13. Road-Warrior Configuration (Voice over IP)

- Telecommuter Configuration (Dual Connection)

The telecommuter configuration of DEFINITY IP Softphone enables telecommuters to use the full DEFINITY ECS feature set from home. In this configuration, you use two connections to the DEFINITY server. Over the first connection (via dial-up networking or a network interface card), you log into the DEFINITY server and control all of your incoming and outgoing calls. Over the second connection (which is a separate telephone line), you speak to and hear the other parties on your calls. The DEFINITY server routes all of your voice calls to the separate telephone line, enabling you to speak to and hear the other parties via the telephone attached to that line. In effect, the telephone acts like the handset on the DEFINITY telephone in your office.

When you make or answer a call with DEFINITY IP Softphone, your separate telephone line is controlled by the DEFINITY server. While the DEFINITY server controls this telephone line, you are unable to place calls from the telephone attached to this telephone line. Any digits you dial from

this telephone will be dialed during the current call. When the DEFINITY server no longer controls the telephone line, you can place calls from the telephone attached to the telephone line.

If you receive an incoming call on your separate telephone line while that line is controlled by the DEFINITY server, that caller will hear either ringing (if you have call waiting on that line) or a busy signal. If you have call waiting, you can answer the incoming call as you would answer any other call waiting call.

For the telecommuter configuration, you must have a separate telephone line available at your remote location. You do not need an H.323 PC audio application such as Microsoft NetMeeting 3.0x.

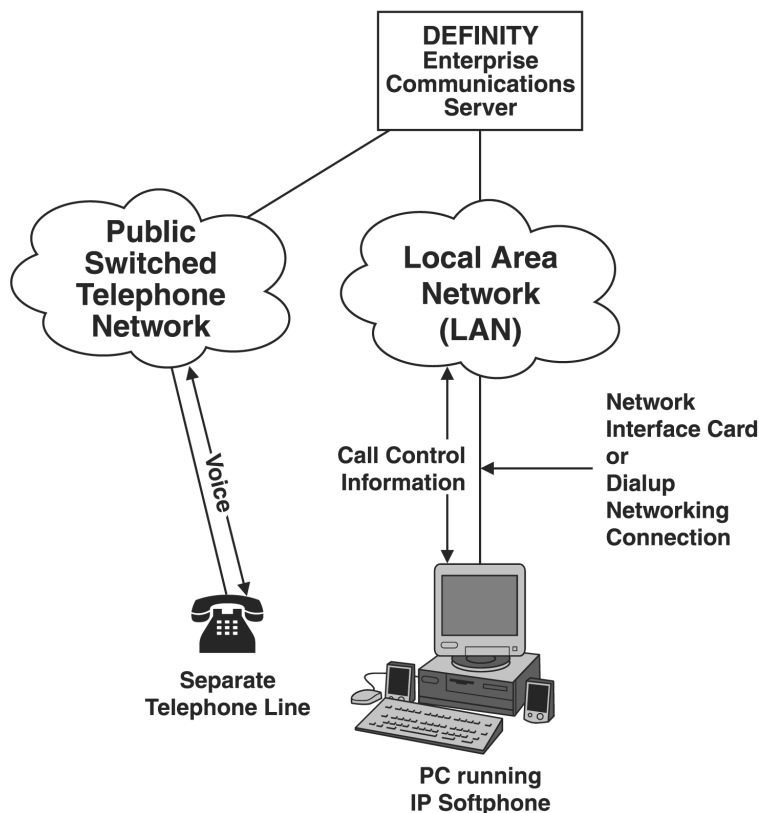


Figure 14. Telecommuter Configuration (Dual Connection)

- Stand-alone H.323 Configuration (Voice over IP)

The stand-alone H.323 configuration enables users to place and receive calls via an H.323 V2-compliant audio application, such as Microsoft NetMeeting 3.0x, running on a PC that is connected to DEFINITY ECS over an IP network. Since this configuration offers limited call-handling capabilities, the stand-alone H.323 configuration is not recommended.

Adding a DEFINITY IP Softphone

DEFINITY IP Softphones enable the end user to control telephone calls directly from a personal computer (PC). An end user can log into your company's DEFINITY server remotely and make and receive telephone calls from the telephone extension.

DEFINITY IP Softphone supports the following two configurations:

- Road-warrior application

You typically use this configuration for laptop users who are travelling. In this configuration, the PC LAN connection carries both the call control signaling and the voice path. Because voice calls are routed over the connection to the PC, you'll need an H.323 v2 PC audio application such as Microsoft® NetMeeting® as well as a headset connected to the PC to speak and hear sounds.

- Telecommuter application or CentreVu IP agent

For the telecommuter or CentreVu IP configuration, you make two separate connections to the DEFINITY server. The signaling path is carried over an IP network and the voice path is carried over the standard circuit-switched telephone network (PSTN). Since you are using a phone for audio, you do not need an H.323 PC audio application.

The telecommuter configuration uses the DEFINITY IP Softphone interface (on the user's PC) and a standard phone. The CentreVu IP agent configuration uses the CentreVu IP Agent interface (on the agent's PC) and a call center phone, such as a CallMaster.

Before you start

Be sure that your system has been enabled to use IP Softphones. Display the System Parameters Customer-Options screen and verify the following field settings:

- Maximum H.323 Stations is greater than 0
- Maximum IP Softphones is greater than 0
- IP Stations field is y

Once you're finished administering your DEFINITY system, you need to install the IP Softphone software on each user's PC. If you are using the road-warrior configuration, you also need to install an H.323 v2 compliant audio software application, such as Microsoft NetMeeting.

Instructions for adding a road-warrior application

You can use the road-warrior application when you have only a single telephone line available to access the DEFINITY system over the IP network.

Let's add a road-warrior application at extension 3001. You have to actually administer two extensions for each road-warrior application, so we will first add an H.323 extension at 3000.

To assign an H.323 extension, complete the following steps:

1. Type **add station 3000** and press RETURN.

The Station screen appears.

```
add station 3000                                     Page 1 of 3
                                     STATION
Extension: 3000      Lock Messages? n                BCC: 0
  Type: H.323      Security Code:                    TN: 1
  Port: X          Coverage Path 1:                  COR: 1
  Name:            Coverage Path 2:                  COS: 1
                  Hunt-to Station:                  Tests? y

STATION OPTIONS
                Loss Group: 2
```

2. In the `Type` field, enter **H.323**.
3. In the `Port` field, enter **x**.
4. Press ENTER to save your work.

Now, you need to administer the phone (DCP) extension. To do so, complete the following steps:

1. Type **add station 3001** and press RETURN.

The Station screen appears.



NOTE:

You choose to change an existing DCP extension by using **change station nnnn** in this step, where nnnn is the existing DCP extension.

```

add station 3001                                     Page 1 of 4
                                     STATION
Extension: 3001      Lock Messages? n              BCC: 0
  Type: 6408D      Security Code: *                TN: 1
  Port: X          Coverage Path 1:                COR: 1
  Name:           Coverage Path 2:                COS: 1
                                     Hunt-to Station:

STATION OPTIONS
  Loss Group: 2      Personalized Ringing Pattern: 1
  Data Module? n    Message Lamp Ext: 3000
  Speakerphone: 2-way Mute Button Enabled? y
  Display Language: english

                                     Media Complex Ext: 3000
                                     IP SoftPhone? y
    
```

2. In the `Type` field, enter the model of phone you want to use, such as **6408D**.
3. In the `Port` field, type **x** for virtual phone, or enter the port number if there is hardware.
4. In the `Security Code` field, enter the password for this remote user, such as **1234321**.
This password can be up to 7 digits in length.
5. In the `Media Complex Ext` field, type **3000**.
This is the H.323 extension we just administered.
6. In the `IP Softphone` field, type **y**.
7. Move to the `Service Link Mode` field, type **as-needed**.
Set this field to **permanent** only for extremely busy remote phone users, such as call center agents.
8. Press ENTER to save your work.

Now you can install and configure the software on the user's PC. In this example, the user will login by entering their DCP extension (3001) and password (1234321).

Instructions for adding a telecommuter application

Assign this configuration to remote users who have two available phone lines. For example, to administer a telecommuter application for a home user at extension 3010, complete the following steps:

1. Type **add station 3010** and press RETURN.

The Station screen appears.

NOTE:

Use the **add station** command if this is a new DCP extension. Use the **change station** command for an existing DCP extension, and ignore steps 2 and 3.)

2. In the `Port` field, type **x** for virtual phone, or enter the port number if there is hardware.
3. In the `Security Code` field, enter the password for this remote user, such as **1234321**.
This password can be up to 7 digits in length.
4. In the `IP Softphone` field, type **y**.
5. Move to the `Service Link Mode` field, type **as-needed**.
Set this field to **permanent** only for extremely busy remote phone users, such as call center agents.
6. Press ENTER to save your work.

Now you can install and configure the software on the user's PC. In this example, the user will login by entering their DCP extension (3010) and password (1234321).

NOTE:

You can use list multimedia ip-softphones to display the available extensions permitted for IP Softphones and the associated media complex extension (the H.323 extension), and the port, where assigned.

IP Support Hardware

TN799/TN799B: Control LAN (C-LAN) Interface

The TN799/TN799B Control LAN Interface simplifies the connections between adjunct equipment and DEFINITY. It provides TCP/IP connectivity over Ethernet or PPP (Point to Point Protocol) connections to applications, such as CMS Call Center, INTUITY™, AUDIX®, and DCS. The TN799B is required to support the

TN802B IP Interface operating in MedPro mode for H.323 functionality. The TN799/TN799B is also required to support ATM interfaces and WAN PNC on DEFINITY R7 or later.

The TN799/TN799B does not support BX.25 connectivity on the R8 or later csi platform. On the other platforms, TN765 Processor Interface and TN577 Packet Gateway circuit packs provide BX.25 connectivity for existing customers as well as TCP/IP service.

The TN799B C-LAN pack provides connectionless UDP sockets for IP solutions support, variable length PING and Traceroute. The pack will also support 500 remote sockets, while current packs support 220.

The TN799 circuit pack is available with R7r, R7si and R7csi and the TN799B C-LAN circuit pack is available with R8r, R8si, and R8csi.

The TN799 supports the *traceroute* and *netstat* network testing commands starting in R8.

TN802/TN802B Medpro: (IP Interface Assembly)

The TN802 Internet Protocol Interface circuit pack supports voice and fax calls from the DEFINITY ECS across a corporate intranet or the Internet. The IP trunking software runs on an embedded PC operating under Windows NT. The TN802B supports IP Solutions including IP Trunking and MedPro (H.323) with IP SoftPhones.

It is implemented using the TN802/TN802B IP Interface, which is a Windows NT server that resides on the IP Interface circuit pack inside the DEFINITY ECS. The TN802B IP Interface operates in two modes: IP Trunk, available with Release 7, and Media Processor (MedPro/H.323), available with Release 8. The TN802B is backwards compatible and defaults to IP Trunk mode. To use it in MedPro mode you activate it through administration to take full advantage of all the R8 IP Solutions features. You also must install a new TN799B C-LAN circuit pack.

C-LAN Connectivity Changes and Enhancements

Changes to reduce the number of IP routes

1. The change in R8 to make automatic subnet routing always enabled means that fewer IP routes need to be defined. In R7, if automatic subnet routing is disabled, an explicit IP route needs to be defined between every pair nodes on the subnet that you want to inter-communicate. In R8, automatic subnet routing is always enabled so you never need to define an explicit IP route within a subnet.
2. In R7, there was no default gateway unless it was explicitly defined in an IP route using the "default" node name as the Destination. With no default gateway, any communication with a remote node on another subnet required two IP routes. One IP route was needed from the C-LAN

(identified as the gateway on the IP route screen) to a router (identified as the destination on the IP route screen). Then the router is identified as the gateway and the remote node as the destination in another IP route. In R8, you can specify a default gateway on the IP Interfaces screen for each IP interface. Then any messages sent to a node on a different subnet will be automatically sent to this gateway. This eliminates the need to create the two IP routes needed in R7.

New Screens

A new screen, IP Interfaces, defines the properties of an interface between a DEFINITY system and an IP network.

IP Interfaces

```
change ip-interfaces                                     Page 1 of 2
```

IP Interfaces

Inter-region IP connectivity allowed? n

Enable	Eth	Type	Slot	Code	Sfx	Node Name	Subnet Mask	Gateway Addr	Net Rgn
y		C-LAN	01A10	TN799	B	clan-a1	255.255.255.0	192.168.10 .100	1
y		MEDPRO	01A13	TN802	B	med-a1	255.255.255.0	192.168.10 .100	1
y		MEDPRO	01A16	TN802	B	med-a2	255.255.255.0	192.168.20 .100	1
y		C-LAN	01A09	TN799	B	clan-a2	255.255.255.0	192.168.10 .100	2
y		MEDPRO	01B13	TN802	B	med-a1	255.255.255.0	192.168.10 .100	2
n							255.255.255.0	. . .	
n							255.255.255.0	. . .	
n							255.255.255.0	. . .	
n							255.255.255.0	. . .	
n							255.255.255.0	. . .	
n							255.255.255.0	. . .	
n							255.255.255.0	. . .	
n							255.255.255.0	. . .	
n							255.255.255.0	. . .	
n							255.255.255.0	. . .	

IP Interfaces screen

- Inter-region IP connectivity allowed?: specifies whether media processor (TN802B) resources can be shared between regions.
- Enable Eth: enable/disable this IP Interface
- Type: type of circuit pack (C-LAN or MedPro) for this IP Interface
- Slot: slot location of the circuit pack
- Code: board code for the circuit pack (TN799 or TN802)
- Sfx: suffix letter for the board code (B)
- Node Name: moved from the ethernet Data Module screen
- Subnet Mask: moved from the ethernet Data Module screen
- Gateway Addr: IP address of the default gateway for this IP Interface
- Net Rgn: the network region this IP Interface is assigned to

Screen Changes

The data module screen has changed for both ethernet and ppp connections.
The IP Routing screen has also changed.

Data Module Screen (Ethernet)

```
add data-module next                                     Page 1 of X
                                                    DATA MODULE
Data Extension: 2377                                     Name: ethernet on link 2
  Type: ethernet
  Port: 01c0817
  Link: 2
Network uses 1's for broadcast addresses? y
```

Data Module (Ethernet) screen

- Fields moved to the IP Interfaces screen:
 - Nodes Names
 - Subnet Mask
- Fields that have been removed and are not needed:
 - Broadcast Address
 - Automatic Subnet Routing?
- New fields:
 - Network uses all 1's for broadcast addresses?: allows you to specify whether the DEFINITY system is to use all 1's or all 0's in the host ID to form broadcast addresses sent on the network by this C-LAN board; similarly, it specifies whether the DEFINITY system recognizes all 1's or all 0's in the host ID as broadcast addresses received.

Data Module Screen (PPP)

```
add data-module 1994                                     Page 1 of x

                                DATA MODULE

Data Extension: 1994                                     Name: ppp on link 4 to node 4      BCC: 2
  Type: ppp                                             COS: 1
  Port: 01c1502                                         COR: 1
  Link: 4                                               TN: 1
Enable Link? n

  Node Name: ppp14
  Subnet Mask: 255.255.255.0

Establish Connection: y

DESTINATION
  Digits: 7241991
  Node Name: ppp41
  CHAP? n
```

Data Module (Ethernet) screen

- New field:

- Subnet Mask: R7 did not allow subnetting of IP addresses of nodes in a ppp connection. The addition of the Subnet Mask Field on the ppp Data Module screen, enables the specification of subnets for these addresses.

IP Routing Screen

```
add ip-route next                                     Page 1 of 1

                                IP ROUTING

Route Number: 2
Destination Node: node-1
Gateway: node-4
C-LAN Board: 1c05
Metric: 0
```

IP Routing screen

- New field:

- Route Type: specifies whether the route defined on this screen is a network or host route.

When to use IP routes

You need to define host and/or network IP routes only in special cases when default gateways are not defined or when you want to limit communication between nodes. This section describes the network configurations that require explicit IP routes.

There are two type of IP routes — *host* and *network*. The following table summarizes when you would use each type:

Connection Type	When IP Routes are Needed:	
	Host IP Routes	Network IP Routes
Ethernet	You want the local host to communicate with only the specified host on a remote subnet — endpoints are on different subnets and no default gateway is defined for the local node.	You want the local host to communicate with any host on the specified remote subnet but not with hosts on other subnets — endpoints are on different subnets and no default gateway is defined for the local node.
PPP	There are one or more intermediate nodes between endpoints.	Network IP routes are not used with ppp connections.

NOTE:

The endpoint nodes are on the same subnet if the following three conditions are met:

- the endpoints are on the same physical subnetwork
- the Subnet Mask field is assigned the same value on the IP Interface screens for the two endpoint nodes
- the network + subnet portions of the IP addresses (as determined by the subnet mask) are the same

To set up an IP route, enter the node names for the destination and the gateway, and enter the slot location of the C-LAN on the local switch. Enter **host** or **network** in the Route Type field. For either type, the destination and gateway node names and their associated IP addresses must be specified on the Node Names screen. An IP address for a network has the network ID in the network portion and 0 in the host portion. For example, 192.168.1.0 is the network address for the 192.168.1 network.

When a network is subnetted, and you want to set up a network IP route to a subnetwork, the IP address of the subnetwork is the first address in that subnetwork, which has all 0's for the host portion of the address. For example, the subnet mask formed by using 2 bits of the host portion of a Class C address is 255.255.255.192 (1100000 = 192). For the 192.168.1 network, this subnet mask

creates 2 usable subnetworks whose IP addresses are 192.168.1.64 (0100000 = 64) and 192.168.1.128 (1000000 = 128)

.Serviceability Tools for C-LAN

- The C-LAN feature of the DEFINITY® system provides TCP/IP connectivity over Ethernet or PPP (Point to Point Protocol) to adjuncts such as CMS Call Center, Intuity™ AUDIX® and for DCS connectivity.
- TCP/IP connectivity over the customer's LAN represents a significant change from BX.25 protocol employed for connectivity prior to C-LAN.

Commands

- These commands are accessible through local and remote SAT access.
 - Traceroute Command
 - Variable Length Ping - Customer option for Packet size (64 or 1.5k bytes)
 - Packet Error History
 - list measurements clan

Trace route

- Provides the ability to trace the route of C-LAN originated packets through the LAN.
- Output shows the IP address of each router or host (hop) that the packets encounter.
- The time elapsed between each hop is also displayed.
- If C-LAN has trouble communicating with a far-end device, trace route can be used to determine how far packets get toward the destination.

Trace Route IP Address

```
trace-route ip-address 192.11.128.111 board 01C07
```

TRACE ROUTE RESULTS

Hop	Time(ms)	IP Address
1	5	135.60.129.1
2	2	135.60.109.4
3	10	135.60.119.5

Trace-route

```
trace-route ip-address 106.245.27.205 board lc11
```

TRACE ROUTE RESULTS

Hop	Time(ms)	IP Address
1	10, 12, 11	124. 9. 5.103
2	15, 15, 12	124. 9. 14. 23
3	23, 26, 26	122.245. 15.111
4	23, 25, 25	122.245. 27.205

Traceroute screens

- Authorization: init, inads, customer and craft logins
- Syntax: traceroute ip-address
- Help: Enter IP address, 'board' 'board location' where IP address is (0-255), (0-255), (0-255), (0-255)
- Errors:
- Hop - The node number as it is sequentially found
- Time(ms) - The time from the board to each intermediate destination
- IP Address - This field is the 32-bit network address.

Variable length ping

- The standard ping implementation issues a packet that is 64 bytes in length.
- Ethernet frames can be as large as 1500 bytes in length.
- Using ping with a packet size of up to 1500 bytes provides a more complete indication of the link status.
- The objective of C-LAN troubleshooting efforts is to quickly and unambiguously determine if the fault lies within the Lucent provided equipment or if the fault is with the LAN or LAN administration to which the DEFINITY® system is connected.
- Provides an enhancement to the ping command.
- Specifying a longer packet to be sent by ping can show if a router or host has a problem fragmenting or integrating transferred packets.

Ping IP Address

```
ping ip-address 192.68.9.26 board 1B07
```

PING RESULTS

End-pt	Port	Port Type	Result	Time(ms)	Error Code
192.68.9.26	1b0701	MED-PT	PASS	205	

Variable Length Ping

```
ping ip-address 192.11.128.111 packet-length 1500
```

PING RESULTS

End-pt IP	Port	Port Type	Result	Time(ms)	Error Code
192.11.128.111	01A0917	ETH-PT	PASS	10	

Variable Length Ping

- Authorization: init, inads, customer and craft logins
- Syntax: ping ip-address xxx.xxx.xxx.xxx [board UUCSS] [packet-length <64-1500>] [repeat <1-100>]
- Help: ping ip-address xxx.xxx.xxx.xxx [board UUCSS] [packet-length <64-1500>] [repeat <1-100>] where IP address is (0-255), (0-255), (0-255), (0-255)
- Errors: 'x' entry out of range, please press help
- End-pt IP - This field is the destination IP address that is being pinged.
- Port - Fixed field giving board and port location for the port that will provide the interface for the route
- Port Type - This field can be either PPP-PT for PPP ports (ports 1-16) or ETH-PT.
- Result - This field is the outcomes of either 'pass', 'fail' or 'abort'
- Time(ms) - This field describes the time in milliseconds.
- Error Code - This field describes the failed attempt codes.



NOTE:

The Result will support all the abort and failure codes from R7.

Packet Error History

- In C-LAN Release 7, the capability of checking packet error was provided, but was simply displayed on the board counters.
- Now there is the capability of having a historical view of packet errors. Packet error history provides a 24-hour history of important packet level statistics that indirectly indicate some LAN performance characteristics.
- Show the current 15-minute intervals (for the last 24 hours) for CRC errors and Collision counts for Ethernet connections.
- High collision counts indicate high traffic on the LAN segment.
- High CRC errors indicate that the LAN connection may be noisy.
 - It could indicate that a wire connection is loose or a wire is frayed or broken.
- Checks are for Invalid Frame errors (error count for invalid frame errors) and Chap Failures (the number of failed attempts for PPP authentication) for PPP connections.
- If the data cannot be retrieved for any 15-minute interval, a N/A will be returned in that field. These error checks will be provided for Ethernet and PPP.

List Measurements - PPP

- The counter value can be up to 7 digits long because of its 16-bit counter on the board.
- The delta and the total will be provided for each error count.
- After the occurrence of an N/A, the delta equals the total

List Measurements CLAN PPP

list measurements clan ppp 1a0901							Page 1	
Switch Name:				Date: 3:50 pm TUE NOV 16, 1999				
C-LAN PPP PERFORMANCE MEASUREMENTS DETAILED REPORT								
Date	Time	CRC CHECK		Invalid Frame		CHAP Failures		
		Total	Delta	Total	Delta	Total	Delta	
7/03	1535	89	0	8011	5	0	0	
7/03	1520	89	0	8006	7	0	0	
7/03	1505	89	0	7999	7	0	0	
7/03	1450	89	0	7992	6	0	0	
7/03	1435	89	0	7986	3	0	0	
7/03	1420	89	0	7983	8	0	0	
7/03	1405	89	0	7975	8	0	0	
7/03	1350	89	0	7967	10	0	0	
7/03	1335	89	0	7957	3	0	0	
7/03	1320	89	0	7954	4	0	0	
7/03	1305	89	0	7950	6	0	0	
7/03	1250	89	0	7944	5	0	0	

List Measurements - PPP

list measurements clan ppp screen

- Authorization: init, inads, customer and craft logins
- Syntax: list measurements clan ppp [cabinet(1-%d)]; carrier(A-E); slot(00-%d); circuit(01-16)
- Help: Enter ['number' 'x'] ['to-number' 'x'] ['count' 'n']
- Date - This field sets the current date that the action was done.
- Time - This field sets the current 15-minute interval the action was performed.
- CRC Check - This field displays the error count for CRC errors.
- Total - The total value of the counter on the board
- Delta - The difference between the current and the previous sample
- Invalid Frame - This is the error count for invalid frame errors.
- CHAP Failures - This field describes the number of failed attempts for PPP authentication.



NOTE:

The counter value can be up to 7 digits long because it is a 316-bit counter on the board. After the occurrence of a N/A, the delta will equal the total. Busy out /release; board/port; reset board; and reseating the board would clear firmware counters.

List Measurements - Ethernet

- Packet error history is a list measurements 24-hour history of error statistics for Ethernet connections.
- The data is retrieved for 15-minute intervals for 24 hours for CRC and Collisions for Ethernet connections.
 - If the data cannot be retrieved for the 15-minute interval, a N/A will be returned in the field.
- The counter value can be up to 11 digits long because of its 32-bit counter on the board.
- The delta and the total are provided for each error count.
- After the occurrence of a N/A, the delta will equal the total.

List Measurement CLAN Ethernet

```
list measurements clan ethernet 1C1017                                     Page 1 of x
Switch Name:                                                            Date:02/02/1999

                                C-LAN ETHERNET PERFORMANCE MEASUREMENT DETAILED REPORT
```

Date	Time	CRC Check		Collision Count	
		total	delta	total	delta
02/01	03:08	650	50	650	250
02/01	02:53	600	600	400	400
02/01	02:38	N/A	N/A	N/A	N/A
02/01	02:23	1000000570	20	10000000570	20
02/01	02:08	10000000550	10000000550	10000000550	10000000550

List measurements clan ethernet screen

- Authorizations: init, inads, customer and craft logins
- Syntax: list measurements clan ethernet [cabinet(1-%d)]; carrier(A-E); slot(00-%d); ['print' or 'schedule']
- Help: Enter ['number' x] [to-number 'x'] [count 'n']
- Errors: 'x' invalid location for clan board
- Date - This field sets the current date that the action was done.
- Time - This field sets the current 15-minute interval the action was performed.
- CRC Check - This field displays the error count for CRC errors.
- Total - The total value of the counter on the board
- Delta - The difference between the current and the previous sample
- Collision Count - This is the error count for collisions on the Ethernet.



NOTE:

The counter value can be up to 11 digits long because of its 32-bit counter on the board. After the occurrence of a N/A, the delta will equal the total. Busy out /release; board/port; reset board; and reseating the board would clear firmware counters.

Upgrades

- Customers that decide to upgrade to the C-LAN solutions in R8 for each of the models (r, and si) may choose to change all, some or none of the existing interfaces for DCS, AUDIX[®], or CMS from BX.25 to TCP/IP.

Networking - ATM

DEFINITY ATM Networking uses open standards to integrate DEFINITY functionality with ATM transmission and switching. This allows you to use DEFINITY ECS applications over an ATM infrastructure throughout an enterprise.

This convergence of voice and data networking is made possible by an ATM Services Card and ATM software that integrate into the DEFINITY ECS. The hardware and software allow the DEFINITY ECS to interoperate with any standards-based ATM switch technologies, further maximizing the value of the DEFINITY system and your network infrastructure.

The DEFINITY ATM (asynchronous transfer mode) combines portions of the DEFINITY platform with a ATM switch platform that meets specific criteria. DEFINITY ATM offers both intraswitch and interswitch ATM solutions. The interswitch solution is called ATM circuit emulation service or ATM-CES, and the intraswitch solution is called the ATM port network connectivity or ATM-PNC.

ATM-Port Network Connectivity (ATM-PNC)

ATM Port Network Connectivity (ATM-PNC) provides an alternative to either the direct connect or Center Stage Switch configurations for connecting the Processor Port Network (PPN) to one or more Expansion Port Networks (EPNs). ATM PNC replaces the Center Stage Switch in a DEFINITY R8r network with an Asynchronous Transfer Mode (ATM) switch. ATM PNC is available with all three DEFINITY ECS reliability options — standard, high, and critical.

ATM PNC integrates delivery of voice, video, and data via ATM over a common large bandwidth LAN, providing reduced infrastructure cost and improved network manageability. ATM PNC uses standards-based open interfaces that can be provisioned with either new or existing DEFINITY ECS systems and is ATM-ready for future expansion.

ATM-Circuit Emulation Service (ATM-CES)

ATM-CES (Circuit-Emulation Service) lets the DEFINITY ECS emulate an ISDN-PRI trunk on an ATM facility. These virtual trunks can serve as integrated access, tandem, or tie trunks. ATM-CES trunk emulation maximizes port network capacities by consolidating trunking. For example, the CES interface can define up to ten virtual circuits for tie-line connectivity, consolidating onto one circuit card network connectivity that usually requires multiple circuit packs.

ATM Wide Area Network PNC (ATM WAN-PNC)

ATM Wide Area Network (ATM WAN) extends the Port Network Connectivity (PNC) beyond a single ATM switch. This allows you to utilize either a private ATM network, public WAN, or a combination of both. Several networked ATM devices can be used as effectively as a single ATM switch for inter-port network connectivity. ATM WAN is supported by the "Multiple Location" feature, where port

network cabinets can be administered as separate locations; however, it is not required. You can use multiple ATM switches without multiple locations administered.

The following figure shows an example of the basic ATM connections for Release 8r and Release 8csi system using ATM-PNC and ATM-CES.

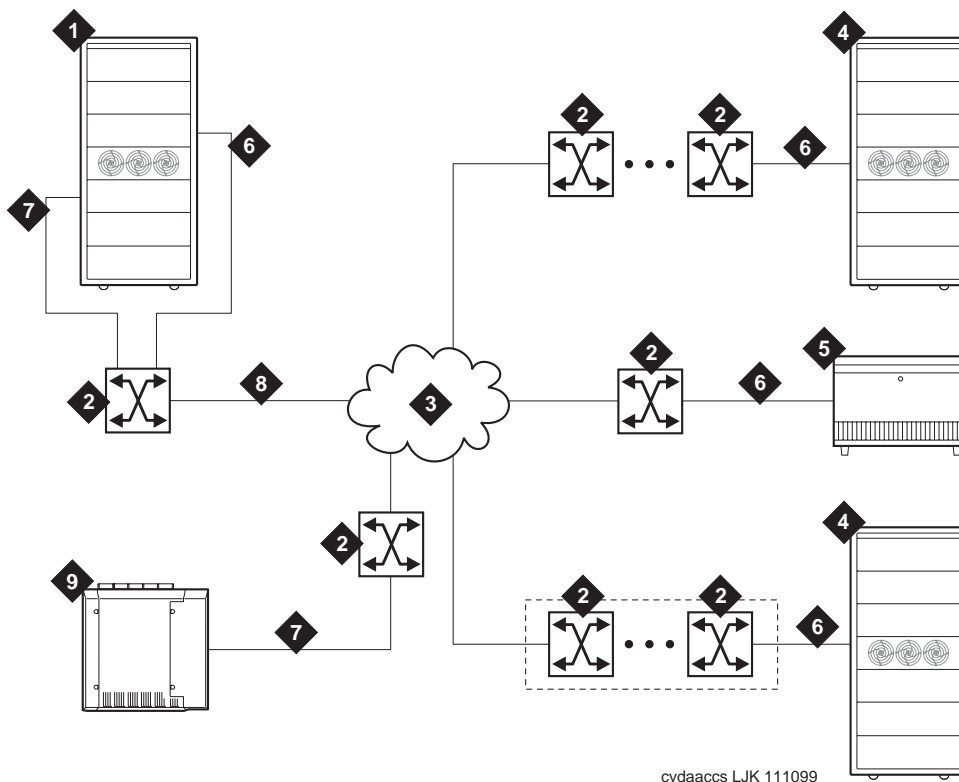


Figure Notes

- | | |
|----------------------------|-----------------------------|
| 1. DEFINITY ECS PPN-1 (r) | 5. DEFINITY ECS EPN in SCC |
| 2. Lucent ATM switch | 6. ATM-PNC |
| 3. ATM network (the cloud) | 7. ATM-CES |
| 4. DEFINITY ECS EPN in MCC | 8. ATM-PNC and ATM-CES |
| | 9. DEFINITY ECS PPN-2 (csi) |

ATM Hardware

Hardware required for ATM-PNC or ATM-CES:

- TN2305 (multimode) or TN2306 (single mode) ATM interface circuit pack

	TN2238	TN2305	TN2306
R6.3	v		
R7	v	v	v
R8.2		v	v

- ATM-based equivalent for the TN570 expansion interface circuit pack or circuit emulation service (CES)
- OC-3/STM-1 fiber connection at 155 Mbps
- Built-in transceivers
- Connections to EPNs
- Switched virtual circuits (SVCs—ATM-PNC only)
- 8-ms near-end echo cancellation
- Synchronization splitter (ATM-PNC only)
 - 400A—T1 100-Ω reduced-range capability (limited availability)
 - 401A—T1 100-Ω full-range capability (includes ICSU)
 - 402A—E1 120-Ω (includes ICSU)
 - 403A—E1 75-Ω (includes ICSU—available in Release 8.2)

For more detailed information, see [“T1 401A/E1 402A/E1 403A Synchronization Splitters” on page 270.](#)

The following equipment is not supported or required for ATM-PNC:

- Direct-connected EPNs.
- Survivable Remote Expansion Port Networks (SREPNs).
- Switch node carrier.
- Hybrid operation of CSS and ATM-based port network connectivity.
- DS1 converter remote port networks.

Administering ATM-CES and ATM-PNC

Acquiring ATM Addresses

Equipment that interfaces with an ATM switch must have a unique 40-digit (20-byte) ATM address. These 40 characters are divided into sectors as [Table 1](#) shows:

Table 1. ATM addressing

Hex digit position	Length (hexadecimal)	Description
1-26	26	Network prefix that identifies a particular ATM switch in the network.
27-38	12	End system identifier (ESI) identifies each end-station connected through an ATM switch. ^a
39-40	2	Selector byte is always 0 for DEFINITY ATM

- a. The ESI is usually the IEEE Media Access Control (MAC) address programmed into the end-station device during its manufacture. The MAC address on the TN2305/TN2306 is located on the MAC address button on the circuit pack (see Figure 5-1).
-

In Release 7 or later, the address of the EPN is automatically derived by the local ATM switch using the address registration procedure defined in the Integrated Local Management Interface (ILMI). The ATM switch uses its 13-byte network address plus the 6-byte ESI, or MAC address, and the 1-byte selector byte of the corresponding TN2305 or TN2306 board(s) in the EPN to formulate the 20-byte ATM address(es).

ATM switches can use any of the address formats listed in [Table 2](#).

NOTE:

The beginning AFI and the last 2 sectors (ESI, and Selector) in [Table 2](#) are all the same length, making it easier to parse the ATM address, regardless of the format.

Table 2. ATM address formats

Format	Length (hex characters)	Description (high to low order)
Data Country Code (DCC)	2	Address Format Identifier (AFI) (39 for DCC)
	4	Data Country Code
	20	High-Order Domain Specific Part (HO-DSP)
	12	End System Identifier (ESI)
	2	Selector
International Code Designator (ICD)	2	Address Format Identifier (AFI) (47 for ICD)
	4	International Code Designator
	20	High-Order Domain Specific Part (HO-DSP)
	12	End System Identifier (ESI)
	2	Selector
ISDN E.164	2	Address Format Identifier (AFI) (45 for E.164)
	16	E.164 address
	8	High-Order Domain Specific Part (HO-DSP)
	12	End System Identifier (ESI)
	2	Selector

Administering the ATM Switch

To administer the ATM switch, refer to your switch's quick reference guide.

Administering DEFINITY ECS

The DEFINITY ECS switch software provides 2 kinds of ATM service:

- [ATM Port Network Connectivity \(ATM-PNC\)](#)
- [ATM Circuit Emulation Service \(ATM-CES\)](#)

[Table 3](#) outlines important concepts to observe when administering ATM.

Table 3. General DEFINITY ECS ATM parameters



Concept	Description
PNC duplication	<p>If you are installing or upgrading a critical reliability system, you must enable the PNC duplication on the system-parameters customer-options screen (init password required) before the feature can be used. If it is enabled, you must further</p> <ul style="list-style-type: none"> ■ Add pnc-a or pnc-b to command strings to clarify which carrier the ATM interface circuit pack is in ■ Specify the location of the ATM interface circuit pack serving as the duplicate <p>The ATM interface circuit pack in the A carrier must be in the same port network as the B carrier. If not, an alarm is issued. You need not administer ATM interface circuit packs serving as backups to each other with the same parameters.</p>
Fiber links	<p>You do not need to administer fiber links for ATM-PNC. Information on the ATM circuit pack screen identifies which packs are used for PNC (endpoint-1), and endpoint-2 is administered on the ATM switch.</p>
Remote switch administration	<p>A remote switch connected with a permanent virtual circuit (PVC). The DEFINITY system recognizes the remote switch as a node with compatible circuit parameters at the other end of an emulated circuit.</p> <p>You can connect a remote switch to the DEFINITY with one or more CES-emulated circuits. In such a case, the emulated circuits remain discrete.</p>
Synchronization	<p>PNC—All synchronization is derived from the ATM switch providing port connectivity. The ATM switch obtains synchronization from either a sync splitter attached to one or two DS1 cards or from the ATM network.</p> <p>CES—Synchronization is derived the same way it is in a non-ATM-PNC or ATM-PNC environment, whichever applies.</p>

ATM Port Network Connectivity (ATM-PNC)

Each DEFINITY port network must be translated in the PPN's screen with the full ATM address (The ATM switch prefix, plus the port network's ESI, plus a Selector of 0) that uniquely identifies it.

Use the procedure in [Table 4](#) to administer ATM-PNC:

Table 4. ATM-PNC administration

√	Step	Action	Command	Description
	1.	Log on DEFINITY ECS switch		Use the <code>init</code> login to log on to the DEFINITY switch.  NOTE: An <code>init</code> login is required to change customer options and will be challenged by Access Security Gateway (ASG). Contact your regional CSA (customer software administrator) to perform this function.
	2.	Turn on ATM-PNC	<code>change system-parameters customer-options</code>	The Customer Options screen (Screen 3) displays.
	3.	Set the mode to yes		Set <code>Async.Transfer Mode (ATM) PNC?</code> field to y .
	4.	If duplicated PNC (critical reliability), set PNC duplication		Set the <code>PNC Duplication?</code> field to y
	5.	Log off the DEFINITY ECS		Log off the DEFINITY system administration terminal.
	6.	Log onto the DEFINITY ECS		Use the <code>craft</code> login to log on to the DEFINITY switch (ignore any alarms at this time).  NOTE: The <code>init</code> login is no longer required.
	7.	Administer ATM PNC		Use the ATM worksheet in Appendix A, "Baselining the Customer's Configuration" to record the configuration.
	8.	Add new PNC	<code>add pnc next</code>	Add the next PNC.
	9.	Set location address		Set the <code>Location:</code> field to the address of the ATM circuit pack (in 01C01 - cabinet, carrier, slot format).
	10.	Assign network name		Type a name in the <code>Name:</code> field to assign a unique, 15-character port network name.

Continued on next page

Table 4. ATM-PNC administration — *Continued*

√	Step	Action	Command	Description
	11.	Set the MAC address (Steps 10-13)		Set <code>Address format</code> to one of the following: E.164 ATM private, DCC, or ICD ⇒ NOTE: If other ATM applications are installed ask the ATM system administrator to determine which address format to use. The system default is DCC.
	12.	Fill in information from worksheet		Set <code>HO DSP</code> : xxxxxxxx , fill in information from the ATM worksheet (Appendix A, "Baselining the Customer's Configuration").
	13.	Enter the ESI address		Type in the ESI from the ATM worksheet (Appendix A, "Baselining the Customer's Configuration") (MAC address) of the TN2305/ TN2306 in the EPN. ⇒ NOTE: The first 13 bytes auto set after initial entry. Just enter the MAC address from each column for each TN2305/TN2306 circuit pack.
	14.	Set the Sel field		Set the <code>Sel</code> : field to 00 . ⇒ NOTE: Steps 11-13 make up the 20-byte ATM address.
	15.	If duplicated PNC (critical reliability), enable PNC	<code>change system-parameters duplication</code>	Set <code>Enable Operation of SPE Duplication?</code> field to y Set <code>Enable Operation of PNC Duplication?</code> field to y (Screen 5)
	16.	Repeat for all PNs		Repeat Steps 7-13 until all port networks are administered.
	17.	Check administration	<code>list atm pnc</code>	Verify that all ATM connections are administered.
	18.	Check links	<code>list sys-link</code>	Types EAL (EPN) and PACL (all PNs) should display.
	19.	Administer the DEFINITY ECS synchronization	<code>change synchronization</code>	Synchronize all signals.
	20.	Set primary timing reference		Set <code>primary</code> : atm-sw .

Table 4. ATM-PNC administration — Continued

√	Step	Action	Command	Description
	21.	Set secondary timing reference		Set secondary: atm-sw . Press Enter to effect the changes.
	22.	Save translations	save translations	Save translations to system disk.

ATM Circuit Emulation Service (ATM-CES)

ATM Circuit Emulation Service (ATM-CES):

- Can be administered
 - as an ATM-CES direct connect
 - through an ATM network using PVCs

Both procedures are in [Table 9](#), under [“ATM-CES administration procedure” on page 223](#).

- Uses ATM emulated circuits instead of physical T1/E1 tie-trunks out to the ATM network.
- Uses up to 8 signaling groups per TN2305/TN2306 to create virtual trunking.
- Uses bearer (B) and signaling (D) channels, similar to ISDN facilities but does not support nonfacility associated signaling (NFAS).
- Uses channels inside each signaling group, which appear as if they were ports on the TN2305/TN2306 circuit pack. Hereafter, these are referred to as ports.
- Uses constant bit-rate (CBR) signalling over permanent virtual connections (PVCs).
- Has a transparent DEFINITY feature set to other locations through narrowband QSIG (N-QSIG) or distributed communications system over ISDN-PRI (DCS+)

⇒ NOTE:

Because ATM trunks always appear as ISDN-emulated trunks, they do not support DCS, only DCS+.

- The TN2305/TN2306 ATM Interfaces must be used with ATM CES, with ATM WAN PNC, and if multiple ATM switches are used.

[Table 5](#) lists the maximum number of TN2305/TN2306 circuit packs allowed in a DEFINITY ECS.

Table 5. Maximum number of ATM interface TN2305/TN2306 circuit packs

Platform	Maximum ATM circuit packs allowed	Description
r	88	2 x 44 port networks (CES)
si, csi, c	6	CES only

ATM-CES rules

[Table 6](#) lists several guidelines for ATM-CES administration.

Table 6. ATM-CES administration rules

Rule	How to administer or confirm
Both ends of the emulated circuit must have the same number of channels	Type change signaling-group <i>siggrpnr</i> and press Enter; go to screen 2
Both ends of the emulated circuit must have the same number of trunks.	Type change trunk-group <i>trkgrpnr</i> and press Enter; go to screen 6.
Both ends of the emulated circuit must use the same channel numbers.	Type change signaling-group <i>siggrpnr</i> and press Enter; go to screen 2. In the port-to-channel mapping, use the same channel numbers for both ends of the trunk.
Each end of the trunk may use different port numbers on the respective ATM circuit packs.	Type change trunk-group <i>trkgrpnr</i> and press Enter; go to screen 6. The same port numbers on the circuit packs need not be used on the two ends of the emulated circuit.
The maximum number of emulated circuits (signaling groups) on a single TN2305 circuit pack is 8.	A circuit pack can have more than one signaling group and more than one D-channel. The D-channel from one emulated circuit cannot signal for the bearer channels of another emulated circuit (no NFAS for virtual circuits).
A minimum of 7 ports must be administered for each emulated circuit (signaling group).	Because this includes the D-channel, at least 6 B-channels and the D channel must be administered in a signaling group.

Continued on next page

Table 6. ATM-CES administration rules — *Continued*

Rule	How to administer or confirm
The D channel must be in a port between 009 and 032.	Type display signaling-group siggrpnr and press Enter; go to screen 2 to verify. The D channel is automatically populated in channel 24 (T1) or in channel 16 (E1).
The TN2305/TN2306 circuit packs provide up to 248 ports for trunking.	A circuit pack has 256 ports, but ports 1 through 8 are reserved for future use. Of the 248 available ports, 8 are for D-channels, and 240 are for B-channels. See Table 7 and Table 8 for details about port and trunk group capacities for the TN2305/TN2306 circuit pack and the DEFINITY ECS switch.
A trunk group may contain either ISDN trunks or ATM trunks, but not both	

Table 7. TN2305/TN2306 CES trunk capacities

Bearer ports per signaling group (B channel)	Signaling groups per TN2305/TN2306 (D-channel)	Total ports used ^a
T1: 23	8	192
E1: 30	8	248

a. The circuit pack provides 256 ports, but ports 1 through 8 are reserved for future use.

Table 8. DEFINITY ECS trunk capacities by model

Capacity	c, csi, si	r
Maximum trunks per trunk group	99	255
Maximum number of trunk groups	99	666
Maximum number of trunks in the system	400	4000

Trunk groups

At least one trunk group is typically created for each remote switch with these attributes (see [Screen 9](#)):

- Group Type field is **isdn**
- Carrier medium field is **atm**
- Dial Access? field is always **n**
- TN2305/TN2306 may have Trunk Access Codes (TAC), but these are for reporting purposes only and cannot be dialed.
- CDR records for ATM trunk calls are maintained.

NOTE:

Release 7 introduced 3-digit port numbers for administering a CES trunk, and CMS cannot read the old 2-digit numbers. Release 8 CMS software can read both the 2- and 3-digit port numbers.

- A trunk group may include ports from more than one of the ATM circuit packs in the system. This enables *same module preference*, the feature that gives preference to outgoing trunks on the same port network as the originator.
- ATM trunk groups can only contain ports on ATM boards, and ports on ATM boards cannot be assigned to non-ATM trunk groups.

ATM-CES trunk board

When administered as a trunk board, the TN2305/TN2306 can have up to 8 ATM signaling groups with up to 30 members per signaling group over an OC-3/STM-1 connection. Although the ATM board does not have physical ports, software

makes the ATM board appear as if it does. This concept is depicted in [Figure 3-1](#). Multiple TN2305/TN2306 circuit packs in a port network could connect to different ATM switches.

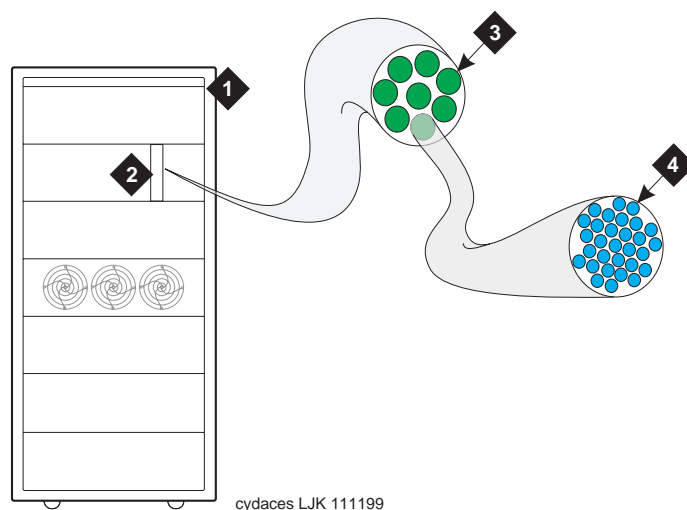


Figure Notes

- | | |
|---|--------------------------------------|
| 1. DEFINITY ECS PPN or EPN | 4. Ports |
| 2. TN2305/TN2306 circuit pack in any port | 24 maximum (T1), including D-channel |
| 3. Signaling groups (maximum 8) | 31 maximum (E1), including D-channel |

Figure 15. DEFINITY signaling groups and members for ATM trunking

- Each of the 248 ports in the ATM trunk group are either B-channels or D-channels.
- Each B-channel has an associated maintenance object with a very simple maintenance strategy. This is done so that any of the virtual ports on a single ATM interface circuit pack can be members of different trunk groups. When a trunk group is busied out, only those ports in that trunk group are taken out of service. Because all ports associated with a single D-channel follow the same PVC, you cannot test them individually.

ATM-CES administration procedure

[Table 9](#) contains ATM-CES administration for

- ATM signaling groups

- ATM-CES direct-connect configurations (two CES circuit packs linked by fiber, with no intervening switches). Observe the note in step 16.



NOTE:

The screens shown are for a Release 8r platform. The screen numbers may differ for Release 8c, Release 8csi, and Release 8si platforms.

Table 9. ATM-CES administration

√	Step	Action	Description
	1.	Set the software version	Type change system-parameters customer-options and press Enter. (init login required) The Customer Options screen (Screen 3) displays.
	2.	Set the version field	Upgrades only: Set the G3 <i>Version</i> field to V8 . Go to screen 2
	3.	Turn on ATM trunking	Set the <i>Async. Transfer Mode (ATM) Trunking?</i> field to y (Screen 3). Go to screen 3.
	4.	Administer as ISDN-PRI	Set the <i>ISDN-PRI</i> field to y (Screen 4). Press Enter to save the changes.
	5.	Log off/on	Log in as craft
	6.	Install the circuit packs	Install the TN2305/TN2306 circuit pack and wait for it to initialize (about 4 minutes)
	7.	Connect the fiber optic cable	Attach the fiber optic cable to the circuit pack and to the ATM switch.
	8.	Look up ATM circuit pack locations	Type list configuration atm and press Enter. The name for this circuit pack appears as ATM INTERFACE (or ATM INTF).
	9.	Administer the circuit pack personality	Type add atm trunk-board UUCSS and press Enter. The ATM Trunk-Board screen (Screen 6) displays.
	10.	Type a name for the ATM board	Type the name of the ATM interface circuit pack (15 alphanumeric characters) in the <i>Name</i> field.




Continued on next page

Table 9. ATM-CES administration — *Continued*

√	Step	Action	Description
	11.	Set the synchronization	<p>Set the <code>Synchronization Capable</code> field to n (default).</p> <p>⇒ NOTE: Up to 44 boards may be designated synchronization capable on R8r and 3 boards on R8si, R8csi, R8c. If you enter y in this field when the maximum number of boards has been designated, the system returns:</p> <p style="padding-left: 40px;"><code>Maximum number of Synchronization Capable boards already administered</code></p> <p>⇒ NOTE: If this ATM trunk circuit pack is a synchronization source, set the field to y. If you change the field to n when the circuit pack is administered as a synchronization source, the system returns:</p> <p style="padding-left: 40px;"><code>Cannot remove synchronization source.</code></p>
	12.	Administer the signaling group	<p>Type add signaling-group and press Enter. The Signaling Group screen (Screen 7) displays.</p> <p>Set the following fields (tabbing goes vertically)</p>
	13.	Set the group type	Set the <code>Group Type</code> field to atm .
	14.	Type port address	Type the <code>D-channel</code> (signaling channel) port address for this new signaling group in UUCSSppp screen at (3-digits required for port addresses). This address must match the slot and port number on an ATM-Interface circuit pack.
	15.	Type a name	Type a name in the <code>Name</code> field (15 alpha-numeric characters).
	16.	Set VPI	The <code>Virtual Path Identifier (VPI)</code> field is set to 0 and is not administrable.

Continued on next page

Table 9. ATM-CES administration — *Continued*

√	Step	Action	Description
	17.	Set VCI	Set the <code>Virtual Channel Identifier (VCI)</code> field (32-1023). This number must be unique among signaling groups that share the same ATM circuit pack.  NOTE: For a direct connection (two fiber-connected CES circuit packs), specify the same VCI at both ends of the emulated circuit. When the CES is connected to an ATM switch, the VCI specified here must match the VCI used for the PVC created on the ATM switch, which is typically provided by the ATM switch administrator.  NOTE: The <code>Signaling Mode</code> field is set to <code>isdn-pri</code> and is not administrable.
	18.	Set idle code	Set the <code>Idle Code</code> field (typically 11111111).
	19.	Set interface companding	Set the <code>Interface Companding</code> field (alaw or mulaw). Once administered, busyout the signaling group before changing.
	20.	Set country protocol	Set the <code>Country Protocol</code> field (Table 10).
	21.	Set protocol version	Set the <code>Protocol Version</code> field (Table 10).
	22.	Set circuit type	Set the <code>Circuit Type</code> field (T1 or E1).  NOTE: Both ends must be the same. It is preferable to set it to E1 even in the United States, Canada, or other T1 countries.
	23.	Set connect	Set the <code>Connect</code> field to pbx . Once administered, busyout the signaling group before changing.
	24.	Set interface	Set the <code>Interface</code> field to user at one end of the emulated circuit and to network at the other end.
	25.	Set DCP/analog bearer capability	Set the <code>DCP/Analog Bearer Capability</code> field to 3.1 kHz .
	26.	Set internetworking message	Set <code>internetworking message</code> field to PROGRESS .

Continued on next page

Table 9. ATM-CES administration — *Continued*

√	Step	Action	Description
	27.	Verify channel settings	<p>Go to screen 2; Screen 8 displays.</p> <p>Depending on the settings in step 18:</p> <ul style="list-style-type: none"> ■ If <code>Circuit Type</code> field is set at T1, then channel 24 is populated. ■ If <code>Circuit Type</code> field is set at E1, then channel 16 is populated. <p>Other details about D-channels:</p> <ul style="list-style-type: none"> ■ All ports for one signaling group must be on the same ATM circuit pack (the same circuit pack that the D channel is on). ■ The D-channel port is never assigned to a trunk group. Because they are all the same, the administrator does not need to type in the circuit pack location for each channel. ■ No port can be assigned to more than one signaling group.
	28.	Type port numbers	<p>Add the port numbers (minimum of 6) from the port-to-channel mapping on signaling group screen 2.</p> <p>See "Changing the port-to-channel mapping" on page 233 if you need to change any administration.</p> <p>When finished, press Enter to save the administration.</p>
	29.	Administer the trunk group	<p>Type add trunk-group <i>trkgrpnbr</i> and press Enter.</p> <p>Screen 9 displays.</p>

Continued on next page

Table 9. ATM-CES administration — *Continued*

√	Step	Action	Description
	30.	Set final fields	<p>Set the following fields (tabbing goes horizontally):</p> <ul style="list-style-type: none"> ■ Group Type: field to isdn. <p>⇒ NOTE: A trunk group can contain either ISDN or ATM trunks, but not both.</p> <ul style="list-style-type: none"> ■ The CDR Reports field defaults to y. ■ The Carrier Medium: field to ATM. ■ The TAC: field to match customer's dial plan. ■ The Dial Access? field defaults to n. ■ The Service Type: field to tie. <p>Go to screen 6 and add the channel-to-port mapping (Screen 11). You must have the same number of entries as port numbers in step 28.</p> <p>Press Enter to save the changes.</p>
	31.	Save translations	Type save translation and press Enter.

ATM-CES administration screens



NOTE:

The screens shown in this section are as they look on a Release 8r platform through a SAT. Your interface and screen page numbers may differ.

```

change system-parameters customer-options                               Page 1 of 6
                                OPTIONAL FEATURES

      G3 Version: V8                                                    Maximum Ports: 300
      Location: 1                                                         Maximum XMOBILE Stations: 0
                                                                           Maximum H.323 Trunks: 0
                                                                           Maximum H.323 Stations: 0
                                                                           Maximum IP SoftPhones: 0
    
```

Screen 2. Optional features—screen 1

2 Networking Enhancements
Networking - ATM

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change system-parameters customer-options Page 2 of 6
OPTIONAL FEATURES

Abbreviated Dialing Enhanced List? y	CAS Branch? y
Access Security Gateway (ASG)? y	CAS Main? y
Analog Trunk Incoming Call ID? y	Change COR by FAC? n
A/D Grp/Sys List Dialing Start at 01? y	Cvg Of Calls Redirected Off-net? y
Answer Supervision by Call Classifier? y	DCS (Basic)? y
ARS? y	DCS Call Coverage? y
ARS/AAR Partitioning? y	DCS with Rerouting? y
ARS/AAR Shortcut Dialing? n	DEFINITY Network Admin? y
ASAI Proprietary Adjunct Links? y	Digital Loss Plan Modification? n
Async. Transfer Mode (ATM) PNC? y	DS1 MSP? y
Async. Transfer Mode (ATM) Trunking? y	Emergency Access to Attendant? y
ATMS? y	Extended Cvg/Fwd Admin? y
Attendant Vectoring? n	External Device Alarm Admin? y
Audible Message Waiting? y	Flexible Billing? y
Authorization Codes? y	Forced Entry of Account Codes? y
	Global Call Classification? y

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

Screen 3. Optional features—screen 2

change system-parameters customer-options Page 3 of 6
OPTIONAL FEATURES

Hospitality (Basic)? y	PNC Duplication? y
Hospitality (G3V3 Enhancements)? y	Processor and System MSP? y
H.323 Trunks? n	Private Networking? y
IP Stations? n	Restrict Call Forward Off Net? y
ISDN Feature Plus? y	Secondary Data Module? y
ISDN-BRI Trunks? y	Station and Trunk MSP? y
ISDN-PRI? y	Tenant Partitioning? y
Malicious Call Trace? y	Terminal Trans. Init. (TTI)? y
Mode Code for Centralized Voice Mail? n	Time of Day Routing? y
Mode Code Interface? y	Uniform Dialing Plan? y
Multifrequency Signaling? y	Usage Allocation Enhancements? y
Multimedia Appl. Server Interface (MASI)? y	Wideband Switching? y
Multimedia Call Handling (Basic)? y	Wireless? y
Multimedia Call Handling (Enhanced)? y	
Multiple Locations? y	
Personal Station Access (PSA)? y	

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

Screen 4. Optional features—screen 3

```
change system-parameters duplication                Page 1 of 1  SPE B
          DUPLICATION RELATED SYSTEM PARAMETERS
```

```
Enable Operation of SPE Duplication? y
Enable Operation of PNC Duplication? y
```

Screen 5. Duplication related system parameters

```
ATM TRUNK-BOARD
```

```
Location: 2E08
```

```
Name: _____ Synchronization Capable: 1
```

Screen 6. ATM trunk board

```
add signaling-group                                Page 1 of 6
```

```
SIGNALING GROUP
```

```
Group Number: 2          Group Type: atm          Name:
                          D-Channel:              Max number of NCA TSC: 0
                                                   Max number of CA TSC: 0
                          Trunk Group for NCA TSC:
```

```
Trunk Group for Channel Selection:
Supplementary Service Protocol: a
```

```
CIRCUIT PARAMETERS
```

```
Virtual Path Identifier: 0
Virtual Channel Identifier:
```

```
          Signaling Mode: isdn-pri          Circuit Type: T1
          Idle Code: 11111111              Connect: pbx
Interface Companding: mulaw
Country Protocol: 1
Protocol Version: a
```

```
DCP/Analog Bearer Capability: 3.1kHz
Interworking Message: PROGRESS
```

Screen 7. Signaling group

Table 10. Country and protocol version supported (for [Screen 7](#))

Value	Country	Protocol supported
1-a	United States, Canada	AT&T TR41449/41459 (tested with AT&T, Canadian, and MCI networks)
1-b	United States	Bellcore TR 1268; NIUF.302; ANSI T1.607
2	Australia	AUSTEL TSO14.1; Telecom Australia TPH 1856
3	Japan	NTT INIS-NET
4	Italy	ETS 300 102
5	Netherlands	ETS 300 102
6	Singapore	ETS 300 102
7	Mexico	ETS 300 102
8	Belgium	ETS 300 102
9	Saudi Arabia	ETS 300 102
10-a	United Kingdom	ETS 300 102 (for connection to DASS II/DPNSS through external converter)
10-b	United Kingdom	ETS 300 102 (Mercury) British Telecom ISDN 30
11	Spain	Telefonica ISDN Specification
12-a	France	VN4 (French National PRI)
12-b	France	ETS 300 102 modified according to P10-20, called Euronumeris
	Argentina	ETS 300 102
	Brazil	ETS 300 102
	China	ETS 300 102
	Germany	ETS 300 102
	Hong Kong	ETS 300 102
	South Korea	ETS 300 102
	New Zealand	ETS 300 102

```

add signaling-group next                               Page 2 of 6
                                                    SIGNALING GROUP

Chan Port Chan Port
__1:  __  __17:  __
__2:  __  __18:  __
__3:  __  __19:  __
__4:  __  __20:  __
__5:  __  __21:  __
__6:  __  __22:  __
__7:  __  __23:  __
__8:  __  __24:  Occ (T1 only)
__9:  __
__10: __
__11: __
__12: __
__13: __
__14: __
__15: __
__16: Occ (E1 only)
    
```

Screen 8. Signaling group D-channel assignments—screen 2

```

add trunk-group                                     Page 1 of 10
                                                    TRUNK GROUP

Group Number: 1                                Group Type: isdn                CDR Reports: y
  Group Name: OUTSIDE CALL                      COR: 1                        TN: 1                TAC:
  Direction: two-way                          Outgoing Display? n          Carrier Medium: ATM
  Dial Access? n                               Busy Threshold: 99           Night Service:
Queue Length: 0
Service Type: tie                               Auth Code? n                 TestCall ITC: rest
                                           Far End Test Line No:

TestCall BCC: 4
TRUNK PARAMETERS
  Codeset to Send Display: 6                   Codeset to Send National IEs: 6
  Max Message Size to Send: 260                Charge Advice: none
  Supplementary Service Protocol: a             Digit Handling (in/out): enbloc/enbloc

  Trunk Hunt: cyclical                         QSIG Value-Added Lucent? n
                                           Digital Loss Group: 13
Calling Number - Delete:                      Insert:                        Numbering Format:
  Bit Rate: 1200                               Synchronization: async       Duplex: full
Disconnect Supervision - In? y Out? n
Answer Supervision Timeout: 0
    
```

Screen 9. Trunk group—screen 1

```
add trunk-group                                     Page 4 of 10
                                                    TRUNK GROUP
                                                    Administered Members (min/max): 0/0
GROUP MEMBER ASSIGNMENTS                        Total Administered Members: 0

   Port      Code Sfx Name          Night          Sig Grp
1:
2:
3:
4:
5:
6:
7:
8:
9:
10:
11:
12:
13:
14:
15:
```

Screen 10. Trunk group—screen 4

Changing the port-to-channel mapping

Use the following steps to change the port-to-channel mapping for emulated circuits:

1. Type **busyout atm signaling-group *siggrpnr*** and press Enter to busyout the signaling group.
2. Type **change trunk-group *trkgrpnr*** and press Enter.
Remove any port administration on screen 6.
3. Type **change signaling-group *siggrpnr***.
Modify the port-to-channel mapping.
4. Type **change trunk-group *trkgrpnr*** and press Enter.
Add the ports back in on screen 6, using new port numbers, if desired.
5. Type **release atm signaling-group *siggrpnr*** and press Enter to release the signaling group.

The display shows the command aborts, but the D-channel and B-channels come up.

DEFINITY commands for CES

Administration commands that are useful in working with CES are listed in [Table 11](#).

Table 11. ATM administration commands

Command	Description
display circuit-packs cabinet	Displays the packs in the cabinet, identifying ATM Trunk as well as ATM PNC EI boards. ATM Interface boards have not been specified as CES or PNC yet.
display atm ports eqloc	Displays the 256 ports on the ATM board with the corresponding signaling and trunk group.
list configuration atm	Lists the ATM boards, identifying equipment location, board code, type, and vintage.
list configuration trunks	Lists boards identifying assigned ports. Although the ATM board is listed, the 256 ports are not.

Additional DEFINITY ECS administration

After administering the ATM-PNC or ATM-CES, you may need to do some further administration, specifically:

- [Locations](#)
- [SVC cache](#) (PNC only)
- [Gain/Loss adjustments](#) (PNC only)

Locations

The DEFINITY ATM-PNC makes it easier for the DEFINITY ECS to have port networks in multiple time zones. To handle distances between the PPN and its EPNs, the system tracks one or more locations by assigning each cabinet a location number. All locations must be in the same country code.

If upgrading from a pre-Release 7, you may need to set the locations for your EPNs. To administer locations, follow the procedure in [Table 12](#).



NOTE:

The screens and field description follow the table.

Table 12. Administer location procedure

√	Step	Action	Description
	1.	Activate the feature	Type change system-parameters customer-options and press Enter. <i>(Init password required.)</i> The Customer Options screen displays. Go to screen 3 (Screen 11).
	2.	Set multiple locations	Change the value in the <code>Multiple locations</code> field to y and press Enter. ⇒ NOTE: The system is limited to a single location unless this field is changed.
	3.	Verify feature activation	Type display system-parameters customer-options and press Enter. The Customer Options screen displays (Screen 11). Go to screen 2 and verify that the <code>Multiple locations</code> field is set to y .
	4.	Change cabinet administration	Type change cabinet n (cabinet number) and press Enter. The Cabinet screen displays (Screen 12).
	5.	Administer cabinet location	Each cabinet in the switch and the port network(s) in that cabinet must be assigned a location number (default location number is 1). Change the <code>Location</code> field to the appropriate number and press Enter.
	6.	Administer location properties	Type change locations and press Enter. The Locations screen displays (Screen 13).
	7.	Set timezone offset	Ensure that the <code>Timezone Offset</code> field for Location 1 is set at 00:00 , indicating the local time. This serves as a system-wide, relative time reference.
	8.	Set Number Plan Area Code	The <code>Number Plan Area Code</code> field (for subnet trunking conversions between 10-digit North American Number Plan (NANP) and 7-digit (local) numbers). Leaving the field blank is the normal setting for a location <ul style="list-style-type: none"> ■ outside the NANP. ■ with no 7-digit local calling plan. These locations may have overlaid area codes requiring 10-digit dialing.

Continued on next page

Table 12. Administer location procedure — Continued

√	Step	Action	Description
	9.	Administer cabinet locations	<p>Administer all other locations as necessary.</p> <p>For example, if the PPN is in New York and a group of EPNs is in Los Angeles, then New York (location 1) is the reference and is administered with no time offset (see step 8). Los Angeles (location 2) is 3 hours earlier than New York time and is administered with a time offset of -03:00.</p> <p>Administer cabinet locations as they are added to the system. More than 1 cabinet can be in the same location.</p> <p>For location-related feature limitations and interactions see Table 12.</p> <p>When finished, press Enter to effect the changes.</p>
	10.	Administer Daylight-Savings Time rules	<p>Type change daylight-savings-rules and press Enter. The Daylight Savings Rules screen displays (Screen 14).</p> <p>Administer as many rules as necessary for all of the administered locations.</p> <p>If you want to deny access for an individual login:</p> <ul style="list-style-type: none"> ■ set the <code>Additional Restrictions</code> field for that login to y, and entering daylight-savings-rules in the <code>Restricted Object List</code> field. ■ set the <code>Administer Features</code> field for that login to n. ■ set the <code>Display Admin. and Maint. Data</code> field for that login to n.
	11.	Verify date and time	<p>Check the system date and time (set time).</p> <p>Ensure that the <code>Type</code> and <code>Daylight Savings Rule</code> fields are administered correctly (Screen 15).</p>
	12.	Administer ARS routing	<p>Administer the <code>Location</code> field on the ARS analysis screen (change ars analysis 0 location all), observing the following conditions:</p> <ul style="list-style-type: none"> ■ ARS customer option (change system-parameters customer-options) must be enabled (<i>init password required</i>). ■ the <code>Multiple Locations</code> customer option must be administered to a value other than all.

```
change system-parameters customer-options                               Page 3 of 6
                                OPTIONAL FEATURES

    Hospitality (Basic)? y                                           PNC Duplication? y
  Hospitality (G3V3 Enhancements)? y
    H.323 Trunks? n                                               Processor and System MSP? y
    IP Stations? n                                               Private Networking? y
    ISDN Feature Plus? y   Restrict Call Forward Off Net? y
    ISDN-BRI Trunks? y   Secondary Data Module? y
    ISDN-PRI? y         Station and Trunk MSP? y
    Malicious Call Trace? y
  Mode Code for Centralized Voice Mail? n   Tenant Partitioning? y
    Mode Code Interface? y   Terminal Trans. Init. (TTI)? y
    Multifrequency Signaling? y   Time of Day Routing? y
  Multimedia Appl. Server Interface (MASI)? y   Uniform Dialing Plan? y
    Multimedia Call Handling (Basic)? y   Usage Allocation Enhancements? y
    Multimedia Call Handling (Enhanced)? y
      Multiple Locations? y                                           Wideband Switching? y
    Personal Station Access (PSA)? y                                           Wireless? y
```

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

Screen 11. Optional features—screen 3

```
change cabinet 1                                                    Page 1 of 1
                                CABINET

CABINET DESCRIPTION
  Cabinet: 1
  Cabinet Layout: five-carrier
  Cabinet Type: processor
  Number of Portnetworks: 1

      Location: 1

          Room:          Floor:          Building:

CARRIER DESCRIPTION
  Carrier      Carrier Type      Number
  C            not-used          PN 01
  B            not-used          PN 01
  A            processor        PN 01
  X            fan
  D            not-used          PN 01
  E            not-used          PN 01
```

Screen 12. Cabinet



NOTE:

With Release 7.1 or later software, all cabinets are automatically assigned location 1.

Field description:

- Location Location number associated with the cabinet (range is 1-44, default 1).
- Field cannot be blank
 - Location is display-only if the Multiple locations field (**change system-parameters customer-options**) is not set to **y**.

change locations Page 1 of 3

LOCATIONS

ARS Prefix 1 Required For 10-Digit NAMP Calls? y

Number	Name	Timezone Offset	Daylight-Savings Rule	Number Plan Area Code
1	Main	+ 00:00	0	
2		:		
3		:		
4		:		
5		:		
6		:		
7		:		
8		:		
9		:		
10		:		
11		:		
12		:		
13		:		
14		:		

Screen 13. Locations—screen 1



NOTE:

Upgrades from a previous software version default to a single location (location 1) with the values shown in [Screen 13](#). The Number Plan Area Code is the home NPA from the dial plan screen of the earlier release.

Field description:

ARS Prefix 1 Required For 10-Digit NAMP Calls? This field is set to **y** if all 10-digit NAMP calls (those marked as FNPA in the ARS analysis tables) must be preceded by a digit '1' (commonly called the toll prefix); otherwise, it is set to **n**.



NOTE:

Many areas in the United States have changed to 10-digit dialing for local calls, and administration for this field and the ARS analysis entries must reflect these changes.

Number This read-only field uniquely identifies a location. Valid values range from 1 to 44.

2 Networking Enhancements
 Networking - ATM

- Name This 15-character field identifies the location but may be blank for any given location (default = y).
- Timezone Offset This field specifies the time offset from the system standard time. Set a value in the ±, hour (0-23), and minute (0-59) areas, as these fields cannot be blank for an administered location.
- Number This field is the number assigned to the administered, customized rule (0 to 15) for this location. Each rule consists of the week, day, month, and time that daylight savings starts and stops.
- Rule 0 is no daylight savings time and is not alterable.
 - Rule 1 defaults to the U.S. daylight savings time rule but may be altered or deleted.
- Number Plan Area Code This field is the numbering plan area code appropriate for this location (can be blank).

change daylight-savings-rules

Page 1 of 2

DAYLIGHT SAVINGS RULES

Rule	Change Day	Month	Date	Time	Increment
0:	No Daylight Savings				
1:	Start: first Sunday	on or after	April	1	at 2 :0
	Stop: first Sunday	on or after	October	25	at 2 :0
2:	Start: first	on or after			at 0 : 0 :
	Stop: first	on or after			at 0 :
3:	Start: first	on or after			at 0 : 0 :
	Stop: first	on or after			at 0 :
4:	Start: first	on or after			at 0 : 0 :
	Stop: first	on or after			at 0 :
5:	Start: first	on or after			at 0 : 0 :
	Stop: first	on or after			at 0 :
6:	Start: first	on or after			at 0 : 0 :
	Stop: first	on or after			at 0 :
7:	Start: first	on or after			at 0 : 0 :
	Stop: first	on or after			at 0 :

Screen 14. Daylight savings rules screen

Field description:

- Change Day The day of the week, or date that the change takes effect.
- Month The month the change takes effect.
- Date The date on or after which the change takes place.
- Time 2 fields that specify the hour (0-23) and minute (0-59) that the change takes effect (in 24-hour format)

Increment 2 fields specifying the number of hours (0-23) and minutes (0-59) the clock is moved ahead to begin daylight savings time (and moved back to return to standard time).



NOTE:

Some states (Arizona) or portions of states (Indiana) do not observe daylight savings time. Verify the daylight savings time status in all your locations.

set time

Page 1 of 1

DATE AND TIME

DATE

Day of the Week: Sunday Month: October
Day of the Month: 24 Year: 1999

TIME

Hour: 8 Minute: 16 Second: 31 **Type: Standard**
Daylight Savings Time Rule: 0

WARNING: Changing the date or time will impact BCMS, CDR and MEASUREMENTS

Screen 15. Date and time screen



NOTE:

When the system clock is upgraded from an earlier release, the daylight savings time rule on the **set time** screen defaults to **0** (no rule). While you can change the daylight savings time rule, the system clock is not adjusted automatically until the next transition of the rule.

Field description:

Type Daylight-savings = daylight savings time
Standard = standard time

Daylight Savings Rule The daylight savings time rule number (0 to 15). Rule 0 is no daylight savings time, and rule 1 defaults to U.S. daylight savings time rule.

SVC cache

DEFINITY ATM software maintains a cache of inter-port-network connections (SVCs), with the preference given to those SVCs with the longest setup times. All cached interconnections are aged to ensure that a large number of them are not kept during nearly idle periods.

You can administer the setup time thresholds, cache holding times, and total cache size, which can mitigate the effects of long interconnection setup delays.

The **change atm svc-cache** command ([Screen 16](#)) is restricted to init, inads, and craft logins.

```
change atm svc-cache
```

Page 1 of 1

```
ATM SVC CACHE
```

```
Algorithm:
```

```
Size: 410
```

```
ALGORITHM PARAMETERS
```

For	Latency	Hold	Time	
>=	50	ms	36	sec
>=	0	ms	12	sec
>=		ms		sec
>=		ms		sec
>=		ms		sec
>=		ms		sec
>=		ms		sec
>=		ms		sec
>=		ms		sec
>=		ms		sec

Screen 16. ATM SVC cache report

Field description:

Algorithm The name of the cache algorithm (default is **lan**). Possible values are **none**, **lan**, **man**, **wan**, and **custom**.

When this field is changed to **none**, **lan**, **man**, or **wan**, the read-only **For Latency** and **Hold Time** field values change to the following (all other entries are blank):

ValueFor Latency (ms)Hold Time (ms)

```

none0 0
lan 50 36
      0 12

man80080
      50 16
      0 8

wan800200
      20036
      50 16
      0 0
    
```

When this field is **custom**, none of the **For Latency** or **Hold Time** field values change, but the fields change from read-only to read/write. For example, if you want a slight variation of the **wan** algorithm, type **wan** first to set the above values, then type **custom** to change the individual parameters.

Size The maximum number of entries simultaneously allowed in the cache (default = 410).

For Latency These fields (10) are read-only if the **Algorithm** field is **none**, **lan**, **man**, or **wan**, but are read/write if **Algorithm** is **custom**. Defaults: **36** for the 1st entry, 0 for the second, blank for the last 8 entries.

Gain/Loss adjustments

Gain and loss are administrable. Calls between a digital (BRI or DCP) phone (6400 and 8400 series) and an analog trunk port require adjustments to the default gain/loss settings to prevent distortion. Use the procedure ([Table 13](#)) to administer digital phones on ATM-PNC DEFINITY ECSs.



NOTE:

Use the following procedure *only for Series 6400 or 8400 digital phones*.

Table 13. Gain/loss adjustment procedure

√	Step	Action	Description
	1.	Set or verify gain/loss administration	Type change system-parameters country-options and press Enter. The System Parameters Country-Options screen displays (Table 17) .
	2.	Set digital loss plan	Domestic: Ensure that the Digital Loss Plan field is 1 . If it is, exit this procedure. International: If the Digital Loss Plan field is anything other than 1, you must change the terminal parameters. Proceed to step 3.
	3.	Customize the digital phone series administration	Type change terminal-parameters termtype and press Enter. (<i>termtype</i> can be 603/302B1 , 6400 or 8400 .) See the 6400-Type Terminal Parameters example (Table 18) display (6400 used for this example). Change the Default Parameter Set field to 1 .
	4.	Change the settings	In the Primary Levels part of the screen, change the following fields to these settings: <ul style="list-style-type: none"> ■ Voice Transmit (dB) = +2.5 ■ Voice Receive (dB) = -2.0 <i>For 6400 and 8400 series phones only:</i> in the BUILT-IN SPEAKER LEVELS part of the screen, change the following fields to these settings: <ul style="list-style-type: none"> ■ Voice Transmit (dB) = 0.0 ■ Voice receive (dB) = 0.0 Press Enter.
	5.	If both series 6400 and 8400 digital phones are used	Repeat steps 3 and 4 for all of the applicable phone series connected to this switch, using 603/302B1 , 6400 or 8400 at the end of the command string. When finished, press Enter to effect the changes.

```
change system-parameters country-options                               Page 1 of 21

                                SYSTEM PARAMETERS COUNTRY-OPTIONS

                                Companding Mode: Mu-Law                Base Tone Generator Set: 1
                                440Hz PBX-dial Tone? n                440Hz Secondary-dial Tone? n
                                Digital Loss Plan: 1
                                Analog Ringing Cadence: 1            Set Layer 1 timer T1 to 30 seconds? n
                                Analog Line Transmission: 1
                                64/84xx Display Character Set? roman
TONE DETECTOR PARAMETERS
                                Tone Detection Mode: 6
                                Interdigit Pause: short
```

Screen 17. System parameters country-options—screen 1

```
change terminal-parameters 6400                                     Page 1 of 1   SPE A

                                6400-TYPE TERMINAL PARAMETERS

                                Default Parameter Set: 1              Customize Parameters? y
OPTIONS
                                Display Mode:
                                Handset Expander Enabled?
                                Volume:
PRIMARY LEVELS
                                Voice Transmit (dB): 2.5
                                Voice Receive (dB): 2.5
                                Voice Transmit (dB):                  Voice Sidetone (dB):
                                Touch Tone Transmit (dB):            Voice Receive (dB):
                                Voice Receive (dB):                  Touch Tone Sidetone (dB):
BUILT-IN SPEAKER LEVELS
                                Voice Transmit (dB):                  Voice Receive (dB):
                                Touch Tone Sidetone (dB):
6402 BUILT-IN SPEAKER LEVELS
                                Voice Receive (dB):                  Touch Tone Sidetone (dB):
```

Screen 18. 6400-type terminal parameters

ATM PNC Reliability

- A DEFINITY® system without duplicated SPEs equipped for ATM Port Network Connectivity (PNC) is supported with duplicated EPN connectivity to disparate points on an ATM network. This process begins with R8r.
- Disparate points may be on separate ATM switches, the same ATM switch, or connected through an ATM wide area network.
 - This provides another level of reliability.

- ATM Network Duplication configurations come equipped with a simplex SPE complex in the PPN, duplicate connectivity over ATM to all PNs and duplicate ATM interfaces in each PN.
- The EPN configuration for ATM Network Duplication is the same as that for an EPN equipped for ATM critical reliability.
- Duplicated paths to the ATM WAN may be through the same ATM switch, different ATM switches, or different points on the ATM WAN.
- ATM Network Duplication is a less expensive alternative to the additional reliability afforded by a DEFINITY[®] system critical reliability configuration, which entails duplicate SPE complexes in the PPN.
- The overall reliability and availability measures associated with the Network Duplication option are better than those associated with the standard reliability option, and worse than those associated with critical reliability.
- DEFINITY[®] system configurators do not support ATM Network Duplication configurations.
- Network Duplication is only available with r ATM PNC configurations.
- All EPNs in a R8r ATM Network Duplication configuration must be configured the same as EPNs in a critical reliability configuration. In this configuration, each PN is equipped with duplicated ATM interfaces, duplicated tone clocks, duplicated fibers, and duplicated ATM connectivity hardware.

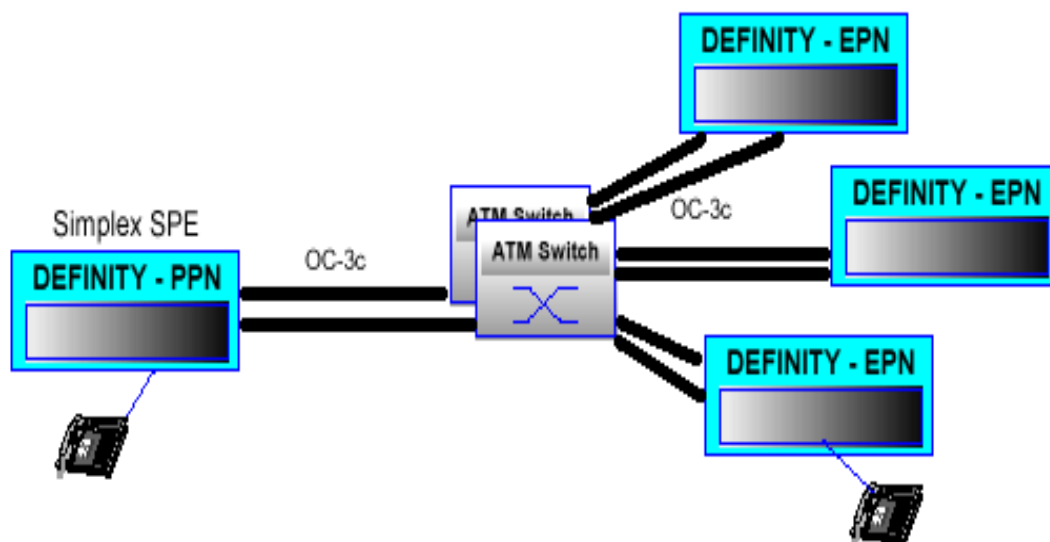


Figure 16. ATM Network Duplication Configuration with Simplex SPE

- Both an expansion control carrier and a port carrier are required in each EPN.
- Each of the two TN2305 or TN2306 ATM interfaces and tone clocks are placed in each of the two carriers.
- A TN771 facility test circuit pack is required in each port network of a R8r system equipped for ATM Network Duplication. DEFINITY® system software does not prevent placement of 2 ATM Interfaces in the same expansion control carrier or port carrier in an ATM Network Duplication EPN.
 - Software does not require that 2 tone clocks be used in an ATM Network Duplication EPN.
- The PPN in a R8r ATM Network Duplication configuration always has only one SPE complex, which is placed in the A cabinet position of an MCC Cabinet.
 - The two required ATM Interfaces in the PPN may both be placed in a port carrier in the B position in slots 2 and 3. One may be placed in the B position port carrier in slot 2 and one may be placed in a C position port carrier in slot 2.
- Directly connected port networks are not supported in an ATM PNC arrangement and are not supported for ATM Network Duplication.

Implementation

- Customers with systems equipped with high reliability (duplex SPE complexes) and who desire greater reliability will migrate to critical reliability.

Procedure

- The command change system parameters customer-options is executed. This will bring up the OPTIONAL FEATURES form. On page 2 of the form is the field PNC Duplication. Enter y to enable PNC duplication for the Network Duplication configuration.
- Insert the second TN2305 or TN2306 ATM Interface in slot 2 of the B position port carrier in each EPN. A TN2182 tone clock must also be present in the slot labeled "tone clock" of the A and B position carriers. Alternatively, the ATM interface and tone clock circuit packs can be administered in software (without hardware insertion) via the standard DEFINITY® system change circuit-packs <cabinet number> command, for example change circuit-packs 2.
- The ATM B-PNC address information is then added via the change atm pnc <n> command, where <n> represents the connection number assigned to each EPN being administered. This will bring up the ATM PNC form.

- Once the ATM Interfaces, tone clocks, and ATM B-PNC addresses have been administered for all EPNs, the change system-parameters duplication command is issued. This brings up the DUPLICATION-RELATED SYSTEM PARAMETERS form. The Enable Operation of PNC Duplication? option is then changed to y. The option for Enable Operation of SPE Duplication? must remain as n.

System Parameters Customer-Options, Page 2

Page 2 of 5

OPTIONAL FEATURES

ISDN-BRI Trunks? n	Restrict Call Forward Off Net? n
ISDN-PRI? n	Secondary Data Module? y
	Station and Trunk MSP? n
Malicious Call Trace? n	Tenant Partitioning? n
Mode Code Interface? n	Terminal Trans. Init. (TTI)? n
Multifrequency Signaling? y	Time of Day Routing? n
Multimedia Appl. Server Interface (MASI)? n	Uniform Dialing Plan? n
Multimedia Call Handling (Basic)? n	Usage Allocation Enhancements? n
Multimedia Call Handling (Enhanced)? n	
Multiple Locations? y	Wideband Switching? n
Personal Station Access (PSA)? n	Wireless? n
PNC Duplicaton? y	
Processor and System MSP? n	
Private Networking? n	

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

Command:

ATM PNC Form - PN With Network Duplication

ATM PNC

Connection Number: 12

A - PNC

Location: 02A01

Name: 123456789012345

Address Format: E.164 ATM Private

AFI: 45

E.164: 1234567890123456

HO-DSP: 12345678

ESI: 123456789012

SEL: 12

B - PNC

Location: 02B02

Name: 123456789012345

Address Format: E.164 ATM Private

AFI: 45

E.164: 1234567890123456

HO-DSP: 12345678

ESI: 123456789012

SEL: 13

Command:

Duplication-Related System Parameters Form

Page 1 of 1

DUPLICATION-RELATED SYSTEM PARAMETERS

Enable Operation of SPE Duplication? n
Enable Operation of PNC Duplication? y

Command:

Duplication configuration

Steps

- busyout pnc-standby
- change sys dup (turn off PNC duplication)
- replace boards on the standby PNC
- change sys dup (turn on PNC duplication)
- reset pnc interchange
- repeat above with other side
- save translations

Upgrades

- TN2305/TN2306 ATM interfaces can be used for R8r ATM PNC capabilities.
- The TN2305/TN2306 ATM Interfaces must be used with ATM CES, with ATM WAN PNC, and if multiple ATM switches are used.

Interworking with Bandwidth Constricted ATM Networks

Conserving ATM Bandwidth Introduction

This is a recommendation to allow DEFINITY[®] R8r systems using ATM Port Network Connectivity to perform optimally when bandwidth is at a premium.

- ATM bandwidth can be conserved, where possible, through optimal DEFINITY[®] resource allocation.
 - Resources that can be optimally placed can include general purpose tone receivers, TN750/B/C multiple announcement boards, and TN771 maintenance test boards.
- When DEFINITY[®] is used together with access concentrators or other ATM switches interfacing to bandwidth constricted or congested facilities on an ATM Wide Area Network call-processing performance can degrade.
- When SVC failures or delays occur more often than normal in a DEFINITY[®] ATM PNC environment, the ATM Network that is out of spec.
 - The solution is to provide tools to manage DEFINITY[®] and the ATM network so that out-of-spec conditions in the ATM network can be detected early, and the network can be restored as quickly as possible.
- Possible strategies entail:
 - Alarming of out-of-spec conditions
 - Detailed SVC latency measurements
 - Providing options to either allow or deny completion of calls that would require a SVC between endpoints which recently experienced a high number of delayed or failed SVCs

Interworking with Bandwidth Constricted Networks

- Port Networks in a DEFINITY[®] ATM PNC configuration, when used together with access concentrators, or other ATM switches interfacing to bandwidth constricted or congested facilities on an ATM Wide Area Network, could incur contention for bandwidth, resulting in failed call completion
- Network throughput with DEFINITY[®] system connections to an ATM Backbone Switch and to the ATM Access Concentrator via the TN2305 or TN2306 ATM interfaces and fiber optic cable is less than OC-3 bandwidth.
 - A few DS1/T1/E1 links are used in the IMA arrangement instead of full OC-3s.

- ATM-PNC operation is the same from the DEFINITY® system perspective as if OC-3 is deployed end-to-end, except when traffic exceeds the carrying capacities of the IMA links, and contention occurs.
 - Similar to what occurs in an ATM network arrangement where IMA is not used but bandwidth is constricted over the ATM network when congestion occurs as the result of traffic over the WAN between destinations other than DEFINITY® endpoints.
 - The DEFINITY® system must coexist with other applications over ATM networks.
 - SVC setups requested by the DEFINITY® system may be denied by the ATM network and associated components, and calls are not completed.

Bandwidth Constricted Network

Example

- Prior to R8.1, when a call is to be extended between PNs distributed over an ATM network, a CCMS message to provide ringing at the destination is sent after the actual SVC setup, but before the SVC connect or release.
 - Until the call is completed, the calling party hears ring back.
- It is possible that the CCMS message goes through and the SVC cannot be established or is delayed.
- The incidence of SVC setup latency has been observed to increase exponentially when very high rates of calls are simulated
 - This would be the equivalent of what would be 280,000 BHC over an EI pair if the rate persisted for an hour.

Improved Resource Allocation

- ATM bandwidth can be conserved, where possible, through optimal DEFINITY® resource allocation.
 - Resources that can be optimally placed can include general purpose tone receivers, TN750/B/C multiple announcement boards, and TN771 maintenance test boards.
- R8r can support up to 10 TN750/B/C integrated announcement boards.
- When an announcement port on one of these boards must be bridged to a call, an ATM SVC connection is required if the integrated announcement board is located in a PN other than where the call is connected.
 - If the integrated announcement board is not at the same location as where the call is connected, a SVC across the ATM WAN is incurred.
 - If the integrated announcement board is at the same location as the call, a SVC does not incur service provider billing.

- Integrated announcement board resources are associated with extension numbers and are often triggered by vector directory numbers (VDNs).
- R8r can be traffic engineered and administered to minimize the number of the expensive SVCs using existing software.
- In a large ATM PNC system distributed over many locations, use of 15A announcement units may be preferable to the use of TN750/B/C.
 - 15A units are connected to analog station or aux trunk ports.
- Multiple units can provide far greater message capacity than 10 integrated announcement boards.
- It may be desirable to enhance DEFINITY[®] software so that the number of integrated announcement boards could be increased.

Performance Management Approach

- When SVC failures or delays occur more often than normally in a DEFINITY[®] ATM PNC environment, this is thought to be evidence of an ATM Network that is out of spec.
 - Hard to predict all of the consequences of a degraded ATM network on the DEFINITY[®] system,
 - It is difficult to predict the effects that could be evidenced when a DEFINITY[®] system processor overload occurs.
- Degraded conditions need to be avoided, as opposed to learning how to live with undesirable circumstances.
- Enhancements include the ability to set thresholds for alarms related to SVC setups, detailed error logs, and new parameters for the determination of state of health.
- Intended that these capabilities will provide administrators with an enhanced ability to isolate, diagnose, and correct problems quickly.

Serviceability Considerations

- This feature is intended to improve serviceability.
- Additional ATM PNC alarm reporting and monitoring capabilities are included.
- Of concern is that customers may set alarm reporting thresholds too low, resulting in an increase of alarms.
 - When an alarm is cleared, ancillary actions will be required in the ATM network.

Improved Resource Allocation

- ATM bandwidth can be conserved, where possible, through optimal DEFINITY[®] resource allocation.
 - Resources that can be optimally placed can include general purpose tone receivers, TN750/B/C multiple announcement boards, and TN771 maintenance test boards.
- R8r can support up to 10 TN750/B/C integrated announcement boards.
 - SVC connection is required if the integrated announcement board is located in a PN other than When an announcement port on one of these boards must be bridged to a call, an ATM where the call is connected.
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 - If the integrated announcement board is at the same location as the call, a SVC does not incur service provider billing.
- Integrated announcement board resources are associated with extension numbers and are often triggered by vector directory numbers (VDNs).
- R8r can be traffic engineered and administered to minimize the number of the expensive SVCs using existing software.

Restarts

- On HOT and SINGLE PROCESS restarts, all impairments, active setup events and alarms are retained.
- On WARM, COLD-2, COLD-1, COOL, REBOOT and EXTENDED REBOOT restarts, all impairments, active setup events, IMP alarms, and system setup alarms are cleared.

Transmission Interface

- TN2305/TN2306 ATM Interfaces may be used for R8r ATM PNC capabilities.
- The TN2305/TN2306 ATM Interfaces must be used with ATM CES, with ATM WAN PNC, and if multiple ATM switches are used.

SETTING THRESHOLDS

- Alarm thresholds are meant to be used for service-affecting conditions only.
- If the system setup alarm is raised, but no individual ATM-IMPs are also alarmed, it is likely that the Activation threshold has been set too high.

- Technicians should consider lowering the Activation threshold, and raising the delay time threshold.
- If an ATM-IMP becomes alarmed unnecessarily, it is preferable to raise the delay threshold rather than raising the Activation threshold.

System Setup Alarms

- A System SETUP alarm uses the existing MO_ATM_IMP maintenance object.
 - Reported as either a minor type alarm or a warning type alarm as determined by the Off-board ATM Network Alarms setting on the Alarm Reporting Options forms and is defaulted to a warning alarm.
- Reported as ATM-NTWK on the ALARM REPORT and HARDWARE ERROR REPORT forms.
- When the number of active SETUP events for all paths becomes greater than 1,500, the System SETUP alarm is raised.
 - The ATM network is alarmed with a warning or minor alarm.
- Resolved when the number of active SETUP events for all paths becomes less than 1,400.
- Special physical name of AT00 is used for this alarm/error.
- If the alarming percentage is 6% and 4 active events for each of 375 ordered pairs exist, the next event is the 1501st event and causes the system alarm, even though none of the individual ordered pairs had yet gotten to 6%.
 - Selection for Activation (%) on the ATM-RELATED SYTEM PARAMETERS form)
 - Interpreted to mean that so many events occurred that the system lost its ability to monitor the percentage of delays and failures on individual connections.
 - A system alarm is created.
 - The system alarm is what might be considered an alarm for non-specified connections.
 - Indicates that network problems may go uncleared.
- If the alarm appears frequently, the administrator may consider changing the threshold.

Set Options Command

- The set options command can be used to change off board ATM Network alarms from warning to minor or from minor to warning.
- This form is displayed via the set options command.

- The Off board ATM Network Alarms field will appear regardless of the option for ATM PNC, but will not appear if the system is a csi or si.
- The set options command can change alarm levels.
- Provides the ability to downgrade the off board ATM-NTWK alarm level from minor to warning level, or vice versa.

Administration

SVC SETUP Alarming Maintenance Object

- Alarms generated due to impaired paths utilize the existing MO_ATM_IMP maintenance object.

SVC Setup Event Alarm Reporting

- ATM-IMP alarms are reported as either a minor type alarm or a warning type alarm as determined by the Offboard ATM Alarms setting and are defaulted to a warning alarm.
- These alarms are reported as ATM-NTWK alarms on the ALARM REPORT and HARDWARE ERROR REPORT forms.

New Commands

- Access to all the new commands may be blocked for a given login by administering the Additional Restriction system-atm on the logins permissions form.
 - Change Command
 - Authorization: Administer Features
 - Syntax: change system-parameters atm
 - Help: Entry is not required
 - Errors: <none>
 - Display Command
 - Authorization: Display Admin. and Maint. Data
 - Syntax: display system-parameters atm ["print" or "schedule"]
 - Help: ["print" or "schedule"]
 - Errors: " " is an invalid identifier; please press HELP

ATM PNC LATENCY HISTOGRAM Form

- This form is accessed with the command list measurements atm latency-histogram [current-hour, last-hour, today-total, and yesterday-total].

- The commands are available if Asynch. Transfer Mode (ATM) PNC? is enabled via the form resulting from the command change systems-parameters customer-options, regardless of the administered software version.

ATM PNC LATENCY HISTOGRAM Form

```
list measurements atm latency-histogram last-hour
```

Page 1 SPE A

Switch Name: _____ Date: 10:27 am TUE OCT 25, 1999

ATM PNC LATENCY HISTOGRAM

Setup Count: 770	Average Latency (msec): 53	Failure Count: 3
AddParty Count: 538	Average Latency (msec): 92	Failure Count: 1
Total Count: 1308	Average Latency (msec): 71	Failure Count: 4

Range (msec)	Setup	AddParty	Total	Range (sec)	Setup	AddParty	Total
0-9	0	0	0	1-1.9	1	0	1
10-19	200	2	202	2-2.9	0	0	0
20-49	400	3	403	3-3.9	0	0	0
50-99	100	1	101	4-4.9	6	1	7
100-149	50	6	56	5-5.9	0	0	0
150-199	10	0	10	6-6.9	0	0	0
200-249	3	0	3	7-7.9	0	0	0
250-299	0	0	0	8-8.9	0	0	0
300-399	0	0	0	9-9.9	0	0	0
400-499	0	0	0	10 +	0	0	0
500-999	0	0	0				

- Access to this command may be blocked for a given login by administering the Additional Restriction "atm-pnc" on the logins permissions form.
- Only those logins having the authorizations will have access to the commands.
- List Command
 - Authorization: System Measurements
 - Syntax: list measurements atm latency-histogram "current-hour" "last-hour" "today-total" or "yesterday -total" ["print" or "schedule"]
 - Help: Enter "current-hour" "last-hour" "today-total" or "yesterday-total" , ["print" or "schedule"]
 - Errors: "x" is an invalid identifier; please press HELP



NOTE:

Execution of the set time command will not affect the output of the list measurements atm latency-histogram commands.

Form Fields

- Setup Count: the number of SVCs setup during the time period indicated. Values: 0 - 9999999

- AddParty Count: the number of ADDPARTY requests processed. Values: 0 - 9999999
- Total Count: the total number of SETUP and ADDPARTY requests. Values: 0 - 9999999
- Average Latency: average latency in milliseconds for SVC type described during the time period indicated. Values: 0 - 8000
- Failure Count: the number of SVCs of the type described that could not be established by the ATM network during the time period. Does not include SVCs that were delayed and the delay exceeds the threshold set for Timeout (msec): on the ATM-RELATED SYSTEM PARAMETERS form.)
- Values: 0 - 9999. The number 9999 will be shown if the value is greater than 9999.
- (msec) Count: the number of SVCs for which delay in milliseconds fell within each range during the time period indicated. Values: 0 - 9999999
- (sec) Count: the number of SVCs for which delay in seconds fell within each range during the time period indicated. Values: 0 - 9999999

ATM PNC Delayed/Failed Setups Form

- This form is accessible with the command list measurements atm setup-events ["x"] or ["from-conn" x] ["to-conn" x] ["print" or "schedule"].
- Available if Asynch. Transfer Mode (ATM) PNC? is enabled via the form resulting from the command systems-parameters customer-options.
- The command syntax allow the administrator to list setup events for a particular connection number or for a range of connection numbers.
- list measurements atm setups-events Command

Command	Result
"list measurements atm setup-events"	Provides a list of up to 100 of the most recent setup events.
"list measurements atm setup-events from-conn 3"	Provides a list of all setup events from connection 3 to all other connections.
"list measurements atm setup-events from-conn 3 to-conn 4"	Provides a list of all setup events from connection 3 to connection 4.
"list measurements atm setup-events to-conn 4"	Provides a list of all setup events from all connections to connection 4.
"list measurements atm setup-events 4"	Provides a list of all setup events from or to connection 4.

ATM PNC DELAYED/FAILED SETUPS Form, Page 1

```
list measurements atm setup-events
```

```
Page 1
```

```
Switch Name:          Date: 2:33 pm TUE DEC 12, 1999
```

ATM PNC DELAYED/FAILED SETUPS						
Date	Time	Event	Delay(msec)	From Conn	To Conn	PNC
12/12/1999	14:32:10	delayed	1765	3	7	A-PNC
12/12/1999	14:31:37	failed		3	7	A-PNC
12/12/1999	14:31:22	delayed	1809	7	8	A-PNC
12/12/1999	14:31:18	delayed	1672	12	6	B-PNC
12/12/1999	14:31:15	delayed	1023	3	7	B-PNC
12/12/1999	14:29:45	delayed	1005	1	3	B-PNC
12/12/1999	14:29:23	failed		3	1	A-PNC
12/12/1999	14:28:12	failed		6	2	A-PNC
12/12/1999	14:27:56	delayed	2088	6	7	A-PNC
12/12/1999	14:27:24	delayed	1308	6	8	B-PNC
12/12/1999	14:27:05	delayed	1011	6	2	A-PNC
12/12/1999	14:26:38	failed		11	4	B-PNC
12/12/1999	14:26:32	delayed	1246	3	7	A-PNC

```
Command:
```

ATM PNC DELAYED/FAILED SETUPS Form

- Access to the new command may be blocked for a given login by administering the Additional Restriction "atm-pnc" on the logins permissions form.
- Only those logins having the authorizations will have access to the commands.
 - Authorization: System Measurements
 - Syntax: list measurements atm setup-events ["x"] or ["from-conn" x] ["to-conn" x] ["print" or "schedule"]
 - Help: Enter [x] or ["from-conn" x] ["to-conn" x], ["print" or "schedule"] (Where x is an administered connection number between 1-44.)
 - Errors: is an invalid identifier; please press HELP
 - Connection number out of range; please press HELP
 - Connection not assigned; please select another
 - The "from" connection number cannot equal the "to" connection number.



NOTE:

*Execution of the set time command will not affect the output of the **list measurements atm setup-events** command.*

Form Pages

- Up to 100 entries will be listed if no selection criteria are indicated, otherwise all available are provided.

Form Fields

- Switch Name: The value administered on the system-parameters features form.
- Date: Time/date that the command is being executed.
- Date: System Date when the event occurred (4-character year field used for Y2K compliance)
- Time: System Time when the event occurred
- Event: The type of event (delayed or failed)
- Delay (msec): The Delay value is only displayed for delayed events. For failed events, this field will be blank.
Values: (5 digits) Values will range from 250 to 20,000
- From Conn: The originating connection number
- To Conn: The destination connection number
- PNC: A-PNC or B-PNC

Changes to Existing Forms and Commands

Modifications to the PERMISSIONS Form

- Adds new Additional Restriction keyword system-atm to the PERMISSIONS form.
- This controls access to the ATM-RELATED SYSTEM PARAMETERS form.

ATM RELATED SYSTEM PARAMETERS

- This form is accessed with the commands change system- parameters atm and display system-parameters atm.
- The entire form is only accessible and visible if the value for Asynch Transfer Mode (ATM) PNC? on the SYSTEM PARAMETERS -CUSTOMER OPTIONS form is set to y, and the G3 Version is V8 or greater.

ATM-Related System Parameters Form

```
change system-parameters atm
Page 1
ATM-RELATED SYSTEM PARAMETERS

      THRESHOLDS OF SETUP EVENTS TO DECLARE IMPAIRMENT
                                Activation (%): 4
                                Deactivation (%): 3
                                Timeout (msec): 1000

      TREATMENT DURING IMPAIRED CONDITIONS

                                Deny Calls Using Impaired Paths? n

Command:
```

- **Activation (%): Field**
 - Values for this field can be any whole number from 1 to 20. The default is 4.
- **Deactivation (%): Field**
 - Values for this field can be any whole number from 0 to 1 less than the number indicated for Activation (%):.
 - The default is 3.
 - If the value for Activation (%): is 5, then the value for Deactivation (%): can be 0, 1, 2, 3, or 4.
- **Timeout (msec): Field**
 - Values for this field are from 250 to 20,000, in increments of 1. The default is 1000.
- **Deny Calls Using Impaired Paths? Field**
 - Values for this field are y or n. The default is n.
- **Activation (%)**
 - Use: This field specifies the percentage of setup events for a path to become impaired. That is, if the percentage of setup events for a given path meets or exceeds this administered value, then it becomes impaired. An alarm is raised against an ATM-IMP when one or more paths originating from that ATM-IMP become impaired.
 - Default: 4
 - Length: 2 characters
 - Help: Enter number between 1 - 20
 - Errors: Entry must be all digits
 - Field cannot be blank

- Deactivation (%)
 - Use: This field specifies the percentage of setup events for a path to come out of the impaired state. That is, if the percentage of setup events for an impaired path meets or drops below this administered value, then it is no longer impaired. An alarm is retired when there are no more impaired paths originating from that ATM-IMP.
 - Default: 3
 - Length: 2 characters
 - Help: Enter number (0-19) which is less than the Activation value.
 - Errors: Entry must be all digits
 - Field cannot be blank
 - Entry must be less than the Activation value
- Timeout (msec):
 - Use: This field specifies the maximum number of milliseconds allowed for a SVC connection to be completed before being counted as delayed.
 - Default: 1000
 - Length: 5 characters
 - Help: Enter number between 250 - 20000
 - Errors: Entry must be all digits
 - Field cannot be blank
- Deny Calls Using Impaired Paths?
 - Use: This field specifies whether to deny calls that use connections over impaired paths. If the option is y, then calls, which require a new SVC on an impaired path on the active PNC, will not be completed. If the option is n, then such calls will be treated as per current operation for ATM PNC.
 - Default: n
 - Length: 1 character
 - Help: n (no) y (yes)
 - Errors: x is an invalid entry; please press HELP

Authorization/Restrictions

- Asynch. Transfer Mode (ATM) PNC? must be enabled on the OPTIONAL FEATURES form to access Conserving ATM Bandwidth capabilities.
- The command to access the form is change system-parameters customer-options.

Upgrades

- Upgrades of R6r and R7r ATM PNC equipped systems to R8r will use the capabilities of conserving ATM Bandwidth without additional feature administration.
- Upgrades of critically duplicated r systems to R8r are call preserving.

Implementation Notes



NOTE:

Not all calls are denied on impaired paths since there are some calls which can be satisfied by SVCs in the cache. Therefore, since a denied call may place a new SVC in the cache (see below), the next call on this connection may succeed when the prior call was denied. The customer could see calls succeeding somewhat randomly.

- To prevent an alarm from staying on forever, DEFINITY Connection Management software (CM) performs the following whenever it denies a call:
 - If a SVC in the coming up state exists in the cache, do nothing.
 - Creates an SVC and immediately put it in the cache (if room), or mark it for destruction (if no room in the cache).
 - These SVC setup attempts will contribute to measurements of delayed & failed SVCs.
 - When this SVC comes up, scan the event log and possibly clear the alarm, if the setup delay is now below the threshold.

Alarm Activation

- SVC SETUP impairment parameters are administered on the ATM-RELATED SYSTEM PARAMETERS form. Notification of an alarm condition for a given ATM-IMP will be recorded in the alarm log when one or more paths originating at that ATM-IMP becomes impaired.



NOTE:

An ATM Interface Module Port (ATM-IMP) is a port on an ATM network to which a DEFINITY ATM Interface circuit pack is attached. The ATM-IMP maintenance object (MO) deals with off-board alarms associated with an ATM Interface Module Port.

- The maximum matrix of PN pairs could be as great as 2x(44x43), depending on the number of PNs - 44 is the maximum on an r.
- A delayed SVC is an SVC that did not complete in an amount of time less than or equal to the parameter set for "Timeout (msec):" on the ATM-RELATED SYSTEM PARAMETERS form.

Alarm Deactivation

- The alarm condition for a given ATM-IMP is resolved whenever the number of active SETUP events drops to or below the Deactivation value on the ATM-RELATED SYSTEM PARAMETERS form for all paths originating at that ATM-IMP (i.e., all paths become unimpaired).
 - If the activation threshold is 5%, then the clearing threshold can be 0%, 1%, 2%, 3%, or 4%.

SVC Latency Measurements

- The ATM PNC LATENCY HISTOGRAM form will be visible when the command list measurements atm latency-histogram (current-hour, last-hour, today-total, and yesterday-total)] is executed.
- The number of setups per time period selected (current-hour, last-hour, today-total, and yesterday-total), the average latency in milliseconds, and the failure count per each range is provided by call processing software to administration software and is available to the user.
- Provide Detail for Active Setup Events
- The ATM PNC DELAYED/FAILED SETUPS form is displayed by the command list measurements atm setup-events. Event date, event time, event type (delayed or failed), delay time in milliseconds, from/to PN connection numbers, and PNC connection (A- PNC or B-PNC) detail for all active SETUP events are provided.

The Option to Deny Calls

- The option to deny calls over impaired paths is administered via the new Deny Calls Using Impaired Paths? field on the ATM-RELATED SYSTEM PARAMETERS form.
- If the option is y, then calls which require a new SVC over an impaired path on the active PNC will not be completed.
- If the option is n, then such calls will be treated as per current operation for ATM PNC.

Alarm Deactivation while Calls Are Being Denied

- When denying calls, call processing software continues to periodically launch SETUPS into the network on impaired paths (but not in sufficient quantity to cause significant further degradation of the path).
 - This is necessary to stimulate the retiring of alarms when network conditions improve.

- If they are not denied, current operation for delayed SVCs means that it is possible that the called party hears ringing, goes off-hook and initially cannot hear the calling party or cannot be heard by the calling party, or both cannot hear each other until one-way SVCs in both directions are established.
- The drop on failure operation is maintained for failed calls.
- If one or more inter-PN connection requests are failed by the ATM network, the affected call will be dropped if it is a two-party call.
 - Otherwise, the failure will be ignored by the switch and must be dropped by the parties to the conference.
- A call is denied if and only if all of the below are true:
 - The Deny Calls Using Impaired Paths? option is on
 - The call requires a path which is impaired on the active PNC
 - A SVC for that path cannot be found in the cache
- No paths on the standby PNC are considered when denying calls.

Call Processing

Performance Management Approach

- Alarming of out of spec conditions
- Detailed SVC latency measurements
- Giving the administrator the option to either allow or deny completion of calls that would require a new SVC over a path which has recently experienced a high number of delayed or failed SVCs

Initial Setup v. Adding a Leg

- When a connection is first set up between two endpoints over an ATM PNC arrangement, it is created with a SETUP request to the network.
 - Later, it is possible to add another endpoint with an ADDPARTY request as the result of adding a conferee to a conference call, for example.
- The ADDPARTY and the SETUP messages are very similar from the call processing perspective.
 - A station user being called, however, would not know which method was used to connect a call.
 - The effect on the talk path is the same.
 - For example, a slow SETUP or a slow ADDPARTY may both cause operational problems.

Interactions between Call Processing Software and Maintenance Software

- Call processing software and maintenance software work together to provide alarm activation and deactivation.
- Call processing software will report event information to maintenance software that can result in maintenance software activating or deactivating alarms.

Configuration/Environment

Hardware

- TN2305/TN2306 ATM Interfaces are to be used for R8r ATM PNC capabilities.
- The TN2305/TN2306 ATM Interfaces must be used with ATM CES, with ATM WAN PNC, and if multiple ATM switches are used.

Performance Management Approach

- When SVC failures or delays occur more often than normally in a DEFINITY[®] ATM PNC environment, this is most often expected to be evidence of an ATM Network that is out of spec.
- Hard to predict all of the consequences of a degraded ATM network on the DEFINITY[®] system,
- DEFINITY[®] system is the platform for call centers, voice messaging, CTI, Telecommuting feature, and many other complex applications and interactions.
- It is difficult to predict the effects that could be evidenced when a DEFINITY[®] system processor overload occurs.
- Such degraded conditions need to be avoided, as opposed to learning how to live with undesirable circumstances.
- Strategy entails:
 - Alarming of out-of-spec conditions
 - Detailing SVC latency measurements
 - Giving the administrator the option to either allow or deny completion of calls which would require a new SVC over a path which has recently experienced a high number of delayed or failed SVCs so that alternative methods of handling these calls is possible, particularly in a call center environment
- Enhancements include the ability to set thresholds for alarms related to SVC setups, detailed error logs, and new parameters for the determination of state of health.

- Intend is that these capabilities will provide administrators with an enhanced ability to isolate, diagnose, and correct problems quickly.

Bandwidth Constricted Network

Example

- Currently, when a call is to be extended between PNs distributed over an ATM network, a CCMS message to provide ringing at the destination is sent after the actual SVC setup, but before the SVC connect or release.
 - Until the call is completed, the calling party hears ring back.
- It is possible that the CCMS message goes through and the SVC cannot be established or is delayed.
- The incidence of SVC setup latency has been observed to increase exponentially when very high rates of calls are simulated
 - Equivalent of what would be 280,000 BHC over an EI pair if the rate persisted for an hour.
- When ATM PNC SVC setups are delayed or failed, the called party may hear ringing, take the receiver offhook, and hear nothing.
- The called party may eventually receive a delayed call, but will never receive a failed call.
- If the call was delayed, the calling party hears silence if one of the SVCs is not up or hears the called party if it is up.
- If the call is never completed, the caller will eventually hang up without knowing why the call was not completed.
- Occurrence is annoying to users on both ends.
- In a call center environment this type of problem results in dissatisfied callers, abandoned calls, wasted time for call center agents, and lost opportunities for business revenue, thereby defeating the purpose of the call center.

Response To System Setup Alarm

- The system setup alarm is raised to alert technicians to the possibility of impaired paths, which are undetected.
- To diagnose the problem, the technicians can find ample information in the error log, the setup-events log, the latency-histogram, and the pnc-latency screens.
- The error log will also enable one to translate an IMP alarm into an impaired path.
- Technicians may decide that the switch is operating normally.
- Technician should consider lowering the Activation threshold.

- If all appears normal in the error log, but the latencies are too high, may need to upgrade the ATM network to solve the problem.



NOTE:

fter the problem is fixed, the alarm will clear, but it will take an indeterminate amount of time.

Alarm Reporting Options Form

```

set options
ALARM REPORTING OPTIONS
Page 1 of 2

                Major Minor
On-board Station Alarms: w   w
Off-board Station Alarms: w   w
On-board Trunk Alarms (Alarm Group 1): y   y
Off-board Trunk Alarms (Alarm Group 1): w   w
On-board Trunk Alarms (Alarm Group 2): w   w
Off-board Trunk Alarms (Alarm Group 2): w   w
On-board Trunk Alarms (Alarm Group 3): w   w
Off-board Trunk Alarms (Alarm Group 3): w   w
On-board Trunk Alarms (Alarm Group 4): w   w
Off-board Trunk Alarms (Alarm Group 4): w   w
On-board Adjunct Link Alarms: w   w
Off-board Adjunct Link Alarms: w   w
Off-board MASI Link Alarms: w
Off-board DS1 Alarms: w   w
Off-board PI-LINK Alarms: w   w
Off-board Alarms (Other): w   w
Off-board ATM Network Alarms: w
    
```

- The ALARM REPORTING OPTIONS form is displayed via the set options command.
- The Off-board ATM Network Alarms field appears regardless of the option for ATM PNC, but will not appear if the system is a csi or si.

Hardware Error Report

```
display errors
HARDWARE ERROR REPORT
Port Mtcce Alt Err Aux First Last Err Err Rt/ Al Ac
  Name      Name Type Data Occur      Occur      Cnt Rt Hr St
AT03A ATM-NTWK      268 412 11/17/15:01 11/17/15:01 1 0 1 n n
AT03B ATM-NTWK      268 412 11/17/15:02 11/17/15:02 1 0 1 n n
AT03A ATM-NTWK      269 413 11/17/15:03 11/17/15:03 1 0 1 n n
AT01A ATM-NTWK      260 104 11/17/15:05 11/17/15:05 1 0 1 n n
AT01B ATM-NTWK      260 104 11/17/15:06 11/17/15:06 1 0 1 n n
AT13A ATM-NTWK      264 1507 11/17/15:07 11/17/15:07 1 0 1 n n
AT22A ATM-NTWK      259 603 11/17/15:08 11/17/15:08 1 0 1 n n
AT22A ATM-NTWK      276 610 11/17/15:09 11/17/15:09 1 0 1 n n
AT00 ATM-NTWK      301      11/16/09:01 11/16/09:01 1 0 1 n n

Command successfully completed
Command:
```

- The Port is deciphered as: ATxy.
 - xy is the connection number, not the port network number.
- The mapping between connection number and port network number can be found with the command list atm pnc.
- The Aux Data field contains the originating/ destination port network numbers.
 - The rightmost two digits are the destination port network number.
 - The remaining leftmost digits are the origination port network number.
- The orig/dest designates the direction of the SVC setup.

Example

```
AT13A   ATM-NTWK   264   1507   11/17/15:01 11/17/15:01   23   0   1 n n
```

```
AT13A:           ATM connection number 13, PNC-A.
  264:           Error type: 256 + destination connection number 8
  1507:
```

Originating port network number is 15. Destination port network number is 7. This all means that the threshold for SVC setup events from connection # 13 to connection # 8 has been exceeded.

- The Err Type field is the value 256+destination connection #.
- This allows multiple entries of the same port name/ Mtce name pair in the error log.
- If the error type value was the same for a given port name/Mtce name pair, only get 1 entry for that port name/Mtce name pair in the error log.
- Up to 6 entries of the same port name/Mtce name pair can be stored in the error log.
- When a new entry with the same port name/Mtce name pair is logged, the oldest one is overwritten.

Alarm Report

```
display alarms
ALARM REPORT
Port      Maintenance On  Alt  Alarm  Svc  Ack?  Date      Date
          Name      Brd? Name  Type  State 1 2   Alarmed  Resolved

AT03A    ATM-NTWK   n   WARN           11/17/15:01 00/00/00:00
AT03B    ATM-NTWK   n   WARN           11/17/15:02 00/00/00:00
AT01A    ATM-NTWK   n   WARN           11/17/15:03 00/00/00:00
AT01B    ATM-NTWK   n   WARN           11/17/15:04 00/00/00:00
AT13A    ATM-NTWK   n   WARN           11/17/15:05 00/00/00:00
AT22A    ATM-NTWK   n   WARN           11/17/15:06 00/00/00:00
AT00     ATM-NTWK   n   WARN           11/17/15:06 00/00/00:00
```

```
Command successfully completed
Command:
```

- The Port is deciphered as: ATxyZ. xy is the connection number, not the port network number.
- The mapping between connection number and port network number can be found with the command list atm pnc.
- The alarm indicates that at least one path originating from the displayed Port, has exceeded the SVC setup threshold.
- The alarm is resolved when SVC setups from all paths that originate from the given Port, have gone below a deactivation threshold for SVC setups.

Test Results for ATM SVC Setups

```
test atm svc-setups from-conn 3 to-conn 5 repeat 100
```

TEST RESULTS

Port	Maint Name	Orig PN	Dest PN	Result	Error Code
AT03A	ATM-NTWK	3	5	PASS	

Command successfully completed

Command:

Changes to Port Network Connectivity Form for Status PNC Command with ATM PNC Enabled

```
status pnc
PORT NETWORK CONNECTIVITY
```

Duplicated? yes

Software Locked? no

Standby Busied? no

Standby Refreshed? no

Interchange Disabled? yes

A-PNC

Mode: standby

State of Health: functional

Inter PN Index: 00.00.00.00

Major Alarms: 0

Minor Alarms: 0

Warning Alarms: 0

B-PNC

Mode: active

State of Health: functional

Inter PN Index: 00.00.00.00

Major Alarms: 0

Minor Alarms: 0

Warning Alarms: 0

T1 401A/E1 402A/E1 403A Synchronization Splitters

A DEFINITY ATM-PNC requires network synchronization for DS1 circuit packs not to slip relative to the LEC/IXC switches. The ATM switch serves as the sync reference source for the DEFINITY. The ATM switch, in turn, derives primary and secondary sync. To accomplish this, the most common option is to use synchronization expanders (splitters).

Connections needing synchronization splitters

If the ATM network does not provide a synchronization expander (splitter), then the ATM configurations may require one that takes a DS1 T1 or E1 signal and redirects it to the

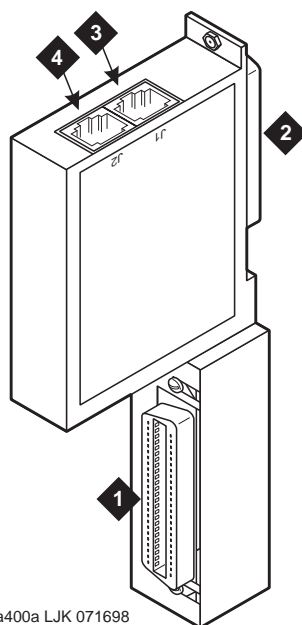
- ATM switch(es), depending on configuration and duplication
- DEFINITY ECS through the DS1 circuit pack

Splitter descriptions

[Table 14](#) describes the 4 splitter models and their capabilities. The drawings show the splitters and their connection points. [Figure 21](#) shows a schematic of the 2 jumper sets and their connections for 401A, 402A, and 403A sync splitters.

Table 14. Synchronization splitter models and attributes

Model	T1/E1	Impedance	Comcode	Drawing	Description/Application
400A	T1	100 Ω	108217795	Figure 17	No ICSU capability
401A	T1	100 Ω	108508078	Figure 18	Limited ICSU capability
402A	E1	120 Ω	108508094	Figure 19	
403A	E1	75 Ω	108508102	Figure 20	



crda400a LJK 071698

Figure Notes:

1. From network interface
2. Amphenol connection to DEFINITY ECS
3. Timing output port (J1) to the ATM switch^a
4. Timing output port (J2) to the ATM switch^a

- a. Ports J1 and J2 provide identical DS1 timing source signals to the ATM switches. The ATM switch can use two separate DS1 timing signals (one at a time from two separate spans).

Figure 17. 400A synchronization splitter

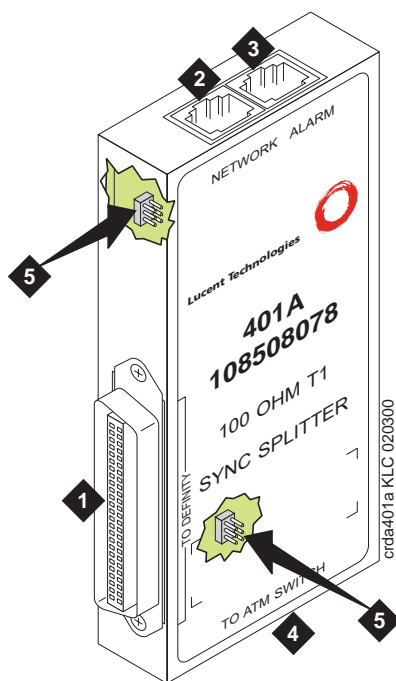


Figure Notes:

- | | |
|---------------------------------------|--|
| 1. Amphenol connector to DEFINITY ECS | 3. Timing alarm lead connection |
| 2. Network timing connection | 4. Timing output ports (RJ45) to ATM switch ^a |
| | 5. Jumpers and capacitors (inside case). See Figure 21 for settings. |

- a. Ports J1 and J2 provide identical DS1 timing source signals to the ATM switches. The ATM switch can use two separate DS1 timing signals (one at a time from two separate spans).

Figure 18. 401A synchronization splitter

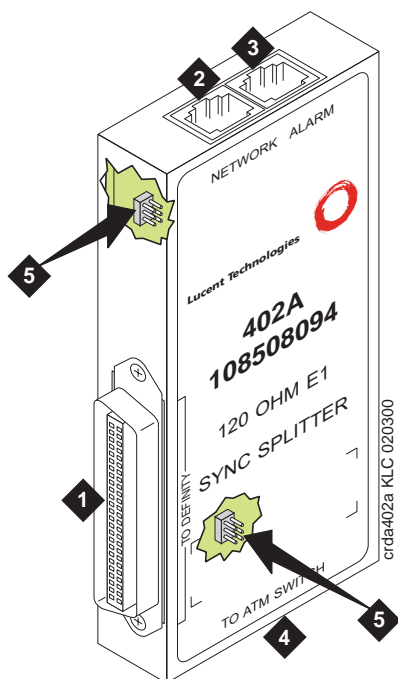


Figure Notes:

- | | |
|---------------------------------------|--|
| 1. Amphenol connector to DEFINITY ECS | 3. Timing alarm lead connection |
| 2. Network timing connection | 4. Timing output ports (RJ45) to ATM switch ^a |
| | 5. Jumpers and capacitors (inside case). See Figure 21 for settings. |

a. Ports J1 and J2 provide identical DS1 timing source signals to the ATM switches. The ATM switch can use two separate DS1 timing signals (one at a time from two separate spans).

Figure 19. 402A synchronization splitter

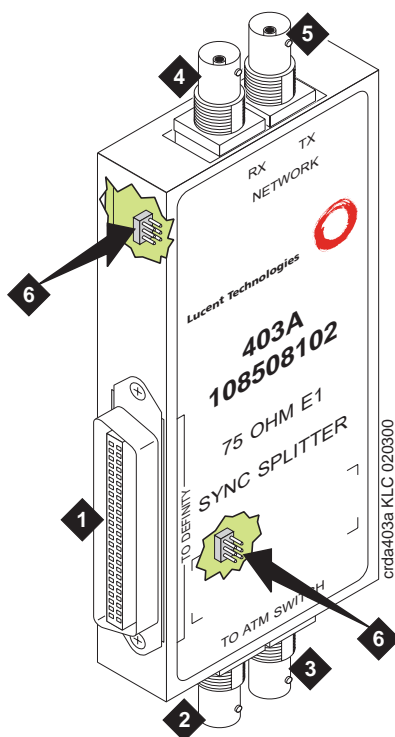


Figure Notes:

1. Amphenol connector to DEFINITY ECS
2. Synchronization source (timing output ports) to ATM switch¹
3. Synchronization source (timing output ports) to ATM switch^a
4. Network receive connection, BNC connector
5. Network transmit connection, BNC connector
6. Jumpers and capacitors (inside case). See [Figure 21](#) for settings.

- a. These are identical DS1 timing source signals to the ATM switches. The ATM switch can use two separate DS1 timing signals (one at a time from two separate spans).

Figure 20. 403A synchronization splitter

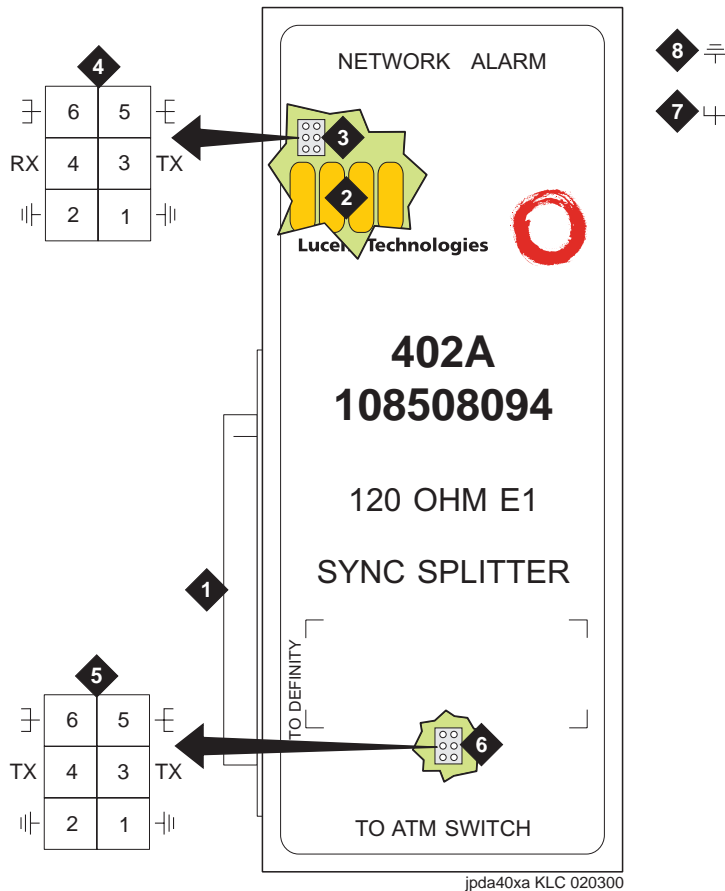


Figure Notes:

1. Amphenol connection to DEFINITY ECS switch
2. Row of capacitors
3. Jumpers for incoming network connections
4. Incoming network transmit and receive connections
5. Jumper 1-2 = true ground
Jumper 5-6 = shield grounded
Jumper 3 = TX cable ground
Jumper 4 = RX cable ground
Default connections = 1-2, 3-5
6. True ground
7. Cable shield grounded
8. Jumper 1-2 = true ground
Jumper 5-6 = shield grounded
Jumper 3 = ATM switch A TX cable ground
Jumper 4 = ATM switch B TX cable ground
Default connections = 3-5, 4-6
9. Output jumpers
10. Timing output to ATM switch A
11. Timing output to ATM switch B

Figure 21. Jumper settings (401A/402A/403A)

Synchronization splitter connections

The splitter connects to a timing source. [Figure 22](#) shows the synchronization connections through a DSU/CSU (400A), and [Figure 23](#) shows the synchronization connections through an ICSU (400A). [Figure 24](#) shows the synchronization connections directly to the timing source (401A, 402A, 403A).

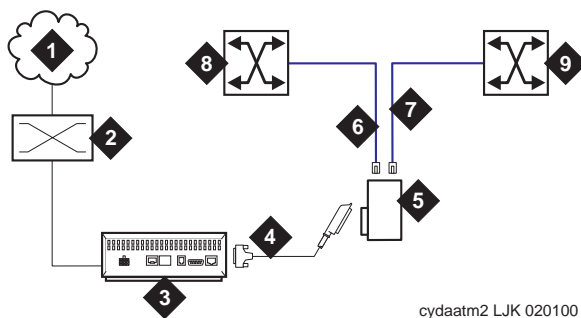


Figure Notes:

- | | |
|--|--|
| 1. Public Switched Telephone Network (PSTN) | 6. H600-383 quad cable from sync splitter to ATM switch A |
| 2. Main distribution frame (MDF) or smart jack. | 7. H600-383 quad cable from sync splitter to ATM switch B (critical reliability/ATM network duplication) |
| 3. Channel service unit (CSU) | 8. Lucent ATM switch A |
| 4. H600-307-GR2 cable | 9. Lucent ATM switch B (critical reliability/ATM network duplication) |
| 5. 400A T1 (100 Ω) splitter connects to the DEFINITY ECS | |

Figure 22. Synchronization connections through an external DSU/CSU (400A)

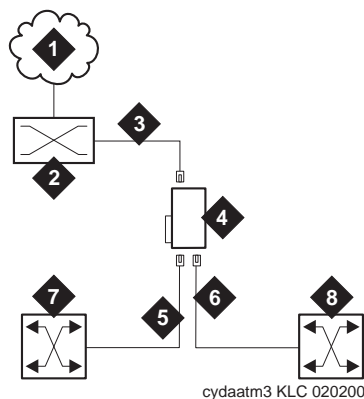


Figure Notes:

- | | |
|---|---|
| 1. Public switched telephone network (PSTN) | 5. H600-383 quad cable from sync splitter to ATM switch A |
| 2. Main distribution frame (MDF) or smart jack | 6. H600-383 quad cable from sync splitter to duplicated ATM switch B (critical reliability/ATM network duplication) |
| 3. H600-383 quad cable | 7. Lucent ATM switch A |
| 4. 401A T1 (100 Ω) splitter, 402A E1 (120 Ω) splitter, or 403A E1 (75 Ω) splitter connected to DEFINITY ECS | 8. Lucent ATM switch B (critical reliability/ATM network duplication) |

Figure 24. Synchronization connections directly to timing source (401A/402A/403A)

Use the information in [Table 15](#) to determine the maximum cable run lengths for the configuration.

Table 15. Maximum cable lengths

Splitter	Used in	Cable	Maximum distance from splitter to ATM switch
400A T1 100 Ω	North America (USA and Canada)	H600-383 ^a shielded twisted pair	250 feet 76 meters
401A T1 100 Ω	North America (USA and Canada)	H600-383 ¹ shielded twisted pair	1310 feet ² 393 meters
402A E1 120 Ω	See Table 16	120-Ω shielded twisted pair	1000 feet ^b 305 meters
403A E1 75 Ω	See Table 16	75-Ω coaxial	1000 feet ² 305 meters

-
- a. Must have RJ45 connectors on each end.
 - b. The loss allowed in the cable is 6 dB. Typical cables have losses of 0.6 dB/100 ft; hence, 1000 feet may be considered a typical distance. Distance varies if cables with different losses are used.

[Table 16](#) lists the splitter model(s) applicable to specific countries.

Table 16. Country-specific splitter applications

Country	Splitter model		
	400A/401A T1 100 Ω	402A E1 120 Ω	403A E1 75 Ω
Argentina			Y
Australia		Y	
Austria		Y	Y
Bahrain		Y	Y
Belgium		Y	Y
Brazil			Y
Canada	Y		
China			Y
Columbia			Y
Denmark		Y	Y
Egypt		Y	Y
Finland		Y	Y
France		Y	Y
Germany		Y	Y
Hong Kong		Y	
India		Y	Y
Indonesia		Y	Y
Ireland		Y	Y
Italy		Y	Y
Japan ^a			
Korea		Y	Y
Luxembourg		Y	Y
Malaysia		Y	Y
Mexico			Y
Netherlands		Y	Y
New Zealand		Y	Y
Norway		Y	Y

Continued on next page

Table 16. Country-specific splitter applications — Continued

Country	Splitter model		
	400A/401A T1 100 Ω	402A E1 120 Ω	403A E1 75 Ω
Philippines		Y	Y
Poland		Y	Y
Portugal		Y	Y
Russia		Y	Y
Saudi Arabia		Y	Y
Singapore		Y	
South Africa		Y	Y
Spain		Y	Y
Sri Lanka		Y	Y
Sweden		Y	Y
Switzerland		Y	Y
Taiwan		Y	Y
Ukraine		Y	Y
UAE		Y	Y
UK		Y	Y
USA	Y		
Uzbekistan		Y	Y
Vietnam		Y	Y

- a. Japan uses both 75-Ω and 120-Ω T1; 402A and 403A sync splitters will also work with T1s.

2 Networking Enhancements
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