Lucent Technologies
Bell Labs Innovations

Lucent Compact Switch Feature Packages Guide

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Software Version 3.11

Notice

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

Mandatory customer information

Safety

Always observe the Safety Instructions when operating the system.

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Ordering information

The order number of this document is 255-400-012R3.11.

Support

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Introduction

This document contains the following sections:

Scope

The *Feature Packages Guide* (255-400-012) is part of the overall switch documentation set.

The *Feature Packages* provides an overview of the key Feature Packages that are supported by the Lucent Compact Switch,

Note: For detailed descriptions and illustrations of the switching system hardware, refer to the *Lucent Gateway Platform Planning and Engineering Guide*.

This introduction describes the *Feature Packages Guide* and how it relates to other manuals within the system documentation set. It also describes the switching system documentation set. Specifically, these topics are described in this introduction:

- Scope and Audience
- Reason for Reissue
- Manual Organization
- Using the CD-ROM
- Switch Documentation

Audience

This manual is intended for personnel who would like to know more about the functions and features supported by the switch software. It may be helpful initially to the customer trying to determine whether or not the switch suits their needs, or after the sale and installation process, to the person responsible for provisioning the switch. Regardless of the person using this manual, these personnel must have a thorough knowledge of telecommunications.

Reason for Reissue

This manual was introduced for system software version 3.11.

Manual Organization

The manual contains detailed descriptions of key Feature Packages features supported by the switching system software, as well as the following information:

- The Introduction contains a description of the system manuals and manual organization, and general information about the system documentation.
- The Class 5 Services section provides details about the Custom Local Area Signaling Services (CLASS) and Custom Calling Services (CCS) that are supported on the Compact Switch.
- The Advanced Routing section describes the advanced routing features that are available on the Compact Switch and explains how to provision the feature.
- The AIN section describes the call processing subsystem architecture flexibility to support both internal (on-board) and external features using the AIN message and parameter set.
- The 9-1-1 Service section describes the Enhanced 9-1-1 Service capabilities that are supported.
- The Communications Assistance Law Enforcement Agency (CALEA) section describes the functional capabilities supported to facilitate criminal communications surveillance in the digital age.
- The Implementing SIP section describes the Session Initiation Protocol and its relationship to VoIP.
- The Packetized VoIP section describes the IP call types supported by various protocols, as well as CODEC negotiation information.

Using the System Documentation CD-ROM

Like many documentation sets today, the switch documentation set is provided a CD-ROM sent to you, the customer, with the switch. When inserted into a drive, the CD-ROM will automatically open to the Main Menu page. As directed by the *Read Me First* page included in your packed materials, you can select any manual listed from the Main Menu page on the CD-ROM. Once you have selected a manual, you can move to different areas of a manual using the bookmarks on the left side of the page. Clicking on the + sign in front of a topic will expand it; clicking on the – sign will minimize it. Table of Contents information in blue font will also move you to that identified topic. Clicking on the cover page or the title of the third page (Manual Contents) will return you to the Main Menu page.

System Documentation

One set of system documentation on CD-ROM (255-400-007) is sent to you with each switch you purchase. Each set consists of, at a minimum, these core manuals and release notes:

System Documentation Set

Product	Part Number	Product Description
Lucent Gateway Platform Operations Manual	255-400-000	Contains the platform provisioning procedures.
Lucent Gateway Platform Maintenance and Troubleshooting Guide	255-400-001	Contains the procedures for adding and upgrading modules, and maintaining and troubleshooting switch alarms.
Lucent Gateway Platform TL1 Commands Reference Guide	255-400-002	Description of all the TL1 commands needed to provision the platform, functional entities and services.
Lucent Gateway Platform Planning and Engineering Guide	255-400-003	Contains the information necessary for designing an installation site including: hardware specifications; cabling schematics; and cabling, floor plan, environmental and power requirements.
Lucent Gateway Platform Product Overview Guide	255-400-004	Contains descriptions of the base switching platform, functional entities (Network Controller, Signaling Gateway; Network Gateway, Compact Switch) and supported provisioning methods.

Product	Part Number	Product Description
Lucent Gateway Platform System Release Notes	255-400-006	Contains new features and feature enhancements, new and modified TL1 commands, hardware and software limitations and other important release-specific information not available elsewhere.
Lucent Gateway Platform Feature Packages Guide	255-400-012	Contains detailed feature package descriptions.
Lucent Gateway Platform Billing and Traffic Collection (BTC) Guide	255-400-403	Contains installation, upgrade, and applications procedures.
Lucent Gateway Platform BTC Release Notes	255-400-404	Contains software features and release-specific information that is not available elsewhere.
Lucent Gateway Platform System Documentation CD-ROM	255-400-007	Contains all of the manuals and the release notes listed above in Adobe Acrobat PDF format.

Printed versions of these documents can be ordered individually, using the part numbers listed.

PlexView Documentation

Other manuals and release notes, which are available upon purchase of additional software include:

PlexView Documentation Set

Product Documentation	Part Number	Product Description
Lucent Gateway Platform Element Management System (EMS) User Guide	255-400-400	EMS provisioning reference guide.
Lucent Gateway Platform Element Management System (EMS) Installation Guide	255-400-401	Installing the EMS software on a Sun workstation.
Lucent Gateway Platform EMS Software Release Notes	255-400-402	Contains software features and release-specific information that is not available elsewhere.

Product Documentation	Part Number	Product Description
Lucent Gateway Platform Billing Traffic Collection (BTC) Guide	255-400-403	Contains installation, upgrade, and applications procedures.
Lucent Gateway Platform BTC Release Notes	255-400-404	Contains software features and release-specific information that is not available elsewhere.
Lucent Gateway Platform EMS/BTC Documentation CD-ROM	255-400-406	Contains: • EMS User Guide 255-400-400 • EMS Installation Guide 255-400-401 • EMS Software Release Notes 255-400-402 • BTC Guide 255-400-403 • BTC Release Notes 255-400-404
Lucent Gateway Platform Advanced Reporting System (ARS) User's Guide	255-400-200	ARS provisioning reference guide.
Lucent Gateway Platform Advanced Reporting System (ARS) Installation Guide	255-400-201	Installation and upgrade information for the ARS software.
Lucent Gateway Platform Advanced Reporting System (ARS) with Advanced Traffic Collector (ATC) Installation Guide	255-400-202	Provides installation and upgrade information for the ATC and ARS in sequential order.
Lucent Gateway Platform ARS Software Release Notes	255-400-203	ARS software features and release-specific information that is not available elsewhere
Lucent Gateway Platform ARS Documentation CD-ROM	255-400-204	Contains: • ARS User Guide 255-400-200 • ARS Installation Guide 255-400-201 • ARS with ATC Installation Guide 255-400-202 • ARS Software Release Notes 255-400-203

Notes:

Class 5 Software Feature Package

This document contains the following sections:

Scope

This section provides details about the Class 5 originating and terminating services. CLASS 5 services include Custom Local Area Signaling Services (CLASS) and Custom Calling Services (CCS).

Note: With a few exceptions, CLASS 5 features are used only by CAS/GR-303 subscribers and are *not* applicable to subscribers served by PRI terminations. If there is an exception, it is noted in that specific feature section.

The words, service and feature, are used interchangeably when referring to CLASS 5 features or services.

Local Processing of Services

Local service logic allows processing of all services except Local Number Portability (LNP), Advanced Intelligent Network (AIN) Toll Free and Intelligent Network (IN) Toll

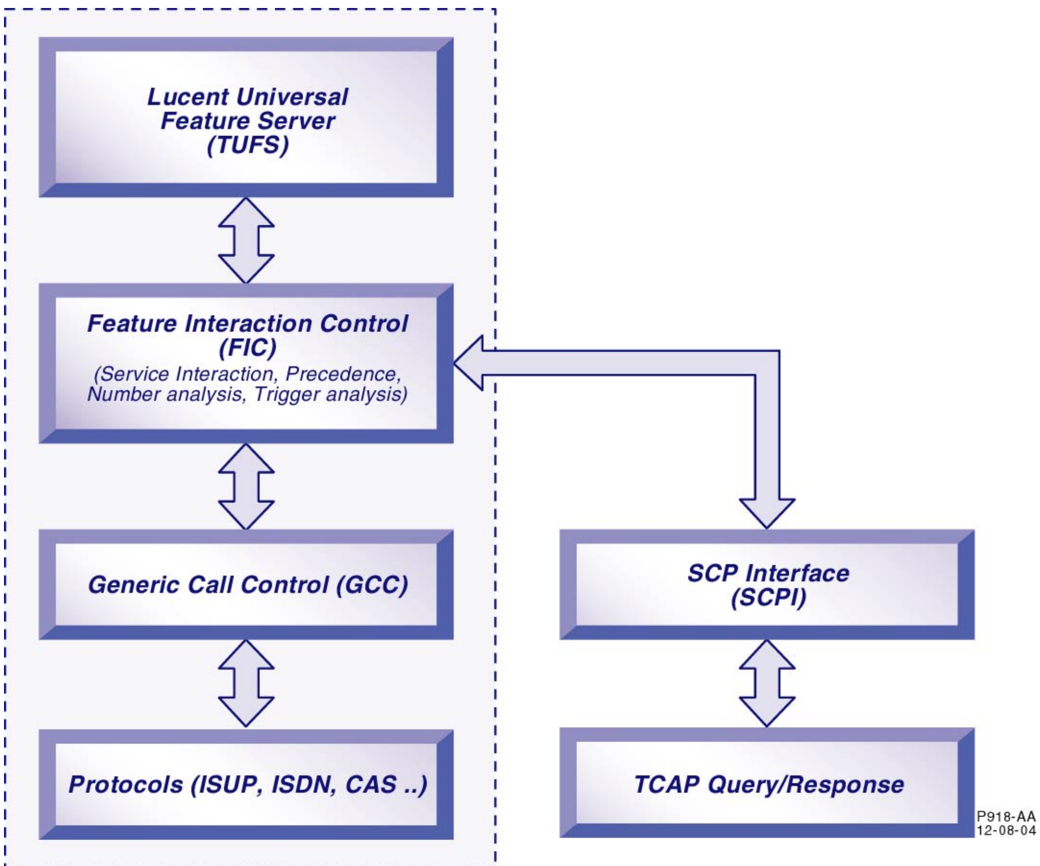
Free, which require Service Control Point (SCP) lookups. LNP and Toll Free are not CLASS services. Calling Name (CNAM) may also be provisioned off-switch at an SCP.

All CLASS services can be processed on the switch reducing the cost of SCP dips. On-switch processing means that the service logic runs on the switch and that relevant data is stored on the switch. SCP-based processing means that the service logic and the relevant data are stored and executed from the SCP. SS7 TCAP queries are directed towards configured SCP routes.

CLASS 5 Service Logic Environment

On-switch processing of services uses a Feature Server (FS). The FS is configured, by default, for all CLASS services. Data associated with the service, such as Anonymous Call Reject, are stored on the switch as part of a subscriber record. No TCAP queries are necessary and no SCP support is required. Figure 1 shows the architecture for service processing.

Figure 1. Feature Processing (Local and Remote)



Supported CLASS 5 Services

Table 1 lists the CLASS 5 services that are supported in this version, along with the minimum software version that is required for each of the individual services.

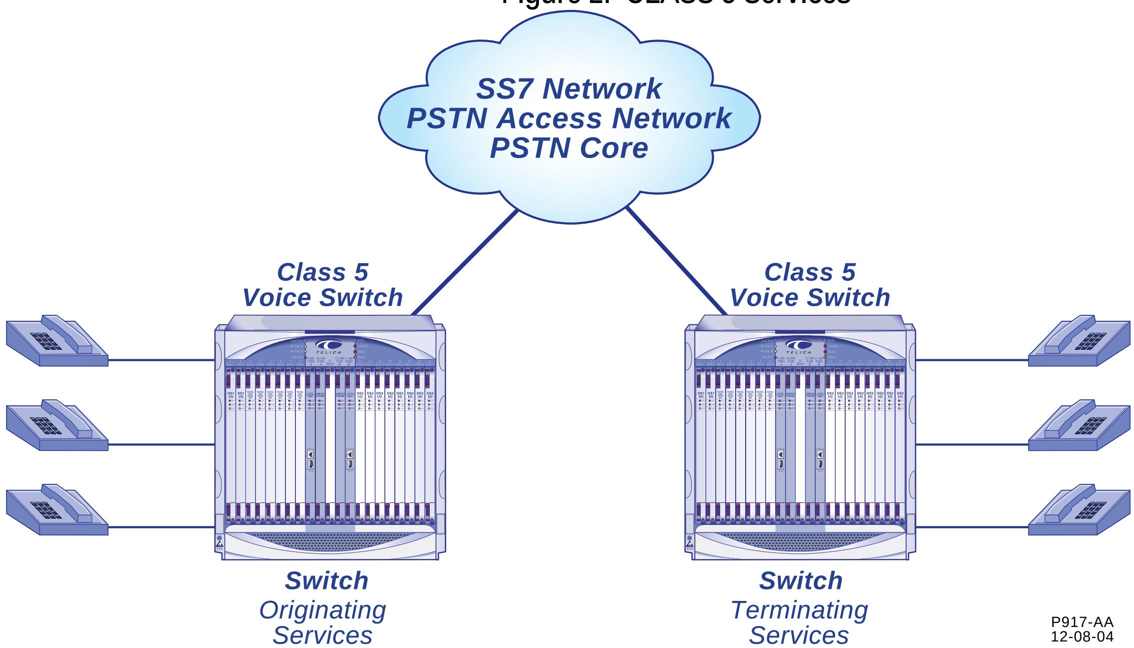
Table 1 - Supported CLASS 5 Services

Service Name	Minimum Software Version
AIN Toll Free	3.8
Anonymous Call Rejection (ACR)	3.7
Automatic Call Back (ACB)	3.8
Automatic Recall (AR)	3.8
Authorization Codes	3.7
BLV (Busy Line Verification)/Busy Line Interruption	3.11
Call Forwarding Busy Line (CFBL)	3.7
Call Forwarding Don’t Answer (CFDA)	3.7
Call Forwarding Variable (CFV)	3.7
Calling Identity Delivery Suppression (CIDS)	3.7
Calling Identity Delivery on Call Waiting (CIDCW)	3.7
Calling Line DN Verification	3.9
Calling Name Blocking (CNAB)	3.7
Calling Name Delivery (CNAM)	3.7
Calling Number Delivery (CND)	3.7
Calling Number Delivery Blocking (CNDB)	3.7
Call Waiting (CWT or CW)	3.7
Cancel Call Waiting (CWC)	3.7
Code Restriction/Diversion Prohibited Code List (CRDPCL)	3.7
Customer Originated Trace (COT)	3.7
Home Intercom (HI)/Barn Phone	3.11
Hot Line (HL)/Automatic Line	3.11
Local Number Portability	3.7
Multi-Way Calling (MWC) or Add On Transfer Conference (AOTC)	3.8
Remote Access to call Forwarding (RAF)	3.8
Residence Distinctive Alerting Service (RDAS) or Distinctive Ring	3.7
Selective Call Acceptance (SCA)	3.10
Selective Call Forwarding (SCF)	3.10
Selective Call Rejection (SCR)	3.10
Speed Calling using one digit (SC1) or SPEED 8	3.7
Speed Calling using two digits (SC2) or SPEED 32	3.7

Service Name	Minimum Software Version
Toll Free (IN)	3.7
Three-Way Calling (TWC)	3.7
Voice Mail System (VMS) – Intra-switch	3.7
Voice Mail System (VMS) – Inter-switch	3.8.3
Warm Line (WL)	3.11

Figure 2 shows a high-level view of originating and terminating switches through which the services interact.

Figure 2. CLASS 5 Services



Subscriber Services and Service Access Codes

You can use either TL1 or the PlexView Element Management System (EMS) to provision subscriber services. Refer to the Subscriber Management Commands section in the *TL1 Commands Reference Guide* or the Provision Subscribers section in the *PlexView Element Management System (EMS) User Guide* for detailed information.

Most services have provisioning requirements under Subscriber Management Commands of TL1 or Subscriber Services section in EMS. Many services have associated service access codes that may also be called Vertical Service Codes (VSCs) for activation and deactivation. Most of the service access codes listed are *industry standard*, but use of activation and deactivation codes may vary from one service provider to another. Codes for the switch can be changed by using the `ED-SERVICE-ACCESSCODE` TL1 command or by selecting the `Service Access Code` screen and changing the code in EMS. If none is provided, you may select a code as long as it does not interfere with another code. Refer to Table 2 for the list of codes and the default activation and deactivation codes.

Table 2. Service Access Codes

Service Access Code	Activation	Deactivation
ACB (Automatic Callback)	*66	*86
ACR (Anonymous Call Rejection)	*77	*87
AIN_TOLLFREE (Advanced Intelligent Network Toll Free)	0 = 0perator	
AR (Automatic Recall)	*69	*89
AUTH or **AUTHCODE** (Authorization Code)		
BLV (Busy Line Verification or Busy Line Interruption)	None	None
CFB or **CFBL** (Call Forward Busy or Call Forward Busy Line)	*90	*91
CFNA or **CFDA** (Call Forward No Answer or Call Forward Don't Answer)	*92	*93
CFV (Call Forward Variable)	*72	*73
CID (Caller ID or Calling Identity Delivery)	*61	
CIDCW (Calling Identity Delivery on Call Waiting)	See Note 1.	
CIS (Calling Identity Suppression)	*65	
Calling Line DN Verification	See Note 3.	See Note 3.
CNAB (Calling Name Blocking)	*63	None
CNAM (Calling Name Delivery)	See Note 1.	
CND (Calling Number Delivery)		
CNDB (Calling Number Delivery Blocking	*67	None
COT (Customer Orginated Trace)	*57	
Screen Block or **CRDPCL** (Code Restriction/Diversion Prohibited Code List)	See Note 1.	
CCW or **CWC** (Cancel Call Waiting or Call Waiting Cancel)	*70	
CW or **CWT** (Call Waiting)	See Note 1.	
LNP (Local Number Portability)	See Note 2.	
HI (Home Intercom or Barn Phone)	None	None
HL (Hot Line or Automatic Line)	None	None
MWC or **AOTC** (Multi-Way Calling or Add On Transfer Conference**)**	Switch-hook or Flash button.	
RAF (Remote Access to call Forwarding), also known as RMT		
RDAS or **DR** (Residence Distinctive Alerting Service or Distinctive Ring)		
RLT (Release Link Trunk)		

Service Access Code	Activation	Deactivation
SC1 or **SPEED_8** (Speed Calling 1 or Speed 8)	*74	
SC2 or **SPEED_32** (Speed Calling 2 or Speed 32)	*75	
SCA (Selective Call Acceptance)	*64	*84
SCF (Selective Call Forwarding)	*62	*83
SCR (Selective Call Rejection)	*60	*80
TOLLFREE (Toll Free, IN version as opposed to AIN)	0 = Operator	
TWC (Three-Way Calling)	Switch-hook or Flash button	
Voicemail or **VMS** (Voice Mail System)	See Note 1.	
WL (Warm Line)	None	None
Note 1: Subscription Service ***Note 2:*** Activated by service provider. ***Note 3:*** Must be provisioned by service provider.		

AIN Toll Free

AIN Toll Free is a switch-wide service similar to the IN Toll Free service but it differs from IN Toll Free in the handling of messaging. AIN Toll Free is an Advanced Intelligent Network (AIN)-based service, as opposed to an Intelligent Network (IN)-based service. AIN Toll Free provisioning is part of SLHR SCP Services.

Provisioning the switch using TL1

To provision AIN Toll Free with DPC-SSN routing, you must establish an SCCP (Signaling Connection Control Part) route to an SSN (sub-system number) on a remote point code using the `ED-SCCP-SSN` command. In the following example the point code to receive the query is 3-3-3 and subsystem number of the providing service to route to is 233.

```
Command Example: ENT-SCCP-SSN::3-3-3-233;
```

You must also establish a service logic host route to a Service Control Point (SCP) using the `ENT-SLHR-SCP` command as shown in the following example. The *slhrid* in the example is 1; the *routetype* is RTE_SSN indicating routing directly to an SCP using a point code;and an SSN (Sub-Service Number); the *dpc* is 3-3-3; the *svcreqssn* and *svcprovssn*, the SSN for the service on the switch and the SCP, is 233.

```
Command Example: ENT-SLHR-SCP::1:DESC= AINTOLLFREE_SERVICE,
                 ROUTETYPE=RTE_SSN,DPC=3-3-3,SVCREQSSN=233,
                 SVCPROVSSN=233;
```

Finally the feature must be enabled using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::TOLLFREE:::
                 DESC=AINTOLLFREE_SERVICE,SLHRID=1:IS;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide*.

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *SS7*, double-click *Intelligent Network Services*. Then right-click *SCCP SSN* and select *Add* to add a route to an SSN or right-click an existing route and select *Modify*. Make entries or changes and select *OK* or *Apply*.

Double click *SLHR SCP Services*. Then right-click *SLHR SCP Services* to add a SCP or right-click an existing SCP to modify. Make any entries or required changes. Select *OK* or *Apply*.

Double-click *System Services* and then *Service Access Code*. Right-click *AINTOLLFREE_* SERVICES and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Anonymous Call Rejection (ACR)

Anonymous Call Rejection allows a customer to reject incoming calls marked private, restricted, anonymous, blocked, or not available. Such marked calls prevent name and number information from passing to the called party. ACR, if used, is billed on a per-usage basis or on a flat rate.

AMA records are generated for each activation or deactivation of ACR for subscribers with usage-sensitive ACR.

How to Use

Lift the receiver and listen for dial tone.

Dial *77 to activate ACR. The following confirmation announcement will be played:

```
“Your anonymous call rejection service is on.”
```

Dial *87 to deactivate ACR. The following confirmation announcement will be played:

```
“Your anonymous call rejection service is OFF.”
```

When ACR is active, the called customer does not receive an alert for a call that is rejected. The call is routed to a denial announcement and subsequently terminated.

Provisioning the switch using TL1

To provision a user for ACR, you must activate ACR and you must provision the codes for user activation and deactivation using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::ACR:::
                 ACTIVATIONCODE=77,DEACTIVATIONCODE=87:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ENT-SUBSCRIBER:: 5084062001:::ACR=US;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *ACR_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *ACR*, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

ACB takes precedence over ACR. If a customer with both ACR and Automatic Callback/Recall (ACB/AR) invokes ACB to attempt a call to a number that is marked private, the switch will not block ring back associated with that ACB/AR request.

For incoming calls to lines with both ACR and a Call Forwarding (CF) feature (except CFDA) the switch checks the presentation status and proceeds as follows:

- If presentation is restricted, the call is routed to the ACR denial announcement.
- If presentation is allowed, the call continues to be processed according to requirements specified for the particular CF feature.

An incoming call forwarded to an ACR customer by a CF feature is checked for presentation status of the originating party's DN. The switch then proceeds as follows:

- If presentation of that DN is restricted, the call is routed to the ACR denial announcement.

- If presentation is allowed, the switch attempts to complete the call to the ACR customer’s line.

For incoming calls to a line with both ACR and CFDA active, the switch checks the presentation status and proceeds as follows:

- If presentation is restricted, the call is routed to the ACR denial announcement.
- If presentation is allowed, the switch offers the call to the customer and invokes CFNA, if applicable.

A customer with CWT and ACR active does not receive CWT tones for incoming calls with presentation restricted. The calling party is provided with busy treatment and the ACR customer receives no alerting for this call.

If a customer has both ACR and RDAS active on a line, the switch checks incoming calls against the RDAS screening list and proceeds as follows:

- If there is a match, the switch does not provide ACR processing for that call; call processing continues according to other features that are active on that customer’s line.
- If there is no match, ACR processing is provided for the call.

For lines with both ACR and SCF active, the switch shall check incoming calls against the customer's SCF screening list, and proceed as follows:

- If there is a match, the call shall be routed to the remote DN specified for the customer's SCF service and there is no ACR processing for the call.
- If there is no match, ACR processing shall be applied to the call.

An incoming call forwarded to an ACR customer by SCF shall be checked for presentation status of the originating party’s DN; the switch shall proceed as follows:

- If presentation of that DN is restricted, the call shall be routed to the ACR denial announcement.
- If presentation is allowed, the switch shall attempt to complete the call to the ACR customer's line.

If incoming calls are to a line with both ACR and SCR active, the switch shall check incoming calls against the customer's SCR screening list, and proceed as follows:

- If there is a match, the call shall be routed to the SCR denial announcement.
- If there is no match, the switch shall apply ACR processing to the call.

When a TWC customer invokes TWC to attempt to include a third party that has ACR, the ACR customer's switch shall check the presentation status of the TWC customer and proceed as follows:

- If presentation is restricted, the call shall be routed to the ACR denial announcement.

- If presentation is allowed, the three-way call setup shall continue

Automatic Call Back (ACB)

Automatic Call Back (ACB) is a service that allows subscribers to redial the last call initiated from their line. If the number is busy, the called line is automatically monitored until the line becomes idle, or until the ACB monitoring period expires (usually 30 minutes). The customer can deactivate ACB monitoring before the callback is complete by dialing the ACB deactivation code. ACB, if used, is billed on a per-usage basis or on a flat rate.

How to Use

Lift the receiver and listen for dial tone.

Dial *66 to activate and the last number called will be redialed.

Dial *86 to deactivate ACB.

Provisioning the switch using TL1

If ACB will be inter-switch, your must assign an SLHR ID signed using the `ED-ACBAR-SYS` command. The SLHRID is initially provisioned with the `ENT-SLHR-SCP` command.

```
Command Example: ED-ACBAR-SYS::SLHRID=5;
```

To enable ACB, you use the `ED-SERVICE-ACCESSCODE` command. The *serviceId* is ACB and *pst* is IS (In-Service).

```
Command Example: ED-SERVICE-ACCESSCODE::ACB:::
                 ACTIVATIONCODE=66,DEACTIVATIONCODE=86:IS;
```

To provision a user for ACB, you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ED-SUBSCRIBER::5084061001:::ACB=US;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *SS7*, double-click *Intelligent Network Services*. Then right-click *Automatic Call-Back./Redial System* and select *Add* or *Modify*. Select the *SLHRID* and then make any required changes. Select *OK* or *Apply*. If the SLHRID has not yet been provisioned, first select *SLHR SCP Services* and right-click, select *Add* to add new SLHRID.

Double-click *System Services* and then *Service Access Code*. Right-click *ACB_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *ACB*, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

If ACB and Call Forwarding Busy Line (CFBL) are active on the same line, ACB takes precedence as follows:

- If the ACB customer's line is idle.
- If the ACB customer's line is busy, ACB special ringing and call setup shall not be forwarded to a remote station if the ACB customer has CFBL active.

ACB and Call Forwarding Variable (CFV) can be assigned to the same line and ACB and Selective Call Forwarding (SCF) can be assigned to the same line.

If ACB and CFV or SCF are active on the same line, then ACB takes precedence as follows:

- If ACB is activated from a line that has SCF or CFV active, the switch shall apply ACB special ringing to the customer's line instead of forwarding the special ringing to a remote station.
- ACB attempts toward a line that has CFV active or that has SCF active, with the calling party's DN on the called party's SCF screening list, result in the originating switch providing short-term denial to the ACB customer. The originating switch receives information on the status of SCF and CFV in the Initial Response Message.

If ACB and Calling Number Delivery Blocking (CNDB) or Calling Name Delivery Blocking (CNAB) or Calling Identity Delivery Suppression (CID/CICS) are active on the same line, then ACB takes precedence as follows:

- A customer may activate a per-call presentation feature for an ACB request by dialing the per-call presentation feature access code before placing the originating call, before activating ACB toward a DN or before reactivating ACB.
- For a given ACB request, the originating switch honors the most recent activation of a per-call presentation feature.

- If a customer does not activate a per-call presentation feature for an ACB request, then the calling DN and name permanent presentation statuses are used to determine the calling party's DN and name presentation statuses for ACB call setup.

Automatic Recall (AR)

Automatic Recall (AR) provides you with the telephone number of the last caller. You will hear the number of the last caller. Press the 1 to return the call. AR, if used, is billed on a per-usage basis or on a flat rate.

How to Use

Lift the receiver and listen for dial tone.

Dial *69 to activate AR.

Dial *89 to deactivate AR.

Provisioning the switch using TL1

If AR will be inter-switch, you must assign an SLHRID using the `ED-ACBAR-SYS` command.

```
Command Example: ED-ACBAR-SYS::SLHID=5;
```

To enable AR, you use the `ED-SERVICE-ACCESSCODE` command. The *serviceId* is AR and *pst* is IS (In-Service). The activation and deactivation codes are the defaults of 69 and 89.

```
Command Example: ED-SERVICE-ACCESSCODE::AR:::SLHRID=5:IS;
```

To provision a user for AR, you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ED-SUBSCRIBER::5084061001:::AR=US;
```

Note: Both one and two stage feature operation is supported (ACTLVL). One stage operation immediately redials the number when *69 is dialed; two stage operation announces the number (PLAYBACK) and then redials the number.

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *SS7*, double-click *Intelligent Network Services*. Then right-click *Automatic Call-Back./Redial System* and select *Add* or *Modify*. Select the *SLHRID* and then make any required changes. Select *OK* or *Apply*. If the SLHRID has not yet been provisioned, first select *SLHR SCP Services* and right-click, select *Add* to add new SLHRID.

Double-click *System Services* and then *Service Access Code*. Right-click *AR_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Note: Both one and two stage feature operation is supported (ACTLVL). One stage operation immediately redials the number when *69 is dialed; two stage operation announces the number (PLAYBACK) and then redials the number.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *AR*, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *PlexView Element Management System User Guide*

Interactions With Other Features

AR takes precedence if the customer also has ACR, CFBL, CFDA, CFV, SCF, CWT or CIDCW.

Authorization Codes (AUTH)

Authorization Codes are validated against an on-switch database before a call is completed. If a code is not valid, the caller is prompted for a valid code. If a valid code is still not entered, the call is released. Authorization Codes can be set per subscriber for all calls or toll calls only.

The switch screens each authorization code input for validity against valid codes assigned to that interface, facility or Directory Number. Authorization Codes range from one to sixteen (16) characters and may include an * character, but not as the initial character.

Account Codes may be paired with Authorization Codes (in place of PINs).

PINS are used in conjunction with Authorization Codes. PIN codes range from 4 to 8 digits, including the # and * characters. The switch supports either a single or two-step code entry process per facility group.

Provisioning the switch using TL1

To enable authorization codes, you use the `ED-SERVICE-ACCESSCODE` command. The *serviceId* is AUTHCODE and *pst* is IS (In-Service).

```
Command Example: ED-SERVICE-ACCESSCODE::AUTHCODE::::IS;
```

You use the `ENT-LIST-AUTHCODE` command to establish an indexed list containing an authorization code, a personal identification number (PIN) code, or an authorization code with an associated PIN code.

```
Command Example: ENT-LIST-AUTHCODE::telica-1:::
                 AUTHCODE=1234,PINCODE=56789;

Command Example: ENT-LIST-AUTHCODE::auth-1:::AUTHCODE=1234;

Command Example: ENT-LIST-AUTHCODE::pin-1:::PINCODE=56789;

Command Example: ENT-LIST-AUTHCODE::maxauthpin-1:::
                 AUTHCODE=1010101,PINCODE=12345678;
```

You then use the `ED-SUB-IF` to provision which calls require an authorization code, how the codes operate on the interface, the name of the list and the code length.

```
Command Example: ED-SUB-IF::5-ISDN:::AUTHTYPE=ALL,
                 AUTHMODE=ONEPIN,AUTHLIST=maxauthpin;

Command Example: ED-SUB-IF::1000-CAS:::AUTHTYPE=ALL,
                 AUTHMODE=ONLYAUTH,AUTHLIST=auth;

Command Example: ED-SUB-IF::2000-CAS:::AUTHTYPE=ALL,
                 AUTHMODE=ONLYPIN,AUTHLIST=pin;

Command Example: ED-SUB-IF::3000-CAS:::AUTHTYPE=ALL,
                 AUTHMODE=ONEPIN,AUTHLIST=telica;

Command Example: ED-SUB-IF::1-MGCP:::AUTHTYPE=TOLL,
                 AUTHMODE=ONLYACNT,ACNTCODELEN=7;

Command Example: ED-SUB-IF::1-GR303:::AUTHTYPE=ALL,
                 AUTHMODE=ACNTAUTH,AUTHLIST=auth,
                 ACNTCODELEN=7;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *Auth_Service* and select *Modify*. Provision *IS.* Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *List Auth Code*. To add a new list, right-click *List Auth Code* and select *Add*. To modify a list, move cursor over an entry, right-click and select *Modify*. Make entries or changes and select *Ok* or *Apply*.

Double-click *MGCP,* then *Line Group* and *Subscriber Interface*. Right-click *Subscriber Interface* and select *Add* to add an interface or right-click an interface and select *Modify* to change an interface. Make entries or changes and select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Busy Line Verification/Busy Line Interrupt (BLV)

The Busy Line Verification (BLV) feature allows an operator to determine if a specific subscriber number is off hook and whether voice conversation is actually taking place. The Busy Line Interrupt (BLI) feature provides the additional capability of interrupting the voice conversation to relay a message to the subscriber. An E&M trunk interface using MF signaling is provided for interconnection to the operator service position. The operator signals the target number to the Compact Switch. Before performing the operation, the Compact Switch checks an Compact Switch-based database to determine if the requested action is allowed on the specific subscriber line. The various operational, call control and signaling conditions that may be encountered in attempting to complete the verification request are handled as described in Table A of GR-531-Core Issue 1, June 2000.

Notes:

- BLV Class of Service is used to provision both BLV and BLI features, since they both look the same to the Compact Switch.
- BLV Class of Service can ONLY be assigned to a CAS Trunk Group.
- Service applies to CAS and GR-303 subscriber lines ONLY. It is NOT applicable to PRI interfaces or MGCP endpoints (BLV denial parameter is initially set to Yes by the system for MGCP endpoints).
- BLV/BLI connections are NOT preserved on a SP/SF failover

Provisioning the switch using TL1

You must first provision the E&M trunk interface that is using MF signaling to provide the interconnection between the Compact Switch and the operator service position.

`Command Example: ED-TRKGRP::1600:::CAS:CLASSOFSERVICE=BLV;`

You must then activate BLV by putting it into service (IS) using the `ED-SERVICE-ACCESSCODE` command.

`Command Example: ED-SERVICE-ACCESSCODE::BLV::::IS;`

The default value for the BLV denial attribute is No, which means that BLV/BLI is allowed. When required, the BLV denial parameter should be set to YES to protect data lines, fax lines or other lines from interruption.

```
Command Example: ENT-SUBSCRIBER::5084062001:::DENYBLV=YES;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *IMT* and then *CAS Trunk Group*. Right-click the desired trunk group and select *Modify*. Select the *Routing Params* tab. Using the pull-down menu for *Class of Service*, select *BLV*. Select *Ok* or *Apply*

Double-click *System Services* and then *Service Access Code*. Right-click *BLV_SERVICE* and select *Modify*. Provision *IS*. Select *Ok* or *Apply*.

The default value for the BLV denial attribute is No, which means that BLV/BLI is allowed. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move the cursor over the subscriber, right-click and select *Modify*. Select the *Other Details* tab and then using the pull-down menu for *BLV*, set the *BLV denial attribute* to *Yes*. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Call Forwarding

Call Forwarding Busy Line (CFBL)

Call Forwarding Busy Line allows a customer to forward calls to another location when the customer’s phone line is busy.

Call forwarding is to one single number that the subscriber specifies at subscription time and is provisioned into the switch at service activation.

You can use different forwarding numbers for CFBL and CFDA. The Service provider only or both the service provider and the subscriber can control the activation or deactivation of the service. Call forwarding, if used, is billed on a per-usage basis or on a flat rate.

How to Use CFBL

Lift the receiver and listen for dial tone.

Dial *90 to activate.

Dial *91 to deactivate.

Provisioning the switch using TL1

To provision a user for CFBL, you must activate CFBL and provision the codes for user activation and deactivation using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::CFBL:::
                 ACTIVATIONCODE=90,DEACTIVATIONCODE=91:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ENT-SUBSCRIBER:: 5084062001:::CFBL=US;
```

For versions earlier than 3.10, the following two commands are used.

```
Command Example: ED-SERVICE-ACCESSCODE::CFBL::::IS;
```

```
Command Example: ED-SUBSCRIBER::5084061001:::CFBL=UNRES,
                 CFNB=17812621234;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *CFB_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *ACB*, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

CFBL can be used if CFV is not activated. CFV takes take precedence.

Call Forwarding Don’t Answer (CFDA)

Call Forwarding Don’t Answer or Call Forwarding No Answer (CFNA) allows a customer to forward calls to another location when the customer does not answer the incoming call. The number of rings before a call is forwarded is provisionable.

Call forwarding is to one single number that the subscriber specifies at subscription time and is provisioned into the switch at service activation.

You can use different forwarding numbers for CFDA and CFBL. The Service provider only or both the service provider and the subscriber can control the activation or deactivation of the service. Call forwarding, if used, is billed on a per-usage basis or on a flat rate.

How to Use CFDA

Lift the receiver and listen for dial tone.

Dial *92 to activate.

Dial *93 to deactivate.

Provisioning the switch using TL1

To provision a user for CFDA, CFDA must be activated and the codes for user activation and deactivation must be provisioned using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::CFDA:::
                 ACTIVATIONCODE=92,DEACTIVATIONCODE=93:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command, entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ENT-SUBSCRIBER::5084061001:::CFDA=US;
```

For versions earlier than 3.10, the following two commands are used.

```
Command Example: ED-SERVICE-ACCESSCODE::CFDA::::IS;
```

```
Command Example: ED-SUBSCRIBER::5084061001:::CFDA=UNRES,
                 CFND=19082226666,CFINT=3;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *CFNA_Service* and select *Modify*. Make any changes and provision *IS.* Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *ACB*, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

CFDA can be used if CFV is not activated. CFV takes take precedence.

Call Forwarding Variable (CFV)

Call Forwarding Variable allows a subscriber to direct incoming calls to his/her phone to another phone number by activating CFV and entering the phone number.

The switch screens the dialed number to determine if it is valid. N11, 950-WXXX, operator calls, blocked numbers, and 800 numbers hosted by the switch are invalid. If the number is invalid, the call is routed to an announcement. CFV, if used, is billed on a per-usage basis or on a flat rate.

An AMA record is activated when CFV is activated and a deactivation AMA record is generated when CFV is deactivated.

How to Use

Lift the receiver and listen for dial tone.

Dial *72 to activate. Dial the number to which you want the calls forwarded. If the call is answered, after hanging up, you will hear a confirmation tone indicating forwarding is activated. If the line is busy or no one answers, hang up, and immediately press *72 again and redial. You should receive a confirmation tone that call forwarding is now activated.

The switch can be provisioned whether to establish a courtesy call to the remote DN, and whether an answer is required to activate he service. The values are: Answer Required with Confirmation indication, Answer Required with No Confirmation indication, No Answer Required, and do Not Establish Courtesy Call.

When you have the CFV feature activated and receive a call, a ring reminder is applied to the base station when idle, which indicates that a call has been received and forwarded.

An off-hook while CFV is activated is regarded as an origination, not an answer. The subscriber cannot answer a call that has been forwarded.

Dial *73 to deactivate. A confirmation tone is then returned followed by dial tone.

Provisioning the switch using TL1

To provision a user for CFV, you must activate and provision the codes for user activation and deactivation using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::CFV:::
                 ACTIVATIONCODE=72,DEACTIVATIONCODE=73:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ED-SUBSCRIBER::5084061001:::CFV=US;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *CFV_Service* and select *Modify*. Make any changes and provision *IS.* Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *CFV*, select the type of service. Select *Ok* or *Apply.*

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

When CFV is deactivated, CFBL can be used, when CFV is activated, CFBL cannot be used.

If a customer has CFV activated, the call is forwarded whether the base station is idle or in use and CWT service is ineffective at the base station. A customer can disable CWT while CFV is activated.

Activation of CFV is not permitted to code restricted numbers. The system gives special error treatment in response to this activation request.

CFV can be used with speed calling codes to forward calls to another DN by dialing a call forwarding access code and then the speed calling code of the DN to which the call is forwarded.

Calling Identity Delivery

Calling Identity Delivery (CID)

CID allows a customer to force the delivery of their name and number to the called party by dialing a delivery feature access code before dialing a complete telephone number. CID can be enabled or disabled by the service provider.

How to Use

Lift the receiver and listen for dial tone.

Dial *61 to activate CID. Wait for recall dial tone and then dial the called party.

Provisioning the switch using TL1

To provision a user for CID, you must activate CID/CIS and the codes for user activation using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::CID:::
                 ACTIVATIONCODE=61:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ED-SUBSCRIBER::5084061001:::CIDS=US;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *CID_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *CIDS*, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Calling Identity Delivery On Call Waiting (CIDCW)

CIDCW allows a customer to see who is trying to call while the customer has an established call. Calling Identity Delivery On Call Waiting works in conjunction with Caller ID (CND and CNAM). This is a subscription-based call termination service available to CAS customers. CIDCW is enabled or disabled by the service provided.

Provisioning the switch using TL1

To provision a user for CIDCW, you must activate using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::CIDCW::::IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is standard calling identity delivery on call waiting.

```
Command Example: ED-SUBSCRIBER::5084061001:::CIDCW=STND;
```

The CIDCW service will not be activated until either or both of the following services are assigned to the subscriber:

```
Command Example: ED-SUBSCRIBER::5084061001:::CND=US;
```

```
Command Example: ED-SUBSCRIBER::5084061001:::CNAM=US;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *CIDCW_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *CIDCW*, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

A CIDCW customer with ACR active is not alerted to incoming calls that have presentation of calling party number restricted. No CID data is transmitted for such calls, and the calling party will be routed to the ACR denial announcement.

CFV takes precedence over CIDCW and calls received for the customer while CFV is active are forwarded to the remote station.

The switch allows a customer to override CIDCW service on a particular outgoing call by using the CCW feature. When CCW is activated for a particular call, calls received for the customer during the course of that call receive a busy treatment. CIDCW alerting and off-hook transmission are not provided during that call.

If a line has both CIDCW and CFBL active, and does not have a party in a waited or held state, the CIDCW service takes precedence over CFBL for any calls that are received while the customer is off-hook and engaged in a call.

If a customer with CIDCW and CFBL active is off-hook but not in a stable call state (that is, the customer is dialing) when a new call is received for that customer, CFBL treatment will apply to that call (call is forwarded).

If a customer with CIDCW and CFBL active is in a talk state and has a party in a waited or held state, any additional calls received for that customer receives CFBL treatment (the calls are forwarded).

If a customer with both CIDCW and CFBL is engaged in a call and has activated CCW for that call, then CIDCW does not apply for calls that are received for that customer and CFBL is invoked (the call is forwarded).

If a customer has CIDCW and CFDA enabled, CFDA service is applied to incoming calls that received CIDCW treatment, if the call remains unanswered for the interval specified for the CFDA service.

Calling Identity Suppression (CIS)

CIS allows a customer to override (once) the delivery of their name and number to the called party by dialing a suppression feature access code before dialing a complete telephone number to temporarily override the presentation statuses of both the caller's DN and the calling name. CID or CIS can be enabled or disabled by the service provider.

How to Use

Lift the receiver and listen for dial tone.

Dial *65 to activate CIS. Wait for recall dial tone and then dial the called party.

Provisioning the switch using TL1

To provision a user for CIS, you must activate CIS and the codes for user activation using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::CIS:::
                 ACTIVATIONCODE=65:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ED-SUBSCRIBER::5084061001:::CIDS=US,;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *CIS_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *CIDS*, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Calling Line DN Verification

Allows the service provider to assign a number on the switch to be called by telephone installers to verify that the phone they just installed is connected to the number they think is should be. When the installer calls the number, the switch reads back the number the phone is connected to.

Provisioning the switch using TL1

Unlike other features on the switch that one turns or off, like call waiting, Calling Line DN Verification has to be provisioned.

First add an Audio Announcement. Requires assigning an Announcement ID (3001-32766), Name, Description, Repeat Counter, Message list ID

```
Command Example: ENT-AUDIO-
                 ANNC::3456:::NAME=cgpn_announcement,DESC=c
                 gpn_announcement,MSGLIST=DIG_GEN,RPTCNT=1;
```

Then add a Treatment. Requires assigning a Name, treatment type, Announcement ID, and an announcement argument set to Calling Party Number.

```
Command Example: ENT-
                 TREATMENT::anncgpn:::TREATTYPE=ANNC,ANNCID
                 =3456,ANNCARGS=CGPN-
                 DEFAULT,ANSIND=N,FAILCND=NULL,DGTMODKEY=NU
                 LL,RTKEY=NULL,NEXTTREAT=NULL;
```

Then add a phone number to route digits that will be directed to the treatment announcement. Requires the phone number along with nature of address and route destination, which is the treatment announcement.

```
Command Example: ENT-ROUTE-DIGITS::DEFAULT-CDPN-NATNUM-
                 5088048906:::RTDEST=TREAT-
                 anncgpn,ACTN=NULL,MINCOST=0,MAXCOST=0,RERO
                 UTE=N;
```

Calling Name

Calling Name Blocking (CNAB)

CNAB toggles the CNAM delivery option (between Allowed or Restricted) associated with a calling party as provisioned in the SCP related to the terminating switch. If the caller's presentation status is normally *restricted*, activating the CNAB will allow presentation of the caller's name for the call in process. Conversely, if the normal presentation status is *allowed*, activating CNAB will suppress presentation of the caller's name for the call in process.

Note: The reverse scenario can be accomplished in the same manner (Calling Party's normal presentation is PRIVATE and he/she wants to present his/her CNAM to the Called party).

How to Use

Lift the receiver and listen for dial tone.

Dial *63 to activate. Wait for recall dial tone and then dial the called party.

Provisioning the switch using TL1

To provision a user for CNAB, you must activate CNAB using the `ED-SERVICE-ACCESSCODE` command.

`Command Example: ED-SERVICE-ACCESSCODE::CNAB::::IS;`

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

`Command Example: ENT-SUBSCRIBER::5084062001:::CNAB=US;`

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *CNAB_Service* and select *Modify*. Make any changes and provision *IS.* Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *CNAB*, select the type of service. Select *Ok* or *Apply.*

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

The presentation status associated with a caller's name has no effect on ACR. ACR only interacts with the presentation status associated with the caller DN.

When a call is forwarded, the remote station receives the presentation statuses associated with the original calling party rather that the presentation statuses associated with the base station.

Calling Name Delivery (CNAM)

Calling Name Delivery allows a customer to receive the name of the calling party and the date and time of the call, unless the calling party has restricted name delivery. If the delivery of the name is restricted, the letter *P* for *private* is displayed. If the name is unavailable from the calling name database, the letter *O* for out-of-area/unavailable is displayed.

CNAM is delivered during the first long silent period during ringing using the Multiple Data Message Format.

CNAM may be provisioned either on the switch or at a remote SCP. For off-switch CNAM, the terminating end office does a SCP dip. The SCP uses the Calling Party DN to determine the CNAM and then provides it to the called party who has subscribed to the service. CNAM, if used, is billed on a per-usage basis or on a flat rate.

Provisioning the switch using TL1

To provision a user for CNAM, you must activate CNAM using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::CNAM::::IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number, the type of service, name presentation and subscriber's name. In the following example, the type of service is usage based and the presentation is public.

```
Command Example: ENT-SUBSCRIBER:: 5084062001:::CNAM=US,
                 CNAMP=PUBLIC,SUBNAME=John Doe;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *CNAM_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Enter the subscriber name if necessary and select *Ok* or *Apply*. Select the *Services* tab and then using the pull-down menu for *CNAM*, select the type of service. Make any changes and provision *IS*. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

CNAM service takes place at the base station for all terminating calls that are not forwarded. If a CNAM subscriber has a CFDA activated, CNAM service takes place at the base station for all incoming calls. If a CNAM subscriber has CFBL activated, CNAM service does not take place at the base station for terminating calls that are forwarded due to a busy condition. If a call is forwarded and the remote station has CNAM service, the calling party name of the originating party is delivered to the remote station.

When a customer that has CNAM active on its line forwards calls to a remote DN using CFV, the base station does not receive the calling party name for incoming calls that are forwarded. If a remote party has CNAM active on its line, the calling name of the originating party is sent to the remote party's CPE.

Calling Number

Calling Number Delivery (CND)

Calling Number Delivery allows a customer to have the calling party's DN displayed. The calling party DN is delivered/displayed to customers who have subscribed to the service. CND, if used, is billed on a per-usage basis or on a flat rate.

Provisioning the switch using TL1

To provision a user for CND, you must activate CND using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::CND::::IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ED-SUBSCRIBER::5084061001:::CND=US,
                 CDNP=PRIVATE;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *CND_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *CND*, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

If CND and ACR are active on the same line, no CND information is sent on a call that received rejection treatment.

If a customer subscribes to both CND and CWT, no CND data is transmitted during or after a CWT tone or during or after any switch hook flashes that occur in response to a CWT tone.

CND shall be enabled on the CIDCW customer's line so that CND data delivery for the waited call takes place.

Calling Number Delivery (CNDB)

CNDB toggles the CND delivery option (between Allowed/Public or Restricted/Private) associated with a calling party as provisioned in the subscriber profile. If the caller's presentation status is normally *private/restricted,* activating the CNDB will allow presentation of the caller's number for the call in process. Conversely, if the normal presentation status is *public/allowed,* activating CNDB will suppress presentation of the caller's number for the call in process.

How to Use

Lift the receiver and listen for dial tone.

Dial *67 to activate. Wait for recall dial tone and dial the called party.

Provisioning the switch using TL1

To provision a user for CNDB, you must activate CNDB and the codes for user activation must be provisioned using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::CNDB:::
                 ACTIVATIONCODE=67:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ED-SUBSCRIBER::5084061001:::CNDP=PRIVATE,
                 CNDB=US;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *CNDB_Service* and select *Modify*. Make any changes and provision *IS.* Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber,

right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for CND*B*, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Call Waiting

Cancel Call Waiting (CCW or CWC)

CCW or CWC allows a customer to disable Call Waiting for the duration of a telephone call. CCW is an addition to the Call Waiting feature.

How to Use

Lift the receiver and listen for dial tone.

Dial *70 and then dial the number to be called. Call waiting is disabled during that call. When the subscriber disconnects from the call, call waiting is again active.

Provisioning the switch using TL1

To provision a user for CWC, you must activate CWC and the codes for user activation must be provisioned using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::CWC:::
                 ACTIVATIONCODE=70:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is CWC is standard.

```
Command Example: ENT-SUBSCRIBER:: 5084062001:::CWC=STND;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *CCW_Service* and select *Modify*. Make any changes and provision *IS.* Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *CWC*, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

If a call terminates by way of CFBL to a busy line and that line is CWT disabled, a busy treatment is provided.

If a line has both CFBL and CWT and CWT is canceled, when a call comes in and the line is busy it gets CFBL treatment.

When activated, CFV takes precedence over CWT. Any attempt to disable CWT by way of the CCS feature while CFV is activated has no effect. Although CCW has no effect in this situation, the switching system still returns a recall dial tone after the customer dials the CCW access code and receives the telephone number dialed by the customer. The system allows a customer to disable CWT while CFV is activated. Customers who subscribe to both CFV and CWT can dial the access codes for both on a single call as long as the CCW access code is dialed first.

When CWT is in effect, it should take precedence over TWC. When CWT is disabled, TWC treatment should be given when the customer flashes. If a customer activates CCW and then originates a TWC, CWT should remain disabled until all connections are torn down. If either of the noncontrolling parties of the TWC disconnect (or are disconnected by the controller), CWT should remain disabled for the remaining two-way connection. If the initiator of TWC hangs up with a party on hold, he/she should be rung back and connected to the held party on answer. If the initiator's CWT was disabled prior to hanging up on the held party, it should remain disabled after the customer answers the ringback.

Call Waiting (CWT or CW)

Call Waiting notifies a customer of an incoming call while another call is already established. The service allows a customer to put the present call on hold and establish a connection with the new caller. CWT is a subscription-based call termination service that does not impact call origination.

Before applying the CWT tone, the switch checks to determine if Cancel CW applies to the standing call on the called line. Also, a busy treatment is returned to a caller who attempts to call a line with CWT when the CWT feature is already applied.

Note: CIDCW and CWT cannot be active simultaneously for a subscriber.

Provisioning the switch using TL1

To provision a user for CWT, you must activate using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::CWT::::IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is standard.

```
Command Example: ED-SUBSCRIBER::5084061001:::CWT=STND;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *CW_Service* and select *Modify*. Make any changes and provision *IS.* Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *CWT*, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

During a call to a 911 attendant, the Call Waiting service is either inhibited (no CW tone) or flashes are not recognized.

Given a line that has both CFBL and CWT and is in a talk state, the first call attempting to terminate is treated as a CWT call. Subsequent termination attempts are call forwarded (that is, CFBL is invoked only if a call is already waiting.)

If CWT treatment cannot be given (because the line is dialing or ringing), then CFBL takes effect.

If a customer has CFV activated, the CWT service is ineffective at the base station. The call is forwarded whether the base station is idle or in use. (Note: the CWT feature of the base station is not transferred to the remote station.)

Code Restriction/Diversion Prohibited Code List (CDRPCL) or Screen Block

Code Restriction/Diversion Prohibited Code List (CDRPCL), also known as Screen Block, allows a customer to block origination specified calls (900/976) originating from their line. This service is available to both PRI and CAS customers.

A Local Service Provider activates screen blocking.

Provisioning the switch using TL1

To provision a user for CDRPCL, you must activate CDRPCL using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::CDRPCL::::IS;
```

You use the `ED/ENT-SUBSCRIBER` command to provision whether the subscriber can or cannot block numbers.

```
Command Example: ED-SUBSCRIBER::5084061001:::CRDPCL=UNRES;
```

You use the `ED/ENT-LIST-SUBSCNBLK` command to create a list containing strings of digits that a subscriber is blocked from calling. A partial list of examples follows.

```
Command Example: ENT-LIST-SUBSCRNBLK::HorizonInt-
                 INTNATIONAL-987654321234567898765;
```

```
Command Example: ENT-LIST-SUBSCRNBLK::
                 HorizonInt-INTNATIONAL-DEFAULT;
```

```
Command Example: ENT-LIST-SUBSCRNBLK::
                 HorizonIntOp-OPINTRN-DEFAULT;
```

```
Command Example: ENT-LIST-SUBSCRNBLK::
                 HorizonOp-OPNAT-DEFAULT;
```

```
Command Example: ENT-LIST-SUBSCRNBLK::
                 HorizonOp-OPNAT-0;
```

```
Command Example: ENT-LIST-SUBSCRNBLK::
                 HorizonOp-OPERATOR-00;
```

```
Command Example: ENT-LIST-SUBSCRNBLK::
                 HorizonOp-OPERATOR-DEFAULT;
```

```
Command Example: ENT-LIST-SUBSCRNBLK::
                 HorizonNat-NATIONAL-9082221234;
```

```
Command Example: ENT-LIST-SUBSCRNBLK::
                 HorizonN11-NATIONAL-211;
```

```
Command Example: ENT-LIST-SUBSCRNBLK::
                 HorizonTollFree-NATIONAL-800;
```

```
Command Example: ENT-LIST-SUBSCRNBLK::
                 HorizonTollCut-CUTTHRU-0008;
```

```
Command Example: ENT-LIST-SUBSCRNBLK::
                 HorizonTollCut-CUTTHRU-DEFAULT;
```

You then use the `ED-SUBSCRIBER` command to provision the primary and secondary screen block list name.

```
Command Example: ED-SUBSCRIBER::5084061001:::
                 SUBSCRNBLKLIST=HorizonInt,
                 SUBSCRNBLKLIST2=HorizonN11;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *SCREENBLOCK_SERVICE* and select *Modify*. Provision as *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *CRDPCL*, select the type of service. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Screen Block List*. To add a new list, right-click *Screen Block List* and select *Add*. To modify a list, move cursor over a list name, right-click and select *Modify*. Make entries or changes and then select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Screen Block Lists* tab and then using the pull-down menu, select the list names. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

Activation of call forwarding is not permitted to a prohibited destination. The switch gives special service error treatment in response to such an activation request.

Customer Originated Trace (COT)

Customer Originated Trace allows a customer to trace the number of a calling party. A customer who wants to trace the number of a harassing phone call uses the feature in a one shot fashion. COT, if used, is billed on a per-usage basis or on a flat rate.

There are two parties involved in a COT:

- The customer who receives the call and requests the trace.
- The agency that receives the output message that contains the trace results.

If a customer is allowed access for a COT, tone indicates that the customer has access to this feature.

If the number in the incoming memory slot is valid (a 7 or 10-digit directory number is available), the customer can dial an activate command (*57) to trace the last call received or to hang up and end the session.

The last call the customer receives is traced and the customer hears a confirmation tone.

The system sends a message that contains the calling party's number and other call-related information over the appropriate data channel. This data is written to the TT database and can be retrieved with the `RTRV-SUB-COT` command.

How to Use

Dial *57 to activate. Wait for confirmation tone.

Retrieve the trace data using the `RTRV-SUB-COT` TL1 command.

```
Command Example: RTRV-SUB-COT::5084061001;
```

Provisioning the switch using TL1

To provision a user for COT, you must activate and provision the codes for user activation using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::COT:::
                 ACTIVATIONCODE=57:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ED-SUBSCRIBER::5084061001:::COT=US;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *COT_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber,

right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for COT, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

If the customer receives a call that was forwarded from another station, the calling party identification used for the trace is the DN of the party originating the call, if available, not the DN of the base station. If the originating DN is not available, the COT request is denied. If the customer has Call Forwarding (CF) active, the incoming memory slot is not updated so the last call received before CF was activated is the only one that can be traced.

If the customer activates a trace after call waiting interrupts a call, the calling party identification used for the trace is the DN of the party that call-waited, regardless of whether the attempt was answered.

If a customer has CND and the calling number is available, it is sent to the customer's premises and does not in any way affect the customer's ability to use COT.

COT takes precedence over CNDB. If a COT customer activates a trace on a CNDB customer's DN that is designated as *private,* a trace is completed even though the CNDB customer's DN cannot be revealed to the called party.

If the customer has SCF active, the calls selectively forwarded do not update the incoming memory slot at the base station. Therefore, these calls cannot be traced at the base station.

Home Intercom (HI)/Barn Phone

This feature allows a subscriber to use any extension at their home or business as an intercom device to call the other extensions that are connected to the same line. When this feature is enabled, the subscriber dials his/her own Directory Number (DN); upon receiving a busy tone, the subscriber hangs up. The Compact Switch then rings the subscriber back. When one of the other extensions connected to the line answers the call, the answering party receives silence. When another extension, connected to the same line goes off-hook, the two extensions are connected. The call is terminated when both of the parties go on-hook. It is provisionable on a subscriber DN basis.

Notes:

- Service is billed on a flat rate basis.
- Service applies to CAS and GR-303 lines only. It is NOT applicable to MGCP endpoints or PRI interfaces.
- If the subscriber goes off-hook before the ring-back occurs then the ring-back will not occur.

- Active Home Intercom calls are not preserved over an SP switch to protection card since the Home Intercom call is not recognized as a stable two party call.
- Service Interactions/Considerations
  - Speed Calling can be use to invoke Home Intercom.
  - The Home Intercom invocation call will not invoke CFBL or CFV.
  - The operator receives a reorder for BLV/BLI of a line with a Home Intercom call in progress, which is the same as a line that is receiving dial tone.
  - External calls to a Home Intercom subscriber with CFBL or CFV are not forwarded during Home Intercom busy tone.
  - The Home Intercom ring-back call will not invoke CFDA.
  - External calls to a Home Intercom subscriber with CFBL or CFV will forward during Home Intercom ring back or during the active Home Intercom call.
  - No CWT of external calls during Home Intercom call.
  - No flash/TWC during Home Intercom call.
  - An Home Intercom subscriber with CNAM does not see their own name on the Home Intercom ring-back.
  - Home Intercom is not compatible with MDNL (Multiple Directory Numbers per Line) with DR (Distinctive Ringing). Home Intercom is invoked on an MDNL w/DR line only if the Primary DN is dialed, and then, only Home Intercom DR is received.

Provisioning the switch using TL1

To provision a user for HI, you must activate and provision the codes for user activation using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::HI::::IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ED-SUBSCRIBER::5084061001:::HI=FR;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *HI Service* and select *Modify*. Provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for HI, select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Hot Line (HL)/Automatic Line

This feature allows a subscriber's telephone to be programmed as a hotline telephone. Hotline telephones can access only one pre-designated destination, which can be any valid directory number (DN) consisting of 1 to 15 digits. When the subscriber lifts the telephone handset, the Compact Switch automatically dials the DN that is associated with the line.

Notes:

- Service is billed on a flat rate basis.
- Service applies to CAS and GR-303 lines and MGCP endpoints. It is NOT applicable to PRI interfaces.
- Hot line calls must be programmed without a 10xxx carrier access code.
- Hotline cannot be provisioned with any feature that requires a hook switch flash and/or entry of digits (only a few terminating features like caller ID or distinctive ring will work with hot line).
- Hotline call origination while the destination is busy will result in a busy signal.

Provisioning the switch using TL1

To provision a user for HL, you must activate and provision the codes for user activation using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::HL::::IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ED-SUBSCRIBER::5084061001:::HOTLN=FR,
                 HOTLNDN=911;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *HL Service* and select *Modify*. Provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber,

right-click and select *Modify*. Select the Select the *Other Details* tab. Update the *Hot Line DN*. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Local Number Portability (LNP)

Local Number Portability (LNP) is a service that allows a customer to retain a telephone number when he moves within a local rate center. The originating Central Office accesses a Service Control Point, which is a database. It contains a path for the LEC providing service to the target phone number so that the originating carrier can hand off the call to the terminating switch.

Provisioning the switch using TL1

To provision for LNP with DPC-SSN routing, you must establish an SCCP (Signaling Connection Control Part) route to a remote point code using the `ENT-SCCP-ROUTE` command.

```
Command Example: ENT-SCCP-ROUTE::COM:::3-3-3;
```

You must establish an SCCP route to an SSN (sub-system number) on a remote point code using the `ED-SCCP-SSN` command. In the following example the point code to receive the query is 3-3-3 and subsystem number of the providing service to route to is 30.

```
Command Example: ENT-SCCP-SSN::3-3-3-30:::ONLINE;
```

You must also establish a service logic host route to a Service Control Point (SCP) using the `ENT-SLHR-SCP` command as shown in the following example. The *slhrid* in the example is 0; the *routetype* is RTE_SSN indicating routing directly to an SCP using a point code; the *dpc* is 3-3-3; the *svcreqssn* is 69 and the *svcprovssn* is 30.

```
Command Example: ENT-SLHR-SCP::0:::DESC=LNP,
                 ROUTETYPE=RTE_SSN,DPC=3-3-3,
                 SVCREQSSN=69,SVCPROVSSN=30;
```

You must add telephone numbers to the list of called numbers for which LNP queries can be issued using the `ENT-LNPSCREEN-DIGITS` command.

```
Command Example: ENT-LNPSCREEN-DIGITS::508408;
```

To provision a user for LNP, you must activate using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::LNP:
                 DESC=LNP_SERVICE,SLHRID=0:IS;
```

You must establish a location routing number to support the service-provider type of location number portability (LNP) using the `ENT-LRN` command.

```
Command Example: ENT-LRN::6177201111:::P;
```

```
Command Example: ENT-LRN::4012221111:::S;
```

Finally you must edit the truck group using the `ED-TRKGRP` command.

```
Command Example: ED-TRKGRP::1600:::SECONDARYLRN=4012221111;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. *Double-click SS7, Intelligent Network Services and SCCP Route.* Right-click *SCCP Route* and select *Add* to add a new route or right-click a route and select *Modify* to change. Make any changes and select *Ok* or *Apply*.

Right-click *SCCP SSN* and select *Add* to add a route to an SSN or right-click an existing route and select *Modify*. Make entries or changes and select *OK* or *Apply*.

Double click *SLHR SCP Services.* Then right-click *SLHR SCP Services* to add a SCP or right-click an existing SCP to modify. Make any entries or required changes. Select *OK* or *Apply*.

Double-click *SS7*, *LNP* and then *LNP Screen*. Right-click *LNP Screen* and select *Add* to add a range. Make entries and select *Ok* or *Apply*.

Double-click *System Services* and then *Service Access Code*. Right-click *LNP_SERVICE* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *SS7*, *LNP* and then *LRN*. Right-click *LRN* and select *Add* to add a number. Make entries and select *Ok* or *Apply*.

Double-click *IMT* and then the *ISUP, SIP, CAS, BICC or SIP-T Trunk Group*. Right-click the desired trunk group and select *Modify*. Ensure you have selected the *Trunk Group Params* tab and the correct *Trunk Group ID*. Using the pull-down menu for *Secondary LRN*, select the correct number. Select *Ok* or *Apply*

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Multi-Way Calling (MWC) or Add On Transfer Conference (AOTC)

Multi-Way Calling (MWC), also known as Add On Transfer Conference (AOTC), allows a subscriber to hold a call and originate an additional call, as well as:

- Form a three-way conference (TWC)
- Transfer a call to another line (CT)

If used, MWC or AOTC is billed on a per-usage basis or on a flat rate.

Provisioning the switch using TL1

To provision a user for MWC, you must activate MWC using the `ED-SERVICE-ACCESSCODE` command.

`Command Example: ED-SERVICE-ACCESSCODE::MWC::::IS;`

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is add-on call transfer billed on per-usage basis.

`Command Example: ED-SUBSCRIBER::5084061001:::MWC=AOTCUS;`

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *AOTC_Service* and select *Modify*. Make any changes and provision *IS.* Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *MWC*, select the type of service. Select *Ok* or *Apply*,

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Remote Access to Call Forwarding (RAF)

When you are away from the office, Remote Access to call Forwarding (RAF) lets you dial in and direct calls to another phone number.

How to Use

To use this feature, you must:

- Dial the Remote Access Number of the Call Forwarding Center.
- Dial the 10-digit number of the RAF subscriber
- Dial your ten-digit PIN number.
- Dial *72 or 72#.
- Answer the questions.

When you return to your phone, dial 73# to de-activate call forwarding.

If RAF is used, it is billed on a per-usage basis or on a flat rate.

Provisioning the switch using TL1

Ensure that CFV has already been provisioned for the subscriber. Provision the Remote access number provided by the carrier using the `ENT-DN-RAF` command.

```
Command Example: ENT-DN-RAF::5084061234
```

You must activate RAF using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::RAF:::
                 ACTIVATIONCODE=72,DEACTIVATIONCODE=73:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number, the remote access code and the type of service. In the following example, the type of service is usage based.

```
Command Example: ENT-SUBSCRIBER-SERVICE::5084061001:::
                 RMTACSCODE=12345,RMT=US;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *RAF* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *RMT* select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Residence Distinctive Alerting Service (RDAS) or Distinctive Ring (DR)

Residence Distinctive Alerting Service (RDAS), also know as Distinctive Ring (DR), allows a customer to associate additional telephone numbers with a single telephone line. One DN receives normal ringing cadence. Each secondary DN can be provisioned to have its own distinctive ringing pattern, DISTINCTINT or Pattern 1 Distinctive Intergroup that is two rings and DISTINCTSP or Pattern 2 Special that is three rings.

The switch supports up to two distinctive Directory Numbers for each physical line terminated on the unit – the primary number and two secondary numbers.

All calls that originate from a line supporting multiple DNs will generate CDR records referencing the primary DN.

All calling features are line-specific rather than DN-specific in the case of lines equipped with multiple DNs.

Provisioning the switch using TL1

To provision a user for RDAS, you must activate RDAS using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::RDAS::::IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and whether RDAS is activated. SCDN means that the RDAS feature is active and DN is assigned as the secondary DN for the RDAS feature.

```
Command Example: ED-SUBSCRIBER::5084062001:::RDAS=SCDN;
```

You use the ENT-SUB-ALTDN command to associate a previously configured alternate directory number with a subscriber line and assign a distinctive ring for it. In the following example the distinctive ring is Pattern 1 Distinctive Intergroup of two rings.

```
Command Example: ENT-SUB-ALTDN::5084062001-5084069999:::
                 RING=DISTINCTINT;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *DR_Service* and select *Modify*. Make any changes and provision *IS.* Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *RDAS*, select the type of service. Select *Ok* or *Apply*. Select the *Alternate DNs* tab to select the *ring*. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

CFV, SCA or SCR take precedence over DR.

Selective Call

Selective Call Acceptance (SCA)

Selective Call Acceptance allows a subscriber to reject calls from any party that is not programmed on the subscriber's Selective Call Acceptance List. The subscriber can create a list of up to 32 telephone numbers. Incoming calls from these numbers are accepted. If a calling number does not match a number in the SCA list, the call will be forwarded to an announcement and the caller can enter a valid PIN of 4 to 8 digits to authorize call completion or the call shall be forwarded to a customer specified directory number or to the customer's voice mail. Disposition of the call are subscription parameters, and are not user-provisionable as part of the feature editing activity. SCA, if used, is billed on a per-usage basis or on a flat rate.

How to Use

Lift the receiver and listen for dial tone.

Dial *64 to activate. There are recorded instructions to allow set-up, review or change the list of phone numbers to be accepted or turn the feature on/off.

Dial *84 to deactivate SCA.

Provisioning the switch using TL1

To provision a user for SCA, you must activate SCA and provision the codes for user activation and deactivation using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::SCA:::
                 ACTIVATIONCODE=64,DEACTIVATIONCODE=84:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number, the type of service, whether subscriber can edit screen list, the PIN number entered by a caller to complete a call to a subscriber with SCA activated, the termination treatment for calls not accepted due to active SCA and the directory number to which unaccepted calls are forwarded. In the following example, the type of service is usage based, SCA is active, the subscriber can edit screen list, there is no PIN number for a caller and treatment for a rejected call is to play an announcement.

```
Command Example: ENT-SUBSCRIBER-SERVICE::
                 5084062001:::TYPE=EO,SCA=US,SCAAC=Y,SCALE=
                 Y,SCAPIN=NULL,SCATT=REJ;
```

To provision numbers that are accepted, you use the `ENT-SUB-SCFLIST` command to establish the association between a subscriber and a selective call feature list. The following example provisions the first number in the list.

```
Command Example: ENT-SUB-SCFLIST::5084062001-SCA-1:::
                 CGPN=5084900909,CGPNPRES=PUBLIC;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *SCA_Service* and select *Modify*. Make any changes and provision *IS.* Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *SCA*, select the type of service. Select *Ok* or *Apply*.

Select the *Selective Call* tab at the top and *Selective Call Parameters* at the bottom. Make required changes and select *Ok* or *Apply*. Select the *Selective Call Lists* tab at the bottom and the *Acceptance tab* near the top. Make required changes and select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

SCA takes precedence over CFBL, CFDA, CFV and CWT

If an SCA customer activates ACB, the call should be rejected unless the calling number is on the accept list.

Selective Call Forwarding (SCF)

SCF allows a subscriber to forward an incoming call if the calling party is on the subscriber's selective call forwarding list. The customer can construct or modify a telephone number screening list by dialing an activation code. The switch will screen incoming calls against the customer's list and forward only those telephone numbers on the list.

This feature allows the subscriber to create a list of up to 32 telephone numbers. Incoming calls from these numbers may be forwarded to another number instead of being completed at the subscriber’s telephone number. All other calls are completed as usual. SCF, if used, is billed on a per-usage basis or on a flat rate.

How to Use

Lift the receiver and listen for dial tone.

Dial *62 to activate. There are recorded instructions to allow set-up, review or change the list of phone numbers to be forward or turn the feature on/off.

Dial *83 to deactivate SCF.

Provisioning the switch using TL1

To provision a user for SCF, you must activate SCF and provision the codes for user activation and deactivation using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::SCF:::
                 ACTIVATIONCODE=62,DEACTIVATIONCODE=83:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number, the type of service, whether subscriber can edit screen list, the directory number to which calls are forwarded, whether to suppress notification of forwarded call and the maximum number of concurrent sessions allowed for SDV and SCA. In the following example, the type of service is usage based, subscriber can edit screen list, calls are forwarded to 5088048123, based station is notified for forwarded calls and the twelve concurrent SCF and SCA sessions are allowed.

```
Command Example: ENT-SUBSCRIBER:: 5084062001:::TYPE=EO,
                 SCF=US,SCFLE=Y,SCFDN=5088048123,SCFRING=N,
                 MAXSCFSES=12;
```

To provision numbers that are accepted, you use the `ENT-SUB-SCFLIST` command to establish the association between a subscriber and a selective call feature list. The following example provisions the first number in the list.

```
Command Example: ENT-SUB-SCFLIST::5084062001-SCF-1:::
                 CGPN=5084900909,CGPNPRES=PUBLIC;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *SCF_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *SCF*, select the type of service. Select *Ok* or *Apply*.

Select the *Selective Call* tab at the top and *Selective Call Parameters* at the bottom. Make required changes and select *Ok* or *Apply*. Select the *Selective Call Lists* tab at the bottom and the *Forwarding tab* near the top. Make required changes and select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

If an SCF customer activates ACB, ringback should always be given to the customer's line and not forwarded, even if the number to which ACB was activated is on the SCF list.

Selective Call Rejection (SCR)

Selective Call Rejection allows a subscriber to reject calls from any party that is programmed on the subscriber's Selective Call Rejection list. Rejected calls are routed to the Selective Call Rejection intercept treatment such as an announcement. All other incoming calls receive normal call treatment.

This feature allows the subscriber to create a list of up to 32 telephone numbers.

How to Use

Lift the receiver and listen for dial tone.

Dial *60 to activate. There are recorded instructions to allow set-up, review or change the list of phone numbers to be accepted or turn the feature on/off.

Dial *80 to deactivate SCR.

Provisioning the switch using TL1

To provision a user for SCR, you must activate SCR and provision the codes for user activation and deactivation using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::SCR:::
                 ACTIVATIONCODE=60,DEACTIVATIONCODE=80:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number, the type of service, whether subscriber can edit screen list, and whether answer supervision (an ANM or off-hook message) is sent to the originating office when a calling party is connected to the SCR rejection announcement. In the following example, the type of service is usage based, SCA is active, the subscriber can edit screen list, and answer supervision is sent.

```
Command Example: ENT-SUBSCRIBER-SERVICE::
                 5084062001:::TYPE=EO,SCR=US,SCRAC=Y,SCRLE=
                 Y,SCRANS=Y;
```

To provision numbers that are accepted, you use the `ENT-SUB-SCFLIST` command to establish the association between a subscriber and a selective call feature list. The following example provisions the first number in the list.

```
Command Example: ENT-SUB-SCFLIST::5084062001-SCR-1:::
                 CGPN=5084900909,CGPNPRES=PUBLIC;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *SCR_Service* and select *Modify*. Make any changes and provision *IS.* Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *SCR*, select the type of service. Select *Ok* or *Apply.*

Select the *Selective Call* tab at the top and *Selective Call Parameters* at the bottom. Make required changes and select *Ok* or *Apply*. Select the *Selective Call Lists* tab at the bottom and the *Rejection* tab near the top. Make required changes and select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

SCR takes precedence over CFBL, CFDA, CFV, DR and SCF

An SCR customer who also has CWT will not receive call-waiting tones for incoming calls that are rejected by the SCR feature.

If the SCR customer has CND, no message should be delivered to the customer on rejected calls.

Normally, SCR customers cannot activate AR to numbers on their SCR lists because incoming calls from such numbers are rejected. However, if a person uses AR to redial a particular number after placing that number on the SCR list, but before receiving another incoming call, AR can be activated.

Speed Calling

Speed calling allows a customer to dial pre-selected telephone numbers using only one or two digits. It is a subscription-based call origination service. The service can be established either statically or dynamically. The customer selected or carrier-provisioned presets must be routable off the switch (`ENT-ROUTE, ENT-ROUTE-ISDNIF, ENT-ROUTE-CARRIER`).

Speed Calling (SC1 or SPEED 8)

SC1 (also known as SPEED 8) allows a customer to dial up to eight pre-selected telephone numbers using only one digit. Valid digits are 2 – 9. This feature can be used in conjunction with SC2 (also known as SPEED 32). To activate SC1:

How to Use

Lift the receiver and listen for dial tone.

Dial *74 to activate.

Listen for the dial tone.

Press the speed code number (2-9) to be programmed.

Dial the telephone number to be programmed.

Listen for a confirmation.

Hang up.

To implement Speed 8, dial the pre-programmed speed code.

Provisioning the switch using TL1

To provision a user for SC1, you must activate SC1 and the codes for user activation must be provisioned using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::SC1:::
                 ACTIVATIONCODE=74:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and type of control. In the following example, the customer controls.

```
Command Example: ENT-SUBSCRIBER::5084062001:::SC1=SC1CC;
```

To provision numbers for speed calling, you use the `ENT-SUB-SPEED` command. In the following example, the first number in the list (2-9) for the subscriber is 5084800909.

```
Command Example: ENT-SUB-SPEED::5084062001-2:::
                 DN=5084800909;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *SPEED_8_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *SC1*, select the type of service. Select *Ok* or *Apply*.

To provision speed call numbers, select the *Speed Codes* tab. Enter the speed code number and the directory number and select *Ok* or *Apply*.

Speed Calling (SC2 or SPEED 32

SC2 (also known as SPEED 32) allows a customer to dial up to thirty, customer pre-selected telephone numbers using two digits. Valid digits are 20 – 49. This feature is available to CAS customers and can be used in conjunction with SC1.

How to Use

Lift the receiver and listen for dial tone.

Dial *75 to activate.

Listen for the dial tone.

Press the speed code number (20-49) to be programmed.

Dial the telephone number to be programmed.

Listen for a confirmation.

Hang up.

To implement SC2, dial the pre-programmed speed code.

Provisioning the switch using TL1

To provision a user for SC2, you must activate SC2 and the codes for user activation must be provisioned using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::SC2:::
                 ACTIVATIONCODE=75:IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and type of control. In the following example, the customer controls.

```
Command Example: ENT-SUBSCRIBER:: 5084062001:::SC2=SC2CC;
```

To provision numbers for speed calling, you use the `ENT-SUB-SPEED` command. In the following example, the first number in the SC2 list (20-49) for the subscriber is 5084800909.

```
Command Example: ENT-SUB-SPEED::5084062001-20:::
                 DN=5084800909;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *SPEED_32_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for *SC2*, select the type of service. Select *Ok* or *Apply*.

To provision speed call numbers, select the *Speed Codes* tab. Enter the speed code number and the directory number and select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

The CCW access code can be an entry in the customer's speed call list.

Speed call list entries that contain NPAs or NNXs that are code restricted for the customer are screened at the time the entry is placed on the speed call list or when the call is placed.

Toll Free

Toll Free is a service in which there is no long distance charge. It differs from AIN Toll Free in the handling of messaging. Toll Free is an Intelligent Network (IN)-based service, as opposed to an Advanced Intelligent Network (AIN)-based service. Toll Free provisioning is part of SLHR SCP Services.

Provisioning the switch using TL1

To provision for tollfree using DPC-SSN routing, you must establish an SCCP (Signaling Connection Control Part) route to an SSN (sub-system number) on a remote point code using the `ED-SCCP-SSN` command. In the following example the point code to receive the query is 3-3-3 and subsystem number of the providing service to route to is 233.

```
Command Example: ENT-SCCP-SSN::3-3-3-254:ONLINE;
```

A service logic host route to a Service Control Point (SCP) must also be established using the `ENT-SLHR-SCP` command as shown in the following example. The *slhrid* in the example is 2; the *routetype* is RTE_SSN indicating routing directly to an SCP using a point code; the *dpc* is 3-3-3; the *svcreqssn* and *svcprovssn*, the SSN for the service on the switch and the SCP, is 254.

```
Command Example: ENT-SLHR-SCP::2:::DESC=800-IN,
                 ROUTETYPE=RTE_SSN,DPC=3-3-3,
                 SVCREQSSN=254,SVCPROVSSN=254;
```

Finally the feature must be enabled using the `ED/ENT-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::TOLLFREE:::
                 DESC=TOLLFREE_SERVICE,SLHRID=2:IS
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *SS7*, double-click *Intelligent Network Services*. Then right-click *SCCP SSN* and select

Add to add a route to an SSN or right-click an existing route and select *Modify*. Make entries or changes and select *OK* or *Apply*.

Double click *SLHR SCP Services*. Then right-click *SLHR SCP Services* to add a SCP or right-click an existing SCP to modify. Make any entries or required changes. Select *OK* or *Apply*.

Double-click *System Services* and then *Service Access Code*. Right-click *TOLLFREE_* SERVICES and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Three-Way Calling (TWC)

Three-way Calling allows three subscribers to converse at the same time. However, unlike CT you cannot use Call Waiting with this service. TWC, if used, is billed on a per-usage basis or on a flat rate

How to Use

Dial the first party, press the flash or switch-hook to get a second dial tone, dial the second number and press the flash or switch-hook again to connect the two calls.

To deactivate TWC, press flash again and the phone automatically reverts to single-party calling.

Provisioning the switch using TL1

To provision a user for TWC, you must activate TWC using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::TWC::::IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ED-SUBSCRIBER::5084061001:::TWC=US;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *TWC_Service* and select *Modify*. Make any changes and provision *IS*. Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for TWC select the type of service. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Interactions With Other Features

TWC does not restrict the CWT capabilities of customers who did not initiate TWC. They should be able to receive CWT calls in either a talking or held state. They should receive the CWT tone and then be able to put the TWC connection or the held connection on hold.

The initiator of TWC should not receive CW calls or the CW tone while in a TWC mode or while a party is on hold.

If a customer with the speed-calling feature is on an existing call, the customer should be able to initiate TWC, receive recall dial tone, and then dial the speed calling code of the party to be added.

Voice Mail

Voice Mail allows a caller to be transferred to a voice mail system (VMS) to leave a message if the called party does not answer or if the line is busy and/or there is no answer. In addition to storing messages, the VMS controls the Message Waiting Indicator (MWI), which is an audible and/ or visual indicator that one or more messages are waiting for the subscriber.

A Simplified Message Desk Interface (SMDI) provides an interface between a local switch and a Message Storage and Retrieval (MSR) system, also known as the voice mail system. SMDI consists of a group of lines, which form the bearer paths for calls forwarded to the VMS and a serial data link that provides the control path between the local switch and the VMS.

The SMDI data link is used to exchange control information with the VMS. When a call is forwarded to a VMS, a call history record is transported to the VMS over the SMDI data link. The call history record contains the message line identifier, the type of call indicator, the forwarding directory number, the calling directory number, the calling directory number presentation status and the calling name.

When a voice message has been stored on the VMS, the VMS sends a MWI Operate message to the local switch over the SMDI data link. This message includes the directory number of the subscriber, which has a message waiting. When the subscriber removes the message, the VMS sends a MWI Remove message to the switch.

Voice Mail is provisioned for a subscriber using the Subscriber and Voice Mail tab or EMS or TL1 commands found in the Subscriber Management or Voice Mail System Provisioning sections of the *TL1 Commands Reference Guide*.

For billing, the connection between the controller and the original party and the connection between the controller and the add-on party are regarded as separate calls. Even after the controller disconnects and a transfer occurs, billing continues for the original call and the add-on call until the transferred call is disconnected.

Provisioning the switch using TL1

The VMS must have been provisioned before provisioning a user. TL1 commands to provision VMS are `ED-VMS-SYS`, `ENT-VMS-LNK` and `ENT-VMS-CKTID.`

To provision a user for voice mail, you must activate VMS using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::VMS::::IS;
```

You must provision the subscriber whether a caller will transfer to voice mail if the subscriber does not answer or if the line is busy and/or there is no answer, the message waiting type and whether the subscriber will receive an abbreviated ring. You do this using the `ED-SUBSCRIBER` command. In the following example, the caller is forwarded to voice mail if the subsubscriber's line is busy or there is no answer, the link ID, defined in `ENT-VMS-LNK`, is 1-1, the message waiting type is both audio and visual message waiting indicators and the subscriber is provisioned with an abbreviated ring.

```
Command Example: ED-SUBSCRIBER::5084061001:::VMSTYPE=BNA,
                 VMSLNKID=1-1,VMSMWITYPE=BOTH,VMSABRRING=Y;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

The VMS must have been provisioned before provisioning a user. Double-click the switch in the switch panel to select it for the main panel. Double Click *Voice Mail* and provision *System, Link* and *Message Line Identifier*.

To provision a user for voice mail, double-click *System Services* and then *Service Access Code*. Right-click *VOICEMAIL_* SERVICES and select *Modify*. Make any changes and provision *IS.* Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Services* tab and then using the pull-down menu for VMS select the type of service. Select *Ok* or *Apply*. Then select the *Voice Mail* tab and make necessary changes. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Warm Line (WL)

This feature is similar to hot line, except an opportunity is given to dial alternate numbers. If no digits are entered within a pre-set time, the pre-provisioned destination number is automatically dialed. The timeout setting has a range of 0 - 30 seconds and is set on a per subscriber basis.

Notes:

- Service is billed on a flat rate basis.
- Service applies to CAS and GR-303 lines and MGCP endpoints. It is NOT applicable to PRI interfaces.
- Warmline can be provisioned with all services that are available, as long as the proper flashes and service codes are entered before the warm line timer expires.
- Warmline provisioned with a timer of zero follows the same design as Hotline and has the same restrictions.
- Warmline call origination while the destination is busy will result in a busy signal.

Provisioning the switch using TL1

To provision a user for WL, you must activate and provision the codes for user activation using the `ED-SERVICE-ACCESSCODE` command.

```
Command Example: ED-SERVICE-ACCESSCODE::WL::::IS;
```

Then you use the `ED/ENT-SUBSCRIBER` command entering the directory number and the type of service. In the following example, the type of service is usage based.

```
Command Example: ED-SUBSCRIBER::5084061001::: WRMLN=FR,
                 WRMLNDN=911,WRMLNTMR=3;
```

For complete command syntax of the above commands, refer to the *TL1 Commands Reference Guide.*

Provisioning the switch using the EMS

Double-click the switch in the switch panel to select it for the main panel. Double-click *System Services* and then *Service Access Code*. Right-click *WL_Service* and select *Modify*. Provision *IS.* Select *Ok* or *Apply*.

Double-click *Subscriber Database* and then *Subscriber*. To add a new subscriber, right-click *Subscriber* and select *Add*. To modify a subscriber, move cursor over subscriber, right-click and select *Modify*. Select the *Other Details* tab. Update the *Warm Line DN* and the *Warm Line Timer*. Select *Ok* or *Apply*.

For a complete description of the EMS screens and parameters, refer to *the PlexView EMS User Guide.*

Billing and CLASS 5

The switch generates BAF Structure Code 1030 for activation or reactivation and the call count of CLASS 5 features and the list size for SCF, SCA and SCR. In the ASCII reporting these are fields 91 to 100 as shown in the following table. For complete description of these fields, refer to the “6. ASCII Billing Reports” of the *Billing and Traffic Collection Guide.*

Field #	Field Description	GR-1100 Table #	Maximum # of Char.	Format
91	CLASS Feature Code	415	3	###
92	CLASS Functions	330	3	###
93	CLASS Feature Status	331	3	###
94	Deactivation date	6	10	MM-DD-YYYY
95	Deactivation time	18	10	HH:MM:SS.T
96	Call Count - Info Delivered	803	5	#####
97	Call Count - Info Anonymous/Unavailable	803	5	#####
98	Screening List Size - SCF SCA	802	3	###
99	Screening List Size – SCR	802	3	###
100	Screening List Size - DR CW	802	3	###

Enhanced Routing Software Feature Package

This document contains the following sections:

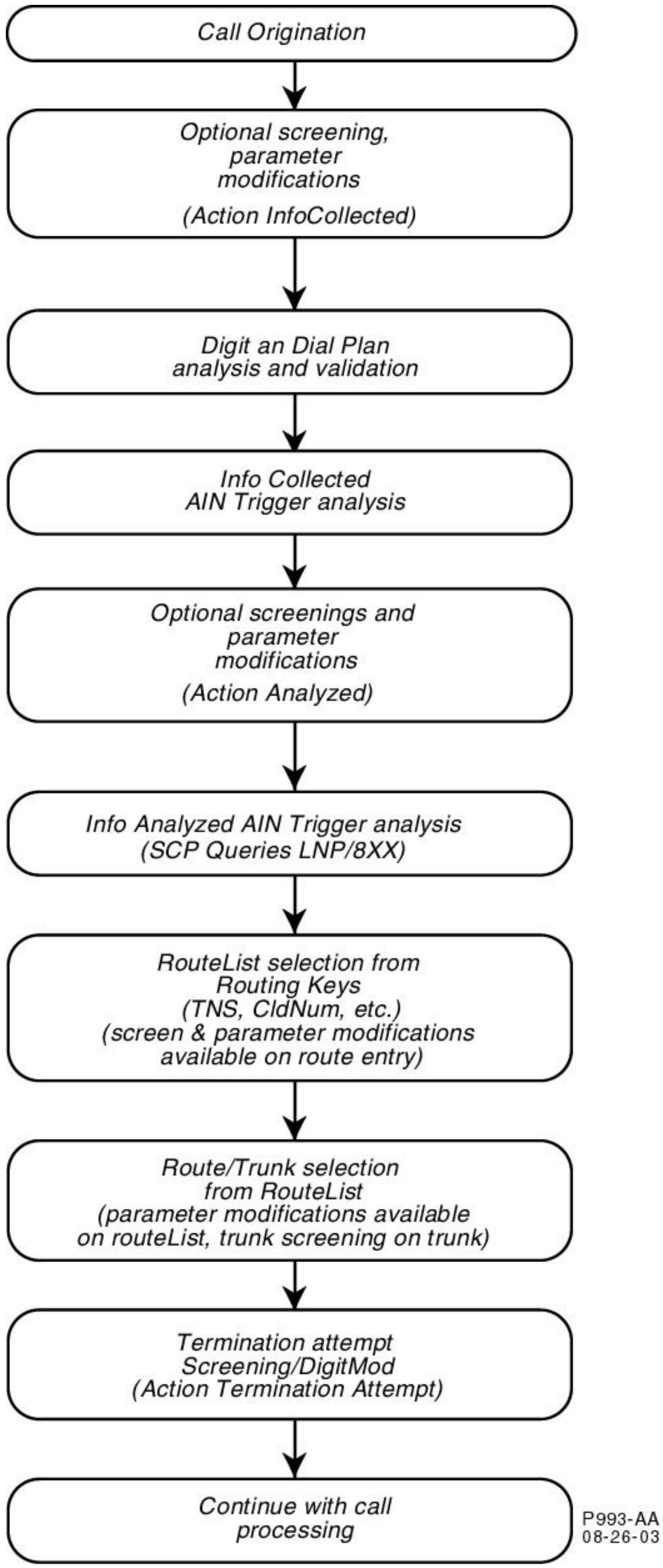
Scope

Routing is the process of matching any incoming call to the switch with a specific egress path to a destination switch. The incoming call parameters, combined with the translation plan, usually drive this process, but in certain circumstances, other parameters may supplement or completely override the number. Such modifying factors include digit manipulation/stripping, screening, and application of treatment when processing the call.

Calls enter and leave the switch on trunks, which are consolidated into trunk groups, which in turn are grouped into routes associated with a translation plan. A route is a path to a destination, not just the next switch, and is defined as a prioritized list of trunk groups and/or interfaces (ISDN, SS7, CAS, SIP, BICC) identified by a route name. On the incoming side, the trunk group on which a call is received is associated with a translation plan that determines how the call is handled -- that is, whether the called number or other parameters will determine the routing. On the egress side, a route is a path to a final destination (the called number). In some cases, there will be several ways to reach the final destination, represented by different next-hop switches directly connected to the switch by trunk groups. Thus, all the trunk groups in a route do not necessarily terminate on the same switch. While each trunk group must be terminated on one far end switch, a route may have trunk groups terminating on several different switches, all representing possible paths to reaching the called number. This provides diversity in reaching the final destination in the event of adjacent switch congestion or trunk failure.

Figure 1 provides a sample of the information analyzed on an incoming call when being routed through the switch. The actual flow of the call and tables containing provisioning information is shown in Figure 2.

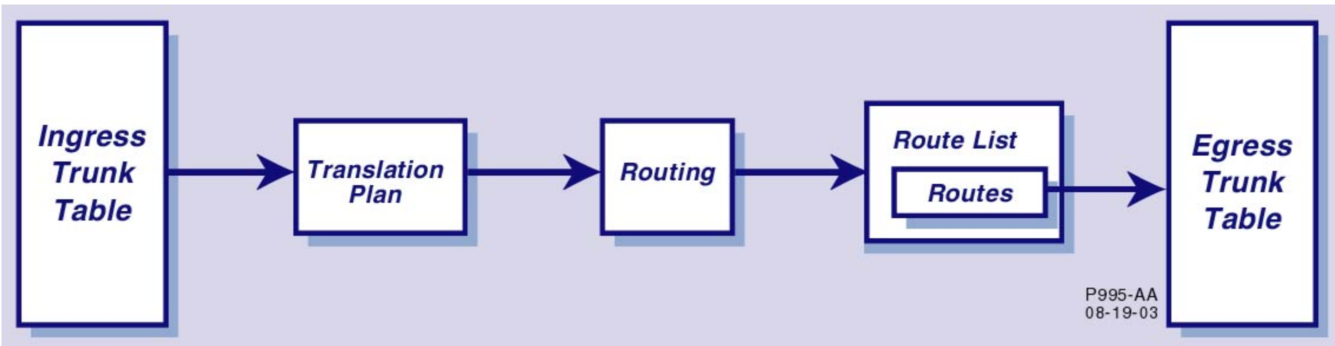
Figure 1. Routing a Call



Routing Flow

When a call comes into the switch, it follows the flow shown in Figure 2. Every ingress trunk group has a unique translation plan assigned to it. The translation plan can be specific to a particular trunk group or be reused as needed by many trunk groups.

Figure 2. Basic Routing Flow



The translation at a minimum defines the routing for the call and may also describe other actions that may be performed on the call, such as screening, digit manipulation, or application of a treatment rather than processing the call. The translation plan specifies up to three routing keys (such as TNS parameter or Called Number, for example) that may be used to route the call in order of priority. If the first key is not matched, then the second will be used, and then the third. In no key is matched, the translation plan specifies the treatment to be applied.

A successful match passes the call to the routing tables as shown in Figure 2. Here each possible key match is paired with a route destination, which will usually be a route list. For example, if the key were TNS, the routing tables would list every TNS supported and pair each with a route list or other destination. These other destinations could be other translation plans or treatments. The routing also allows a cost or weight to be specified for the call when selecting possible trunk groups from the route list.

The route list contains routes and their associated properties, including hunting algorithms, overflow treatment and re-routing instructions if no route is available. The route list is used to select routes from the route table. Percent allocation of calls among routes is also specified here.

The illustration shown in Figure 2 provides a high-level example of the basic flow of a call as it is routed through the switch. This illustration represents the flow of a call that has no special treatments or actions applied to it but, rather, is sent directly to the route list.

From the illustration, you can see that the Ingress Trunk Group Table points to the Translation Plan Table.

Each step in the process is depicted and represented by these Translation Tables:

- Ingress Trunk Table
- Trunk Group Table
- Translation Plan Table
- Routing Table
- Route List Table
- Routes Table
- Egress Trunk Table

The Translation Plan Table is shown in Table I and is represented in the switch by a database table containing various routing information.

Translation Plan

The Translation Plan Table lists up to three entries, or keys, on which to base the call for routing. The Translation Plan key can be any one of the following:

- Called Party Number (CDPN) – nature of address + digits
- Calling Party Number (CGPN) – nature of address + digits
- Charge Number (CHRGN) – nature of address + digits
- Generic Digits – (GENDGTS) – type + digits
- Generic Address – (GAP) – type + digits
- Calling Party Category (CPC) - value
- Originating Line Information (OLI) - value
- Bearer/Trunk Group (BGN) – type + number
- Transit Network Service (TNS) – CIC + CktCode/1NX/0ZZ + prot
- Carrier Identification Parameter (CIP) - CIC
- Carrier Selection Information (CSI) – (presubdial, notpresub)
- LATA – value
- Jurisdiction (JIP) - value
- Call Type (InterLATA, IntraLATA, International)
- Subscriber

When a call comes in on a trunk group, the Translation Plan does a search of the switch database based on the three keys entered into the Translation Plan. The search is conducted using the keys, in the order in which they are listed in the database. For instance, suppose that the three keys consist of the following:

- Route1-TNS
- Route1-CDPN
- Route2-CDPN

The switch first performs a search on Route1-TNS. If no match can be found for that key, then the switch proceeds to the next key and begins a search based on that criteria (Route1-CDPN, in this case). The switch searches the database until a match can be found for the key.

Note: In the various tables, you provision a parameter of NOTPRESENT in case no match can be found. When provisioning NOTPRESENT, you specify an alternate number to which the call can be placed in case a match cannot be found.

Once a match is found, the call is placed through routing, with possible treatments or actions applied, and sent to a Route List and then a Route, where it is then sent out through the Egress Trunk Group Table. This is all based on parameters previously provisioned for that particular Translation Plan.

An alternate Translation Plan key can also be entered. The alternate Translation Plan key overrides the original Translation Plan key and goes into effect as soon as it is entered into the Translation Plan Table.

Translation of the incoming call information ultimately results in the selection of a route to use to transport the call.

The route selected is also associated with the Translation Table via a Route key stored in the Translation Table. The Translation information, then, is used by the Routing Table to determine routing instructions. The Routing Table, using its Route Destination and analyzing the various parameters, knows what actions to perform on the route, or to send the call to a route list.

Often, an action, such as digit modification, will be performed on the incoming call. These actions are also used for screening Class 4 subscribers, performing digit modification for TNS, or applying a treatment, such as a cause code or announcement. Up to six actions can be applied to one call. For more information on various actions, refer to section Translation Plan Actions.

Refer to Table I for a list of Translation Plan parameters.

Routing

Along with Translation keys, the Routing key must be provisioned. The Routing key is used in addition to the Translation key to determine the NPA-NXX number. This information is taken from the Routing Table and is used as the second key when routing the call. From the Routing Table, routing instructions for different key and value matches can be found. The Routing key can consist of one of these four Route Destinations:

- Route List
- Next Routing Partition/Key
- New Translation Plan
- Treatment

This means that the call is either passed on to a Route List, sent back to a Translation Plan, or sent to the next Route Destination. The destination is determined by the Route Partition portion of the Route Destination key. The Route Partition is the set or characters prior to the first hyphen in the Route Destination key. For example:

RTLIST-SBC3 – RTLIST is the Route Destination.

Minimum and maximum costs of trunk groups are also determined by information entered into the Routing Table. Cost information for trunk groups enables you to include only trunks within a certain range, such as low-cost trunks, for example.

Note: Minimum and maximum cost criteria can also be entered into the Subscriber Table.

Refer to Table L for a description of Route Table parameters.

Route Partitioning

With the router, call processing supports partition-based routing. Routing partitions are assignable on the authorization code, ANI, CIC, trunk group, gateway or network level. If a match is not found in a particular partition, a default routing partition will apply.

Call processing also supports the redirection of a call from one routing partition to another. Instead of a route, the routing entry in the partition points to another partition. During a redirection from one partition to another, call processing allows manipulation of the dialed digits either partially or entirely prior to entering the new partition.

Route List

The Route List Table, as defined in Table K, lists the common properties associated with a list of routes. The hunting algorithm, or allocation method, for the route is provided in this table. Using a specified hunting algorithm against an ingress trunk group, the switch routing process selects an available egress trunk group and trunk from the database table to carry the call. Refer to Table K for a description of Route List parameters.

Overflow treatment of calls is also available through this table, enabling you to reroute calls or have two routes inside of one route list in case of overflow.

Routes

Once the Route List is identified, the call is sent to a specific route, as provisioned in the Route Table. The Route Table contains an ordered list of routes which support ISUP, SIP, CAS and BICC trunks, and ISDN lines. A route can also be sent to another Route List, as in the case of overflow, when calls need to be re-routed.

Refer to section 3.4 for a description of the parameters in the Route Table.

Trunk Groups

Calls enter and exit the via trunk groups. Incoming calls assume values set up in the Ingress Trunk table, and outgoing calls exit the switch based on routing criteria and information stored in the Egress Trunk table. Specifically, the Ingress Trunk table points to a Translation Plan, which passes the call through the switch with previously established values associated with the intended Translation Plan. Refer to Table H for an explanation of the Ingress Trunk Table.

The Egress Trunk table contains fields that help determine the outgoing reroute. Specifically, the information entered into the Digit Screen field identifies the outgoing trunk group screening table name. The cost of the trunk group also helps determine on what trunk the call will exit the switch, as well as the bearer cap, which identifies bearer capabilities for the trunk group. The information in the ActionTermAttmpt field identifies the action to take after the trunk group has been chosen on an outgoing interface for the call. Refer to Table T for an explanation of the Egress Trunk Table.

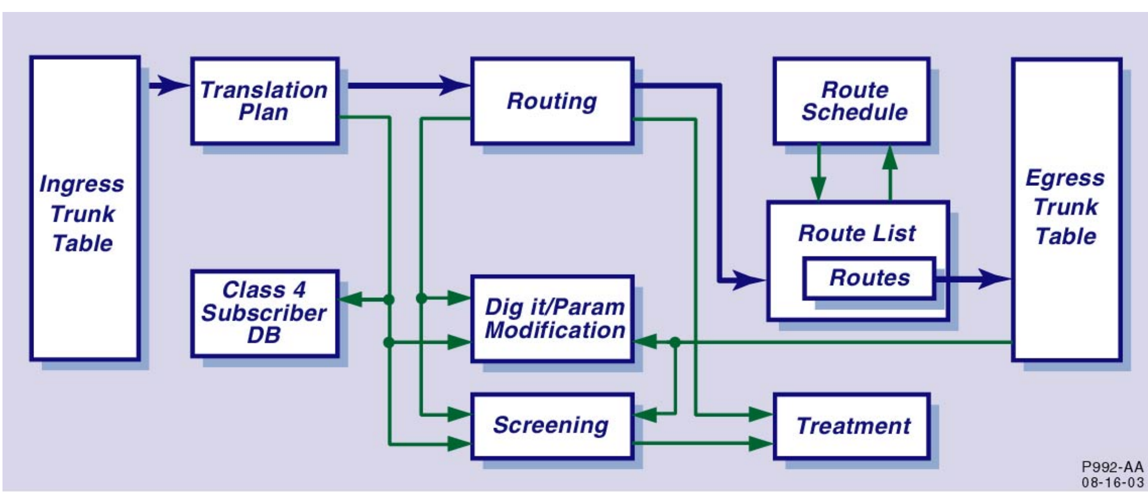
Translation Plan Actions

The Translation Plan assigns criteria to various calls, depending on the type of call. For instance, from the Translation Plan Table, these tables can be referenced:

- Class 4 Subscriber Table
- Screening Table
- Digit Modification Table
- Treatment Table

Refer to Figure 3 for a sample route flow which includes Class 4 subscriber parameters, screening, digit modification and special treatment.

Figure 3. Translation Plan Applying Class 4 Subscriber, Screening, Treatment and Digit/Param Modification Values



Class 4 Subscriber

Class 4 subscribers, which rely on aggregate ANI-based services and controls, are identified by the dialed number and call type or carrier. If the Translation Plan is to reference the Class 4 Subscriber Table, as shown in Figure 3, it is specified to do so in the Action Info Analyzed field of the Translation Plan Table.

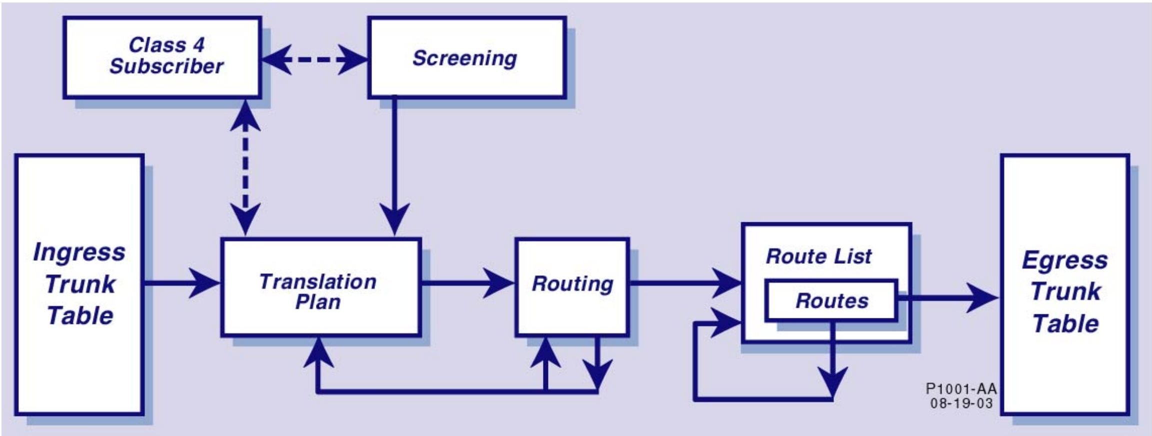
Information assigned to Class 4 subscribers is listed in Table M.

Other information specified in the Class 4 Subscriber Table includes:

- Authorization code and/or account code verifications
- Minimum/maximum trunk group cost
- Billing number
- Screening

Refer to Figure 4 for a sample of the call flow for routes referencing Class 4 subscriber parameters.

Figure 4. Class 4 Subscriber Screening



Call Screening

The Screening Table is also referenced by the Translation Plan Table. If the Translation Plan is to reference screening, as shown in Figure 5, it is specified to do so in the Info Collected or Info Analyzed fields of the Translation Plan Table.

Calls can be screened on the following information:

- Called Party Number (CDPN) – nature of address + digits
- Calling Party Number (CGPN) – nature of address + digits
- Charge Number (CHRGN) – nature of address + digits
- Generic Digits (GENDGTS) – type + digits
- Generic Address (GAP) – type + digits
- Calling Party Category (CPC) – value

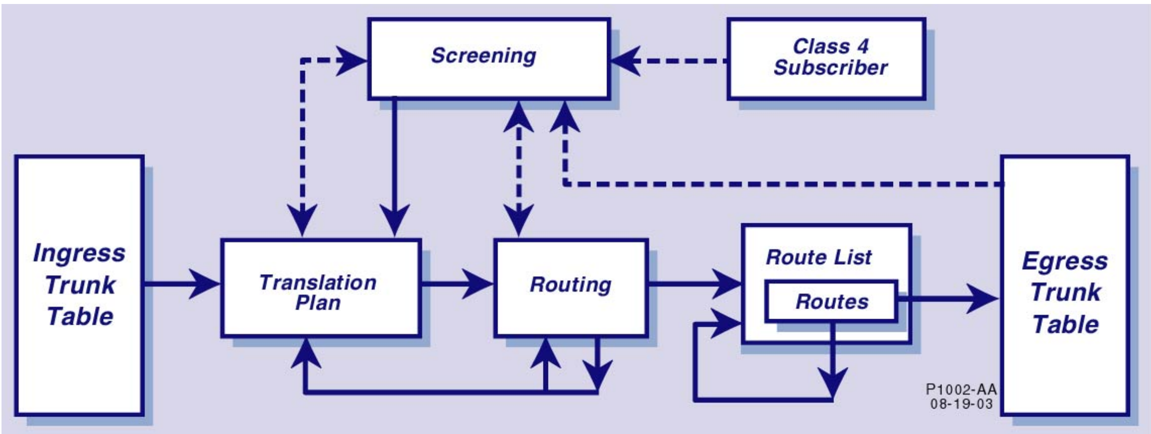
- Originating Line Information (OLI) – value
- Bearer/Trunk Group (BGN) – type + number
- Transit Network Service (TNS) – CIC+CktCode/1NX/0ZZ + port
- Carrier Identification Parameter (CIP) – CIC
- Carrier Selection Information (CSI) – presubdial, notpresub
- LATA – value
- Jurisdiction (JIP) – value
- Call Type (InterLATA, IntraLATA, International)
- SubStatus – IS, OOS, NOTINDB

Screening calls enables the router to perform a number of actions against a call if it meets certain criteria. Specifically, the router can perform these actions on a call:

- Pass/Fail
- Assign Treatment
- Screen for Authorization/Account Codes
- Mark as fraud call
- Assign new Translation Plan
- Send to Next Screen Table

Screening can also be referenced from the Routing Table and the Terminating Trunk Group. Screening keys are listed in Table N.

Figure 5. Screening



Digit Modification

Switch routing provides the ability to generically modify (strip/insert), replace or delete digits from the following:

- Called Party Number
- Calling Party Number
- Charge Number

- GAP
- Generic Digits
- CPC
- OLI
- JIP
- Call Type
- CIP
- LATA
- TNS

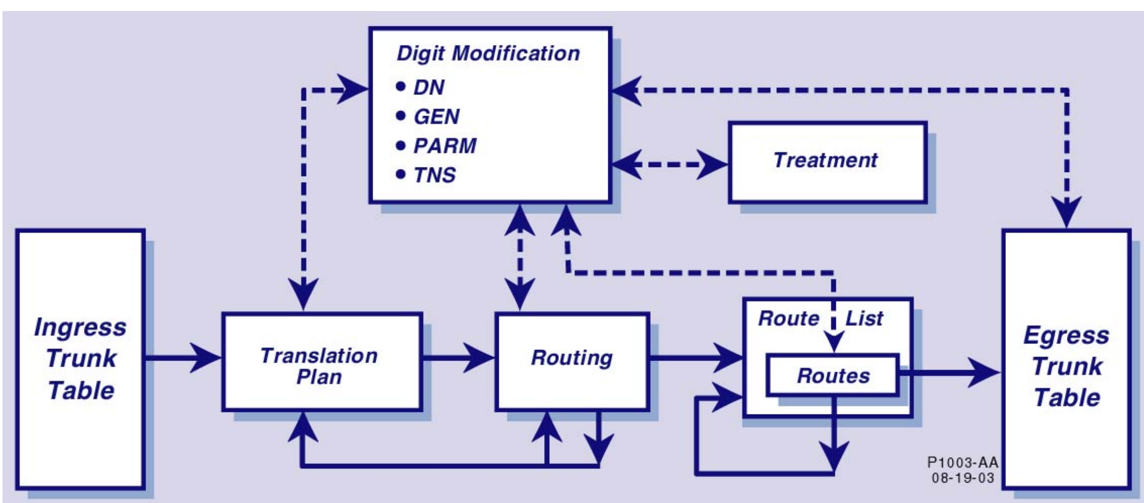
You can strip digits from the beginning, middle or end of a number. The switch is capable of prefixing or stripping digits to/from the called party number after selecting the route list and before sending the outgoing message. Stripped digits can be stored for later re-use.

Digit modification can be done on the following criteria, grouped into four separate tables:

- Dialed number
- Generic parameters
- TNS parameters
- Other parameters

Refer to Figure 6 for an illustrated example of call flow using digit modification.

Figure 6. Digit Modification Parameters



The Digit Modification, Dialed Number Table (Table O) provides the ability to generically modify (strip/insert), replace, or delete digits from the Calling Party Number, Called Party Number or Charge Number.

The Digit Modification, Generic Parameters Table (Table P) provides the ability to generically modify (strip/insert), replace, or delete digits from the GAP and Generic Digits.

The Digit Modification, Parameters Table (Table Q) provides the ability to generically modify (strip/insert), replace or delete digits from the following:

- CPC
- OLI
- JIP
- Call Type
- CIP
- LATA

The Digit Modification, TNS Table (Table R) provides the ability to generically modify (strip/insert), replace or delete digits from the TNS.

Treatment of Calls

A treatment, on the other hand, is used to route calls to another source for call handling. For instance, if you want to send a call to an announcement, you assign the call, from the Route List, to a tones/announcement box.

If there are no special treatments or actions to be performed against the call, the call is sent directly to a route list. It is then that the hunting algorithm is applied to the call.

The treatment of a call determines what happens to the call if it is to be handled other than by sending it to a route list. For instance, if a call is to be routed as an announcement, then the play tone/announcement treatment is applied. If a call is set up with a fail condition, then it will not be routed.

These are the treatments that can be applied to incoming calls:

- Play tone/announcement
- Failure condition
- Re-route (with digit modification, if required)
- Next treatment

Play Tone/Announcement

Play tone/announcement sends the call to a tones/announcement box where an announcement can be played. After the tone or announcement is played, the call can be released or sent to another treatment-specified parameter.

The announcement ID is stored in the Treatment Table and is entered using the RTRV-AUDIO-ANNC TL1 command. If the announcement requires variable arguments, such as time (TME_12, for instance), or phone number (DIG_NDN, for example), then you must provide the source of the variable field in the "annc args" of the treatment. For example, to create an announcement ID of 3000, provision that ID to play a time announcement and provide a treatment that references the announcement and specifies the time to be played as the current time, you would use the following TL1 command:

ENT-AUDIO-ANNC::3000:::name=dateandtime,desc=DATETIMEAUDIOFILE,msglist=TME_T12;

ENT-TREATMENT::TREATTOD:::TREATYPE=annc,anncid=3000,anncarg=TIME-CURRTIME;

As another example, you could create an announcement ID of 3001 which plays "<CalledNumber> is busy" and use the announcement in a treatment. As an example, you could use the following TL1 commands to use the announcement in a treatment.

ENT-AUDIO-ANNC::3001:::name=busymsg,desc=NUMISBUSY,msglist=DIG_NDN-28;

ENT-TREATMENT::TREATTOD:::TREATYPE-annc,anncid=3001,anncarg=CDPN;

Failure Condition

The release cause of a failed call is defined by a Failure Condition and Translation Plan, which enables you to enter overriding failure condition parameters. Specifically, the Failure Condition Table contains the fail type, treatment, release cause, release location and release standard for each route key configured in the Translation Plan Table. The Translation Plan Table also contains a Failure Condition Plan Profile which identifies the release failure profile which overrides the default failure condition profile for a Translation Plan.

Re-route

Re-routing of calls occurs during overflow. When overflow occurs, calls are re-routed to another Route List where they are cycled through the route process again.

Next Treatment

'Next Treatment' applies only to announcements (treatType=ANNC) and enables you to provision call processing to continue after an announcement ends. To provision for this, you assign CONTINUE to the nextTreat parameter. CONTINUE is only valid for screening.

Matching a Translation Plan to a Route Name

Ingress trunks are matched to a Translation Plan via the Ingress Trunk Table. In this table, a trunk group is assigned to a specific Translation Plan. The plan, then, corresponds to a Route Label and Key.

In the simplified view of the Translation Table, the Plan ID is associated with at least one Route Label and Key; however, you can enter up to three Route Label/Key combinations. The router searches on the first Route Label/Key entered, and continues searching until a match is found. If a match is not found for the first key, then the router moves on to the second key. If a match is not found for the second key, the router searches on the third key. The other fields in the Translation Plan Table are used with key fields in the Routing Table to determine the destination of the call. Table A illustrates a sample of the Translation Plan Table.

Table A. Sample Information Taken from the Translation Plan Table

Plan ID	Route Label/Key	Translation Modifiers
Basic	SBC-TNS & BOSTON-CDPN & ASIA-GENDGTS	●●●●●●

The Route Label/Key then, is matched with the Routing table, defined in Table B. This table contains a list of label keys/values, route destinations and corresponding actions. This simplified view highlights the two primary parameters involved: the Label and the route destination to which it points. In the switch, every label key must be paired with a Route Destination in order to route calls.

Table B. Linking the Routing Table to the Route List Table

Label Key and Values	Route Destination	Route Modifiers
SBC-TNS-0211-DEFAULT	RTLIST-SBC	
BOSTON-CDPN-INTNATNUM-91	RTLIST-BOSTON	
ASIA-GENDGTS-DEFAULT	TRANS-ASIA	●●●●●●

In the Table B example, the label key is used as the route key, and every route destination to which the switch can route calls is entered in the table and associated with a label key.

The rest of the columns shown in Table B consist of a large number of parameters labeled Route Modifiers. These Route Modifiers define how the route will be used for that specific Route Destination only. Keep in mind that all information stored in the

Route lists associated with a route label are identified in the Route Destination field of the Routing Table. Details for each route list identified in the Route Table are stored in the Route List Table. In Table B above, you can see that SBC-TNS-0211-DEFAULT is associated with RTLIST-SBC. In Table C, below, you can see that RTLIST-SBC is assigned an allocation method of circular and that overflow treatment does not apply. For BOSTON-CDPN-INTNATNUM-91, the associated route list is RTLIST-BOSTON, the allocation method is Distribution and overflow treatment does apply. This means that Overflow Treatment is used if the call set up on this route fails because no bearers are available.

Table C. Linking the Translation Plan Table to the Routing Table

Route List Name	Allocation Method	Overflow Treatment
RTLIST-SBC	Circular	
RTLIST-BOSTON	Distribution	Y

A route list is then associated with a route, where the bearer group type and group number are assigned to the route. This information, combined with information stored in the Egress Trunk Table, help determine where the call is finally to be routed. Refer to Table D to see the Route Table fields.

Table D. Linking the Routing Table to the Route List Table

Route List Name and Index	Bearer Group Type	Group Number	Route List ID	Digit Modify ID	Weight
RTLIST-SBC-1	TGN	203			
RTLIST-BOSTON-20	ISDNIF	1400			

Time of Day Routing

Route Lists can be provisioned to point to Time of Day routing parameters. When TIMEOFDAY is selected as the allocation method on the Route List Table, the router accesses the Route Schedule Table, as shown in Table E.

In the switch, using a Profile Name enables scheduling of different routes for an incoming call based on time of day for each day of the week and for specific dates. Table E, for example, illustrates various profile names and the route lists with which they are associated.

Table E. Time of Day Route Schedule Table

Route List Name	Profile Name	Route List
RL_TOD	TODP_Holidays	RL_WCA
RL_TOD	TODP_Weekdays	RL_WCA
TL_TOD	TODP_Weeknights	RL_VZ
RL_TOD	TODP_Weekends	RL_VZ
RL_TOD	TODP_Default	RL_VZ

Table F shows each profile and the date, day and time each profile is to be applied to the calls associated with the matching Route List.

Profiles may be created either by time of day/day of week or for specific days of the year, allowing different routing treatment for days such as holidays.

Table F. Time of Day Schedule Profile Table

Profile Name	Date/Day/Time
TODP_Holidays	12/25,10/31,1/1
TODP_Weekdays	MTWRF-6am-6pm
TODP_Weeknights	MTWRF-6pm-6am
TODP_Weekends	SS-all hours
TODP_Default	Add days, all hours

Matching a Route List Name to a Bearer Group

A selected route, in turn, points to a Route List Table that lists all route lists and the allocation (hunting algorithm) used to route each call. Each route list consists of routes that exist in the Route Table. The Route List Name and Index in the Route List correspond with the Route List ID, which determines which route list the call will use. The Bearer Group Type and Bearer Group Number are also key fields in the Route Table which help to determine over what interface or protocol (CAS, ISUP, GR-303, for instance) the call will be routed.

If a bearer group and group number are not used to route the call, then the call is assigned to a route list ID. Specifications within the route list are what is used to route the call.

The Route Table contains an ordered list of trunk groups, lines and other routes.

Table G. Route Table

Route List Name and Index	Bearer Group Type	Group Number	Route List ID
Route1-1	TGN	203	
Route1-10	ISDNIF	14000	
Route1-20			Route2

Routing Tables

The following tables display the various parameters and information needed to successfully route a call through the switch. These tables are described in this section:

- Ingress Trunk Table
- Translation Plan Table
- Routing Table
- Route List Table
- Route Table
- Class 4 Subscriber Table
- Screening Table
- Digit Modification, Dialed Numbers Table
- Digit Modification, Generic Table
- Digit Modification, Parameters Table
- Digit Modification, TNS Table
- Treatment Table
- Egress Table

Ingress Trunk Table

Incoming calls assume values set up in the Ingress Trunk Table, in order to be routed through the switch. The Ingress Table (Table H) associates a trunk group with a translation plan.

Table H. Ingress Trunk Table

Field	Default Value	Description
trkGrpNum	Default Value:	The number of the incoming trunk group to be associated with a Translation Plan.
plan		The Translation Plan name.
dgtScrn	Default Value: NULL	Identifies the outgoing trunk group screening table name, previously defined with ENT-DIGIT-SCREEN and which contains a list of restricted trunk groups. This parameter prevents incoming/outgoing trunk combinations
routeCost	Default Value: 0	Identifies the cost of the trunk group. The value 101 indicates that the trunk group is not available for routing.
bearerCap	Default Values: SPEECH&UNRESDIGITAL&VIDEO&DIGITALWITH TONES (for BICC/ISUP/SIP)	Identifies the bearer capabilities for the trunk group, with ampersands (&) to separate multiple bearer capabilities.
ActionTermAttmpt	Default Value: NULL	Identifies the action to take after the trunk group has been chosen as an outgoing interface for the call.
UseSwitchID	Default Value: N	Identifies whether to enable DMS switch identification routing/transport.

Translation Plan Table

Table I shows the complete set of parameters that can be entered as modifiers in the Translation Plan Table. These parameters modify the behavior of the route with which they are associated. Digit manipulation and screening is also available at this level. This allows digits to be added or deleted to the DN when this particular Trunk Group is accessed by this Route.

Table I. Translation Plan Table

TRANSLATION PLAN		
Field	**Default Value**	**Description**
plan	Default Value:	The dial or numbering plan to be applied before the call is set up.
actnInfoCol	Default Value: NULL	Action Information Collected is done before number analysis/dial plan analysis and defines the various manipulations that need to be performed on this partition. The string format is specified as a separate section, named Action String.
actnInfoAnlz	Default Value: NULL	Action Information Analyzed is performed prior to AIN trigger analysis and defines the various manipulations that need to be

		performed on this partition. The string format is specified as a separate section named Action String.
RtLbl & rtKey	Default Value:	This defines the Routing Table label, that is, the Routing key used for routing. This key combination must exist in the ROUTE-DIGITS Table.
rlsFailCndPrfl	Default Value: NULL	Any exceptions defined in the Fail Condition Profile Table will override the default exception behaviors. This must be previously defined using the PRFL-FAILCND command.
altPlan	Default Value: NULL	Alternate Translation Plan. If this is not NULL, the parameters in this plan are ignored, and the parameters in the identified plan are used instead. This can be used for emergency procedures.

Routing Table

Table J represents the Routing Table. This table defines parameters necessary to route the call to its destination. The Row ID contains a partition that reflects the Translation Plan to be applied.

Table J. Routing Table

ROUTING		
Field	**Default Value**	**Description**
rowId	Default Value:	Consists of the Translation Plan ID, type of call, type of number.
rtDest		Route Destination must be defined as a Translation Plan, Treatment ID, or Route Partition and key to use.
actn	Default Value: NULL	The various actions to be performed on values that match this route.
minCost	Default Value: 0	The minimum cost of the outgoing trunk group.
maxCost		The maximum cost of the outgoing trunk group.
reRoute		Indicates whether calls getting to this route are rerouted on call setup failure on a downstream switch. Options are: Y, N

Route List Table

The keys in the Route List Table (Table K) identify the hunting algorithm to be applied to the call, as well as whether or not to support call overflow.

Table K. Route List Table

ROUTE LIST		
Field	**Default Value**	**Description**
rtListName	Default Value:	The Route List Name is the name of the route list associated with a specific hunting algorithm. All numbers entered into this route list are treated in the same manner.
alloc		The Allocation Method indicates the hunting algorithm to be used when searching for a route. Options are: • Ascending • Descending • Circular • % Distribution • Least Cost • TimeofDay • Random
ovrFlowTreat	Default Value: NULL	An entry in this field determines whether or not overflow treatment applies to calls included in this route list. Overflow Treatment should be used if the call set up on this route fails because no bearers are available.

Route Table

The keys in the Route Table (Table J) identify the Bearer Group Interface Type and Number.

Table L. Route Table

ROUTE		
Field	**Default Value**	**Description**
RtList & index	Default Value:	The Route List name and Index identify the name of the route list to which this route belongs or to which this route entry points.
bgnType		The Bearer Group Interface Type. The Bearer Group must be provisioned before the interface type. The trunk group number covers

		ISUP, BICC, CAS and SIP trunks. Options are: CASIF GR303 ISDNIF MGCP TGN
bgnNum	Default Value:	The Bearer Group Number covers ISUP, BICC, CAS and SIP trunks. Options are: CASIF range: 1-2147483647 GR303 range: 1-32767 ISDNIF range: 1-32767 MGCP range: 1-10000 TGN range: 1-9999
rtList	Default Value: NULL	The Route List entry points to another route that needs to be used. It cannot be used when the Bearer Group Number is used.
DgtModKey		The optional Digit Modification key is used to change parameters that reach this route.
weight	Default Value: 10	This field indicates the relative weight of this route, if the hunting method for the route list is set to DISTRIBUTION.

Class 4 Subscriber Table

The keys in the Class 4 Subscriber Table (Table M) identify the number of an incoming subscriber call, and the carrier for which the subscriber assignment parameters apply.

Table M. Class 4 Subscriber Table

CLASS 4 SUBSCRIBER		
Field	**Default Value**	**Description**
SubscriberId & callType	Default Value:	The Subscriber ID is the NPA-NXX-XXXX of the subscriber. The call type reflects the call type for which the subscriber assignment parameters apply. Options are: InterLATA IntraLATA International
carrier	Default Value:	This field indicates the carrier for which the subscriber assignment parameters apply.

subStatus		This field indicates the status fo the subscriber. Options are: ACTIVE INACTIVE

Screening Table

The Screening Table (Table N) lists the parameters for screening for authorization, blocking and identification of fraud calls.

Table N. Screening Table

SCREENING		
Field	**Default Value**	**Description**
resultType	Default Value: FAIL	The Result specified what, of the following actions, has to be done. Options are: PASS – call has passed screening FAIL – call will be disconnected TREATMENT – specified treatment will be performed AUTHCODE – call must be authorized – can only be used if called from InfoAnalyze SCREEN – more screening needs to be done TRANSPLAN – change translation plans during a screening step. Only valid if the screening is called as an InfoCollected action.
fraudTrapPrfl	Default Value: NULL	This field represents the fraud trap profile to use if this is a fraud situation.
treatment	Default Value: NULL	This field cannot be NULL if the result type is TREATMENT. It should not be specified otherwise.
dgtScrnKey	Default Value: NULL	This field indicates the next screening operation to be performed. This field cannot be NULL if the result type is SCREEN. It should not be specified otherwise.
authPrompt	Default Value: ANNC	This field specifies the type of prompt for the user to enter the authorization code. This can be either a tone or an announcement

		message. Options are: 350 400 ANNC
authMode	Default Value: NONE	This field indicates the mode of authcode operation, which can be: - One pinCode per one authCode - Many pins per one - Only authCodes - Only pinCodes - Account code length >0 - Account code length >0 and authCodes cannot be NONE if result type is AUTHCODE Options are: ONEPIN ONLYAUTH NONE ONLYPIN ONLYACNT ACNTAUTH
AcntCodeLen	Default Value: 0	The length of the account code. 0 implies this option is disabled. The entry in this field cannot be 0 if authMode is ONLY ACNT or ACNT AUTH.
AuthList	Default Value: NULL	An authorization list is created previously with ENT-LIST-AUTHCODE. The entry in this field cannot be NULL if authMode is ONEPIN, ONLYAUTH, ONLYPIN or ACNTAUTH.
Transplant	Default Value: NULL	If specified, this is the translation plan that is to be used on a screening match. The entry in this field cannot be NULL if resultType is TRANSPLAN. It should not be specified otherwise.

Digit Modification Tables

The Digit Modification Tables consist of these tables:

- Dialed Numbers
- Generic
- Parameters
- TNS

Digit Modification, Dialed Numbers Table

The Dialed Numbers Table (Table O) identifies the Nature of Address Indicator, modification type, strip digits and insert digits.

Table O. Digit Modification, Dialed Number Table

DIALED NUMBER		
Field	**Default Value**	**Description**
dnLbl	Default Value:	The digit modification label.
nai	Default Value: DEFAULT	The nature of address indicator, different for the three types.
minDgts	Default Value: 0	The minimum length of the incoming addressing digits (CDPN, CGPN, CHRGN) for this rule to be matched.
modType	Default Value: NOACTION	The type of manipulation to perform on the primary key. This value validates the rest of the parameters present. Options are: STRIPONLY – removes the number of digits specified by STRIPDGTS parameter STRIPINSERTDGTS – removes the number of digits and inserts new digits STRIPINSERTSRC – removes the number of digits and inserts from the specified source INSERTDGTS – inserts specified digits INSERTSRC – inserts digits from the specified source REPLACEDGTS – replaces digits with those specified in the INSERTDGTS parameter REPLACENAI – used in cases where the digits are not changed, just the NAI. DELETE – removes the parameter NOACTION – no action taken.
altModType	Default Value: NOACTION	The type of manipulation to perform on the alternate parameter. Options are: REPLACE DELETE NOACTION
maxDgts	Default Value: 31	Maximum length to match this rule.
stripDirn	Default Value:	Specifies where to start from before

	LEFT	moving StripPos number of digits over. LEFT starts at the leftmost digit and works inwards to the right. RIGHT starts at the right most digits and works inwards to the left.
stripPos	Default Value: 0	Specifies the digit position in the string where stripping will begin (for example, 1 means starting at the first digit). This entry cannot be 0 when using STRIP mode.
stripDgts	Default Value: 0	The number of digits to strip. This entry is ignored if only inserting. This entry cannot be 0 when using STRIP mode.
stripDest	Default Value: NULL	Identifies a register or parameter to store the stripped digits. REG1-3 are scratchpad registers for holding digits during a translation. These registers are only valid during one call translation, that is, they cannot be passed from one call to another.
insertDirn	Default Value: LEFT	Identifies whether to insert on the left side or the right side of the specified position.
insertPos	Default Value: 0	Specifies the digit position in the string where inserting will begin (for example, 1 means starting at the first digit). This entry cannot be 0 when using INSERT mode.
insertDgts	Default Value: NULL	Specifies digits to insert. This entry cannot be NULL when using INSERT mode.
insertSrc	Default Value: NULL	Identifies another parameter/call-related value to prefix. Options are: OLI CPC REG1 REG2 REG3
newNai	Default Value: NULL	Specifies the new NAI for this number. NULL leaves the NAI unchanged.

altParamValue	Default Value: NULL	A key and value. This is a way of changing non-key parameters on a match on the key field. This entry cannot be NULL if altModType is REPLACE.
nextDgtMod	Default Value: NULL	The digit modification table label to call next on a match for this row. This entry cannot be NULL when both modType and altModType are NOACTION.

Digit Modification, Generic Table

The Generic Table (Table P) identifies the parameters used for identifying calls with generic digits or generic address parameters.

Table P. Digit Modification, Generic Table

GENERIC		
Field	**Default Value**	**Description**
dmLbl	Default Value:	The digit modification table label.
genType		The key to use for matching. Options are: GAP GENDGTS
nmbrType		The key to use for matching. Options are: 0-31 for GAP 0-255 for GENDGTS
digits	Default Value: DEFAULT	The leading digits of the number to match. Depending on the type, it could be hexadecimal, ASCII or BCD.
modType	Default Value: NOACTION	The type of manipulation to perform on the primary key. This value validates the rest of the parameters present. STRIPONLY – removes the number of digits specified by STRIPDGTS parameter STRIPINSERTDGTS – removes the number of digits and inserts new digits STRIPINSERTSRC – removes the number of digits, inserts digits from the specified source INSERTDGTS – inserts specified digits INSERTSRC – inserts digits from

		the specified source REPLACEDGTS – replaces digits with those specified in the INSERTDGTS parameter DELETE – removes the parameter NOACTION – no action is taken
altModType	Default Value: NOACTION	The type of manipulation to perform on the Alternate parameter. Options are: REPLACE – replaces the parameter digits specified in AltParam DELETE – removes the parameter specified in AltParam NOACTION – no action is taken
stripDirn	Default Value: LEFT	Specifies where to start from before moving StripPos number of digits over. LEFT starts at the leftmost digit and works inwards to the right. RIGHT starts at the rightmost digits and works inwards to the left.
stripPos	Default Value: 0	Specifies the digit position in the string where stripping will begin (for example, “1” means starting at the first digit). The entry cannot be 0 when using STRIP mode.
stripDgts	Default Value: 0	The number of digits to strip. This entry is ignored if only inserting. The entry cannot be 0 when using STRIP mode.
stripDest	Default Value: NULL	Identifies a register or parameter to store the stripped digits. REG1-3 are scratchpad registers for holding digits during a translation. These registers are only valid during one call translation; that is, they cannot be passed from one call to another.
insertDirn	Default Value: LEFT	Identifies whether to insert on the left side or the right side of the specified position.
insertPos	Default Value: 0	Specifies the digit position in the string where inserting will begin (for example, “1” means starting at the first digit). The entry cannot be 0 when using INSERT mode.
insertDgts	Default Value: NULL	Digits to insert. The entry cannot be NULL when using INSERT mode.
insertSrc	Default Value: NULL	Identifies another parameter/call-related value to prefix. Options are: NULL OLI CPC REG1

		REG2 REG3
altParamValue	Default Value: NULL	A key and value. This is a way of changing non-key parameters on a match on the key field. The entry cannot be NULL if altModType is REPLACE.
newValue	Default Value:	Specifies the new value the parameter key should take. If the newValue is equal to NONE, the key is removed. The length is 2 for CPC/OLI, 4 for CIP/PCIC. For TNS if it is 4, only the carrier value is changed; if it is 7, the carrier and OZZ/CktCode values are changed. Other lengths are invalid.
nextDgtMod	Default Value: NULL	The digit modification table label to call next on a match for this row. This entry cannot be NULL when both modType and altModType are NOACTION.

Digit Modification, Parameters Table

The Parameters Table (Table Q) lists the parameters used for identifying calls with various call parameters.

Table Q. Digit Modification, Parameters Table

PARAMETERS		
Field	**Default Value**	**Description**
dmLbl	Default Value:	The digit modification table label.
modType	Default Value: NOACTION	The type of manipulation to perform. It validates the other non-null parameters. REPLACE: replaces the primary key. Cannot be used for CALLTYPE or BGN; that is, modType=NOACTION for these two keys. DELETE: removes the parameter NOACTION: skip this digit medication
altModType	Default Value: NOACTION	The type of manipulation to perform on the Alternate parameter. REPLACE: replaces the parameter digits specified in AltParam DELETE: removes the parameter specified in AltParam NOACTION: skip this digit modification

newValue	Default Value: NULL	The new value for this parameter. Cannot be specified for CALLTYPE or BGN; that is, you cannot change these two parameters. This entry cannot be NULL when modType is REPLACE.
altParamValue	Default Value: NULL	A key and value. This is a way of changing non-key parameter on a match on the key field. This entry cannot be NULL when altModType is REPLACE.
altValue		Specifies the new value the parameter key should take. Options are: TNS 0-9999255 CSI: PRESUBDIAL PRESUBNOTDIAL PRESUBUNKNOWN NOTPRESUB UNKNOWN
nextDgtMod		The digit modification table label to call next on a match for this row. This entry cannot be NULL when modType and altModType are both NOACTION.

Digit Modification, TNS Table

The TNS Table (Table R) identifies the parameters used for the transit network selection (TNS), which is the circuit code for ISUP calls, and the Carrier or 0ZZ codes for CAS Feature group D.

Table R. Digit Modification, TNS Table

TNS		
Field	**Default Value**	**Description**
dmLbl	Default Value:	The digit modification table label.
callTypeCode		The CAS 0ZZ,1NX code or ISUP Circuit Code. DEFAULT means CallType ignored.
prtcl		The incoming trunk group protocol.
modType	Default Value: NOACTION	The type of manipulation to perform. It validates the other non-null parameters. REPLACETNS: replaces both the Carrier ID and the Call Type Code REPLACECARRIER: replaces the Carrier ID REPLACECALLTYPE: replaces the Call Type Code DELETE: removes the parameter NOACTION: skip this digit

		modification
altModType	Default Value: NOACTION	REPLACE: replaces digits with those specified in the newTns parameter DELETE: removes the parameter NOACTION: skip this digit modification
newCarrier	Default Value: NULL	The new value of TNS on matching the TNS-0ZZ-prtcl. NULL implies no change. Can't be NULL for modTypes REPLACETNS and REPLACECARRIER
newCallTypeCode	Default Value: NULL	The new value of 0ZZ code or circuit code. Can't be NULL for modTypes REPLACETNS and REPLACECALLTYPE
altParamValue	Default Value: NULL	A key and value. This is a way of changing non-key parameter on a match on the key field. Can't be NULL if altModType is REPLACE
altValue		Specifies the new value the parameter key should take. Options are: CPC, OLI: 00-FF TNS: 0-9999255 CIP: 0-9999 CSI: PRESUBDIAL PRESUBNOTDIAL PRESUBUNKNOWN NOTPRESUB UNKNOWN
nextDgtMod	Default Value: NULL	Refer to section 1.1 for parameter details. The digit modification table label to call next on a match for this row. Can't be NULL when both modType and altModType are NOACTION

Treatment Table

The Treatment Table (Table S) contains the parameters that identify whether a call should receive tone/announcement treatment, failed condition, or some other treatment.

Table S. Treatment Table

TREATMENT		
Field	**Default Value**	**Description**
treatmentId	Default Value:	The identification of this treatment.
treatType		The type of treatment that is to be provided. ANNC: play an announcement REROUTE: Use the dgtModKey to reroute FAILCND: A condition to be generated.
anncId	Default Value: NULL	The ID of the announcement to be played. The announcement IDs must be in the range of the internal switch announcements (must be predefined by AUDIO-ANNC or ANNC-EXT). Only valid when treatType = ANNC.
failCnd	Default Value: NULL	The condition, previously defined in the condition table, triggered for this treatment. Use RTRV-FAILCND to determine the existing conditions. Usually used with routing to a treatment, which then uses a condition. This entry cannot be NULL if treatType is FAILCND.
dgtModKey	Default Value: NULL	The digit modification object that is used to change parameters for a reroute. Only valid when treatType = REROUTE.
nextTreat	Default Value: NULL	Only valid when treatType = ANNC. CONTINUE is a “special” treatment that lets call processing continue after the announcement ends. CONTINUE is only valid for screening.

Egress Trunk Group Table

The Egress Trunk Group Table (Table T) associates a trunk group with a route screen, bearer cap and cost.

Table T. Egress Table

EGRESS		
Field	**Default Value**	**Description**
trkGrpNum	Default Value:	The number of the outgoing trunk group.
dgtScrn	Default Value: NULL	Identifies the outgoing trunk group screening table name, previously defined with ENT-

		DIGIT-SCREEN and which contains a list of restricted trunk groups. This parameter prevents incoming/outgoing trunk combinations
routeCost	Default Value: 0	Identifies the cost of the trunk group. The value 101 indicates that the trunk group is not available for routing.
bearerCap	Default Values: SPEECH&UNRESDIGITAL&VIDEO&DIGITALWITHTONES (for BICC/ISUP/SIP)	Identifies the bearer capabilities for the trunk group, with ampersands (&) to separate multiple bearer capabilities.
ActionTermAttmpt	Default Value: NULL	Identifies the action to take after the trunk group has been chosen as an outgoing interface for the call.

Routing Scenarios

Re-routing of Incoming CGPNs

Scenario: The switch supports re-routing of specific incoming Calling Party Number (CGPNs) to alternate destinations, including treatment to customized announcements on a Carrier Identification Code (CIC) basis.

Conditions: Treatment1 sends a call to announcement 3000 and TreatFail1 fails the call if there is a vacant code. If the fail condition is applied, the call is released to PRIVNETRU-CCITT.

1. Establish routes for various trunk groups, assigning route lists to each. For example:

   ENT-ROUTE::RL18-1:::BGN=TGN-1029;
   ENT-ROUTE::RL19-1:::BGN=TGN-1030;
   ENT-ROUTE::RL20-1:::BGN=TGN-1020;

2. Create announcement 3000 using this command:

   ENT-AUDIO-
   ANNC::3000:::name=dateandtime,desc=DATETIMEAUDIOFILE,msglist=DAT_MD-TME_T12;

   OR

   ENT-ROUTE-DIGIT::RD49-TNS-0234:::RtDest=RTLIST-RL18;

3. Create a profile for the fail condition for VACANT_CODE using this command:

   ENT-PRFL-FAILCND::PCICFAIL1-VACANT_CODE-ISUP:::FailType=RELEASE,Release=2-
   PRIVNETRU-CCITT;\

4. Define TreatFail1 using this command and applying the conditions stated for this scenario:

   ENT-TREATMENT::TreatFail1:::TreatType=FAILCND,failcnd=VACANT_CODE;

5. Define Treatment1 using this command and applying the conditions stated for this scenario:

   ENT-TREATMENT::Treatment1:::TreatType=ANNC,AnncId=3000,ANNCARGS=DATE-
   CURRDATE&TIME-CURRTIME,NextTreat=TreatFail1;

6. Apply route digit destinations to a specific CGPN (5086808001) and to the default, and to the TNS of 0234 and the default. For example:

   ENT-ROUTE-DIGITS::RD48-CGPN-DEFAULT:::RtDest=RTLIST=RL11;
   ENT-ROUTE-DIGITS::RD48-CGPN-UNINATNUM-5086808001:::RtDest=RTKEY-RD49-TNS;
   ENT-ROUTE-DIGITS::RD49-TNS-DEFAULT:::RtDest=RTLIST=RL19;

ENT-ROUTE-DIGITS::RD49-TNS-0234:::RtDest=TREAT-Treatment1;

7. Create translation plan 47, establishing a release fail condition profile. For example:

   ENT-TRANS-PLAN::TP47:::rtKeyList=RD48-CGPN,RLSFAILCNDPRFL=RTFAILED;

8. Enter a trunk group and associate it with Translation Plan 47 using this command:

   ENT-TRKGRP::46:::ISUP:Name=TGN46,DPC=2-2-2,TGPROFILE=1,transplant=TP47;

9. Establish SS7 trunk 46 using this command:

   ENT-SS7-TRK::46-46&&-46:::IOMPORTDS0=IOM-3-T3-2-PORT-3-T0-22;

Time of Day Routing

Scenario: The switch supports Time of Day, Day of Week and Holiday routing filters that can be applied post routing.

1. Assign specific days and/or dates with specific times to each schedule. For example:

   ENT-PRFL-SCHED::SchWkEND:::SAT=0000-2400,SUN=0000-2400;
   ENT-PRFL-SCHED::SchHDAY:::DATE1=01-01,DATE2=07-04,DATE3=05-26,DATE4=09-01,DATE5=11-23,DATE6=11-24,DATE7=12-25,TIME=0000-2400;
   ENT-PRFL-SCHED::SchOPK1:::MON=0000-0700,TUE=0000-0700,WED=0000-0700,THU=0000-0700,FRI=0000-0700;
   ENT-PRFL-SCHED::SchPEAK::: MON=0000-2100,TUE=0000-2100,WED=0000-2100,THU=0000-2100,FRI=0000-2100;
   ENT-PRFL-SCHED::SchOPK2::: MON=2100-2400,TUE=2100-2400,WED=2100-2400,THU=2100-2400,FRI=2100-2400;

2. Designate route list RLSCHED1 as a Time of Day list using the following command:

   ENT-ROUTE-LIST::RLSCHED1:::ALLOC=TIMEOFDAY;

3. Assign different schedules already associated with RLSSCHED1 to various destination route lists. For example:

   ENT-ROUTE-SCHED::RLSCHED1-SchWkEND:::RTLIST=RTLIST17;
   ENT-ROUTE-SCHED::RLSCHED1-SchHDAY:::RTLIST=RTLIST18;
   ENT-ROUTE-SCHED::RLSCHED1-SchOPK1:::RTLIST=RTLIST19;
   ENT-ROUTE-SCHED::RLSCHED1-SchPEAK:::RTLIST=RTLIST20;
   ENT-ROUTE-SCHED::RLSCHED1-SchOPK2:::RTLSIT=RTLIST21;
   ENT-ROUTE-SCHED::RLSCHED1-SchDEFAULT:::RTLIST=RTLIST22;

4. Associate the route schedule RLSSCHED1 with a Called Party Number default. For example:

   ENT-ROUTE-DIGIT::RD36-CDPN-DEFAULT:::RtDest=RTLIST-RLSCHED1;

5. Associate a Translation Plan with a route destination for a Called Party Number. For example:

   ENT-TRANS-PLAN=TP37:::rtKeyList=RD36-CDPN;

6. Enter an ISUP trunk number of 36, assigning a DPC of 2-2-2, a trunk group profile of 1, and a Translation Plan of TP37. For example:

   ENT-TRKGRP::36:::ISUP:Name=TGN36,DPC=2-2-2, TGPROFILE=1,transplan=TP37;

Route Plans and Route Schedules

Scenario: The switch can support different routing plans depending on the Time, Date or holidays assigned to a route schedule.

1. Enter specific routing information for any Called Party, Calling Party, TNS, etc., associating it with a specific route list containing specific route schedules and times. For example:

   ENT-ROUTE-DIGITS::RD70-CGPN-UNINATNUM-5084808001:::rtdest=RTLIST-RL23;
   ENT-ROUTE-DIGITS::RD70-CGPN-UNINATNUM-DEFAULT:::rtdest=RTLIST-RL25;
   ENT-ROUTE-DIGITS::RD18-CDPN-NATNUM-908:::RTDest=RTLIST-RL11;
   ENT-ROUTE-DIGITS::RD18-CDPN-NATNUM-DEFAULT:::RTDest=RTLIST-RL12;
   ENT-ROUTE-DIGITS::RD19_1-TNS-111:::RTDest=RTLIST-RL13;
   ENT-ROUTE-DIGITS::RD19_1-TNS-DEFAULT:::RTDest=RTLIST-RL14;

2. Assign specific digit modification parameters to Called Party Number 408-8001. Other sample input which can be provided with this command is shown in the sample command:

   ENT-DIGITMOD-DN::REP908-CDPN-NATNUM-4088001:::Modtype=STRIPINSERTDGTS,strippos=1,stripdgts=3,InsertPos=1,insertDgts=908,maxdgts=31;

3. Assign a route list to each translation plan. For example:

   ENT-TRANS-PLAN::TP5013:::rtKeyList=RD19_1-TNS;
   ENT-TRANS-PLAN::TP5014:::actninfocol=DgtModKey-PRE908-CDPN,rtKeyList=RD18-CDPN;
   ENT-TRANS-PLAN::TP5015:::rtKeyList=RD70-CGPN;

   Note that a digit modification parameter has been set up so that digit modification is performed on all called party numbers having NPA 908, during the times associated with translation plan 5014.

4. Assign days, dates and times to schedule profiles already associated to translation plans. For example:

   ENT-PRFL-SCHED::SchPrfl1:::SAT=0000-2400,SUN=0000-2400;
   ENT-PRFL-SCHED::SchPrfl2:::DATE1=01-01,DATE2=07-04,DATE3=05-26,DATE4=09-01,
   DATE5=11-23,DATE6=11-24,DATE7=12-25,TIME=0000-2400;
   ENT-PRFL-SCHED::SchPrfl3:::MON=0000-0700,TUE=0000-0700,WED=0000-0700,THU=0000-0700,FRI=0000-0700;
   ENT-PRFL-SCHED::SchPrfl4:::MON=0700-2100,TUE=0700-2100,WED=0700-2100,THU=0700-2100,FRI=0700-2100;
   ENT-PRFL-SCHED::SchPrfl5:::MON=2100-2400,TUE=2100-2400,WED=2100-2400,THU=2100-2400,FRI=2100-2400;

5. Assign Translation Plan and Schedule Profile to a Translation Plan combining the two. For example:

```
ENT-TRANS-SCHED::TSC1-SCHPRFL5:::Transplan=TP5014;
ENT-TRANS-SCHED::TSC1-SCHPRFL4:::Transplan=TP5015;
ENT-TRANS-SCHED::TSC1-SCHPRFL3:::Transplan=TP5014;
ENT-TRANS-SCHED::TSC1-SCHPRFL2:::Transplan=TP5013;
ENT-TRANS-SCHED::TSC1-SCHPRFL1:::Transplan=TP5013;
ENT-TRANS-SCHED::TSC1-DEFAULT:::Transplan=TP5013;
```

6. Associate ISUP trunk group 5013 with DPC 2-2-2, trunk group profile 1, and Translation Plan TSC1. For example:

```
ENT-TRKGRP::5013:::ISUP:Name=TGN5013,DPC=2-2-2,TGPROFILE=1,transplan=TSC1;
```

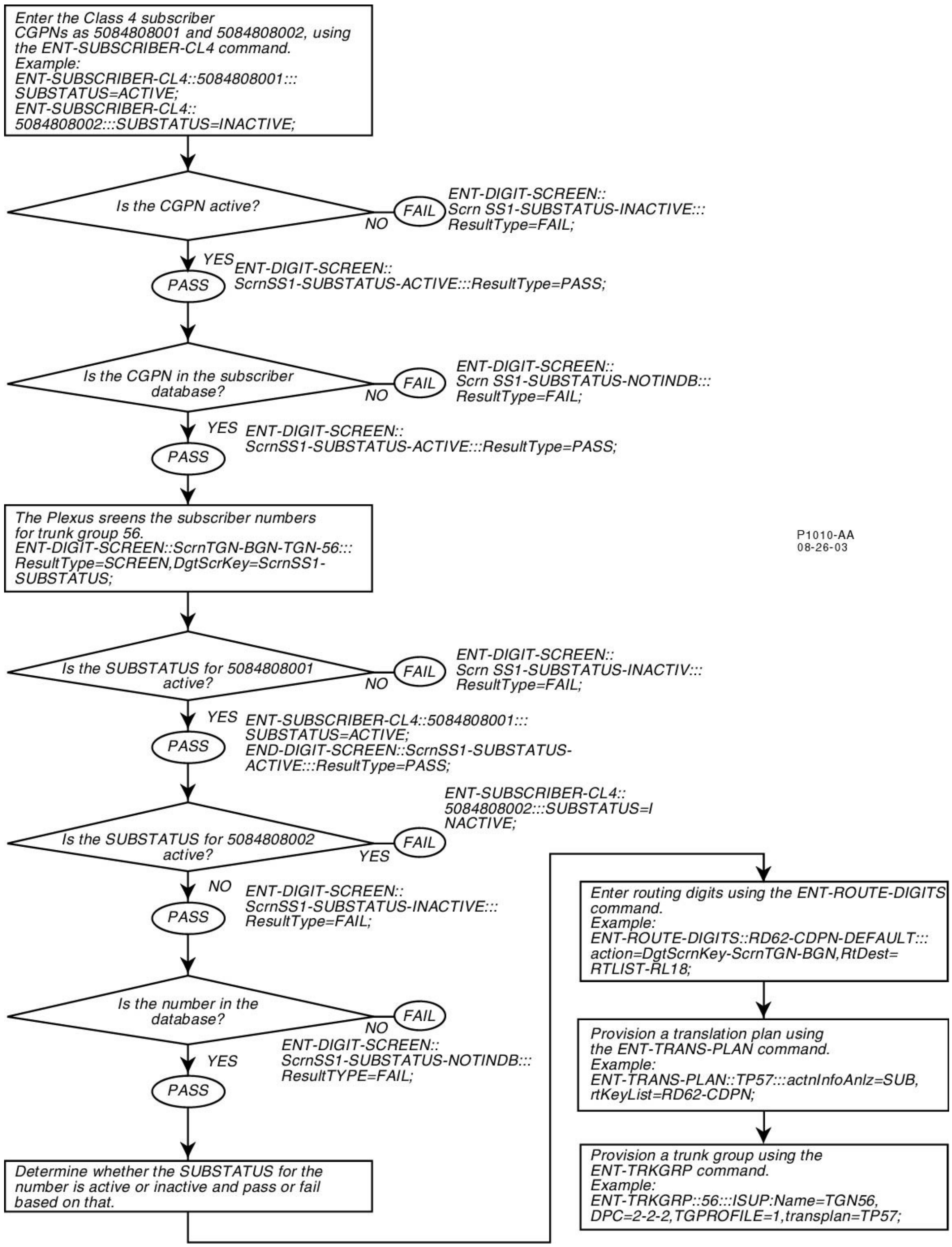
Casual vs. Non-Casual Routing on Trunk Group

Scenario: The switch enables and disables both casual and non-casual routing on a trunk group basis.

Conditions: The Calling Party Numbers (CGPNs) are equal to 5084808001 and 5084808002. The Called Party Number (CDPN) is equal to 6084808001.

By modifying the ResultType in the DIGIT-SCREEN command for SUBSTATUS, the carrier can change the decision of allowing casual routing.

Expected Result: The switch fails the calls if the CGPN is not in the subscriber database and is not in service.


Enter the Class 4 subscriber CGPNs as 5084808001 and 5084808002, using the ENT-SUBSCRIBER-CL4 command.
Example:
ENT-SUBSCRIBER-CL4::5084808001:::SUBSTATUS=ACTIVE;
ENT-SUBSCRIBER-CL4::5084808002:::SUBSTATUS=INACTIVE;
Is the CGPN active?
NO
FAIL
ENT-DIGIT-SCREEN::Scrn SS1-SUBSTATUS-INACTIVE:::ResultType=FAIL;
YES
PASS
ENT-DIGIT-SCREEN::ScrnSS1-SUBSTATUS-ACTIVE:::ResultType=PASS;
Is the CGPN in the subscriber database?
NO
FAIL
ENT-DIGIT-SCREEN::Scrn SS1-SUBSTATUS-NOTINDB:::ResultType=FAIL;
YES
PASS
ENT-DIGIT-SCREEN::ScrnSS1-SUBSTATUS-ACTIVE:::ResultType=PASS;
The Plexus sreens the subscriber numbers for trunk group 56.
ENT-DIGIT-SCREEN::ScrnTGN-BGN-TGN-56:::ResultType=SCREEN,DgtScrKey=ScrnSS1-SUBSTATUS;
P1010-AA
08-26-03
Is the SUBSTATUS for 5084808001 active?
NO
FAIL
ENT-DIGIT-SCREEN::Scrn SS1-SUBSTATUS-INACTIV:::ResultType=FAIL;
YES
PASS
ENT-SUBSCRIBER-CL4::5084808001:::SUBSTATUS=ACTIVE;
END-DIGIT-SCREEN::ScrnSS1-SUBSTATUS-ACTIVE:::ResultType=PASS;
Is the SUBSTATUS for 5084808002 active?
YES
FAIL
ENT-SUBSCRIBER-CL4::5084808002:::SUBSTATUS=I NACTIVE;
NO
PASS
ENT-DIGIT-SCREEN::ScrnSS1-SUBSTATUS-INACTIVE:::ResultType=FAIL;
Is the number in the database?
NO
FAIL
YES
PASS
ENT-DIGIT-SCREEN::ScrnSS1-SUBSTATUS-NOTINDB:::ResultTYPE=FAIL;
Determine whether the SUBSTATUS for the number is active or inactive and pass or fail based on that.
Enter routing digits using the ENT-ROUTE-DIGITS command.
Example:
ENT-ROUTE-DIGITS::RD62-CDPN-DEFAULT:::action=DgtScrnKey-ScrnTGN-BGN,RtDest=RTLIST-RL18;
Provision a translation plan using the ENT-TRANS-PLAN command.
Example:
ENT-TRANS-PLAN::TP57:::actnInfoAnlz=SUB,rtKeyList=RD62-CDPN;
Provision a trunk group using the ENT-TRKGRP command.
Example:
ENT-TRKGRP::56:::ISUP:Name=TGN56,DPC=2-2-2,TGPROFILE=1,transplan=TP57;

ANI Screening

Scenario: The switch supports originating ANI screening with NPA, NPA-NXX and NPA-NXX-XXXX granularity.

Conditions: In the following scenario, the Called Party Number (CDPN) NPA is 908, and the XXX is 480. The Calling Party Number NPA is 508, the XXX is 480, and the XXXX is 4001.

1. Apply ANI screening to the CDPN NPA by associating it with a treatment. For example:

   ENT-DIGIT-SCREEN::ANINPA-CDPN-UNINATNUM-908:::actn=SCREEN,DgtScrnKey=ANINPA908-CGPN;

2. Apply ANI screening to the CGPN NPA by associating it with a treatment. For example:

   ENT-DIGIT-SCREEN::ANINPA908-CGPN-UNINATNUM-508480:::actn=Treat,Treatment=Treat908;

3. Assign a treatment to a route key for an NPA CDPN. For example:

   ENT-TREATMENT::TRT908480:::treatType=REROUTE,rtKey=RTLBL1-CDPN;

4. Apply ANI screening to the CDPN NPA-NXX by associating it with a treatment. For example:

   ENT-DIGIT-SCREEN::ANINPANXXX-CDPN-UNINATNUM-908480:::actn=SCREEN,DgtScrKey=ANINPANXX-CGPN;

5. Apply ANI screening to the CGPN NPA-NXX by associating it with a treatment. For example:

   ENT-DIGIT-SCREEN::ANINPANXX-CGPN-UNINATNUM-508480:::actn=TRT908,Treatment=TRT908408;

6. Assign a treatment to a route key for an NPA-NXX CDPN. For example:

   ENT-TREATMENT::TRT908480:::treatType=REROUTE,rtKey=RTLBL1-CDPN-908;

7. Apply ANI screening to the CDPN NPA-NXX-XXXX by associating it with a treatment. For example:

   ENT-DIGIT-SCREEN::ANINPANXXX-CDPN-UNINATNUM-908480:::actn=SCREEN,DgtScrKey=ANINPANXX-CGPN;

8. Apply ANI screening to the CGPN NPA-NXX-XXXX by associating it with a treatment. For example:

ENT-DIGIT-SCREEN::ANINPANXX-CGPN-UNINATNUM-5084804001:::actn=Treat,Treatment=TRT9084808;

9. Assign a treatment to a route key for an NPA-NXX-XXXX CDPN. For example:

   ENT-TREATMENT::TRT908480:::treatType=REROUTE,rtKey=RTLBL1-CDPN;

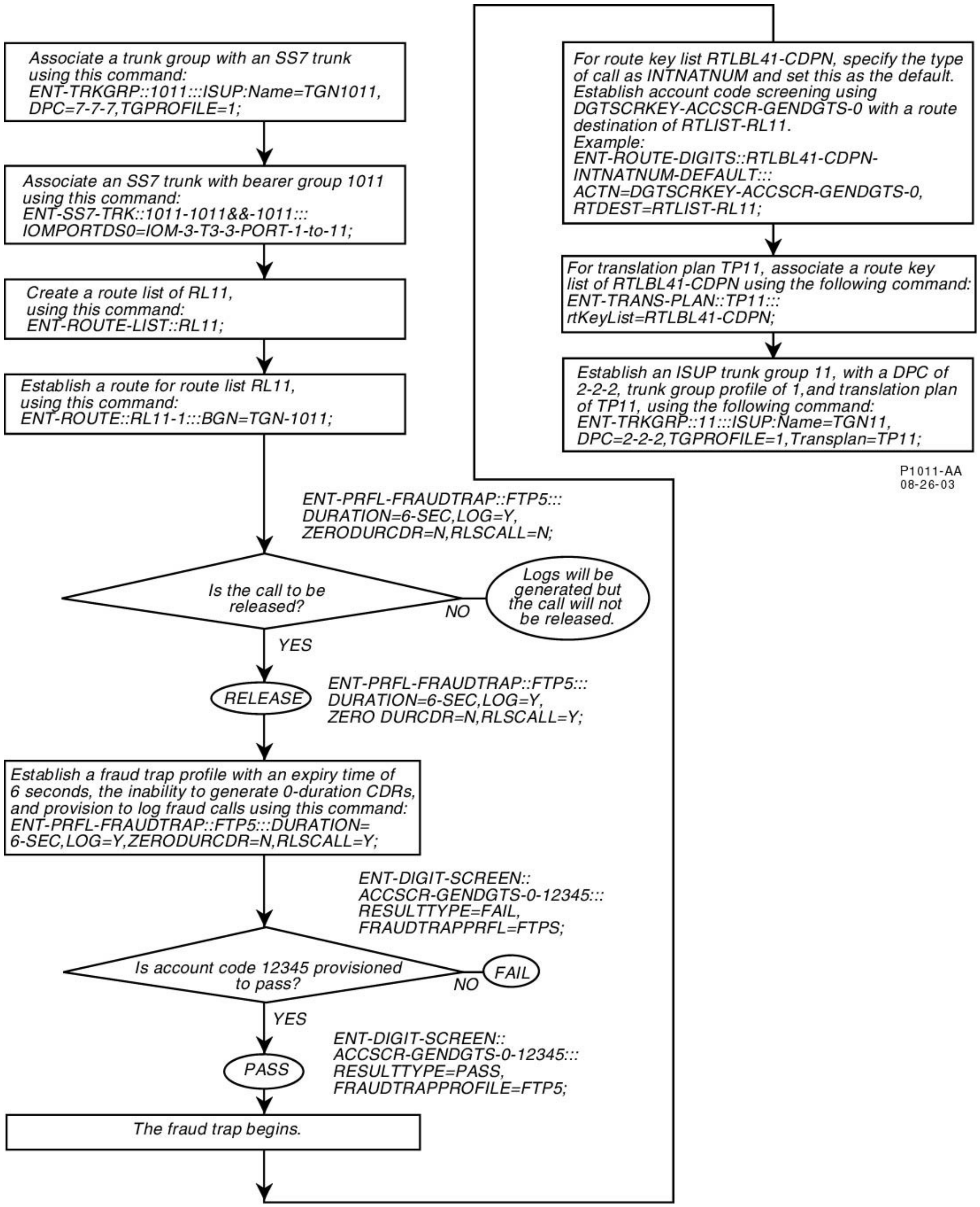
Fraud Traps

Scenario: The switch enables you to set the fraud trap on an account code basis, and keep a record of all calls in progress that meet the fraud trap criteria, enabling you to either manually release the call or set a parameter under which a call should be automatically released.

Conditions: The Carrier can release a fraud call, which is in progress by RLS-CALL, providing just the call Aid. The called number is revealed using the RTRV-FRAUDCALL command and the log for fraud call, also.

The switch allows the Carrier to set the timer in the FraudTrap Profile. After the timer expires and RELEASE=Y, the call will be released automatically. If RELEASE=N, the call will not be released.

The switch keeps record of all fraud calls. All fraud call records can be retrieved using the RTRV-FRAUDCALLS command.


Associate a trunk group with an SS7 trunk using this command: ENT-TRKGRP::1011:::ISUP:Name=TGN1011, DPC=7-7-7,TGPROFILE=1;
Associate an SS7 trunk with bearer group 1011 using this command: ENT-SS7-TRK::1011-1011&&-1011::: IOMPORTDS0=IOM-3-T3-3-PORT-1-to-11;
Create a route list of RL11, using this command: ENT-ROUTE-LIST::RL11;
Establish a route for route list RL11, using this command: ENT-ROUTE::RL11-1:::BGN=TGN-1011;
ENT-PRFL-FRAUDTRAP::FTP5::: DURATION=6-SEC,LOG=Y, ZERODURCDR=N,RLSCALL=N;
Is the call to be released?
NO
Logs will be generated but the call will not be released.
YES
RELEASE
ENT-PRFL-FRAUDTRAP::FTP5::: DURATION=6-SEC,LOG=Y, ZERO DURCDR=N,RLSCALL=Y;
Establish a fraud trap profile with an expiry time of 6 seconds, the inability to generate 0-duration CDRs, and provision to log fraud calls using this command: ENT-PRFL-FRAUDTRAP::FTP5:::DURATION= 6-SEC,LOG=Y,ZERODURCDR=N,RLSCALL=Y;
ENT-DIGIT-SCREEN:: ACCSCR-GENDGTS-0-12345::: RESULTTYPE=FAIL, FRAUDTRAPPRFL=FTPS;
Is account code 12345 provisioned to pass?
NO
FAIL
YES
PASS
ENT-DIGIT-SCREEN:: ACCSCR-GENDGTS-0-12345::: RESULTTYPE=PASS, FRAUDTRAPPROFILE=FTP5;
The fraud trap begins.
For route key list RTLBL41-CDPN, specify the type of call as INTNATNUM and set this as the default. Establish account code screening using DGTSCRKEY-ACCSCR-GENDGTS-0 with a route destination of RTLIST-RL11. Example: ENT-ROUTE-DIGITS::RTLBL41-CDPN- INTNATNUM-DEFAULT::: ACTN=DGTSCRKEY-ACCSCR-GENDGTS-0, RTDEST=RTLIST-RL11;
For translation plan TP11, associate a route key list of RTLBL41-CDPN using the following command: ENT-TRANS-PLAN::TP11::: rtKeyList=RTLBL41-CDPN;
Establish an ISUP trunk group 11, with a DPC of 2-2-2, trunk group profile of 1,and translation plan of TP11, using the following command: ENT-TRKGRP::11:::ISUP:Name=TGN11, DPC=2-2-2,TGPROFILE=1,Transplan=TP11;
P1011-AA
08-26-03

Call Processing Hierarchy

Scenario: When a call comes into the switch, the switch performs call processing on that call. The call processing scenario varies, as the call processing hierarchy changes, depending on the call processing characteristics associated with the incoming call. The following scenario provides a basic example of the switch call processing hierarchy.

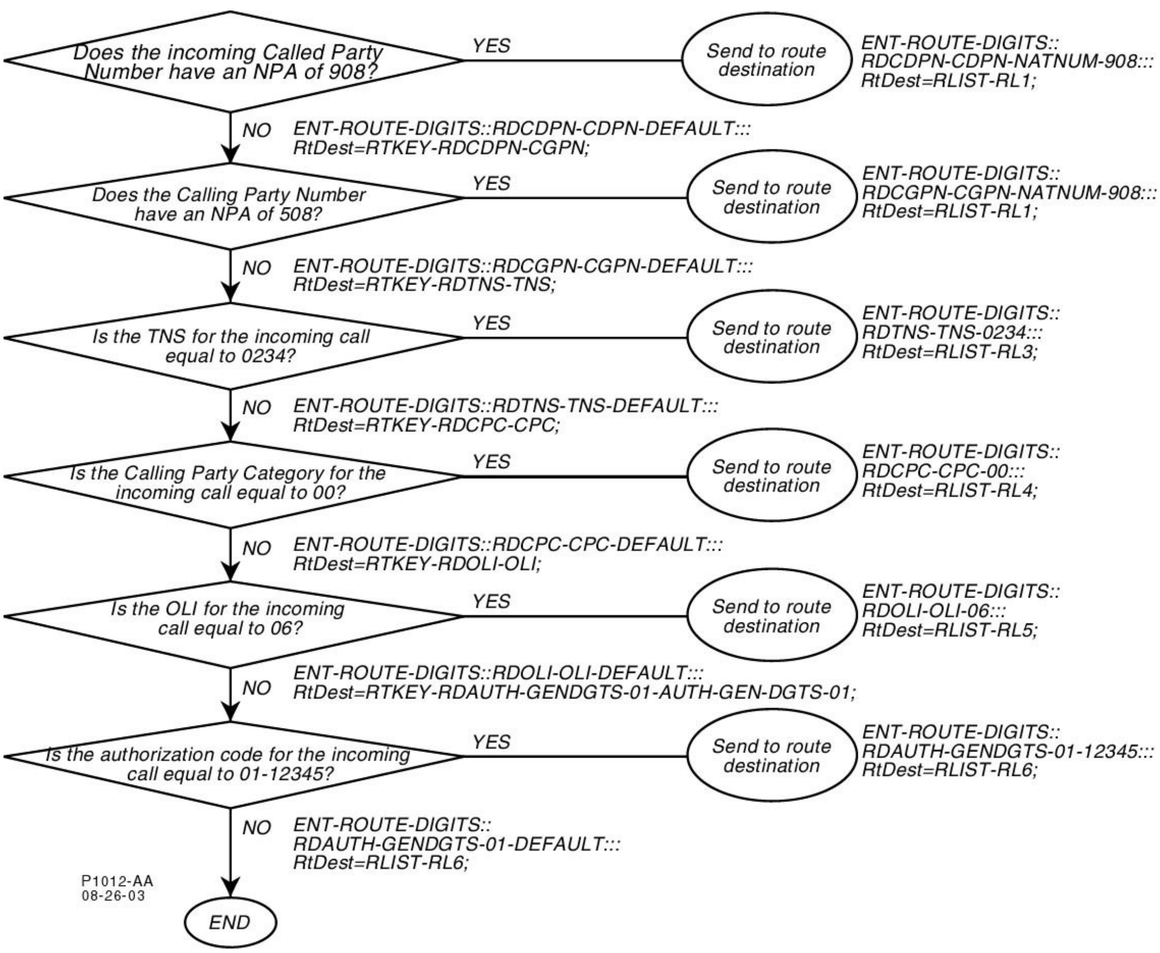
Conditions: ISUP trunk group 506 is provisioned on the switch with a DPC of 4-4-4, trunk group profile of 1 and Translation Plan of TP506. For example:

ENT-TRKGRP::506:::ISUP:Name=ICTGN506,DPC=4-4-4,TGPROFILE=1,Transplan=TP506;

Translation Plan 506 is associated with route key list RD506-CDPN. For example:

ENT-TRANS-PLAN::TP506:::RtKeyList=RD506-CDPN;

When the call comes in, the following analysis is done on that call:



Advanced Intelligent Network Feature Package

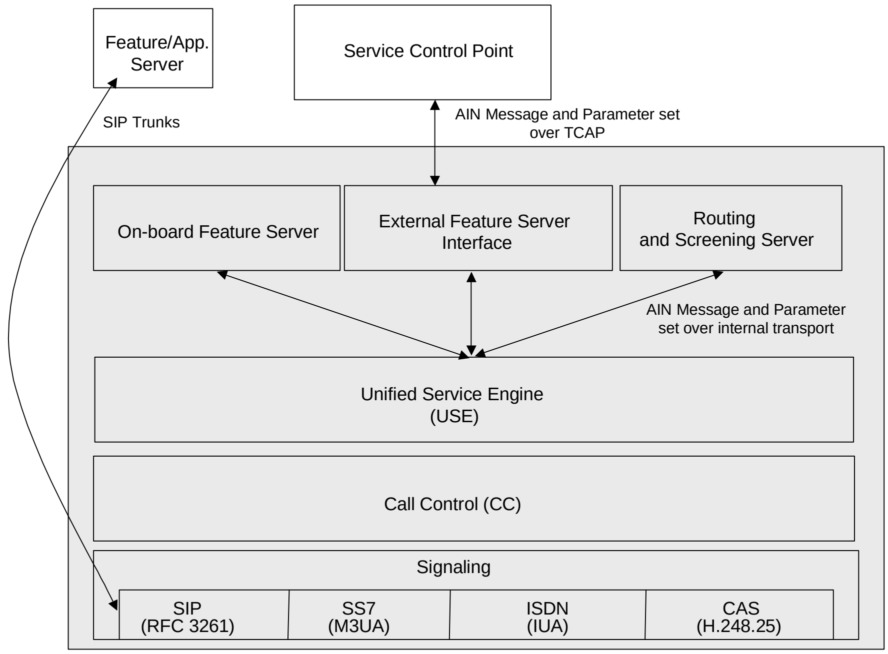
This document contains the following sections:

Scope

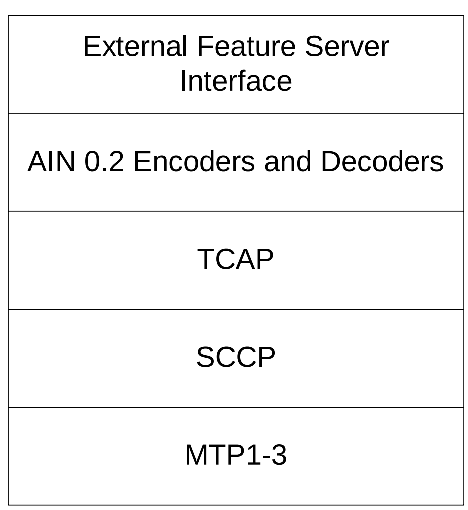
The Lucent Compact Switch's internal call processing subsystem architecture uses an AIN framework to provide the flexibility to support both internal (on-board) and external features using the AIN message and parameter set. This architecture allows the service provider to easily migrate features between on-board and external databases with just provisioning changes and without any additional software development and compilation. The Compact Switch recognizes calls that requires AIN processing without making any assumptions about the service being provided. The Compact Switch does this, after encountering an AIN trigger, by suspending call processing temporarily, then either querying its internal database or assembling and launching an external query to service logic located remotely at a Service Control Point (SCP). The subsequent database reply gives the Compact Switch information on how to continue processing the call. Some features, like CNAM (Caller Name delivery), LNP (Local Number Portability) and Toll-free, must be accessed from an SCP.

As shown in the following diagram, the Call Control (CC) and Unified Service Engine (USE) combine together to provide the entire set of AIN framework features, which includes:

- PIC(s) (Point-In-Call) in both Originating and Terminating portions of the call.
- Detection Points at above PIC(s), which can be armed as triggers statically as Trigger Detection Points (TDP(s)) or dynamically within a transaction as Event Detection Points (EDP(s)).
- Query and Response processing using both persistent and non-persistent TCAP transactions.
- Caller Interaction using on-board announcement and digit collection capabilities.
- Automatic Code Gapping.
- AIN AMA Record Generation.



The Unified Service Engine (USE) Layer manages the precedence and interaction rules for triggers used and armed by multiple features in the On-board Feature Server (OFS), or features distributed between OFS, SCP, and the Routing and Screening server. The USE layer also uses an AIN 0.2 compliant message and parameter set to interface with any of the three entities. For features residing remotely on an SCP, the External Feature Server Interface layer performs the translation of messages and parameters for transport using TCAP over SS7 as shown below.



The Compact Switch supports a subset of AIN 0.2 features for a Service Switching Point (SSP). The following sections describe the supported functionality.

AIN Triggers

Triggering is the process of identifying calls that require AIN handling. Triggers can be subscribed (line-based), group-based, or office-based. The Compact Switch supports AIN triggers including:

- Public Feature Code (subscribed)
- Specific Feature Code (subscribed)
- International Prefix (subscribed / office-based)
- Specific Digit String (office-based) and
- Local Number Portability (office-based)

The Compact Switch also supports the call-processing triggers listed below. It sends a message (with the same name as the trigger) to the SCP when it encounters one of these triggers, after which the SCP replies with an AnalyzeRoute, a Continue, or a SendToResource message.

- O_Called_Party_Busy
- O_No_Answer
- T_Busy
- T_No_Answer
- Network_Busy

These triggers allow AIN services to detect a busy condition on the originating or terminating end of a call, and to detect when the called party does not answer on the originating or terminating end of a call. These new triggers provide AIN with the capability to redirect calls on busy/no answer.

When it detects an active trigger, the Compact Switch suspends normal call processing until it completes communications with its internal database or an SCP.

The Compact Switch conforms to the following basic AIN trigger rules:

- AIN trigger points are configurable on both the line and trunk side of the Compact Switch.
- For each provisionable trigger, the *service provider* can designate whether to use external or internal processing.
- External processing is via an SCP using Transaction Capabilities Application Part (TCAP) messages.
- Point codes designating the SCP(s) to which TCAP queries will be directed are provisioned on the Compact Switch.

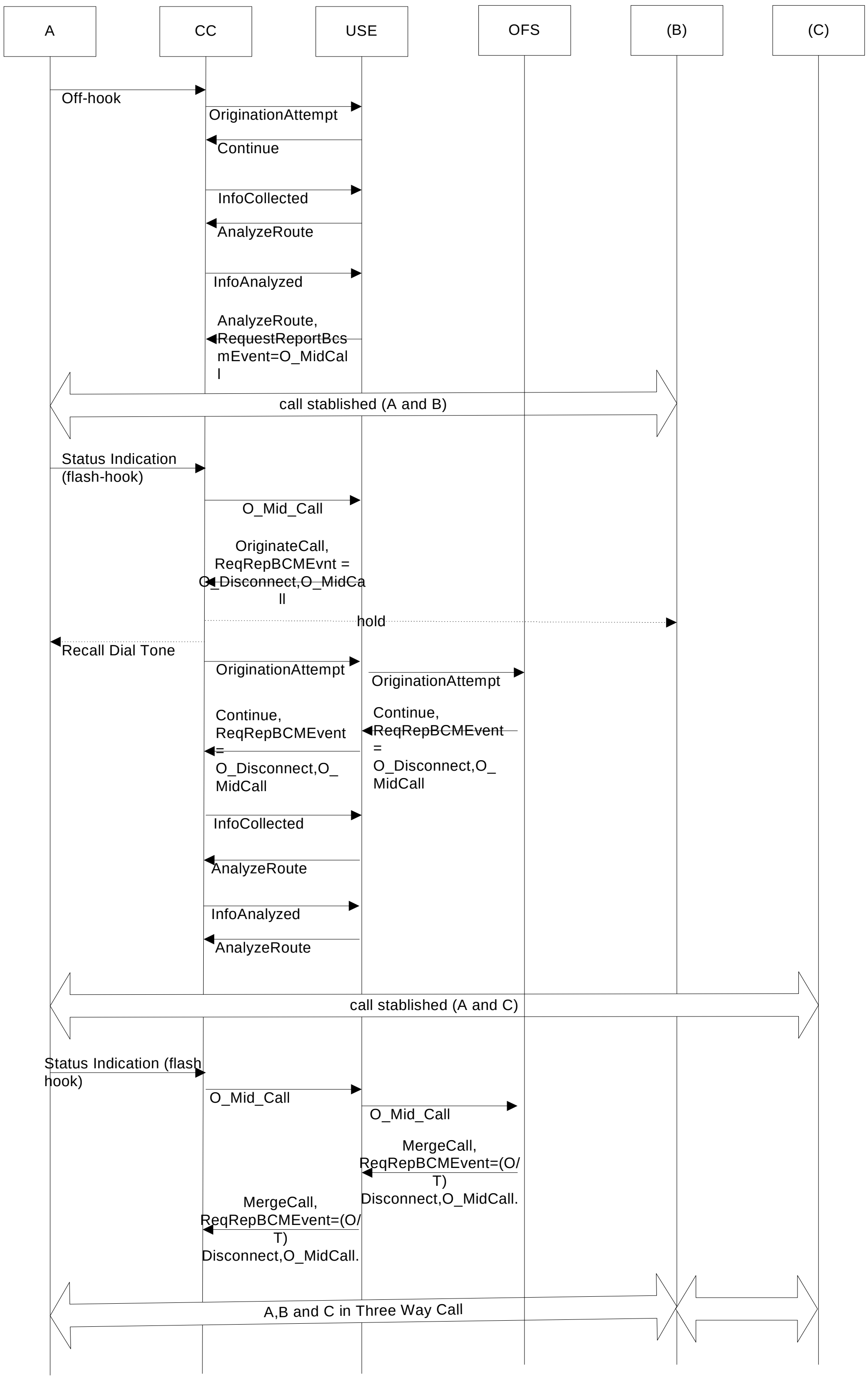
After detecting a trigger, the Compact Switch checks for the presence of applicable, active code-gapping controls to prevent SCP overload. If code-gapping controls apply, the Compact Switch gives the call final treatment; otherwise, it begins querying. Currently, the Compact Switch provides code-gapping support only for LNP and Toll-free calls.

Triggers for Internal Features

Trigger Name	Internal Features Using Trigger
Origination_Attempt	Speed Dial, Voice Mail, Add-on-Transfer-Conference, Three-way-call, Cancel Call Waiting
Info_Analyzed	Auth Codes, Remote Access to Call Forwarding, Voice Mail, Add-on-Transfer-Conference, Three-way-call, Cancel Call Waiting, Tollfree, Local Number Portability
Termination_Attempt	Anonymous Call Rejection, Call Forwarding Variable, Caller Name, Caller Number Delivery, Voice Mail
O_Mid_Call, T_Mid_Call	Add-on-transfer-conference, Three-way-call
T_Busy	Call Waiting, Call Forwarding on Busy, Voice Mail
T_No_Answer	Call Forwarding on No Answer

The message sequence diagram shown in Figure 1 illustrates the message flow between CC, USE and OFS to provide a three-way call service.

Figure 1. Message Flow Between CC, USE and OFS



Triggers for External Features

The following triggers are available for outside AIN support, at the associated "Info_Analyzed" TDP:

- Specific Digit String:
  Applies to called party numbers using NANP and can be set on three to 10 digits of an NPA-NXX-XXXX number.

- International Prefix:
  Applies to called party numbers in one of the following formats:
  - 011 + 7-15 digits
  - 01+7-15 digits
  - 101xxxx 011+7-15 digits
  - 101xxxx 01+7-15 digits

- One-Plus Prefix:
  Applies to called party numbers in one of the following formats:
  - 1+NPA-NXX-XXXX
  - 101XXXX 1+ NPA-NXX-XXXX

- Customized Dialing Plan:
  Applies when a certain 1-7 digit intercom code is dialed within a customized dialing plan.

- Operator Services:
  Applies to called party numbers in one of the following formats:
  - 0-
  - 00-
  - 101xxxx 0-
  - 101xxxx 00-
  - 0+
  - 01+
  - 101xxxx 0+
  - 101xxxx 01+

Note that although the provisionable triggers for remote features are currently a subset of the triggers supported for internal features, the development required for extending this list involves the following:

- Adding the provisioning support to expose the trigger's configuration.
- Adding the message translation code for transfer using TCAP.

Triggers armed using Next Event List

The Event Detection Point trigger events are arm(able) as requests or notifications through the RequestReportBCM message. When one of these events is detected, the Compact Switch sends the corresponding Request or Notification message to the requesting feature in the OFS or an SCP. The following EDP triggers are supported for use by remote features:

- Network_Busy
- O_Term_Seized
- O_Called_Party_Busy
- O_Answer
- O_No_Answer
- O_DTMF_Answered
- Switch_Hook_Flash
- Timeout
- O_Disconnect
- O_Disconnect_Called

AIN Messages Support

In addition to supporting the AIN messages specific to both TDP and EDP triggers mentioned above, the Compact Switch also supports the following SCP messages that affect call processing.

SSP → SCP

The following messages go from the Compact Switch to an SCP:

- Close
- Resource_Clear
- CTR_Clear

SCP → SSP

The following messages go from an SCP to the Compact Switch and are used by internal features:

- Acknowledge
- Analyze_Route
- Authorize_Termination
- Close
- Continue
- Create_Call
- Disconnect
- Disconnect_Leg
- Forward_Call
- Merge_Call

- Move_Leg
- Offer_Call
- Originate_Call
- Reconnect
- Send_To_Resource

The following messages go from an SCP to the Compact Switch and are used by external features:

- Acknowledge
- Analyze_Route
- Close
- Continue
- Disconnect
- Send_To_Resource

Again, note that the development required for extending the list of messages for support by remote features, just involves adding the message translation code for transfer using TCAP.

AIN Message Extension Flexibility

The Compact Switch can support the extension of AIN messages to overcome AIN inadequacy due to the development of new call-affecting features.

Extension Parameter

Most AIN messages have an optional parameter called "*ExtensionParameter*". This parameter allows the service provider to add parameters to messages sent between an SSP and an SCP. The AIN encoding and decoding software in the Compact Switch recognizes this as a valid optional parameter in each message. The development required to allow the use of this parameter by a custom feature in a service provider's network involves only two steps:

- Interpreting the data contained within this parameter.
- Performing the required protocol inter-working and/or call-affecting action as desired by the service provider.

Optional Parameters

The presence of optional parameters allows a protocol to be both forward and backwards compatible, as is the case with the AIN encoders/decoders used by the Compact Switch. With minor modifications to the message definitions, these AIN encoders/decoders in the Compact Switch can be updated to accept some optional parameters desired by service providers, which are otherwise not present in those messages. Again, after completing the encoder/decoder modifications, the additional development necessary to use an optional parameter by a custom feature in a service provider's network involves only two steps:

- Interpreting the data contained within this parameter.
- Performing the required protocol inter-working and/or call-affecting action as desired by the service provider.

AMA Record Generation

The Compact Switch can generate both SCP-based structures (Structure 220 and 221) and switch-based structures for calls requiring AIN interactions with an SCP. The Compact Switch selects between the two depending upon whether the "*AMAslpId*" parameter is present in the message returned by SCP. The Compact Switch supports the "Multiple Record" paradigm when exposed to multiple InfoAnalyzed triggers.

The Compact Switch currently supports recording of following AMA specific parameters that might be present in various messages from the SCP:

- AMAslpId
- AMADigitsDialedWC

Intelligent Network Digit Modification

This feature provides the ability to support digit modification (prefixing or stripping of digits) from the Called Party Number (CdPN) field after the Intelligent Network (IN) trigger point has been reached, but before the actual IN message is sent. **Note:** "IN" is used generically (all services available via ENT-LIST-AINTRIGGER).

Up to a 20 digit prefix can be inserted before the CdPN and up to 31 digits can be stripped from the CdPN (digit stripping takes place from left to right). When both digit stripping and prefixing are enabled, digit stripping takes precedence over prefixing. A commit parameter is supported that, when enabled, makes the CdPN modifications permanent and when disabled, restores the CdPN to its original value if the IN query fails.

AIN Queries

Querying is the process of assembling a TCAP Query message and sending it to an SCP over the Common Channel Signaling (CCS) network using SS7 signaling. The Query messages correspond to the Trigger Detection Points (TDPs): Origination_Attempt, Info_Collected, Info_Analyzed, Network_Busy, and Termination_Attempt. The content of the Query message depends on the type of trigger encountered and the parameters of the call (e.g., terminating party address and originating line information).

The SCP may request the Compact Switch to obtain additional information from the caller using a TCAP Conversation with Permission message. The Compact Switch prompts (through an announcement) and collects the information from the caller (e.g., Dual-Tone Multifrequency [DTMF], dial pulse digits, or D-channel INFOrmation messages), and returns that information to the SCP in a TCAP Conversation Package.

The SCP may request the Compact Switch to activate or deactivate certain triggers using a TCAP Query or Conversation Package. The Compact Switch responds with either a Response or Conversation Package, respectively, indicating whether the activation or deactivation was successful. Response processing consists of interpreting and carrying out the instructions in the TCAP Response message received from the SCP.

The SCP may request the Compact Switch to route the call, redirect the call, disconnect the call, play an interactive announcement to the caller, route the call to an announcement, or provide special terminating treatment (e.g., distinctive alerting or display information). Specifically, the Compact Switch supports response processing for AnalyzeRoute, Continue, Close, Disconnect, RequestReportBCMEvent, and Send_To_Resource. The SCP Response may include a request to be notified when the call ends, in which case the Compact Switch notifies the SCP when the call is disconnected or cleared using a TCAP Unidirectional message. Additionally, the SCP may send the Compact Switch a TCAP Query message, requesting it to monitor the state of certain facilities, in which case the Compact Switch reports the state of the designated facilities using TCAP Conversation and Response messages.

You can use the EMS to provision subscribed triggers so that any calls originating from (or terminating to) the subscriber's line encounter the trigger. Office-based triggers are available to subscribers connected to the telephone switching office or who have access to the North America Numbering Plan (NANP).

Deployments

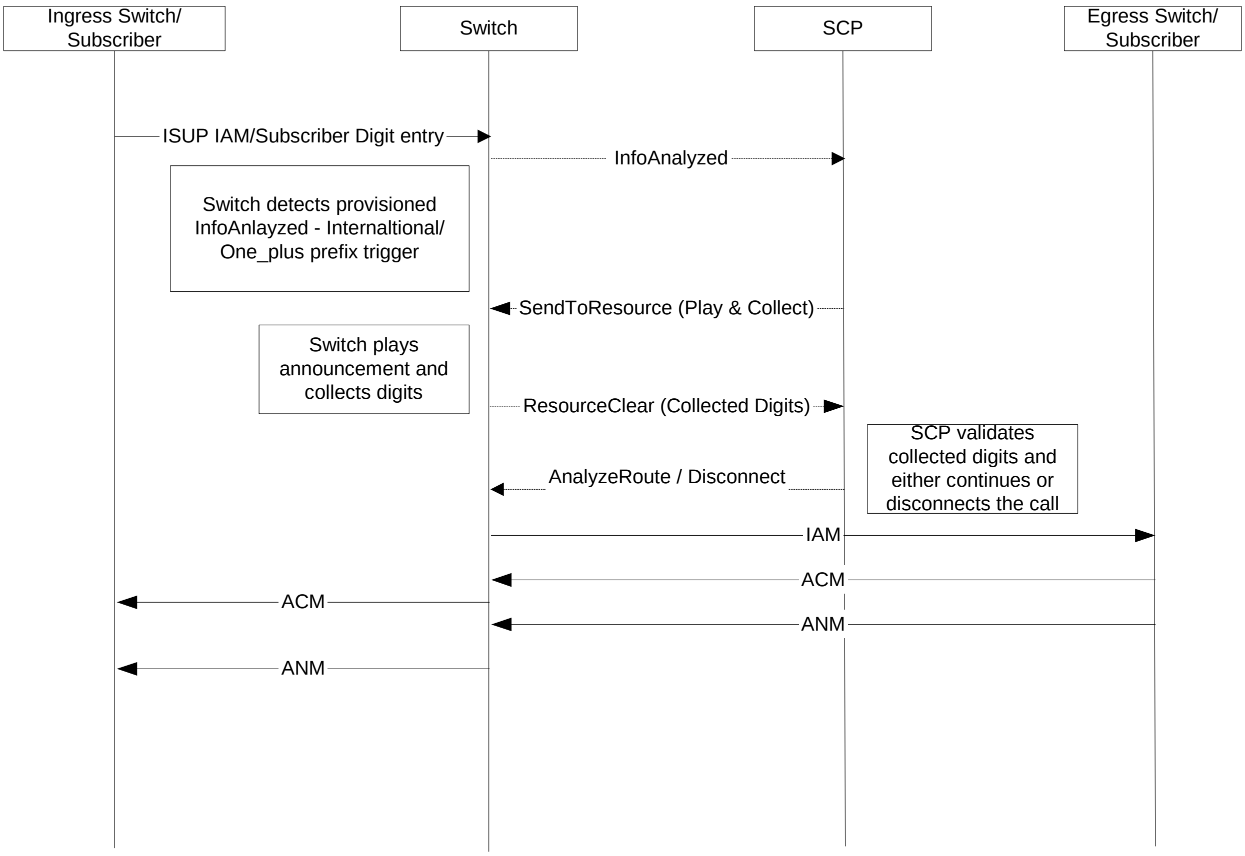
The Compact Switch can process an AIN-triggered call based on these service features:

- Authorization Code
- Terminating Toll-Free
- Call Forwarding Busy/No Answer
- Custom Dial Plan (Intercom/VPN Dialing)
- Single Call Setup and Release by Calling Party with Balance Update
- Multi-Call Setup and Release with Mid-Call Re-origination by Calling Party
- Multi-Call Setup and Release with Re-origination Using DTMF Entry After Called Party Disconnects
- Multi-Call Setup and Release with Re-origination After Called Party Disconnects
- AIN Toll-Free and Local Number Portability

Authorization Code

The Compact Switch can process an AIN-triggered call based on Authorization Code. A sample of this processing is shown in Figure 2.

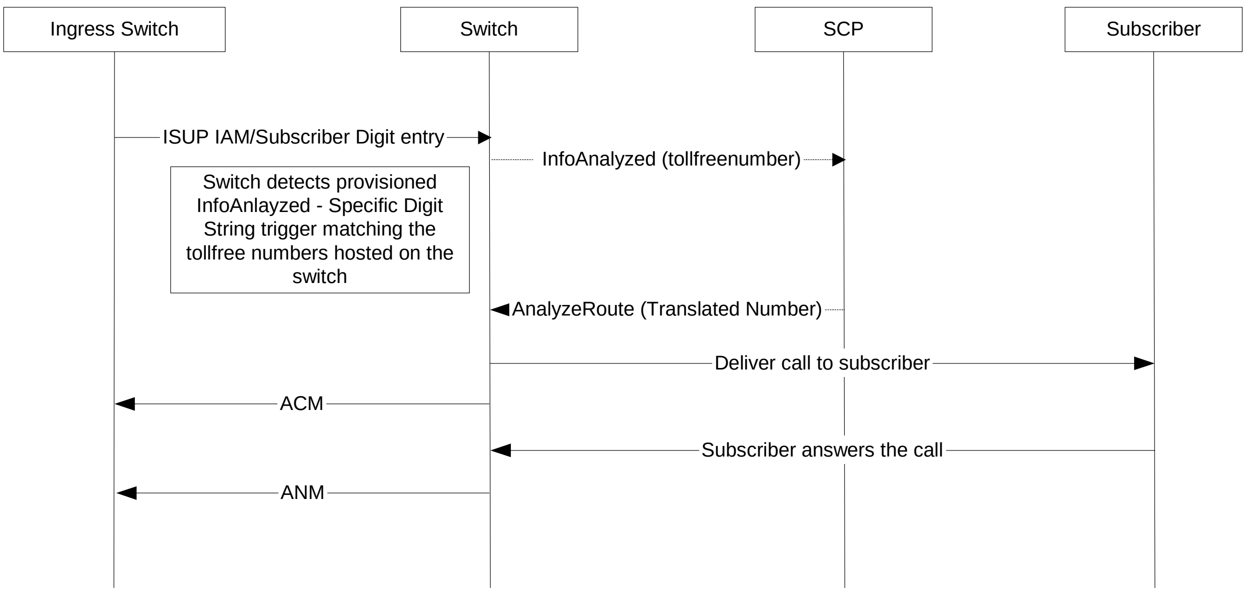
Figure 2. AIN Call Processing by Authorization Code



Terminating Toll-Free

The Compact Switch can process an AIN-triggered call based on the Terminating Toll-Free service. A sample of this processing is shown in Figure 3.

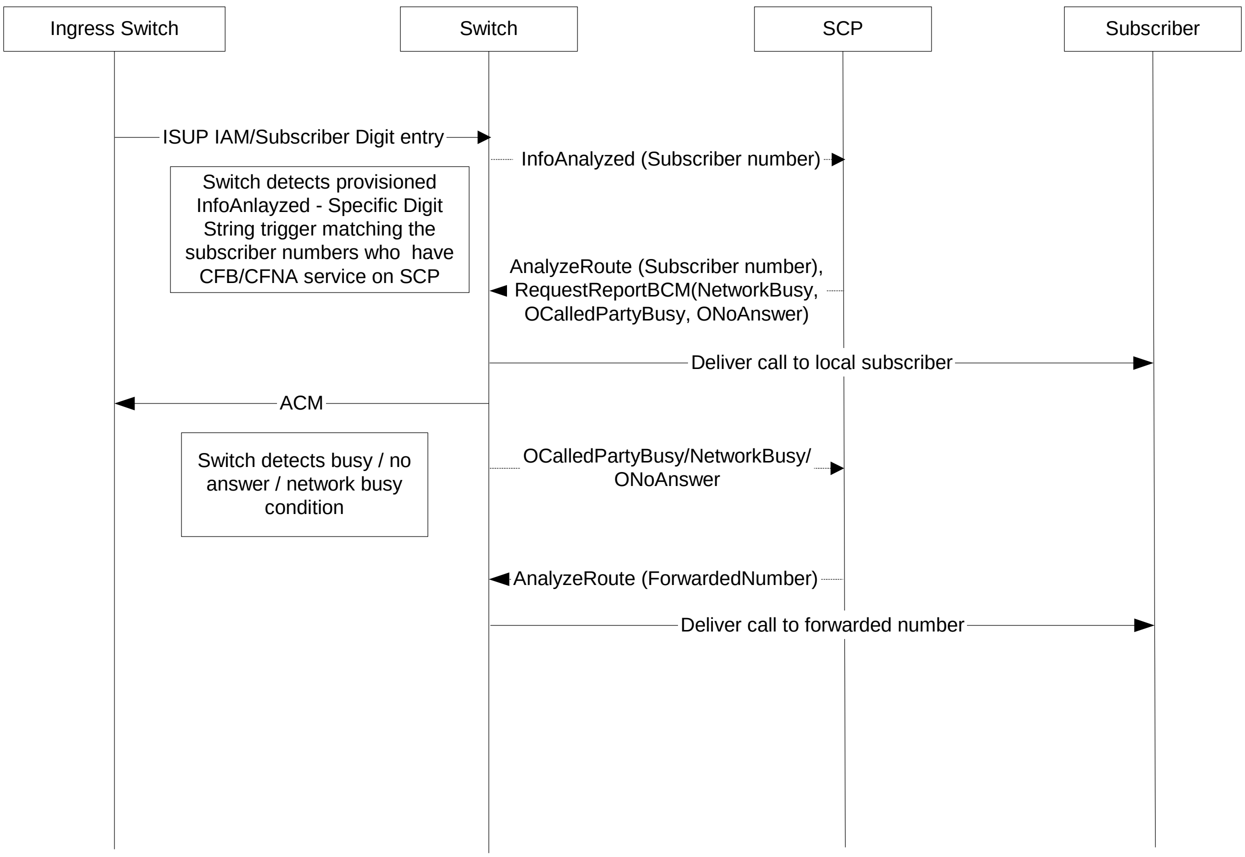
Figure 3. AIN Processing by Terminating Toll-Free Service



Call Forwarding Busy/No Answer

The Compact Switch can process an AIN-triggered call based on the Call Forwarding Busy/No Answer service. A sample of this processing is shown in Figure 4.

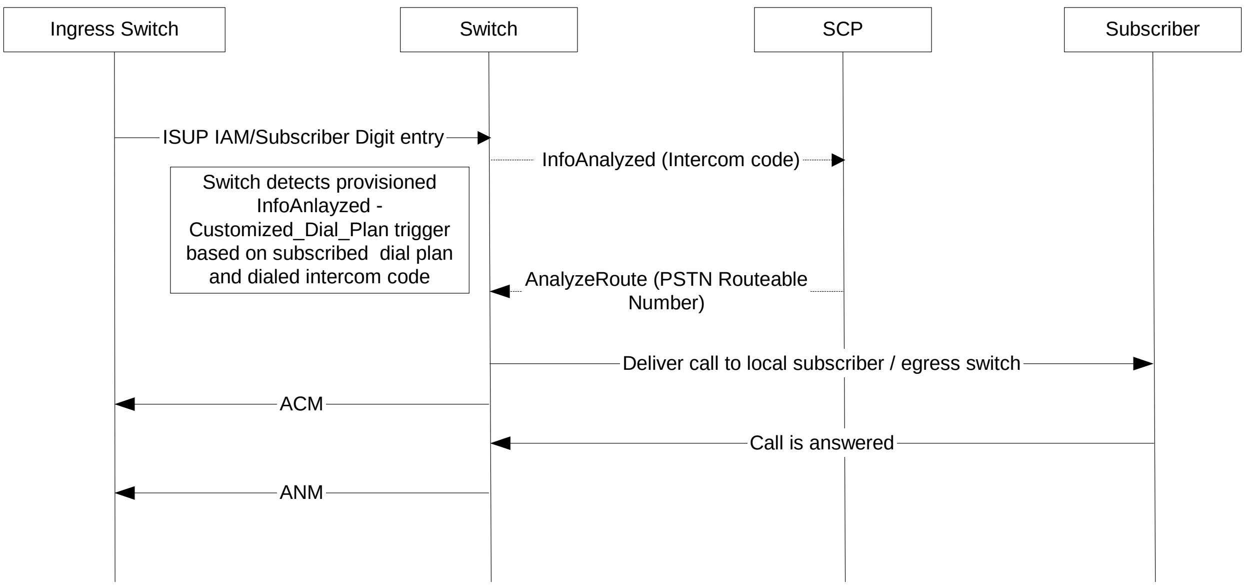
Figure 4. AIN Processing by Call Forwarding Busy/No Answer Service



Custom Dial Plan (Intercom/VPN Dialing)

The Compact Switch can process an AIN-triggered call based on Customer Dial Plan (Intercom/VPN Dialing). A sample of this processing is shown in Figure 5.

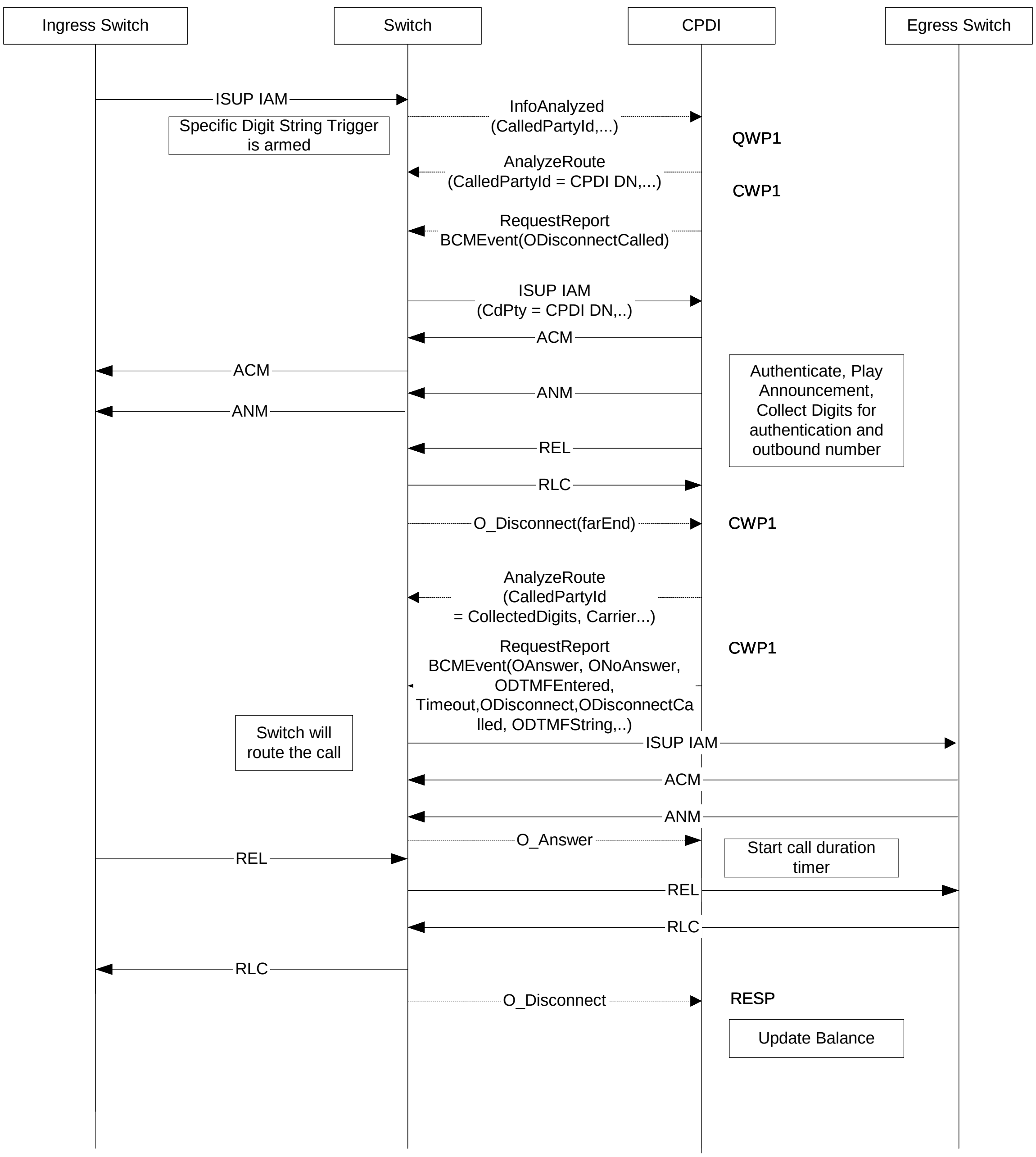
Figure 5. AIN Processing by Custom Dial Plan (Intercom/VPN Dialing)



Single Call Setup and Release by Calling Party with Balance Update

The Compact Switch can process an AIN-triggered call based on single call setup and release by calling party with a balance update. A sample of this processing is shown in Figure 6.

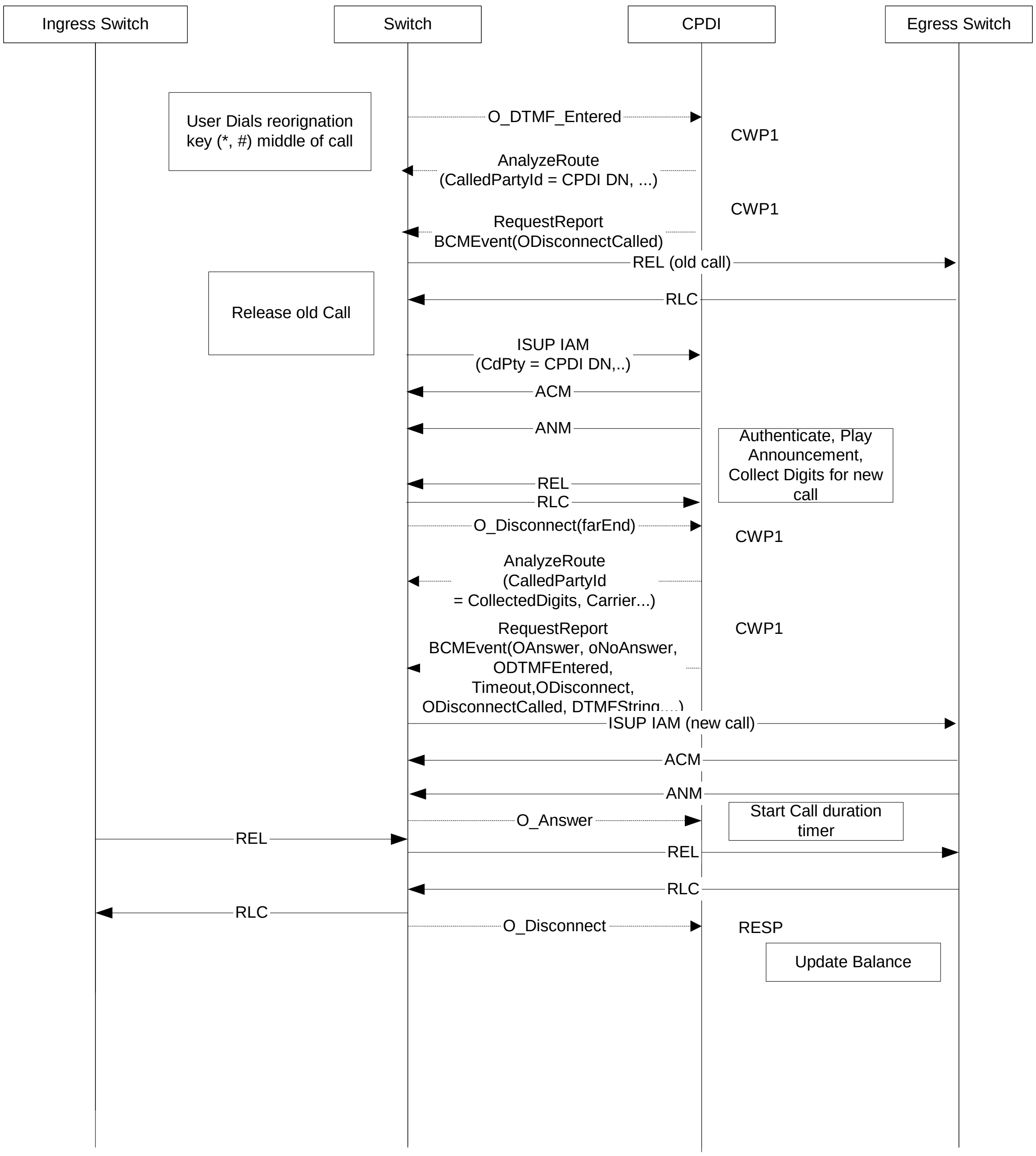
Figure 6. AIN Processing Based on Single Call Setup and Release by Calling Party with Balance Update



Multi-Call Setup and Release with Mid-Call Re-origination by Calling Party

The Compact Switch can process an AIN-triggered call based on multi-call setup and release with mid-call re-origination by calling party. A processing sample is shown in Figure 7.

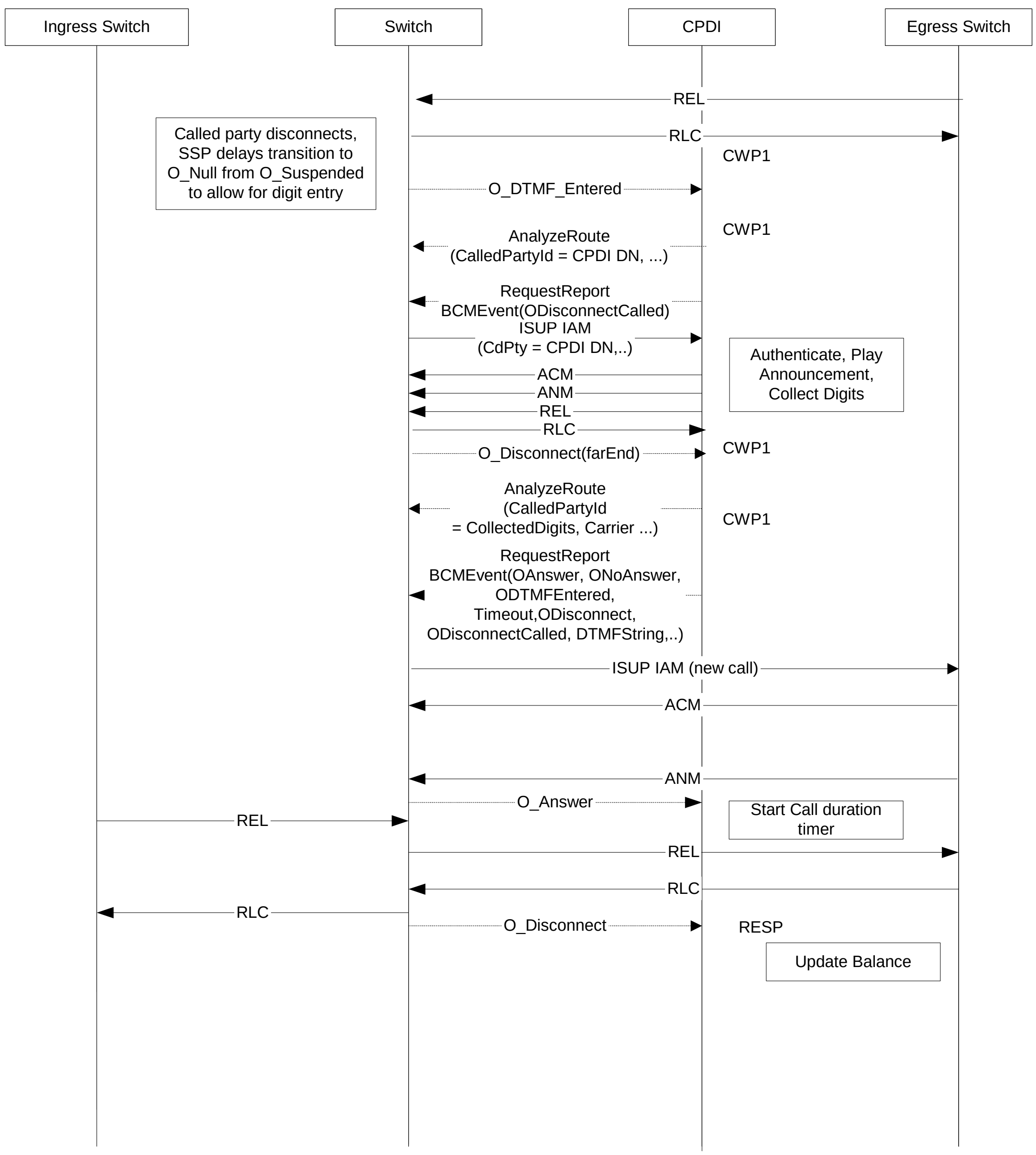
Figure 7. AIN Processing Based on Multi-Call Setup and Release with Mid-Call Re-origination by Calling Party



Multi-Call Setup and Release with Re-origination Using DTMF Entry After Called Party Disconnects

The Compact Switch can process an AIN-triggered call based on multi-call setup and release with re-origination using DTMF entry after called party disconnects. A sample of this processing is shown in Figure 8.

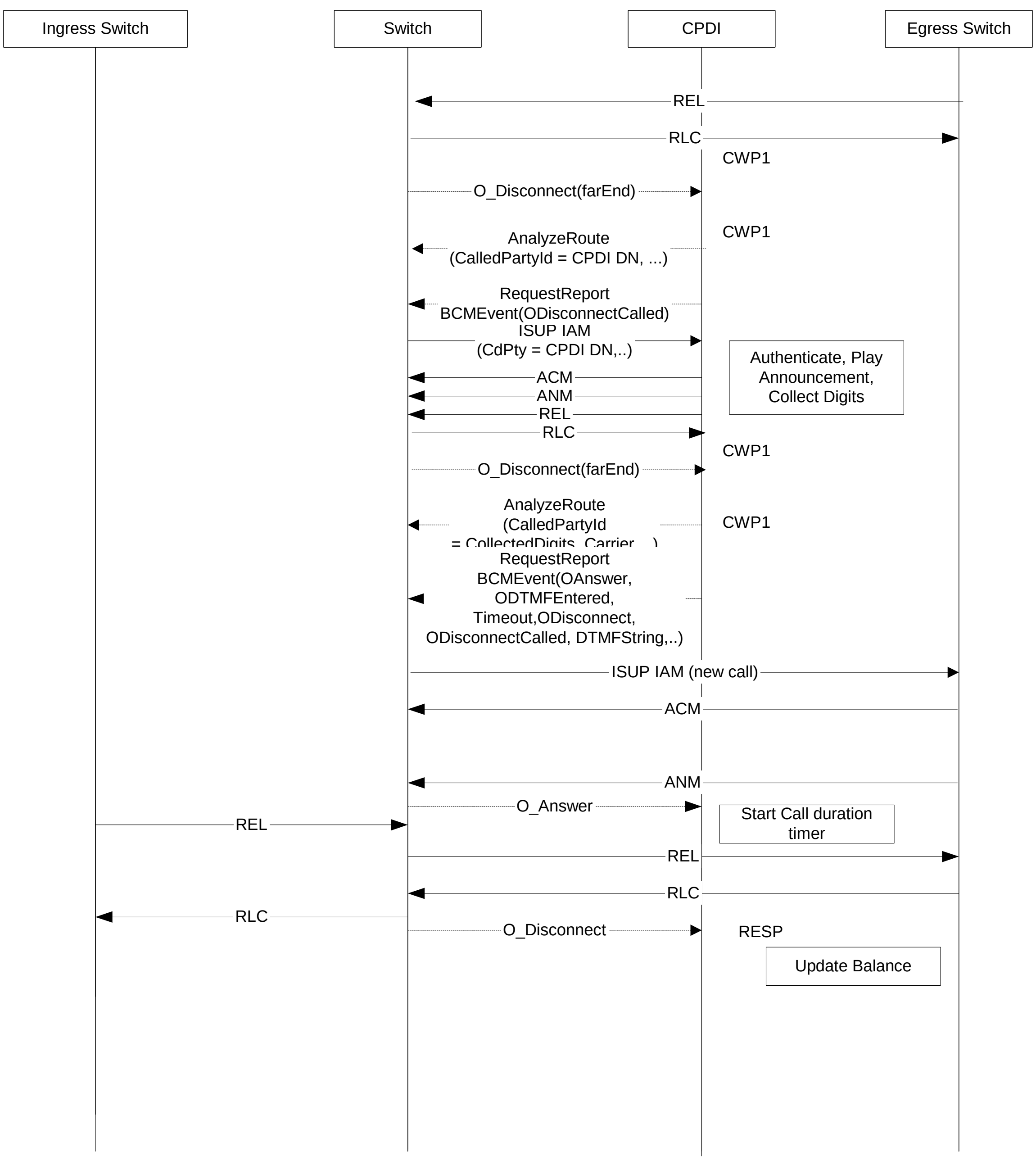
Figure 8. AIN Processing Based on Multi-Call Setup and Release with Re-origination Using DTMF Entry After called Party Disconnects



Multi-Call Setup and Release with Re-origination After Called Party Disconnects

The Compact Switch can process an AIN-triggered call based on multi-call setup and release with re-origination after called party disconnects. A sample of this processing is shown in Figure 9.

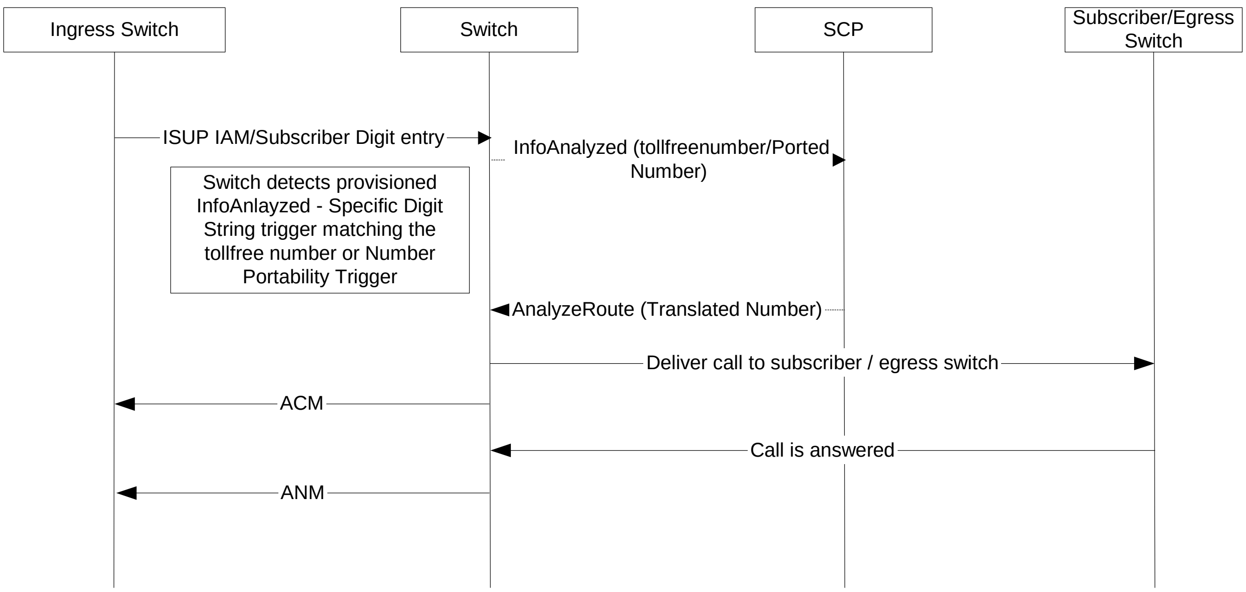
Figure 9. AIN Processing Based on Multi-Call Setup and Release with Re-origination After Called Party Disconnects



AIN Toll-Free and Local Number Portability

The Compact Switch can process an AIN-triggered call based on AIN Toll-Free and Local Number Portability. A sample of this processing is shown in Figure 10.

Figure 10. AIN Processing Based on AIN Toll-Free and Local Number Portability

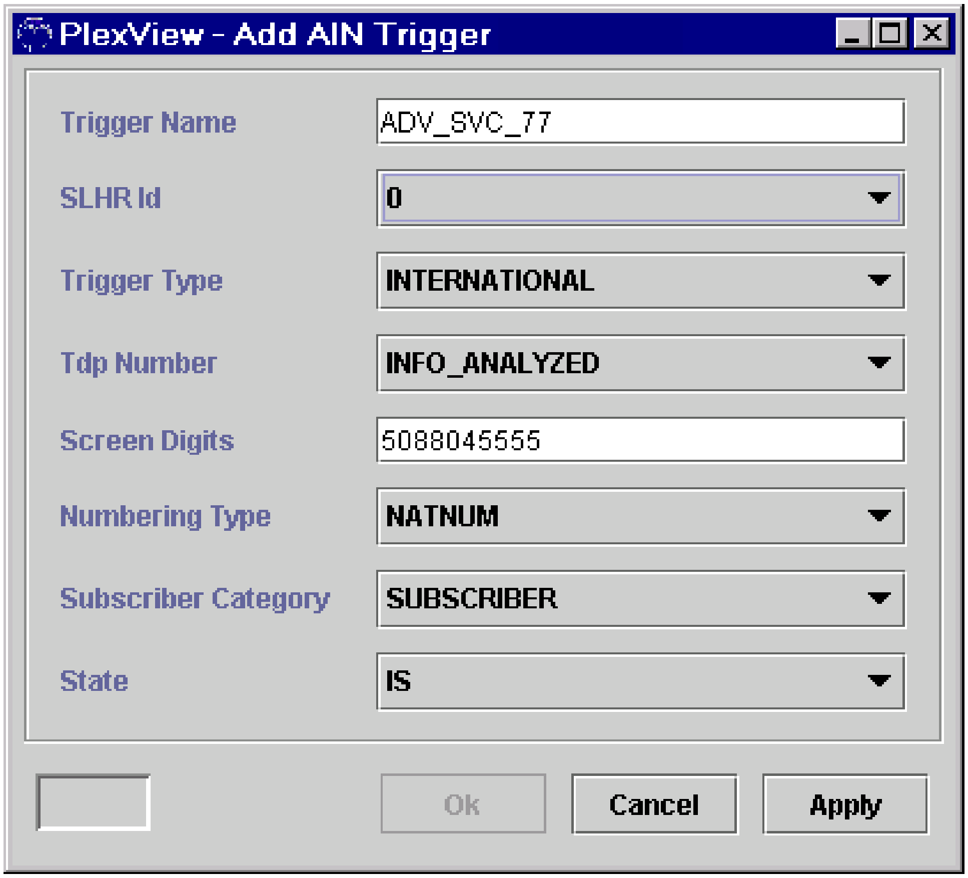


Provisioning AIN Triggers Using the PlexView® Element Management System (EMS)

You can provision the Compact Switch for AIN triggers (and services) using the PlexView EMS. You first create the trigger, using the Add AIN Trigger screen shown in Figure 11, and then provision the trigger per subscriber using the Add Subscriber → AIN Assignments screen as shown in Figure 12.

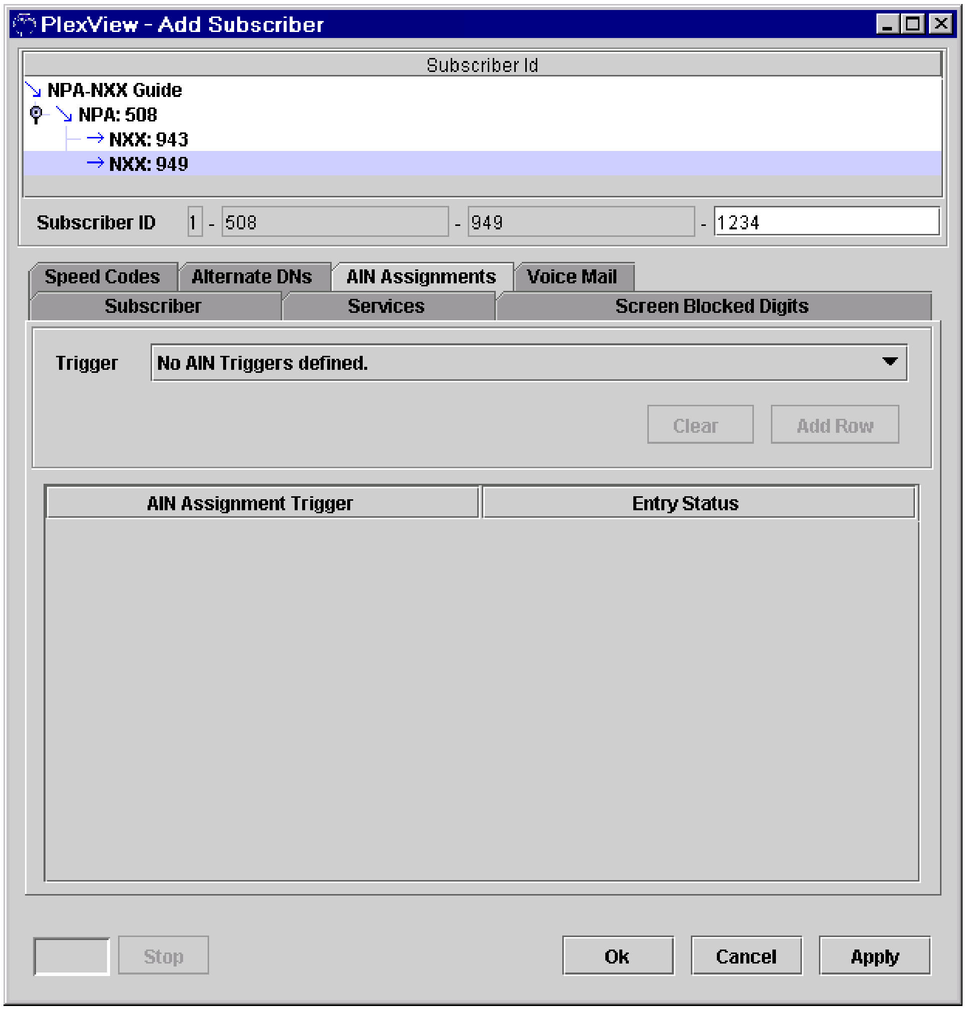
For details about provisioning AIN parameters and settings, refer to the *PlexView Element Management System (EMS) User's Guide*.

Figure 11. Add AIN Trigger



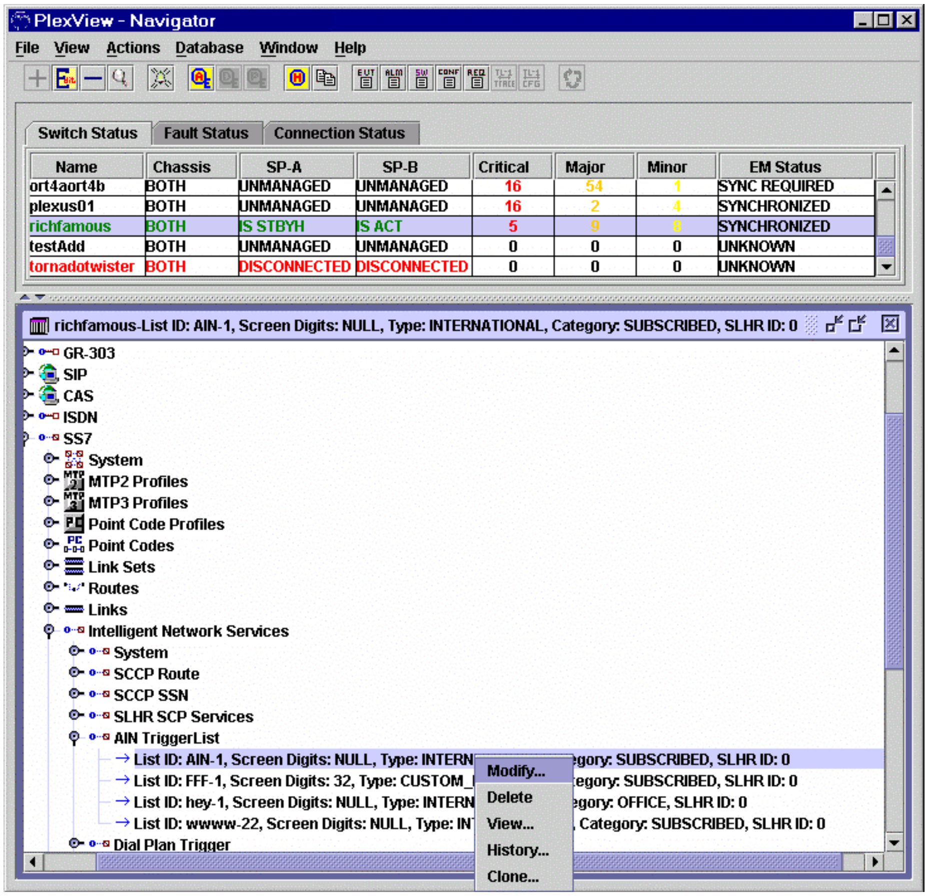
The Trigger Detection Point (TDP) number identifies the point in the basic call processing (based on the Basic Call Model), which identifies when a trigger can be detected and reported to an SCP.

Figure 12. Add Subscriber > AIN Assignments



Prior to provisioning an AIN trigger for a subscriber, you must first provision or add the AIN Trigger name located at SS7 → Intelligent Networks on the EMS Navigator. See Figure 13 for an example.

Figure 13. AIN Trigger



Related Documents

- *EMS User's Guide*

9-1-1 Service

This document contains the following sections:

9-1-1 Service Overview

What is Basic 9-1-1

Initially, there was Basic 9-1-1, where the end office (EO) was programmed to point all 9-1-1 calls to a single destination (PSAP). The good news was that all 9-1-1 calls from any telephones served by the end office were completed to the PSAP. The bad news was that end office serving areas rarely line up with political jurisdictions and the original Basic 9-1-1 systems provided no identification of the caller.

What is E 9-1-1

The introduction of ANI allowed the caller's telephone number to be delivered with the call and displayed at the PSAP so that it could be used to identify the caller, and also be used for callback purposes. Having access to the caller's telephone number also meant that the caller's name and address could also automatically be made available by querying a shared ALI database. The feature that separates Basic 9-1-1 from Enhanced 9-1-1 is Selective Routing. Selective Routing is the automatic routing of a 9-1-1 call to the proper PSAP based upon the location of the caller. The introduction of ANI allowed the caller's telephone number to be delivered with the call and displayed at the PSAP so that it could be used to identify the caller, and also be used for callback purposes. Having access to the caller's telephone number also meant that the caller's name and address could also automatically be made available by querying a shared ALI database. Selective Routing is controlled by the Emergency Service Number (ESN), which is derived from the customer location. An ESN is a three to five digit number representing a unique combination of emergency service agencies (Law Enforcement, Fire and Emergency

Medical Service) that serve a specific range of addresses in a particular geographical area.

Scope of this Document

This document focuses primarily on the E 9-1-1 capabilities of a Lucent switch serving as an EO or a tandem and connecting to either an E 9-1-1 tandem, or in some cases, directly to the PSAP, since the original 9-1-1 service called Basic 9-1-1 (B9-1-1) is rapidly being phased out.

Enhanced 9-1-1 in Lucent Compact Switch

Supported Subscriber Types

All subscriber types on the Compact Switch support E 9-1-1 requirements for US markets, including:

- CAS
- GR-303
- ISDN
- MGCP

Supported Trunk Types

All trunks and trunk groups on the Compact Switch support E 9-1-1 requirements for US markets, including:

- BICC
- CAS
- ISUP
- SIP
- SIP-T

Dedicated and Shared Trunks

Trunk requirements reflect dedicated E 9-1-1 trunks from an EO to an E 9-1-1 tandem. When calls share trunks with non 9-1-1 calls, the Simulated Facilities Groups (SFG) feature is required.

E 9-1-1 Tandem

The term "E 9-1-1 tandem" is referenced throughout this document. The E 9-1-1 tandem carries out many specialized functions. This document addresses the switch serving as an EO or local tandem for 9-1-1- calls. But the Lucent switch does support trunking directly to PSAPs and making routing decisions about what PSAP a call should go to. Additionally, multi-party lines and coin services, each having specialized E 9-1-1- requirements, are not addressed in this document.

PSAP

A PSAP (Public Safety Answering Point) can receive calls routed by E 9-1-1 tandems or directly from switches on dedicated 911 trunks. The Compact Switch can route directly to a PSAP.

E-9-1-1 Operation

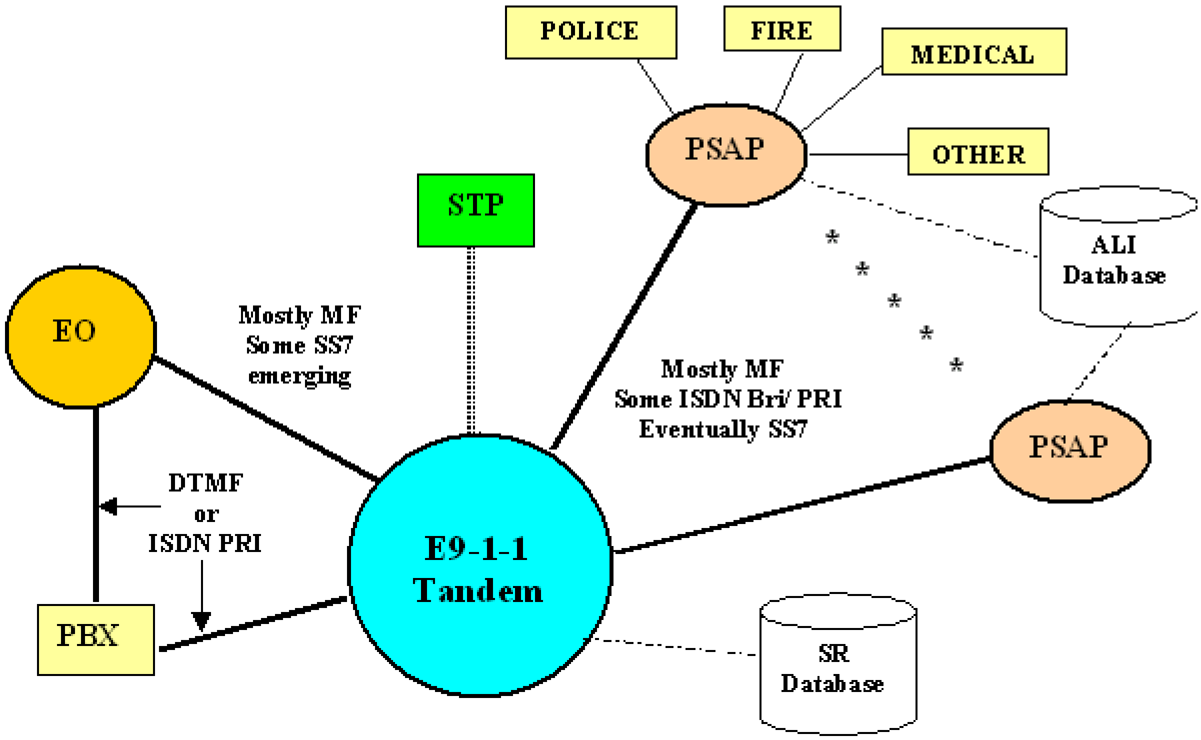
Figure 1 shows a representative architecture for delivering E 9-1-1 calls showing network elements, interfaces, and interworking points for the E 9-1-1 signaling architectures.

Example E 9-1-1 CALL FLOW

1. The caller dials 9-1-1 at the originating station.
2. The EO routes the call to the E 9-1-1 tandem on a dedicated MF trunk for E 9-1-1 service outpulsing the 7-digit Automatic Number Identification (ANI), which identifies the originating station. (Ten-digit ANI is an option)

   End offices are homed on at least two E 9-1-1 tandems: a primary tandem, and a secondary tandem in case the call cannot be completed to the primary tandem. The following figure does not show the secondary tandem. The EO (Lucent switch) can route directly to the PSAP when the Selective Routing (SR) database resides in the Lucent switch. The following figure does not show that call flow.

Figure 1. E 9-1-1 Call Flow



ALI	=	Automatic Location Identification
EO	=	End Office
PBX	=	Private Branch Exchange
PSAP	=	Public Service Answering Position
SR	=	Selective Routing
STP	=	Signal Transfer Point

3. The E 9-1-1 tandem or Lucent switch determines how to properly route the call. The E 9-1-1 tandem uses the incoming trunk group identifier to determine the Numbering Plan Area (NPA). The NPA and 7-digit ANI are used in the SR database to determine the proper PSAP responsible for that originating station in the municipality.
4. The E 9-1-1 tandem or Lucent switch routes the call over dedicated trunks to the E 9-1-1 PSAP.
5. If the E 9-1-1 tandem or switch is unsuccessful in routing the call to the primary PSAP, in most cases it the can reroute the call to the designated alternate PSAP. The tandem or switch has the capability to automatically reroute calls if the first PSAP is down, due to facility outages or PSAP outages. It is also possible to set up routing to alternate PSAPs based on date and time schedules. For example, you can route calls to the designated alternate after 6 PM until 6 AM the next morning.
6. Typically, the PSAP CPE launches the ANI/ALI (Automatic Location Identification) query to get the street address (and usually name) and the closest Police, Fire, or

Emergency Medical Facility for that address and delivers the data to the PSAP attendant. Mobile phone location information typically is triangulated between receiving base stations.

7. The PSAP attendant talks to the 9-1-1 caller and determines the need for Police, Fire, and/or Emergency Medical treatment and then passes the call and related location information to the proper agency (agencies) for dispatch. Whether agencies, e.g., Police, Fire or Emergency Medical can or cannot query the ALI database is determined at a state level. In some states only PSAPs can access ALI by state legislation/regulation. In other states, Police, Fire, Emergency Medical and perhaps other designated agencies can also query the ALI database.
8. If the responding PSAP determines that the call should have been sent to a different PSAP, it may reroute the call to another PSAP using the Central Office Transfer functionality at the E 9-1-1 tandem.

If the PSAP attendant gets disconnected from the caller, the PSAP attendant may make an outgoing call using the information provided to construct the 10-digit ANI for dialing.

Routing

The routing algorithm can key off the called party number (CDPN) and route the call to a dedicated 911 trunk or recursively key off the calling party number (CGPN) to determine how to route E 9-1-1 calls depending on NPA. This supports routing calls to E 9-1-1 tandems or PSAPs in environments where:

- Multiple PSAPS may serve a single NPA
- Multiple NPAs may report to a single PSAP

A call flow diagram of multi-key routing is shown in the following figure.

For 911 calls, the route can go directly to a 911 trunk or operate recursively to add the calling party information (CGPN) to its already known called party information (CDPN). Calling party information is critical for trunk selection on 911 calls..

Provisioning

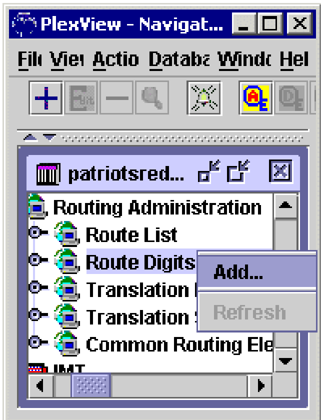
An example of provisioning required for multi-key routing using the PlexView Element Management System is shown here. This is not intended to show a complete or comprehensive routing example. It shows selecting 911, keying off the CDPN 911 to the CGPN route partition key, and where the router finds the appropriate CGPN NPA and routes the call to the desired trunk group.

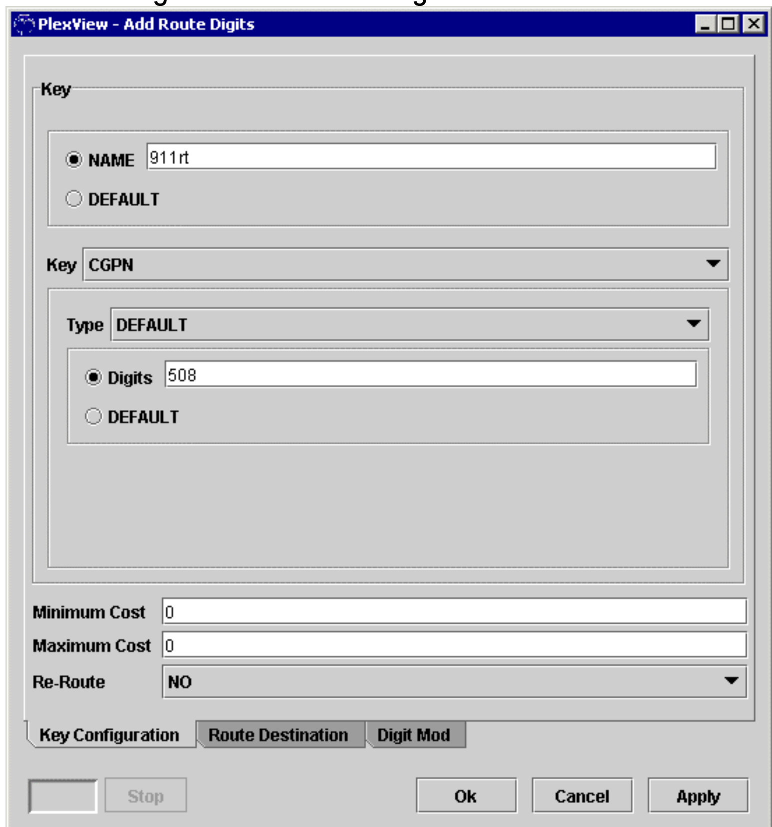
The first command is Add Route Digits. The TL1 equivalent command is ENT-ROUTE-DIGITS. Select Add Route Digits, as shown in the following figure. In the Add Route Digits screen, set the following:

- Key off the CGPN

- Type of DEFAULT
- Digits – enter 508 (the NPA)

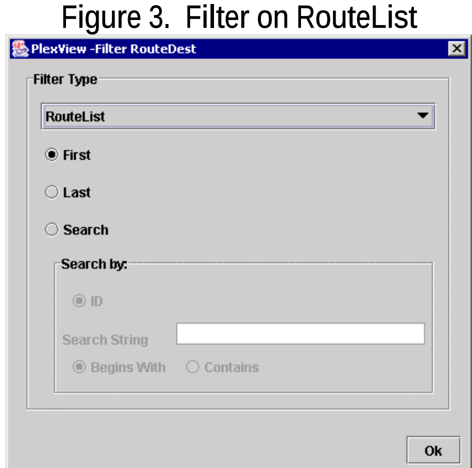
Figure 2. Add Route Digits





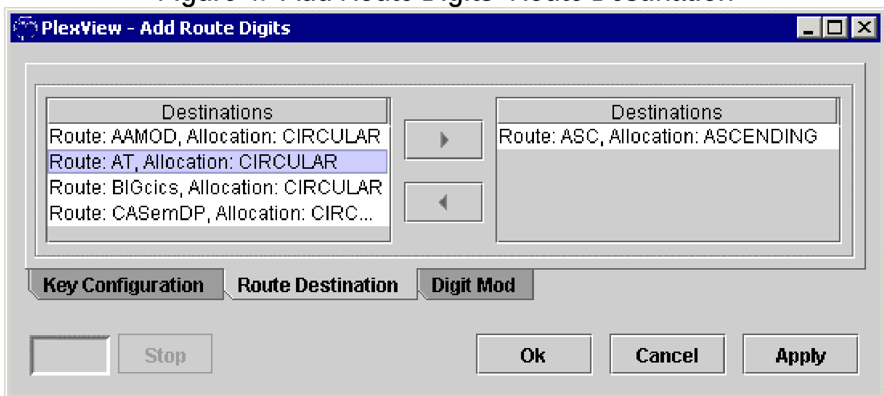
Click the Route Destination tab. In the left (available) field, right-click and select Refresh. Apply the filter as appropriate for your application. In this case, filter on RouteList. Available route lists will display.

Figure 3. Filter on RouteList



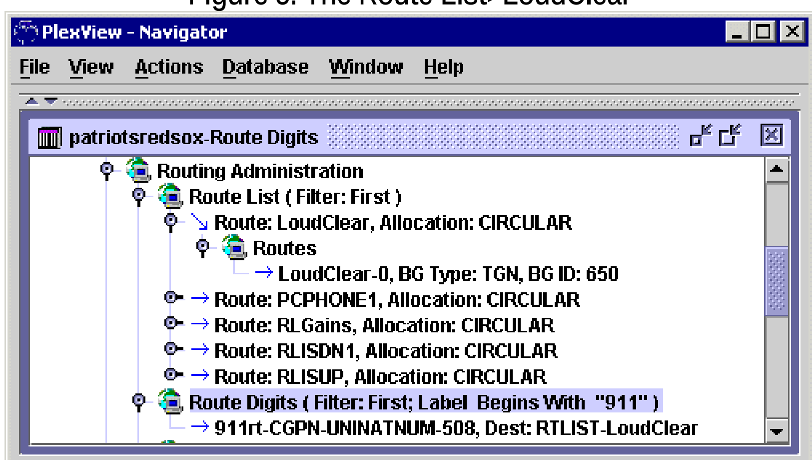
Select the appropriate 911 destination, as shown in the following figure, and select Ok.

Figure 4. Add Route Digits>Route Destination



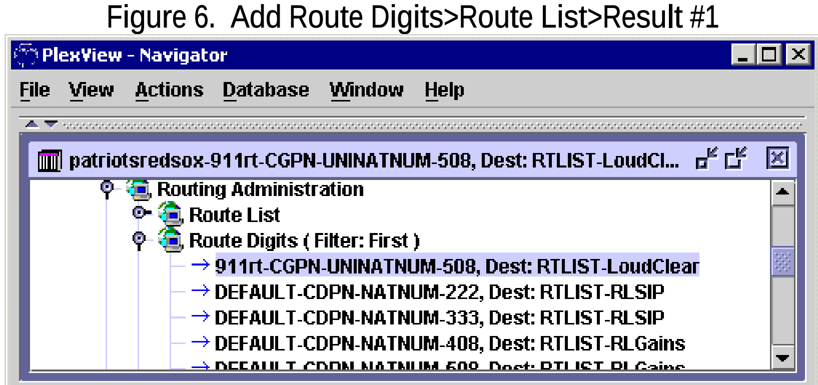
The Route List>Route>Routes>LoudClear hierarchy is shown in the following figure under Route List.

Figure 5. The Route List>LoudClear



The result of the operation is shown in the following figure.

Figure 6. Add Route Digits>Route List>Result #1

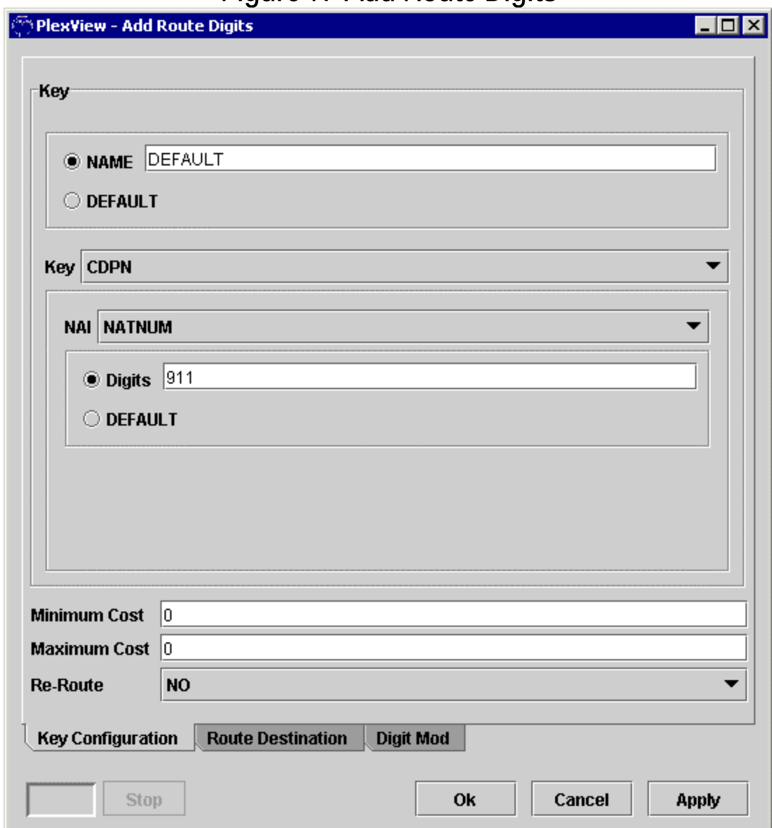


The first part of the highlighted information string shows the route name and that the route keys off CGPN, and is associated with the route list LoudClear.

For 911 calls, calling party information is required by the E 9-1-1 tandem or PSAP and has now been provided. Called party routing is also required for applications where routing decisions must be made. This information is especially true in this example where multiple NPAs use the E 911 tandem and/or PSAP and the call must contain that calling party information.

The routing algorithm operates recursively to add this information. For this example, return to the routing software and add route digits again, keying off CDPN as shown in the following figure.

Figure 7. Add Route Digits

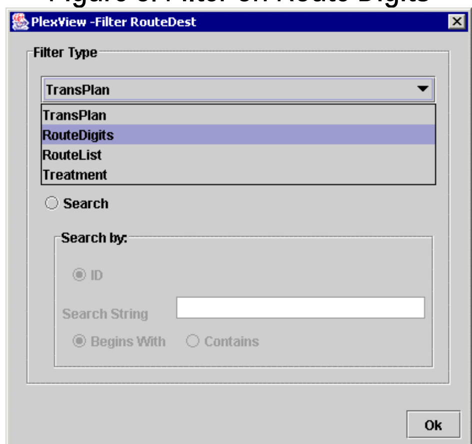


Configure the following:

- Give the route a name
- Key off CDPN
- NAI of NATNUM
- Digits – enter 911

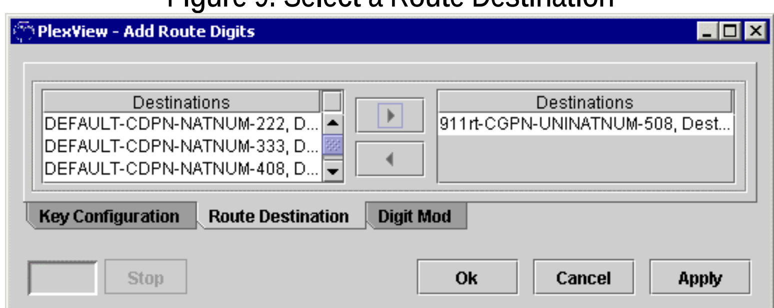
Under the Route Destination tab, right-click and select Refresh. Use the filter with a Filter Type of Route Digits to find the route for your application, as shown in the following figure.

Figure 8. Filter on Route Digits



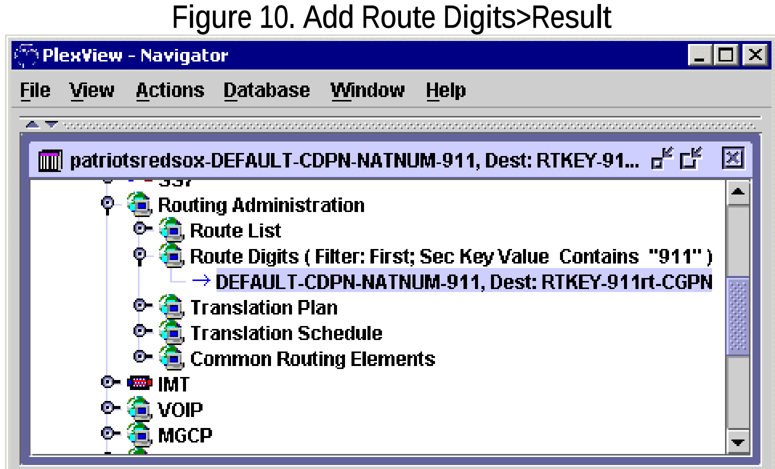
Select the appropriate destination as shown in the following figure and click Ok.

Figure 9. Select a Route Destination



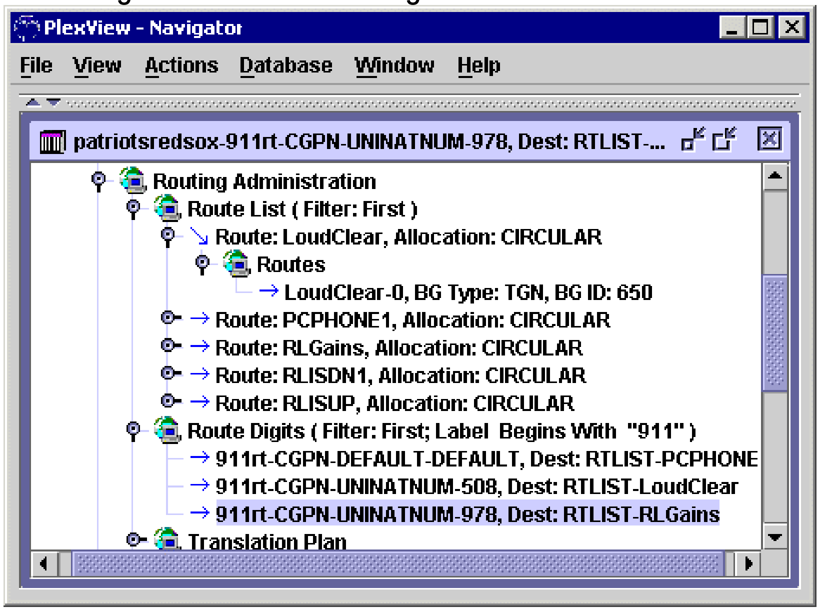
The main screen will show the new route and key, as shown in the following figure.

Figure 10. Add Route Digits>Result



If you want to have a second NPA go to the same RouteList, configure the route again, starting from Figure 2. Add Route Digits and ending with Figure 4. Add Route Digits>Route Destination. In this example the second NPA on the route list is 978. The result looks like the following figure.

Figure 11. . Add Route Digits>Route List>Result #2



This routing example now keys two NPAs to a destination. This example is complete.

Release 3.11 Enhancements

Basic 911 Ring Back

This feature allows the Public Safety Answering Point (PSAP) to alert the originator of the 911 call, by providing a ringing condition to a terminal that is returned to an on-hook condition, or a receiver off-hook tone (ROH) to a terminal which is not in the on-hook condition (reference: ANSI Standard T1.628-2000 "Emergency Calling Service").

When a subscriber dials 911 he/she is connected to a 911 operator and call control is transferred to the Called Party. If the subscriber that dialed 911 hangs up, the connection to the 911 operator is held up until the 911 operator position hangs up the call. During this period the operator can cause alerting signals to be applied to the callers phone by signaling the Compact Switch by applying a wink to the trunk connecting the PSAP to the Compact Switch.

Notes:

- The B911_Trunk Call Model, previously named the 911_Trunk Call Model, is only applicable for a CAS Trunk Group.
- The PSAP hangs directly off the CAS trunk. It is not a CAS trunk to another switch where the PSAP is homed. Nor can the PSAP be homed directly on the Compact Switch, via, for example, a CAS Line.

CAS and GR-303 lines are supported as the B911 originator but MGCP endpoints are NOT supported. It is NOT applicable to PRI interfaces.

The feature can be activated/deactivated via the ED-SERVICE-ACCESSCODE TL1 command for service B911. If you are using the EMS, this feature can be activated/deactivated by double-clicking the *System Services* and then *Service Access Code*. Right-click *BLV_SERVICE* and select *Modify*. Provision *IS*. Select *Ok* or *Apply*.

Notes:

CALEA Feature Package

This document contains the following sections:

Scope

Communications Assistance Law Enforcement Agency (CALEA) is a federal law that requires service providers to facilitate criminal communications surveillance in the digital age. Section 103 of the act states the requirements for telecommunications carriers to support law enforcement agencies (LEA) in the conduct of lawfully authorized electronic surveillance (LAES). Pursuant to a court order or other lawful authorization, carriers must be able to:

- Isolate all wire and electronic communications of a target subject transmitted by the carrier within its service area.
- Isolate call-identifying information of a target.
- Provide intercepted communications and call-identifying information to law enforcement.
- Carry out intercepts discreetly, so targets are not made aware of the electronic surveillance, and in a manner that does not compromise the privacy and security of other communications.

In addition to the above requirements for typical surveillances involving two parties, the Department of Justice (DOJ) includes additional capabilities to deal with advanced services available from modern Class 5 switches and a digital network. The Lucent Compact Switch currently meets the following additional functional requirements (besides simple intercept and monitoring) required by the DOJ:

- *Content of subject-initiated conference calls* – the capability that permits an LEA to monitor the content of conversations by all parties connected via a conference call when the facilities under surveillance maintain a circuit connection to the call.
- *Party drop, remove, join on conference calls* – the capability that permits an LEA to identify the parties involved in a conference call conversation at all times.
- *Subject-initiated dialing and signaling information* – the capability that permits an LEA to be informed when a subject under surveillance uses services that provide call-identifying information, such as call forwarding, call waiting, call hold, and three-way calling. It excludes signals generated by customer premises equipment when no network signal is generated.
- *In-band and out-of-band signaling* – the capability that permits an LEA to be informed when a network message that provides call-identifying information (for example, ringing, busy, call waiting signal, message light) is generated or sent by the Intercept Access Point (IAP) switch to a subject using facilities under surveillance. It excludes signals generated by customer premises equipment when no network signal is generated.
- *Timing information* – the capability that permits an LEA to associate call-identifying information with the content of a call. A call-identifying message must be sent from the carrier's IAP to the LEA's Collection Function within eight seconds of receipt of that message by the IAP at least 95% of the time, and with the call event time stamped to an accuracy of at least 200 milliseconds.

Definitions

The following definitions are used throughout this document.

Associate: A telecommunications user whose telecommunications facilities are communicating with a subject (see Subject).

Call Content Channel (CCC): For a given surveillance, call content (speech) is delivered from the switch serving the subject to an LEA over a CCC. CCCs are established via PSTN connections. When the Intercept Access Point (IAP) switch needs to deliver call content, it replicates the content from the switching matrix (conversation path) and places a copy onto the call path to the appropriate LEA(s). Each time a call content channel is assigned to deliver call content, a message indicating channel identity is sent over the call data channel to the LEA.

Call Data Channel (CDC): A data communications link connecting the service provider to law enforcement. Messages related to the progress of a call under surveillance are delivered on the CDC to law enforcement. Most interceptions require only a CDC, but some require a CDC and a CCC.

Intercept Access Point (IAP): The point within a service provider's network where call-identifying information or call content is accessed; that is, the Compact Switch serving the subject of surveillance.

Law Enforcement Agency (LEA): A government agency (for example, the FBI or a local police department) with legal authority to conduct electronic surveillance.

Subject: A telecommunications user whose communications, call identifying information, or both, have been authorized by a court to be intercepted and delivered to an LEA. Identification of the subject is limited to identifiers used to access the particular equipment, i.e. terminal identity.

Lucent Compact Switch CALEA Implementation

CALEA is limited to Compact Switch applications that terminate subscriber lines; that is, PBX interfaces, channel banks, integrated access devices, and GR-303, for example. Moreover, calls to or from the subject must be identifiable to an individual subscriber line termination.

General Capabilities

- Implements the access and delivery functions (The carrier does not need external CALEA equipment to perform the delivery functions.)
- Intercepts both originating and terminating calls to or from a subject directory number (DN)
- Allows up to 386 CDCs (call identifying interceptions).

- Allows up to 75 simultaneous CCCs (call content interceptions), each broadcast to as many as five independent surveilling agencies.
- Can provision up to three DNs per LEA, per case.
- Combines CCCs into a single DS0 for delivery to the LEA. An internal bridge accesses the transmit side of the subject's conversation, and the transmit side of the associate(s) transmit path. Those two paths will be mixed on a single outgoing CCC DS-0 for the delivery function. The CCC delivery function will send the combined CCC to the Law Enforcement Agencies (LEAs) via a regular PSTN call per the J-STD's "Static Directory Number CCC Delivery".
- Sends call data to the LEA via IP packets on the CDCs.
- Sends a CDC and CCC to as many as five LEAs for a particular surveillance. The DN correlates the CDC and CCC at the LEA.
- Accesses communications among associates on a subject-initiated conference call (Conference bridging is limited to three-way calling.)
- Assigns surveillance either by DN or by physical termination (port). For DNs which represent multiple lines or PRIs, each DS0 will be monitored as one of the 75 maximum supported simultaneous CCCs.
- Maintains transparency of LAES to the subject and any associates participating in the call.

Surveillance Initialization, Activation and Deactivation

The Compact Switch initiates, modifies, and terminates surveillance manually using TL1 commands. The Compact Switch does not access the content of an active call if it cannot do so transparently. In this case, call content access begins with the next call related to the surveillance. Deactivation of surveillance on a subject's line immediately terminates access to call-identifying information and call content. The Compact Switch makes a second attempt to establish a CCC to an LEA when network congestion prevents the first attempt to establish the CCC. If the second attempt fails, an event report is sent to the carrier's network management system. If a CCC delivery attempt results in a ring no-answer condition, the Compact Switch times out and sends an event report to the carrier's network management system. The timeout interval is provisioned by the carrier.

CCC Delivery

Delivery of Call Content to LEAs is provided using dial-up calls made by the Compact Switch to each LEA. No external equipment is required. When a suspect goes off hook, the Compact Switch seizes one of its own DS0 channels and places a normal POTS call to the LEA(s), using DNs provisioned in the Compact Switch for that surveillance. The call path is one-way from the Compact Switch to the LEA.

CDC Delivery

Delivery of call-identifying information on a CDC is unidirectional from the Compact Switch network management or signaling Ethernet port to the LEA. The port connects to an IP router with dial-up capability (dial-on-demand). The router, provided by the customer (carrier), initiates PSTN modem calls to the LEA for CDC delivery.

Notes:

- IP routers with dial-on-demand capability are common products that provide multiple simultaneous dial-up sessions.
- The router establishes a PPP session to the LEA CDC equipment.
- The router maps IP addresses to NANP numbers to the LEAs.
- The carrier obtains and provisions IP addresses to support TCP/IP communications (these could be private IP addresses, given the dedicated PPP link application).
- A firewall should be used to protect the Compact Switch from unwanted communications

There are one primary and two alternative methods available to deliver the CDC modem call.

Primary method:

This method, as shown in in the Figure 1, uses the Compact Switch to dial out to a dial-in modem at a DN anywhere in the PSTN network. Connect to another Class 5 switch using an analog (voice frequency) line. A direct connection is made if the router supports VF lines (RJ-11 jacks). Otherwise, a channel bank is needed between the router and the Class 5 switch.

Example:

1. Establish the route for the dial-out modem:

   ```
   ENT-IPROUTE:::::DEFAULTGATEWAY=10.17.0.5,
     ROUTE1=10.17.0.6,
     DESTINATION1=172.16.10.2;
   ```

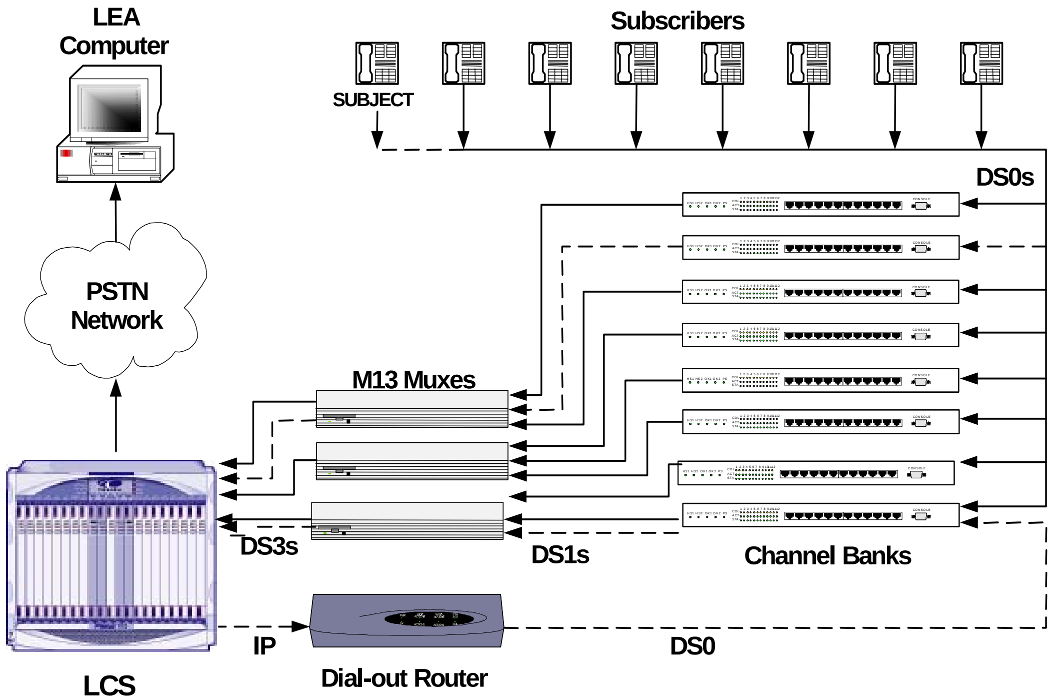
   Where DEFAULTGATEWAY is the IP address of the dial-out modem, ROUTE1 is the IP address of the dial-in modem, and DESTINATION1is the IP address of the CDC server.

2. Establish the CDC:

   ```
   ENT-CDC::FBI-001:::IPADDR=172.16.10.2,
     PORT=8001:IS;
   ```

   Where FBI-001 is the CDC ID, IPADDR the IP address of the CDC server, and PORT is the receiving port of the CDC server.

Figure 1. Primary Method to Deliver the CDC Modem Call



Alternative method 1:

This option uses the Compact Switch as both the dial-out and dial-in switch. Route the modem output to a channel bank after multipexing into a DS-1 signal. Connect the DS-1 to a Compact Switch channel-associated signaling (CAS) DS-1 port.

Alternative method 2:

This option is the direct IP delivery method, in which the Compact Switch delivers CDC messages to the defined IP address of a CDC server anywhere in the TCP/IP network. Connect from the router to the Internet or to a managed IP network. Dial-on-demand software is not needed for this option.

Security

The Lucent Compact Switch

- Supports a separate database for CALEA provisioning, managed by the carrier's CALEA Administrator

- Requires CALEA-specific identification and authentication for CALEA administration
- Allows only identified and authenticated users to establish CALEA sessions
- Enables the system administrator to establish the maximum number of invalid session setup attempts allowed (such as illegal user IDs or passwords) before an intrusion attempt is suspected and alert messages are generated.

Notes:

SIP Software Feature Description

This document contains the following sections:

Introduction

The Lucent Compact Switch offers SIP trunking over IP networks and also terminates SIP calls to PSTN subscribers. The Compact Switch allows SIP callers to reach Compact Switch subscribers and allows Compact Switch subscribers to reach VoIP recipients. The Compact Switch carries calls to/from subscribers on SIP trunks. SIP subscribers are not supported on the Compact Switch itself. The SIP access interface in the Compact Switch connects to an IP feature server. The feature server controls the CPE hardware (IADs and IP phones) and provides functions such as CLASS features and IP Centrex to SIP subscribers. The Compact Switch acts as the PSTN Trunking Gateway in this application.

The SIP Software Feature Package includes the following capabilities:

- PSTN gateway for SIP initiated calls
- IP Gateway for PSTN calls
- International and inter-national voice calls over SIP trunks for toll avoidance
- Class 4 IP tandem trunking
- Class 5 Voice termination

The SIP trunking interfaces to external SIP feature servers enables the following features/services:

- IP Centrex
- Calling Card
- Unified Messaging

- Conferencing
- Call Center
- Attended or semi-attended directory assistance

Intended Audiences

This document has three intended audiences. The first audience wants high-level information on what VoIP is, what SIP is to VoIP, and how the switches support SIP. For this audience, read continuously from the beginning through the section titled SIP Trunking Applications Using the Compact Switch.

The second audience is network engineers interested in how the Compact Switch supports SIP, what's required in the Compact Switch for SIP support, and how that support is set up. That audience should include SIP Trunking Applications, SIP Calls, and SIP Support. Also, if interested in the mechanics of how the interfaces communicate with external servers, review the section titled Configuring SIP using the EMS or TL1, a section that is quite detailed, especially in communications protocol choices for Ethernet cards and Voice Server cards.

The third audience is interested in how switches will integrate into their networks, what interfaces are supported, and how to provision the Compact Switch to communicate with external servers and trunk SIP calls. For that audience, checking the SIP Trunking Applications section, and then moving to Switch Requirements to Support SIP, and then the Configuring SIP using the EMS or TL1 provides useful technical guidance.

SIP Overview

The Session Initiation Protocol (SIP) initiates, modifies and terminates VoIP sessions plus performs call control. It does not attempt device control for those sessions, like MGCP and Megaco do. SIP is a scalable, extensible protocol that works well with other protocols. Sessions that SIP establishes can be as simple as a two-way telephone call or they can be as complex as a collaborative multi-media conference session. As a request-response protocol, SIP closely resembles two other key Internet protocols, HTTP, the primary engine of the World Wide Web, and SMTP, the primary email protocol. Using SIP, telephony becomes another IP application and integrates easily into other Internet services. SIP has become the protocol of choice for IP signaling in VoIP applications because of its flexibility and ease of use.

Major Functions

SIP provides four basic functions:

- User location
- Feature negotiations
- Call management
- Session changes

SIP transparently supports name mapping and redirection services, supporting the implementation of ISDN and Intelligent Network telephony subscriber services. These facilities also enable personal mobility, allowing end users to originate and receive calls and access subscribed telecommunication services on any terminal in any location. The network can still identify end users as they move.

Callers and call recipients are identified by SIP addresses. When a SIP caller makes a call, the SIP phone contacts its homed server and then sends a SIP request. The most common SIP request is the invitation (INVITE). Instead of directly reaching the intended call recipient, a SIP request may be redirected or may trigger a chain of new SIP requests by proxies before reaching the recipient.

Types of Network Implementations

Peer-to-Peer VoIP

Voice over IP (VoIP) is a way of providing phone service over IP networks at prices lower than the prices offered by traditional voice service providers. Services such as Skype, created by the founders of KaZaA, offer peer-to-peer (P2P) VoIP freeware and billable premium services, with each end connecting to a PC. Free World Dialup is another example of the possibilities of P2P VoIP. This is the kind of VoIP is often hyped in the media, and these services run entirely over the IP network.

Voice over an All IP Network

Cable operators offer voice over their cable systems. Some cable modems come with Ethernet ports for phone connections. Voice over DSL broadband is also available with similar connectors. Microsoft now includes SIP in its Windows XP operating system and the company has expressed an interest in providing voice services between PCs. Though voice services over an all IP network have been around for some time, and those voice services are steadily increasing in quality, revenue opportunities have not matured and the market for all VoIP telephony has not yet reached its potential.

Voice over IP that Terminates in the PSTN

VoIP calls have to interact with the PSTN to terminate to traditional phones or cell phones. Because the PSTN is highly regulated, many features that go beyond call setup and voice quality must be dealt with before voice features are seamless on IP networks. PSTN quality offers something better than best-effort IP service. VoIP is still struggling with how to meet latency, echo cancellation, and jitter quality of the PSTN. Some other PSTN requirements include:

- Emergency (911) services
- CALEA (law enforcement monitoring)
- Directory assistance
- Life Line (works during power outages)

The telephone system is transparent to end-users. Users just dial someone and they either connect or don’t connect to them. As new users grow up with cell phones, text messaging and cell phone video, and mobility becomes more of an issue, phone and Internet services interworking will become even more important.

Compact Switch SIP Capabilities

SIP Trunking

The Compact Switch performs class 4 inter-machine SIP trunking and interfaces with external feature servers to assist with class 5 voice call functions for calls initiated by SIP devices that terminate to Compact Switch subscribers (MGCP, CAS, GR-303, ISDN).

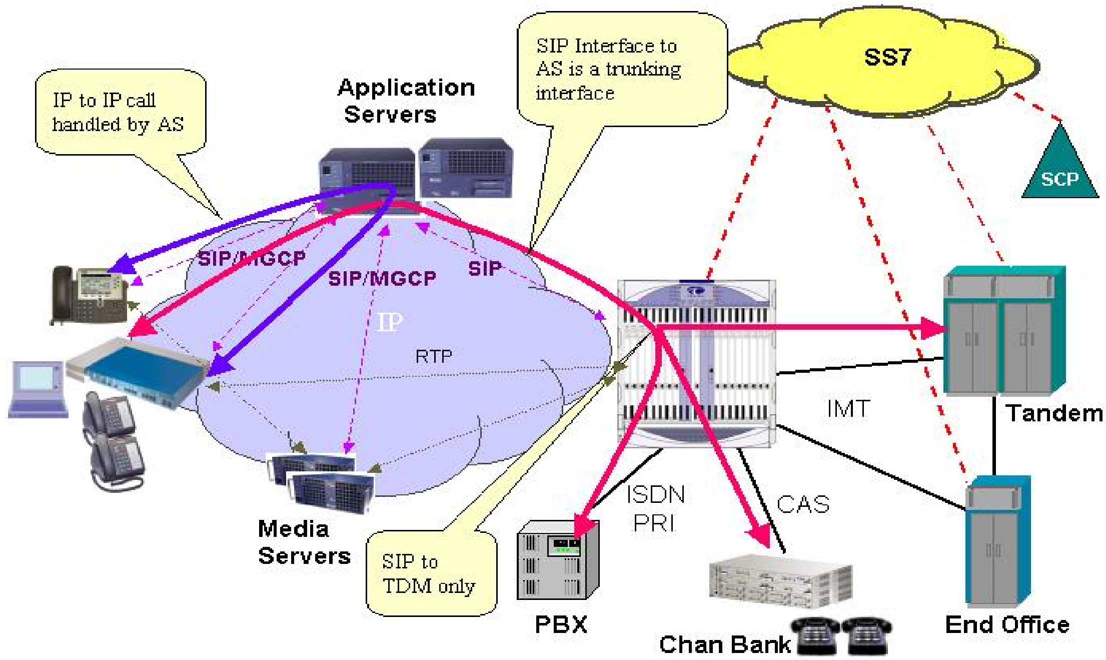
A SIP trunk group uses SIP call signaling protocols to establish and tear down calls. The Compact Switch supports a SIP trunk interface and not a SIP line interface. No SIP line side subscribers can be directly configured on the Compact Switch. Connecting with this interface is the basis for the support of the various applications such as SIP trunking for wholesaling and long distance.

SIP Subscriber Access

The SIP trunking interface can also connect to a feature server. The feature server controls the CPE hardware (IADs and IP phones) and provides features such as CLASS and IP Centrex to subscribers. The Compact Switch acts as the PSTN Trunking Gateway in this SIP application.

Another way of looking at it is that the combination of the Compact Switch and the feature server comprise a decomposed Class 5 switch. The feature server handles all calls from the SIP controlled IAD or IP phone. If the destination of the call is to another IP subscriber, it routes the call within the IP network. If the call is destined for the PSTN (based on NPA-NXX), then the call is routed via SIP to the Compact Switch for PSTN termination. The following figure shows the architecture of the Compact Switch connecting to a feature server for providing IP Centrex features and PSTN access to IP subscribers.

Figure 1 - The Switch Provides IP Centrex and PSTN Access for IP



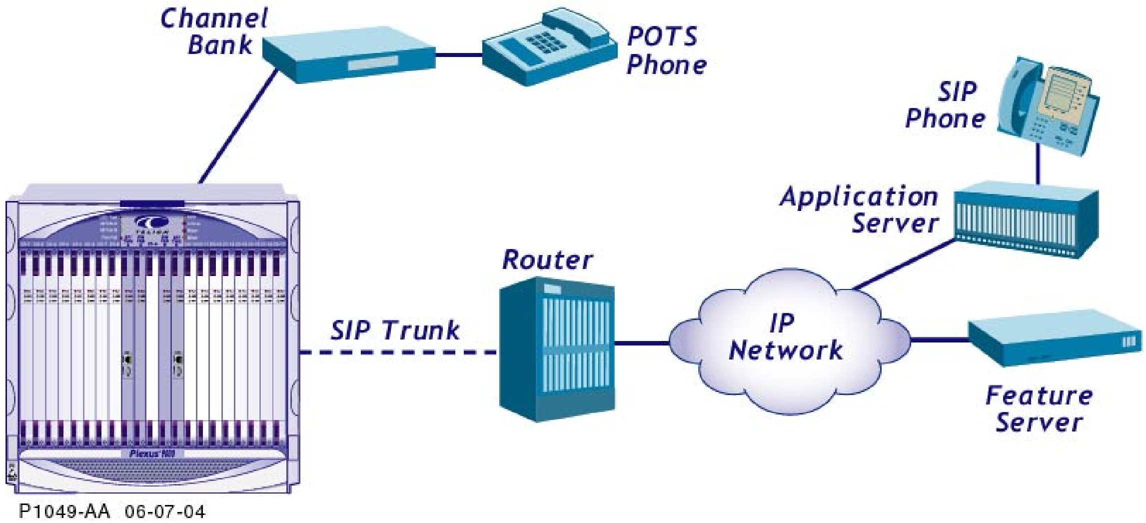
For inter-LATA, intra-LATA and international calls, the external connected server must supply a carrier ID to the Compact Switch. The Compact Switch can then route the calls by carrier.

The Compact Switch allows SIP callers to reach Compact Switch subscribers and the Compact Switch allows Compact Switch callers to reach VoIP subscribers. The interface between the Compact Switch and a feature server is configured as a SIP trunk. Line side IP subscribers are supported on the feature server side of the trunk. The Compact Switch offers trunk services to the feature server, such as call routing into the PSTN, but not line features such as dial tone or CLASS features. SIP subscribers are configured outside the Compact Switch on the IP side. All subscriber specific information is passed from the feature server to the Compact Switch in the SIP INVITE message. The Compact Switch handles the traditional PSTN functions of LNP, 8xx toll-free numbers, operator access and E-9-1-1. Since the subscribers are on the feature server, it handles obligations like CALEA features. When a SIP call for a SIP subscriber arrives at the Compact Switch, it forwards that call to the local application server, feature server, or media gateway for call setup.

SIP Trunking Applications Using the Compact Switch

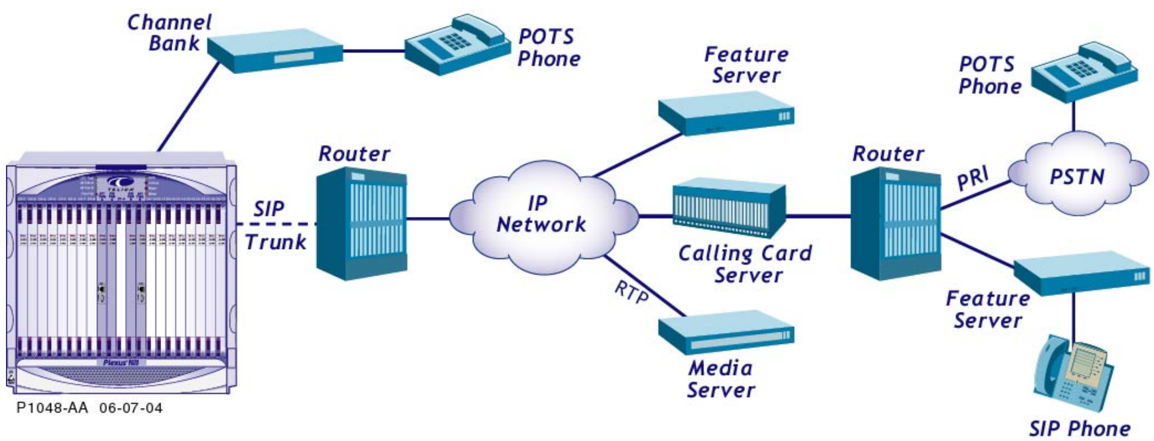
SIP trunks are typically used in IP-IP connnections, as shown in the Figure -2.

Figure -2. Intermachine IP Trunking



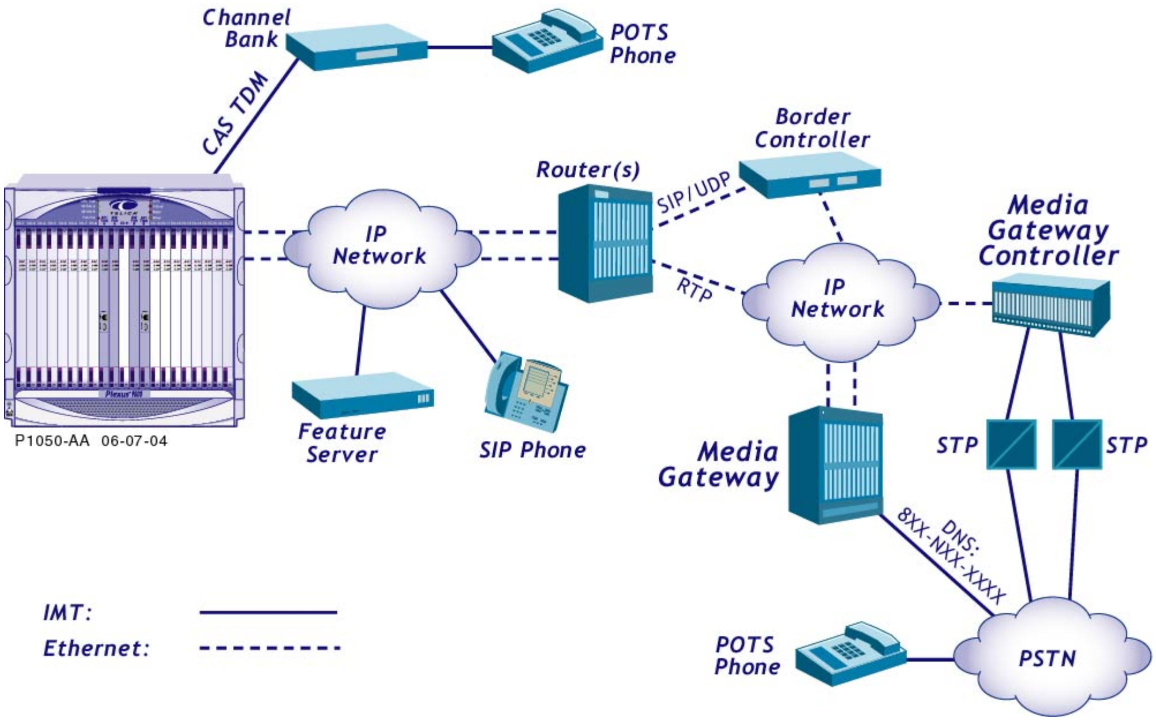
SIP trunking also supports international long distance calling, as shown in Figure 3.

Figure 3. International Long Distance



SIP trunking can be configured so the Compact Switch acts as a bridge between the PSTN and IP worlds. For example, if a calling card over PSTN service wants to interoperate with an IP based carrier, and both have existing networks, the Compact Switch in the calling card carrier's network can facilitate interoperation, as shown in the Figure 4.

Figure 4. Carrier Interoperation



SIP Calls Using the Compact Switch

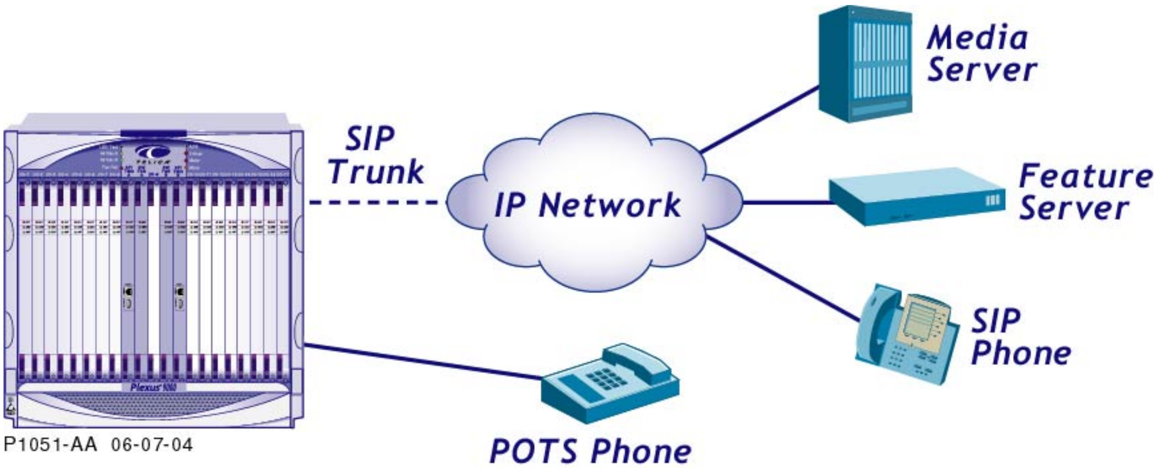
Basic SIP Call Setup

This section describes a simplified SIP call setup for a call initiated by a feature server that terminates to a TDM subscriber on the Compact Switch. It does not address IP to IP SIP calls, calls in large networks with stateful and stateless proxies, complex call setups, multiple INVITE requests, call forking, or other call setup types.

To initiate a session from a SIP phone to a Compact Switch subscriber, the caller sends an INVITE request to the called party (Compact Switch subscriber). The INVITE is not routed directly to the called party, but to the SIP phone's homed server and possibly other external servers for call processing. The external server type depends on the network architecture and the size and scale of the network.

That server, or series of servers in the IP network, forwards the INVITE to other servers or, if the server knows about the called party, forwards it to the called party, as shown in Figure 5.

Figure 5. Simple Call Setup



The called party is addressed as a URL, IP address, or phone number. The Compact Switch requires NPA-NXX numbers, so the feature server must be able to match the called party to a phone number. In the case of a SIP to TDM call, the phone number of the called party is used to route the call to the called party through a SIP trunk.

Coming into the Compact Switch on a SIP trunk, the call goes through the Compact Switch to the called party. The called party responds to the caller, either by answering the call, having an answering machine pick up the call or not answering at all, and the response is forwarded back on the IP network to the calling party. An acknowledgement is sent and then the bearer channel is established for voice communication. The call itself uses RTP and goes over a bearer channel that takes a different path from the setup signaling.

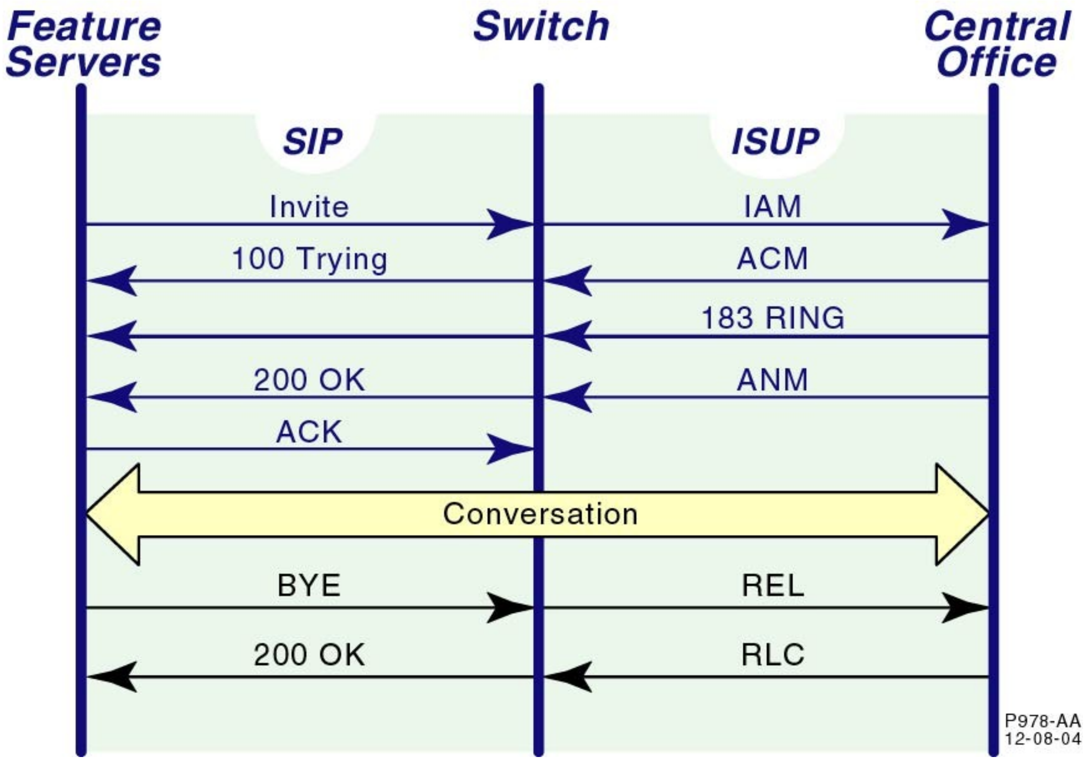
SIP Call Termination

The Compact Switch supports TDM (ISUP, ISDN, CAS) call termination from a SIP phone and feature server(s) in the IP network to the Compact Switch or vice versa. It initiates a TDM call that can terminate in a SIP network when the feature server, application server or media gateway provides the necessary call translation. That outboard server can also add CLASS features to the call.

SIP to ISUP Call INVITE

In the SIP to ISUP circuit as shown in Figure 6, an INVITE message is sent from the calling SIP phone to the feature server, and from the feature server to the local Compact Switch. The Compact Switch converts the INVITE into an IAM (initial address message). The IAM contains the destination and CIC (carrier ID code).

Figure 6. SIP and ISUP Call Setup



The called party responds to the Compact Switch with an ACM (address complete message) indicating that all addressing signals have been received and the phone is ringing. The Compact Switch sends a 100 TRYING message to the feature server, which forwards it to the SIP phone.

Between the 100 TRYING message and the first 200 OK message, the Compact Switch updates the feature server with code 183 SESSION PROGRESS (SDP-3) status messages. The feature server also forwards the 183 status messages to the SIP phone so the caller hears ringing.

When the called party answers and the phone is off hook, an ANM (answer message) is returned to the Compact Switch. The Compact Switch sends a 200 OK message to the feature server, and the feature server forwards it to the SIP phone. The SIP phone sends an ACK (acknowledge) message to the feature server. The feature server sends the ACK to the Compact Switch. The Compact Switch connects the ACK to the called party and now the two parties can communicate. The conversation is on a separate bearer channel. The SIP session setup is complete.

SIP to Switch-ISUP Call REFER

Call REFER redirects calls. Refer can have many scenarios, including automated REFER, semi-attended REFER, and manual call to operator with REFER. For simplicity's sake, this setup describes a call that will be set up and then referred. After the initial call setup to a Directory Assistance Automated Attendant (DAAA), the attendant automatically (automated REFER) connects the caller to the called party. From a network standpoint, the DAAA may be multiple devices, with multiple IP addresses. But from the standpoint of the Compact Switch, the DAAA is one component and communicates over a single interface. The DAAA will be described that way for this simplified call.

Assuming that the caller inputs valid called party information when prompted by the DAAA, the caller will be prompted about whether they want automatic connection, the response will be YES, and the call will be automatically connected by the DAAA to the called party through a SIP REFER. The description will not include external lookups outside the DAAA, handoffs to other servers in the IP network or devices in the PSTN or prompting the caller for semi-automated or operator assisted call referral.

SIP to TDM Call REFER

If the SIP caller calls directory assistance, a DAAA will acknowledge and accept the call, negotiate CODECs with the caller, set up an RTP media session, and then prompt the caller for the called party’s name, city, and state. If the DAAA receives a valid response from the caller and it finds the number, it will forward a SIP REFER to the called party. The called party will respond and the CODECs of the caller and called party will negotiate and the call will take place over a separate media session. The DAAA then drops its connection.

SIP to TDM Call RELEASE

If the SIP subscriber hangs up first, the SIP release session begins. The feature server sends a BYE message to the Compact Switch. The Compact Switch passes on an ISUP REL (release) message to the called party. The called party responds with a RLC (release complete) message to the Compact Switch. The Compact Switch sends a 200 OK message to the feature server. The trunk and line side circuits, and the IP path all go idle and signal that they are available.

TDM to SIP Call RELEASE

If the TDM subscriber hangs up first, the ISUP REL message is interworked by the Compact Switch and converted into a SIP BYE message and sent to the feature server. The call tear-down continues with a RLC message sent and then a 200 OK message sent to the feature server, releasing the call.

Switch Requirements to Support SIP

This section describes the hardware, signaling, interfaces and configuration required to support SIP on the Compact Switch.

Required Hardware for SIP

The Compact Switch requires these cards to support SIP:

- Gigabit Ethernet IOMs
- Voice Server IOM(s)
- DS3 or DS1I/O cards (TDM IOMs)

Hardware Capacities for SIP

The three cards required for SIP have the following per card capacities:

- Gigabit Ethernet IOMs – Four Ports
- Voice Server IOM(s)
  - 2688 G.711, G.723 and G.726 Voice Channels
  - 816 G.729 Voice Channels
- DS3 or DS1I/O cards (TDM IOMs)
  - DS3 – 3 or 8 Ports
  - DS1 – 28 Ports

Required Software for SIP

SIP trunk configuration can be accessed through TL1 or the EMS (Element Management System). The EMS provides a GUI, and interacts with the Compact Switch through TL1.

Software Capacities for SIP

SIP trunks are carried over the Gigabit Ethernet interface. Up to 21,504 SIP calls can be supported on each Compact Switch.

SIP Support

The Compact Switch supports an ever-increasing set of SIP features, including:

- SIP Trunking between VoIP gateways and the Compact Switch
- IP subscriber access (IAD and IP Phones) through feature servers
- SIP calling card use
- SIP calling card re-origination for Pactolus calling cards
- SIP conferencing applications
- Compact Switch to Compact Switch IP trunking
- SIP Interworking with other protocols
  - ➢ SIP-to-SIP - with a voice gateway calling a subscriber on another voice gateway or feature server
  - ➢ SIP-to-MGCP – with a feature server subscriber calling a MGCP subscriber
  - ➢ SIP-to-BICC - with a voice gateway or feature server subscriber calling a subscriber on another Compact Switch
  - ➢ SIP-to-GR-303 – with a voice gateway or feature server subscriber calling a GR-303 subscriber

Calls originating from a SIP interface, with that interface typically being a feature server or voice gateway, can terminate to all supported interfaces on the Compact Switch.

SIP Feature Highlights

SIP-T

SIP-T (SIP for Telephones) provides a framework for the integration of legacy telephony signaling into SIP messages that enables the preservation of received SS7 information within SIP requests at the originating gateway and the reuse of the SS7 information when signaling to the PSTN at the terminating gateway. In addition, SIP-T preserves the routability of SIP requests allowing a SIP request that sets up a telephone call to contain sufficient information in its headers to enable it to be appropriately routed to its destination by proxy servers within the SIP network.

The support for SIP-T, as defined in RFC3372, enables standards-based SIP bridging for the transport of call signaling information across a packet network with third-party MGCs/softswitches, Lucent Compact Switches, running version 3.10.1 or later, and Lucent Network Controllers (MGCs), running version 5.1 or later.

SIP-T Feature Highlights

- Parameter Translation
  - The ISUP parameters are translated into the SIP header per RFC3398
  - The SIP header to ISUP mapping is performed per RFC3392.
  - The information from the SIP header (calling party number, called party number) overrides the equivalent parameters within the ISUP message, since these values may be changed by intermediate SIP proxies.
  - Non-ISUP ingress call legs are mapped to ISUP prior to parameter translation and message encapsulation.
- Message Mapping
  - The ISUP IAM message is carried as a MIME enclosure by the SIP INVITE request;
  - The ISUP ACM is carried as a MIME enclosure by the SIP 18x status message; and
  - The ISUP ANM message is carried as a MIME enclosure by the SIP 200 status message.
  - The ISUP REL message is carried as a MIME enclosure by the SIP BYE or CANCEL message.
  - **Note:** ISUP-to-SIP-T-to-ISUP mid-call signaling is not supported in this version
- Message Encapsulation
  - The encapsulation of the ISUP message within MIME enclosures is in accordance with RFC3204.
- Sending of messages
  - Supported SIP Content Negotiation Types - The sending of the following Content-types is supported for MIME-encapsulated ISUP: application/isup (Content-type: application/isup); required (Content-disposition: signal; handling=required); and optional (Content-disposition: signal; handling=optional - allows the other side to silently discard the MIME-encapsulated ISUP if it does not support it)

- Receipt of messages
  - A MIME-encapsulated ISUP message with a "415 Unsupported Media Type" is rejected if the incoming trunk group is not configured for SIP-T and the content-disposition handling is set to "required".
  - If the incoming trunk group is not configured for SIP-T and the content-disposition handling is set to "optional", the MIME-encapsulated ISUP message is silently discarded and the rest of the SIP message is processed.
  - The receipt of a SIP message from a SIP-T trunk group without a MIME encapsulated ISUP body is treated as if it were received from a SIP trunk group.
- Trunk Group Support
  - A SIP-T trunk group supports all the parameters that were previously supported by SIP trunk groups.

SIP SUBSCRIBE/NOTIFY method for DTMF Relay

SIP SUBSCRIBE/NOTIFY messages transport DTMF tones in SIP without requiring a bearer traffic channel. The applications for this feature include:

- Standardizing calling card call re-origination
- Logging into a feature server from a cell phone and using features formerly only available locally

SIP INFO Method For Generating DTMF Tones

This feature supports speed-dial when initiated from the TDM side of the Compact Switch. The application server tells the Compact Switch to generate DTMF tones toward the PSTN, so a media server is not needed for speed dial support.

SIP REFER and SIP REFER with REPLACES

SIP REFER and SIP REFER with REPLACES provide more efficient ways to transfer calls than using SIP re-INVITEs. They can also be used for replacing or collapsing call legs (similar to ISUP RLT or ISDN TBCT for TDM interfaces).

Supports RFC3515 (SIP REFER), RFC3892 (REFERRED-BY) and RFC3891 (REPLACES) with the following restrictions:

- SIP REFER (unattended call transfer)
  - ➢ Must reference an existing call leg that is inside a call/session (outside of a call/session SIP REFERs are not supported).
  - ➢ The party initiating the transfer (Transferor) must be a SIP call leg.
  - ➢ The party being transferred to the Transfer Target (Transferee) can be a SIP or TDM (ISUP, ISDN, CAS) call leg.
  - ➢ The new party being introduced into the call/session with the Transferee (Transfer Target) can be a SIP or TDM (ISUP, ISDN, CAS) call leg.
  - ➢ The Transfer Target can only be transferred to by initiating a new call (no sending of REFER).

- The SIP REFER routing decisions must be done on the Compact Switch. Using an external device, such as a SIP Proxy, for routing of the INVITE that was initiated by the REFER is not currently supported. This is true for all cases of REFER.

- SIP REPLACES (attended call transfer)
  - The call leg to the Transfer Target that is being replaced may or may not go through the Compact Switch.
  - The Transfer Target can be SIP or TDM (ISUP, ISDN, CAS).
  - If the call leg that is being replaced goes through the Compact Switch, which is always the case for TDM call legs, the transfer replace must be done locally.
  - If the call leg being replaced is a SIP call leg, there is an additional option to have the Transfer Target do the transfer replace (by sending out an INVITE REPLACES).
  - The SIP REPLACES routing decisions must be done on the Compact Switch. Using an external device, such as a SIP Proxy, for routing of the INVITE that was initiated by the REPLACES is not currently supported. This is true for all cases of REPLACES.

SIP REFER/REFER with REPLACES to DTMF Release Link Trunk (new in 3.11)

This feature enhancement supports DTMF Release Link Trunking by sending a “Flash Hook” (a wink) down the originating MF or DTMF CAS trunk, accompanied by the destination number signaled using DTMF. This interface is defined in the 5ESS Flexent/Autoplex Wireless Network Information Services Gateway (ISG). Upon receiving a SIP REFER, the call transfer is invoked via a Flash-Hook followed by DTMF digits. It is configurable on a per trunk group basis. The xferToDtmf and xferPrefix parameters in the ED / ENT-TRKGRP TL1 command are used to enable this capability. In addition, the xferOut parameter of the CAS trunk group must also be enabled to indicate that remote transfers (SIP REFER) will be accepted when this trunk group is the ingress/egress leg of a call.

SIP Session Timer

The session timer audits active calls to make sure they have not terminated on only one side. This cleans up lost sessions and frees up system resources.

SIP PRACK

PRACK (provisional acknowledgements) is a SIP command like INVITE and BYE and it has a unique answer. PRACK enhances the reliability of call provisioning by assuring the reliable transmission of provisional acknowledgements.

SIP Announcement

SIP Announcement supports playing announcements back to a SIP call originator, including playing a spoken phone number at the end of the announcement.

SIP SUBJECT Header Passthrough

With this feature, the Compact Switch passes information from the incoming SIP trunk to the outgoing SIP trunk transparently.

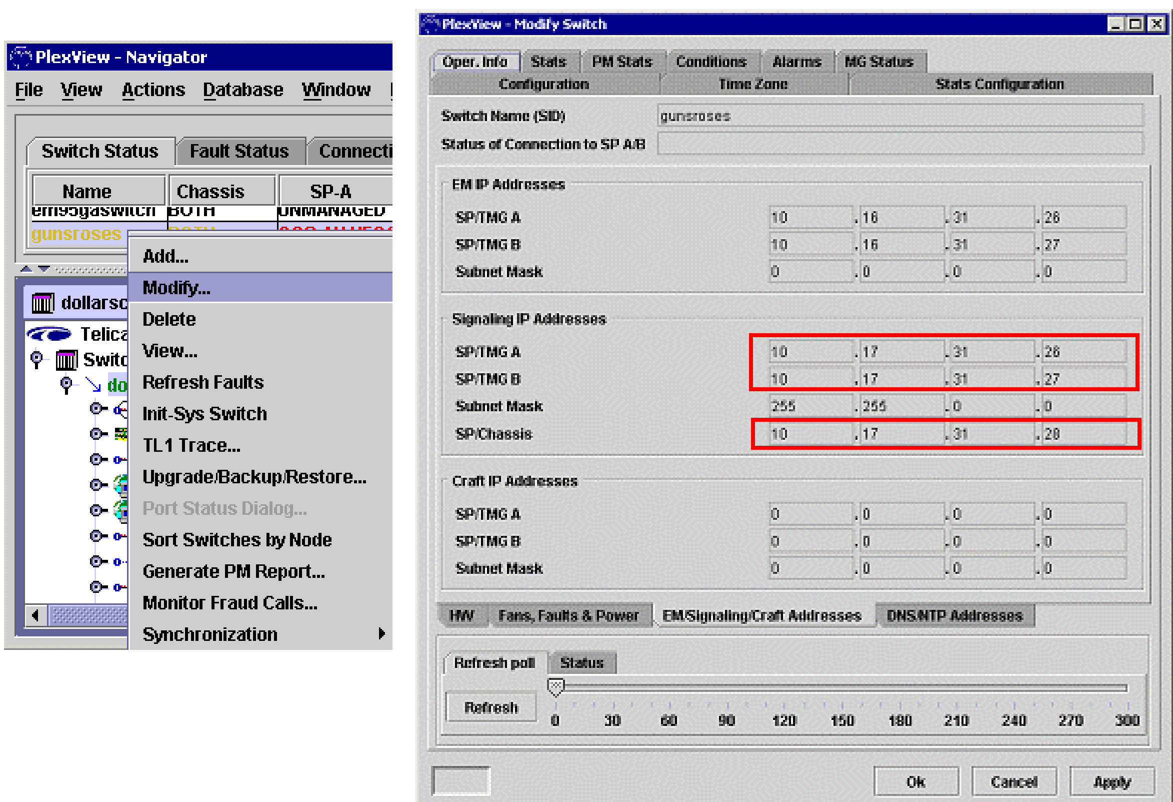
Configuring SIP using the EMS or TL1

This section presents an overview of the important things to consider when configuring SIP on the Compact Switch. Screen shots with key parameters highlighted and briefly explained are shown. This is a guideline, not a comprehensive step-by-step configuration guide.

System Processors (SP) Provisioning

IP addresses, subnet masks and routes must be configured in the System Processor (SP) modules for the signaling ports to operate. SP/TMG-A and SP/TMG-B for the signaling ports must have different addresses, with each address being specific to a signaling port. The SP/Chassis IP address is different and is reachable regardless of active SP. It is configured in devices that do not support IP failover.

Figure 7. EMS Modify SP Dialog



The equivalent TL1 commands are provided in the following table.

Table 1. TL1 Commands for Signaling IP Addresses Provisioning

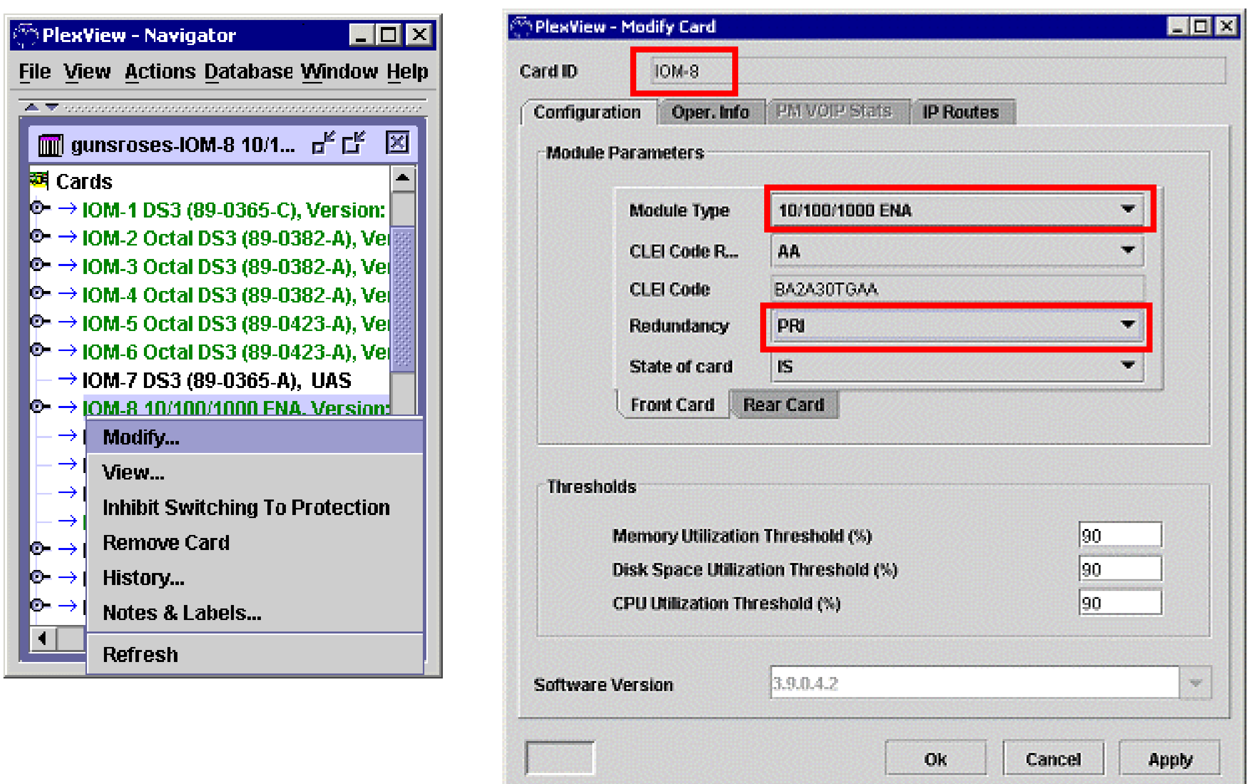
Command and Description	Definition and Parameters
ED-CHASSIS-EQPT	Edit the Signaling IP Addresses for the SP
TL1 equivalent to the red highlighted items to be configured:	signalingIPSPA (SP/TMG A)
	signalingIPSPB (SP/TMG B)
	SignalingIPChassis (SP/Chassis)

Ethernet IOM Provisioning

IP addresses, subnet masks and port information must be configured in the Ethernet Network Adapter IOM ports for bearer traffic.

These Ethernet IOMs reside in slot 8 and 10 of the Compact Switch. Slot 8 must be set to PRI for primary and slot 10 must be set to SEC for secondary. If a failover occurs, the IP address of the slot 8 port becomes the address of the matching slot 10 port.

Figure 8. Ethernet IOM Module Front Card



The equivalent TL1 commands are provided in the following table.

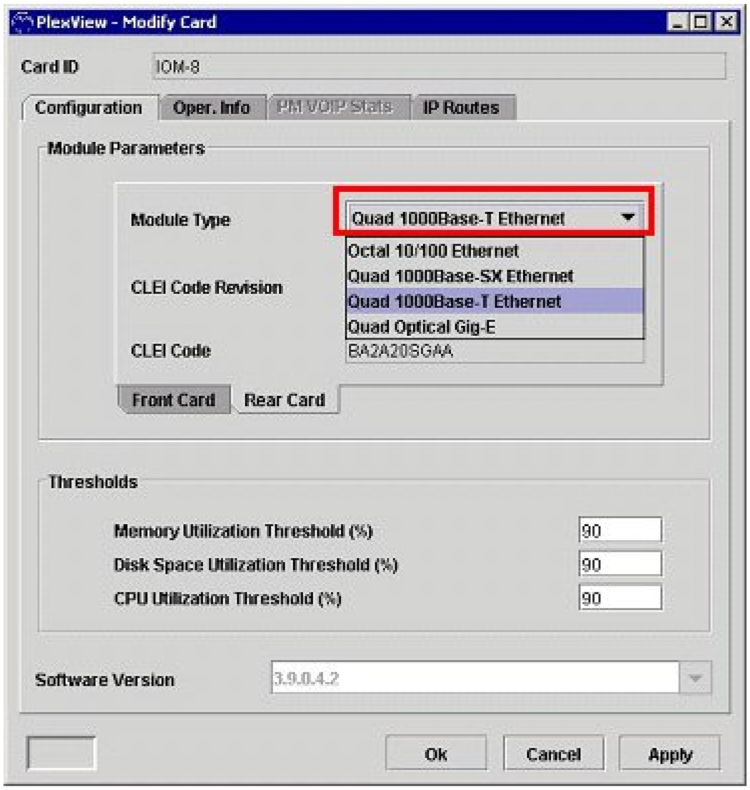
Table 2. TL1 Commands for Ethernet IOM Front and Rear Card Tabs Provisioning

Command and Description	Definition and Parameters
ED/ENT-EQPT	Edit or initialize the properties of an I/O Module.
TL1 equivalent to the red	IoModule (Card ID)

Command and Description	Definition and Parameters
highlighted items to be configured:	IoModuleType (Module Type) Rn (Redundancy) moduleType (Module Type from the following figure)

Click the Rear Card tab. The Ethernet Network Adapter rear card contains the actual physical interface and must be provisioned to match the physical fiber type for the software and interface to match.

Figure 9. Ethernet IOM Module Rear Card

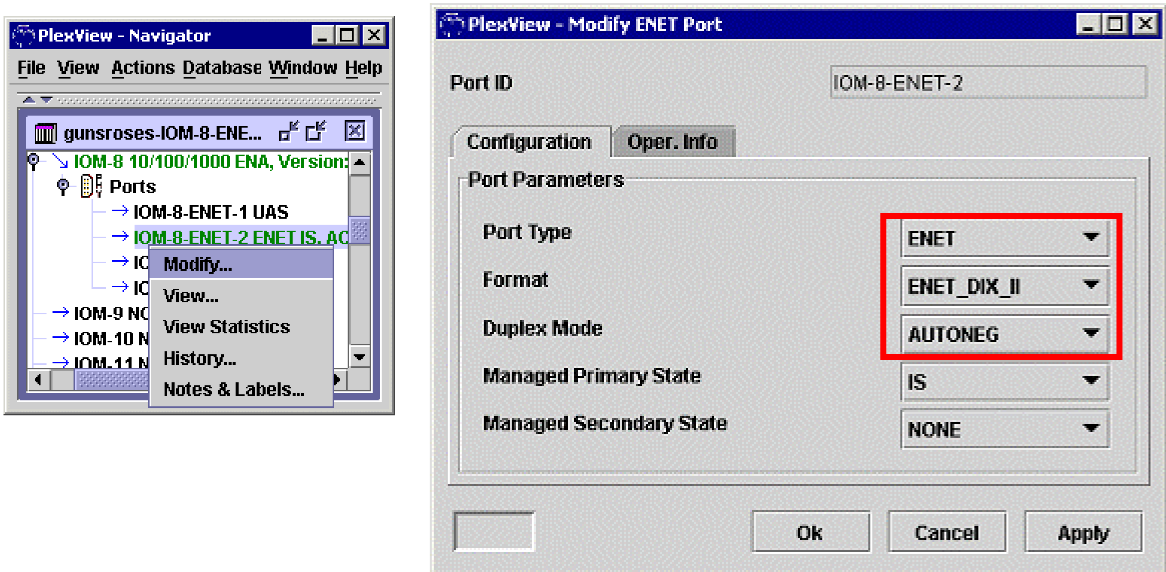


See the table in the previous section for the equivalent TL1 command.

Ethernet Ports Provisioning

All Ethernet ports must use the same settings.

Figure 10. GigE Ports



- Set the Port Type to ENET
- The Format must be ENET_DIX_II frame format or ENET_802_3. Format ENET_802_3 is also supported, but requires the AAL5 Encoding Type to be set to LLC_SNAP in VoIP system provisioning
- Set Duplex Mode to auto-negotiate (AUTONEG)
- Set the far-end Duplex Mode for Full Duplex

The equivalent TL1 commands are provided in the following table.

Table 3. TL1 Commands for Ethernet Ports Provisioning

Command and Description	**Definition and Parameters**
ED/ENT-ENET	Edit or establish the parameters associated with an Ethernet port on an I/O module.
TL1 equivalent to the red highlighted items to be configured:	enetID (Port Type) format (Format) mode (Duplex Mode)

Ethernet IP Address Provisioning:

Ethernet IP Address provisioning works as follows:

- The IP Address is the IP address of the slot 8 Ethernet port
- The Mate IP address is the IP address of slot 10 corresponding port when in standby

Note: If a failover occurs, the IP address of the slot 8 port becomes the IP address of the matching slot 10 port An unsolicited ARP response will update the ARP cache of the router attached to the slot 10 Ethernet port to complete the failover update.

- The Subnet Mask determines what part of the IP address identifies the network and what part identifies the host
- The Default Gateway defines the IP address of the default router for the Ethernet port. The router should support HSRP or VRRP

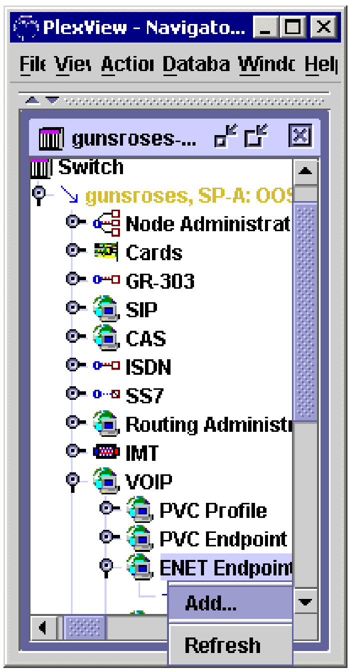
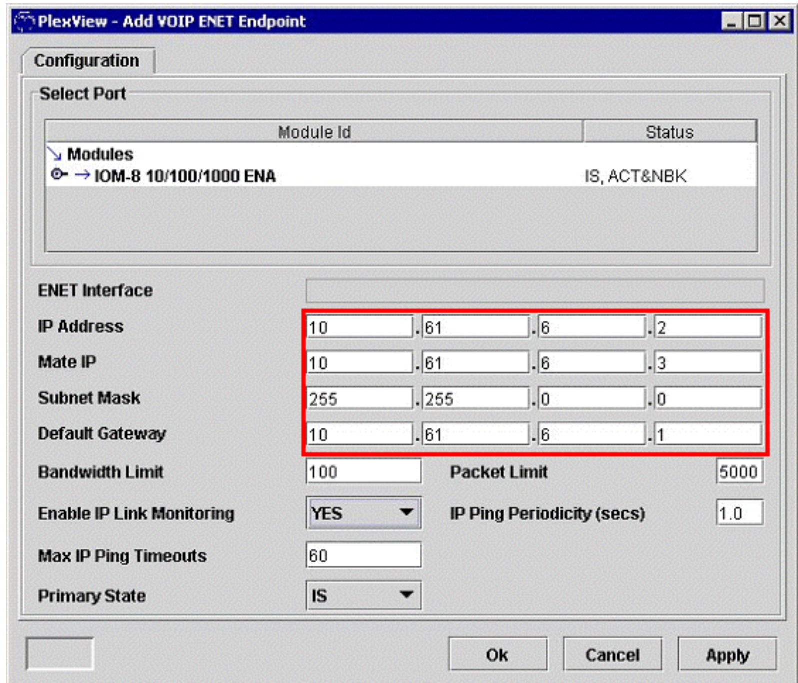


Figure 11. Ethernet IP Address Provisioning



The equivalent TL1 commands are provided in the following table.

Table 4. TL1 Commands for Ethernet IP Address Provisioning

Command and Description	Definition and Parameters
ED/ENT-ENET-ENDPTVOIP	Associate an Ethernet port with an Ethernet endpoint. ED/ENT-ENET must have already been issued for this command to work.
TL1 equivalent to the red highlighted items to be configured:	ipaddress (IP Address)
	mateipaddress (Mate IP)
	subnetmask (Subnet Mask)
	defaultgateway (Default Gateway)

Voice Server IOM Provisioning

The primary VSM (Voice Server Module) can reside in any open slot and must be set to PRI. Slot 17 is the dedicated VSM protection slot.

Configuration Tab

The Configuration tab is the default tab when you open the IOM screen. The highlighted parameters are as follows:

- The Card ID is the slot number of the VSM
- The Module Type is the card type
- Redundancy is PRI always

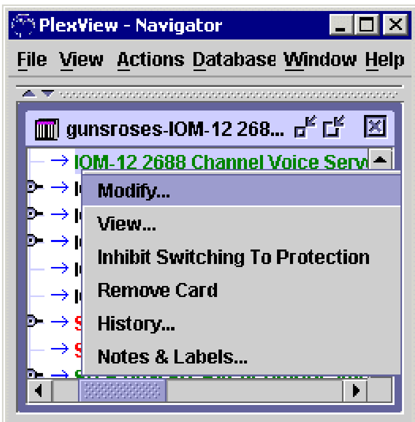
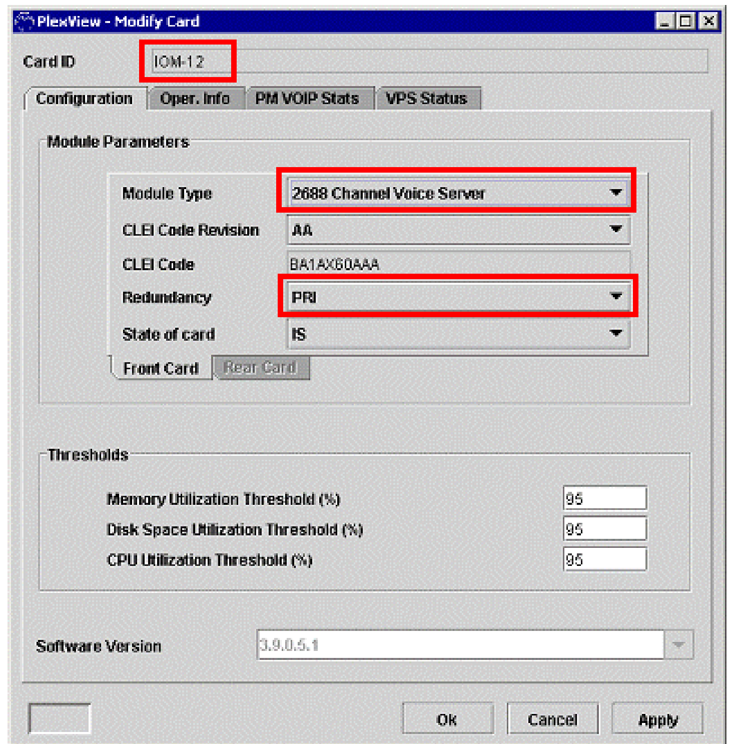


Figure 12. The Primary Voice Server Module



The equivalent TL1 commands are provided in the following table.

Table 5.TL1 Commands for Ethernet IP Address Provisioning

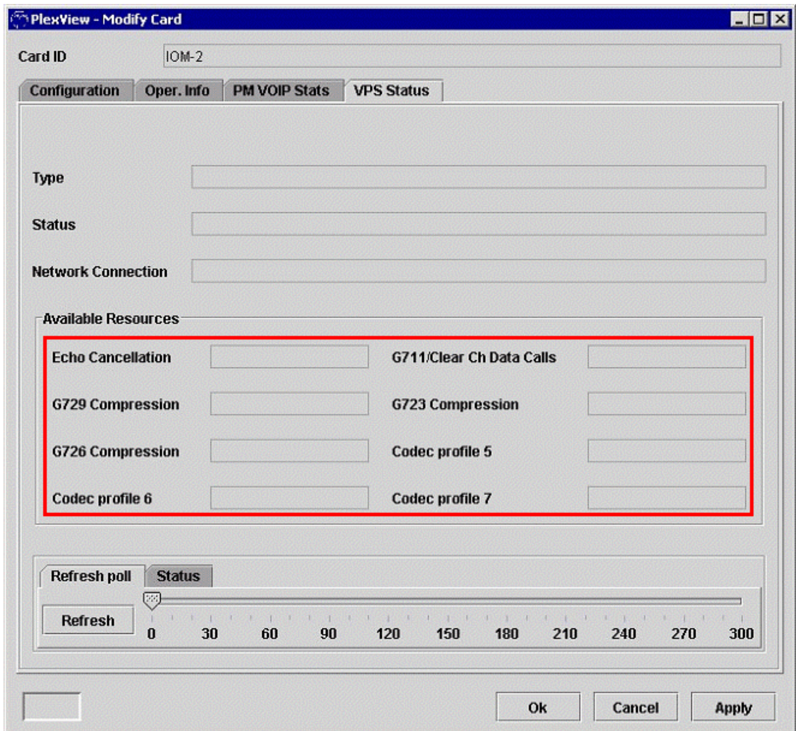
Command and Description	Definition and Parameters
ED/ENT/RTRV-EQPT	Edit or initialize the properties of an I/O Module.
TL1 equivalent to the red highlighted items to be configured:	IoModule (Card ID) IoModuleType (Module Type) Rn (Redundancy)

The VPS Status Tab

Click the VPS Status tab. The supported voice compression algorithms are:

- G.711 a-law/Clear Channel (µ-law to a-law conversion supported) (64k)
- G.729a (8k)
- G.726 (32k)(Cisco big endian and little endian supported)
- G.723.1 (6.4k)

Figure 13. Voice Compression Algorithms



The Compact Switch negotiates compression algorithms with connected devices in descending order of quality. This is set in the SDP profile, not under the VSM tab. SDP provisioning is covered later in this document.

Each VSM adds resources for IP calls and the resources vary depending on the CODEC.

CODEC	VSM2 Server Module
G.711	2688
G.723.1	672
G.729AB	816 (32 ms tail) 768 (> 32 ms tail)

Note: One G729 call on a VSM2 sets the call limits to 768 or 816 depending on the EC tail setting.

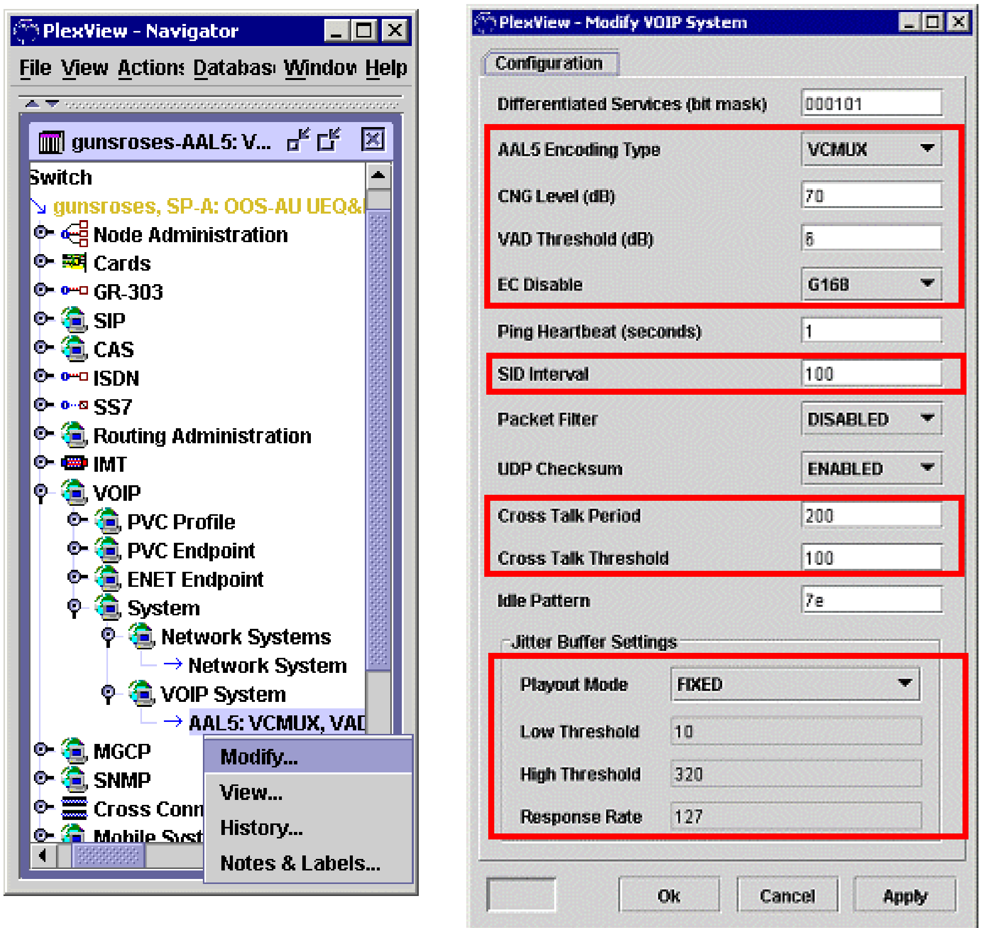
DS3 or DS1I/O cards (TDM IOMs)

Standard TDM IOM configuration is all that is required.

VoIP System Provisioning

The VoIP system comes with defaults that can be modified, as shown in the following figure.

Figure 14. VoIP System Settings



Encoding and activity levels:

- AAL5 encoding should be set to VCMUX. LLCSNAP is also supported, but all GigE ports must be provisioned as ENET_802_3 for it to work
- CNG Level (dB)(comfort noise generation) is the silence threshold before the Compact Switch generates comfort noise
- VAD Threshold (dB)(voice activity detect) is the lowest dB level of audio input considered for packetization (not silence)
- EC Disable discontinues echo cancellation. G168 is typical, G165 and G164 are also supported.
- SID Interval is the rate silence packets are transmitted at

Cross Talk levels:

IP cross talk can occur when a call terminates locally, but the remote end continues to send a packet stream. If the port is activated for a new call, two streams can be received on a Compact Switch port from different remote IP address/port combinations.

- Cross Talk Period: The number of packets received to be considered a valid VoIP stream
- Cross Talk Threshold: The number of packets from multiple sources received on a single port that will generate a cross talk event message

Jitter Buffer Settings:

Jitter buffer settings ensure that once an audio stream begins, it can be sustained even during variations in packet arrival rates. It also provides optimal performance when dealing with silence periods.

- Playout Mode identifies the jitter buffer's playout mode as fixed or adaptive
- Low Threshold is identified in milliseconds
- High Threshold is identified in milliseconds and must be higher than the low threshold
- Response Rate is the time constant. It determines the speed of adaptation with the following tradeoff: a longer time constant slows adaptation but results in a more stable jitter buffer nominal threshold variable. A shorter time constant causes faster adaptation but at the same time may result in frequent changes in the jitter buffer nominal threshold variable. For jitters in bursts, expect a better trade off when using a smaller time constant; for random jitter a longer time constant is more appropriate. The choice of optimal parameters is system dependent and should be selected based on the measure of jitter in the network. However, assuming random jitter and desire for small changes in delay over short instances, setting the value of jitBufLowThresh to "2", jitBufHighThresh to "16", and jitBufRspRt to "127" can be used as a start. The factory default is "127".

The equivalent TL1 commands are provided in the following table.

Table 6. TL1 Commands for Modifying the VOIP System

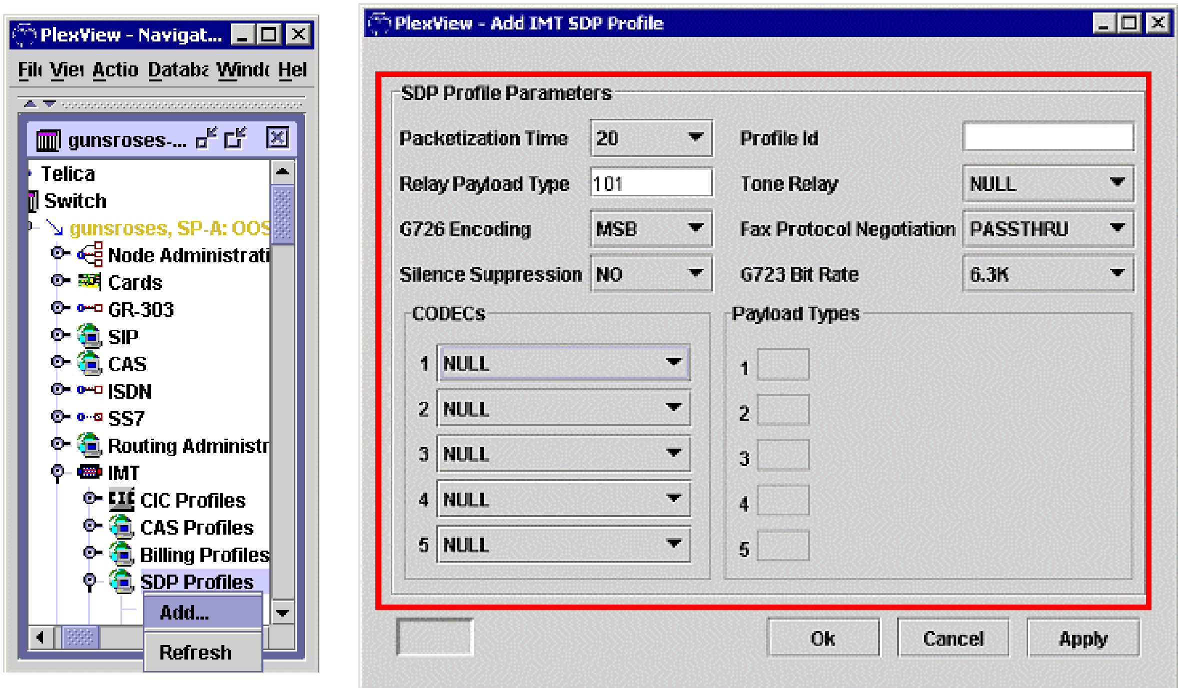
Command and Description	**Definition and Parameters**
ED/RTRV-VOIP-SYS	Edit general system-wide parameters for the DSP resources in the Compact Switch. Edits affect only new circuit setups after a VPS module has been removed and restored to service.

Command and Description	Definition and Parameters
TL1 equivalent to the red highlighted items to be configured:	Aal5Enc (AAL5 Encoding Type)
	cng (CNG Level (dB) (0, 30-77)
	vadThresh (VAD Threshold (dB)
	(Selection #) (Upper Threshold (dB)) (Lower Threshold (dB))
	1 -27 -39
	2 -30 -42
	3 -33 -45
	4 -36 -48
	5 -39 -51
	6 -42 -54
	7 -45 -57
	8 -48 -60
	9 -51 -63
	10 -54 –66
	ecDisable (EC Disable)
	sidTxInterval (SID Interval)
	xtalkPeriod (Cross Talk Period)
	xtalkThresh (Cross Talk Threshold)
	jitBufPlMod (Playout Mode)
	jitBufLowThresh (Low Threshold)
	jitBufHighThresh (High Threshold)
	jitBufRspRt (Response Rate)

Session Description Profile Provisioning

An SDP (Session Description Protocol) profile must be configured to apply to a SIP trunk. An SDP profile must have at least one codec specified for the RTP session that will be set up if the SIP call connects. Specified codecs must be contiguous and must start with codec1. The SDP Profile screen is shown in the following figure.

Figure 15. The SDP Profile Screen



- Packetization Time is the size in milliseconds of the packet containing the audio sample, or the time over which data accumulates before it is assembled into a packet. A value of NONE allows the media gateway to determine the packetization time.
- Relay Payload Type identifies the dynamic range of the RTP payload type for tone carrying packets. Each payload type is mapped to a single codec and cannot be co-assigned to another codec in the same SDP profile.
- G726 Encoding on Most Significant Bit (MSB) or Least Significant Bit (LSB)
- Silence Suppression is disabled (NO) or enabled (YES)
- Profile ID is the number of the Profile ID (1-255)
- Tone Relay determines whether to carry DTMF tones in RTP packets
- Fax Protocol Negotiation determines how to handle faxes
- G.723 Bit Rate sets G.723 for 6.3k (better fidelity) or 5.3k (bandwidth savings)
- CODECs: This section determines the hierarchical order of codec negotiation with the far-end with 1 being the highest level
- Payload Types determines the dynamic range of the RTP payload and corresponds to the codec. So Payload Type 1 would correspond to CODEC1.

The equivalent TL1 commands are provided in the following table.

Table 7. TL1 Commands for Adding an IMT SDP

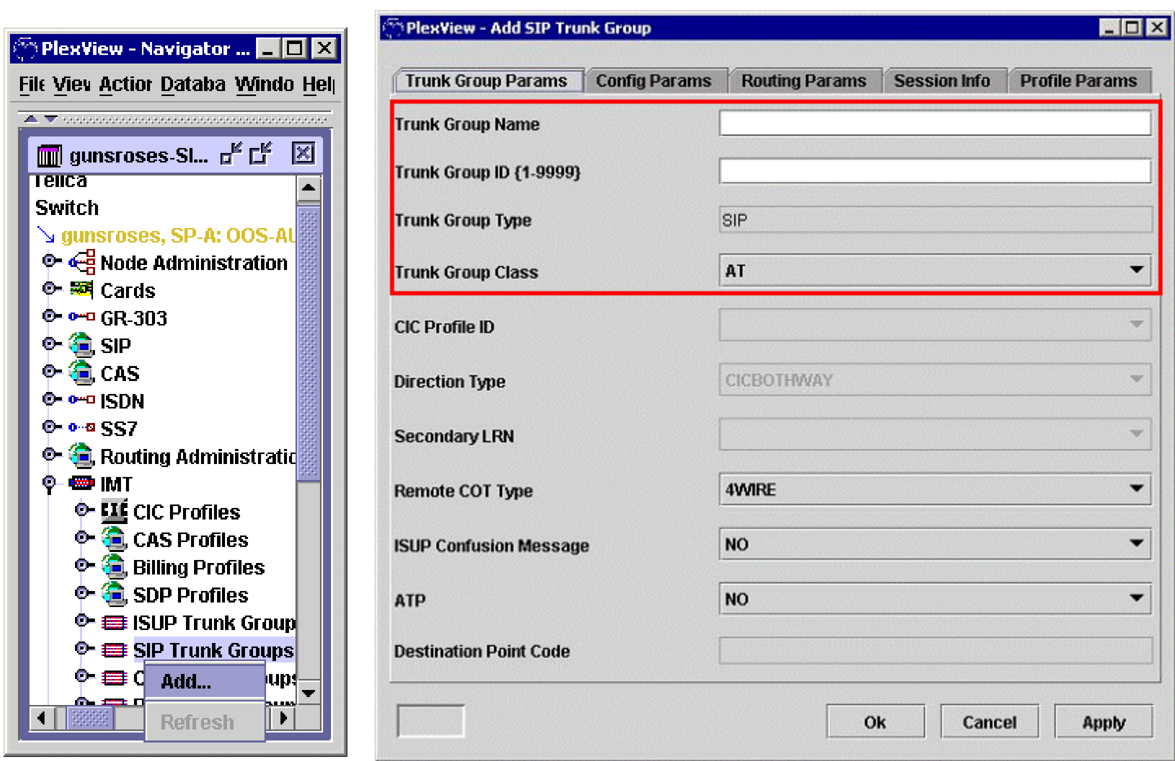
Command and Description	Definition and Parameters
ED/ENT-PRFL-SDP	Edits or establishes an SDP (Session Description Protocol) profile applicable to a SIP trunk. An SDP profile must have at least one codec specified and others must be contiguous.

Command and Description	Definition and Parameters
TL1 equivalent to the red highlighted items to be configured:	pTime (Packetization Time) relayPtype (Relay Payload Type) silSup (Silence Suppression) sdpPrflID (Profile ID) toneRelay (Tone Relay) codec1-codec5 ptype1-ptype5

SIP Trunk Group Provisioning

SIP trunk groups carry SIP calls to and from the Compact Switch. The calls are carried in RTP and are somewhat configured in the SDP profile, but the trunk groups must be fully configured for the bearer channels to work. Trunk group screens are similar for all supported trunk group types. Those menu items not relevant to SIP or SIP-T are grayed out. Also, the pull-down menu items only display selections that are relevant to SIP trunk groups. The first Add Trunk Group Params screen is shown in the following figure.

Figure 16. Add SIP Trunk Group



Initial Trunk Group Setup

This trunk group setup highlights the most important trunk group configurable parameters. Other parameters that are not highlighted must also be configured for SIP trunks.

- Assign the Trunk Group Name and ID
- The Trunk Group Type is preset before you get to this screen
- Trunk Group Class is the class of the remote switch the trunk group terminates to

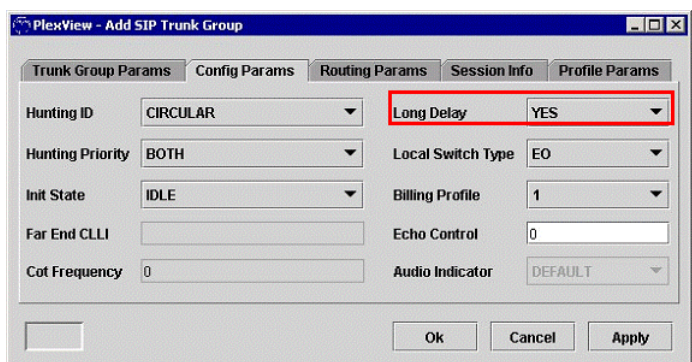
The equivalent TL1 commands are provided in the following table.

Table 8. TL1 Commands for the Initial Trunk Group Params

Command and Description	Definition and Parameters
ED/ENT-TRKGRP	Establish a trunk group.
TL1 equivalent to the red highlighted items to be configured:	TrkGrpName (Trunk Group Name)
	Tgn (Trunk Group ID) (1-9999)
	SigType (Trunk Group Type)
	TrkGrpClass (Trunk Group Class)
	LongDelay (Long Delay under Config Params tab)
	IpAddr (IP Address under Session Info tab)
	MaxIpCalls (Maximum IP Calls under Session Info tab)

Click the Config Params tab and the next screen appears, as shown in the following figure.

Figure 17. SIP Trunk Group Config Params

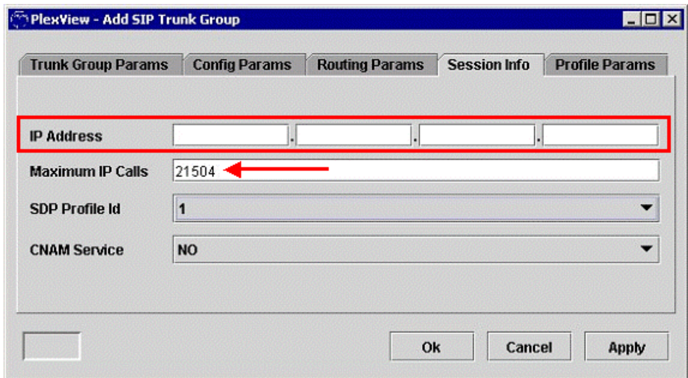


Long Delay enables echo cancellation.

Click the Session Info tab and more relevant configurable parameters appear, as shown in the following figure. They are only available when the Trunk Group Type is set to SIP.

- The IP address for this field is the IP address of the local feature server, application server, media gateway, or network server
- The Maximum IP Calls is the maximum number of calls supported on the trunk group

Figure 18. SIP Session Information

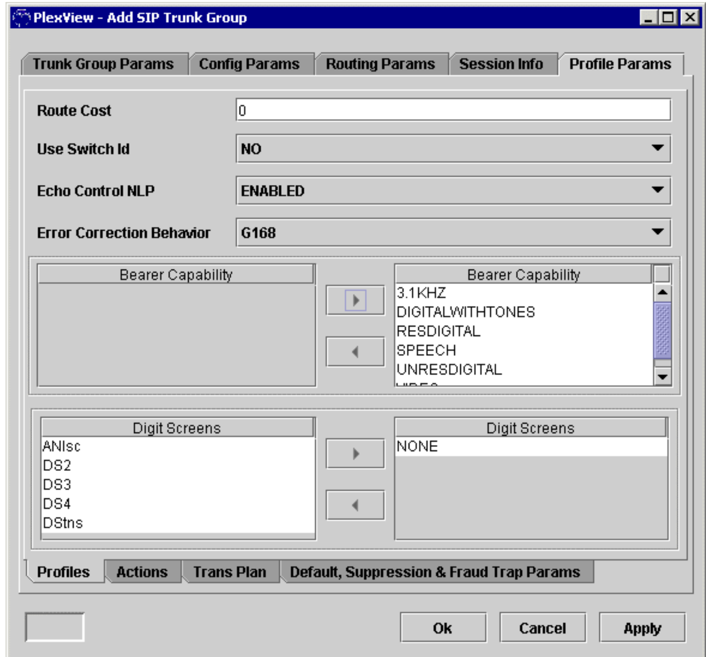


The profile parameters screen has call termination and treatment information that becomes relevant after the path of the trunk is chosen. The screen includes:

- The type of information to be carried in the bearer channel
- Digit screening
- The action to take after the trunk group is chosen
- Behavior profiles
- Translation plans
- Default profiles
- Parameter suppression profiles
- Fraud trap profiles

Many of the configurable items are under the tabs at the bottom of the screen, as shown in the following figure.

Figure 19. SIP Profile Parameters



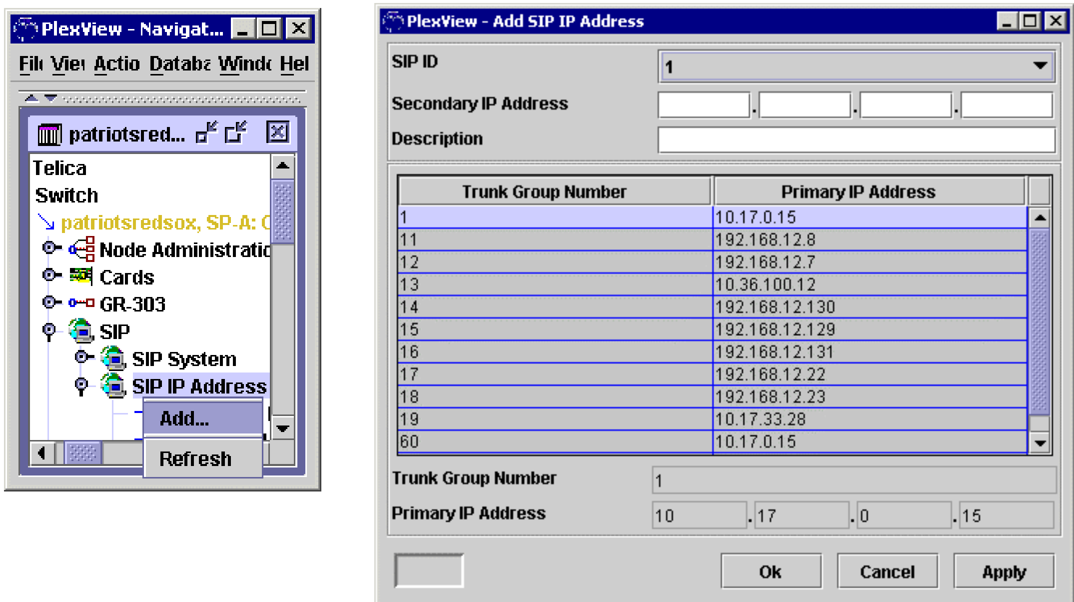
SIP IP Address Provisioning

The SIP IP Address establishes a primary and optionally a secondary IP address to access the session access node (feature server or equivalent). The SIP trunk group must be established prior to configuring the SIP IP Address. All items pertain to SIP, so no items are highlighted with a red outline.

- The SIP ID is the Session Access Node
- The Secondary IP Address optionally provides a secondary IP path to the feature server
- The Description identifies the interface
- The Trunk Group Number is the number assigned in Add SIP Trunk Group
- The Primary IP Address is the address to access the feature server.

The Add SIP IP Address screen is shown in the following figure.

Figure 20. SIP IP Address



The equivalent TL1 commands are provided in the following table.

Table 9. TL1 Commands for SIP IP Address

Command and Description	Definition and Parameters
ED/ENT-SIP-IPADDR	Edit or establish the primary and (optional) secondary IP address and description for a trunk group to a specific SIP session access node (feature server or equivalent).
TL1 equivalents to the red highlighted items to be configured:	tgnSipID (SIP ID & Trunk Group Number)
	secIPAddr (Secondary IP Address)
	description (Description)
	primIPAddr (Primary IP Address)

SIP System Provisioning

The SIP System node modifies the SIP stack from the system defaults for the number of supported connections and various timers. Details are beyond the scope of this document.

SIP CDR Generation with a Feature Server

When using the switch with a feature server to support enhanced IP features, two CDRs are created – one from the switch and one from the feature server. The same SIP Call ID appears on each CDR, identifying them for the same call. This facilitates billing correlation between the switch and feature server with regard to CDRs.

Related Documents

- *TL1 Commands Reference Guide*
- *EMS User's Guide*

Notes:

Voice Over Packet Software Package

This document contains the following sections:

Overview

The Lucent Compact Switch can be deployed to a variety of VoIP network topologies delivering either or both IP access and IP trunking services. For these applications, the Compact Switch can be connected to another Compact Switch or to an IP access device such as an IAD. Several protocols are available to support these applications, including

MGCP for controlling IADs, as well as BICC and SIP for supporting call control between multiple Compact Switch without the requirement of using the SS7 network. For simpler IP-to-IP calling scenarios, SIP may be used between an Compact Switch-based gateway and smaller gateway devices that do not support BICC/SS7.

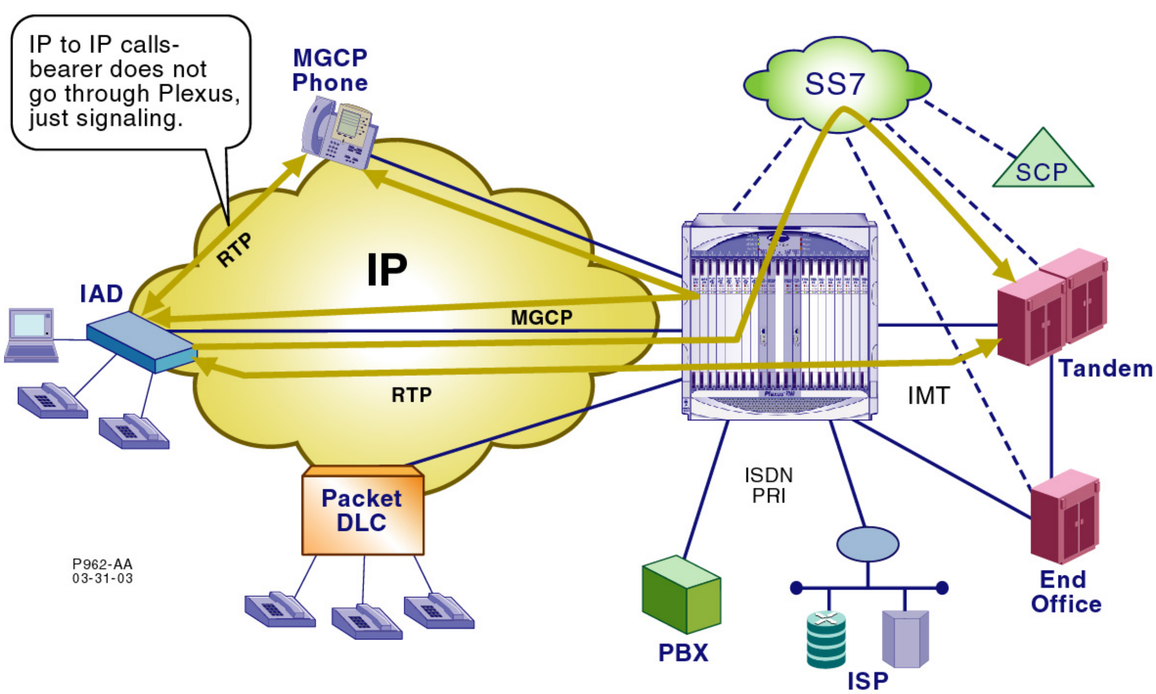
MGCP

The Compact Switch supports the Media Gateway Control Protocol (MGCP) for controlling VoIP access devices such as Integrated Access Devices (IADs) and MGCP IP Phones located at the customer premises, and Packet Digital Loop Carriers (DLCs) located at a remote terminal office or central office. The Compact Switch's direct support of the VoIP devices provides a cost-effective alternative to an external application feature server, for those service providers who only need to offer CLASS/Custom calling services to their IP subscribers. They can still deploy external application feature servers for IP Centrex and other enhanced IP services. In this capacity, IP-to-IP or IP-to-PSTN access is supported.

Subscribers can access the services and features of the Compact Switch with traditional analog phones connected to an IAD or Packet DLC or with new MGCP IP phones. The Compact Switch controls these devices using the MGCP protocol.

See Figure 1 or a sample depiction of an MGCP-supported scenario.

Figure 1. MGCP/Compact Switch Sample Architecture



The diagram in Figure 1 shows the Compact Switch providing IP access services to conventional “black phone” subscribers using IADs, or to subscribers with MGCP telephones “directly attached” to the Compact Switch. In this diagram, both signaling and bearer paths are carried by an IP network providing the converged voice/data local loop.

In all cases, signaling information is communicated between the subscriber and the Compact Switch as shown by the dotted lines in the diagram in Figure 1. However, in some cases, the bearer (speech) path may not involve the Compact Switch. For one IP subscriber calling another, the bearer path will be established directly between the two subscriber end points (IAD and MGCP phone, for example) using signaling information passed between the Compact Switch and each endpoint. Other call scenarios involving PSTN end points will pass through the Compact Switch which will be equipped with Voice Server Modules to provide voice-to-packet conversion.

Using MGCP, the same line-side services are available to IP access subscribers as currently supported by the Compact Switch for analog access customers (CAS and GR-303, for example). These include Class 5 features E 9-1-1, operator, LNP, authorization and account codes, and voice mail. (In the case of CALEA, there is currently a restriction. For more information, refer to section 1.3.3, MGCP-Supported Communication Assistance for Law Enforcement Act (CALEA).

IP access devices can also support other VoIP signaling protocols such as SIP or H.323. The Compact Switch does not directly support these devices. For SIP devices, an external application feature server is required for support. For H.323 devices, an external H.323/SIP converter is needed in conjunction with the external application feature server.

MGCP Features

MGCP on the Compact Switch enables service providers to offer the benefits of converged data/voice services, along with the Class 5 features with which subscribers are already familiar.

Note: There currently is no support for MGCP trunking, including the control of an MGCP trunking gateway and the support for MF trunks on the access gateway for PBX.

MGCP-Supported Class 5 Features

The Class 5 features supported by the Compact Switch when using MGCP include:

- Caller ID/Name
- Call Waiting
- Call Waiting with Caller ID
- Cancel Call Waiting

- Call Forwarding
  - Variable
  - No Answer
- Busy
- Remote
- Add-on Conference and Transfer
- Customer Originated Trace (COT)
- Anonymous Call Rejection
- Distinctive Ring
- Speed Calling 8 and 30
- Auto Recall
- Auto Callback
- 3-Way Conferencing
- SMDI Voice Mail
- Hotline Automatic Line – added in Release 3.11
- Warmline – added in Release 3.11

For a description of Class 5 features, refer to the CLASS 5 Features section of the *Product Overview Guide*.

Communication Assistance for Law Enforcement Act (CALEA) Support for MGCP

Calls, when analyzed for CALEA purposes, are scrutinized for two things: call data (called number, time stamps), which consists of the signaling information used to make the call and is often the only information required by the law enforcement agency (LEA), and call content, which consists of the actual voice content of the call and is less frequently authorized by the courts. The Compact Switch enables law enforcement agencies to collect both types of information from CALEA TDM calls; however, for subscribers supported by MGCP, only call data is currently available. Content of an MGCP call cannot be copied to a law enforcement agency. Call content support will be available in a future software release.

Voice Mail for MGCP Subscribers

Voice Mail is supported for MGCP subscribers. Service is provided using a conventional voice mail system connected to the Compact Switch using the SMDI protocol. Provisioning of Voice Mail is done through the PlexView EMS. Refer to the *PlexView EMS User Guide* for provisioning details.

MGCP CODECs Supported

The Compact Switch supports the following voice CODECs for MGCP controlled devices:

- G.711 µLaw (64kbps)
- G.711 aLaw (64kbps)
- G.726 (32kbps)
- G.729a (8kbps)

DTMF tone relay is supported via RFC2833 telephony signaling events. However, in a feature server environment, the Compact Switch needs to connect to a media server to provide announcements and digit collection. Since some low-cost media servers do not have DTMF tone detection functions, the Compact Switch must detect the DTMF tones and send them to the media server via RTP-named telephone events.

In addition, Fax Relay is supported via upspeeding to G.711 when the Fax modem tones are detected. For more information regarding Compact Switch-supported CODECs, refer to the CODEC Support section.

MGCP Interworking

Calls can be originated from an MGCP subscriber to and from all supported Compact Switch interfaces and protocols. This includes the following:

TDM Interfaces/Protocols

- MGCP to/from ISUP
- MGCP to/from PRI
- MGCP to/from CAS and MF
- MGCP to/from GR-303
- MGCP to/from MGCP

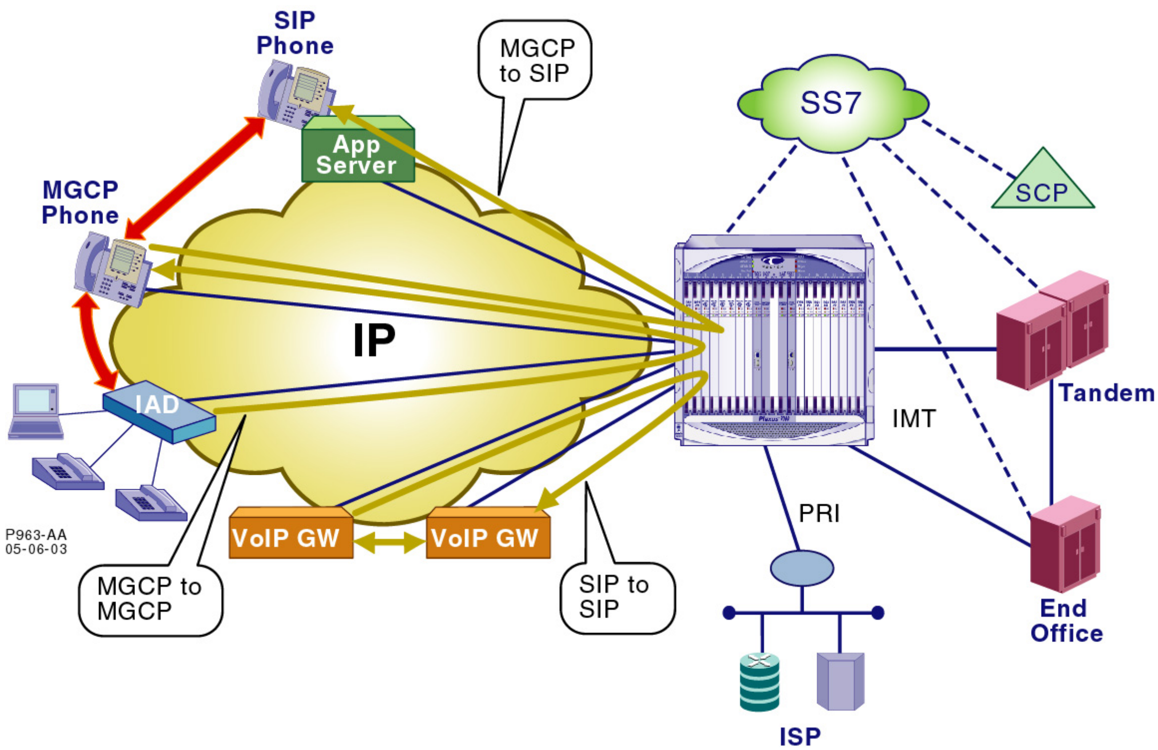
Packet Interfaces/Protocols

- MGCP to/from MGCP – on-net IP call (IP subscriber calling another IP subscriber)
- MGCP to/from SIP trunking(e.g., Feature Server, SIP-based VoIP Gateway) – off-net IP call (IP subscriber caling a remote attached subscriber), or Compact Switch IP subscriber calling a Feature Server subscriber
- MGCP to/from BICC (i.e., Compact Switch-to-Compact Switch Trunking) – off-net IP call (IP subscriber calling a subscriber -- either TDM or IP – that is attached to a remote Compact Switch). Refer to the BICC Over IP Support section for more details.

Figure 2 shows the MGCP and SIP interworking.

Note: That the bearer of IP-to-IP calls does not go through the Compact Switch, but rather directly between the two end-points.

Figure 2. Compact Switch MGCP IP Interworking



Configuring for MGCP Using the EMS

Using the PlexView EMS to configure the Compact Switch for MGCP involves the following steps:

1. Adding an MGCP Gateway Link for each MGCP device (e.g., IAD, IP Phone, Packet DLC).
2. Adding MGCP Line Groups
3. Adding MGCP Lines to the MGCP Line Groups and Gateways

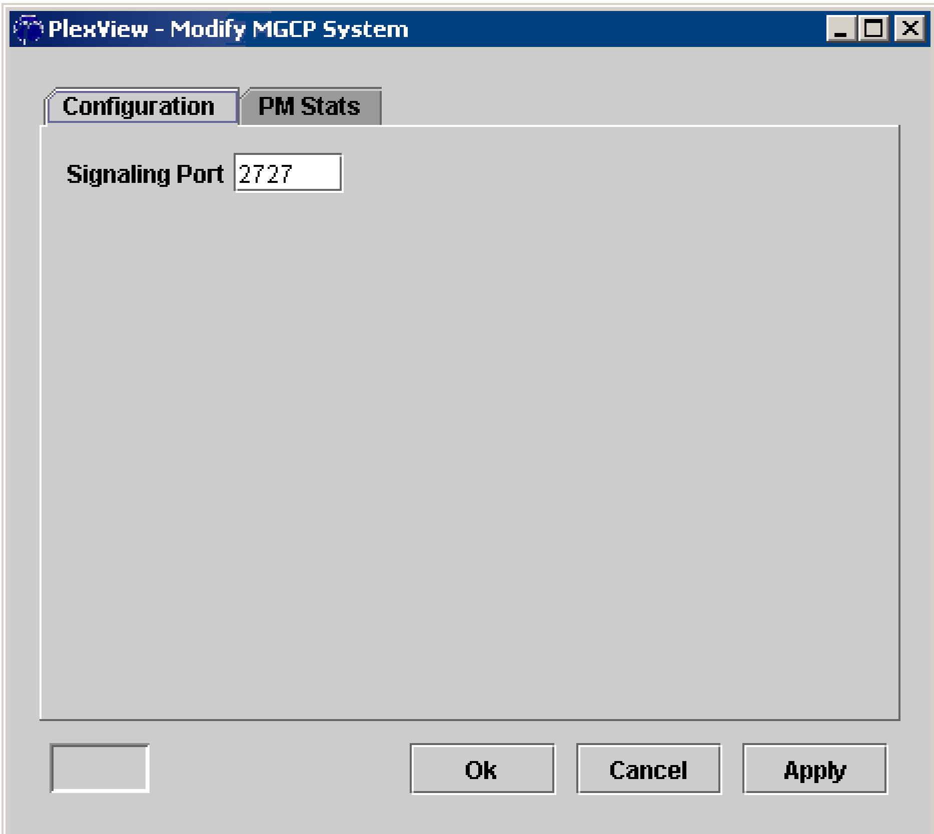
Note: A brief description of the EMS screens used to provision the Compact Switch for MGCP support are provided in this section. For a more detailed explanation, refer to the *PlexView EMS User Guide*.

The steps above should be performed in the order listed; however, you may want to change the default signaling port before performing any provisioning.

Note: There is no ordering requirement on changing the MGCP port.

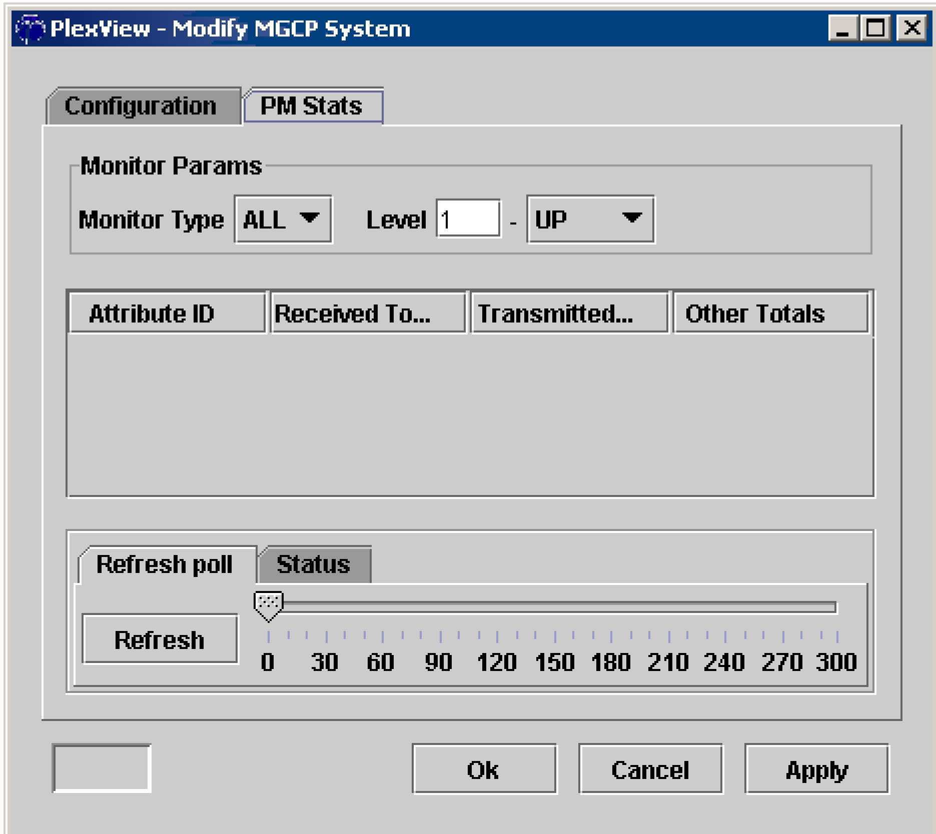
You can view or modify the MGCP signaling port using the Modify or View MGCP System screen. The default signaling port is 2727, but you can use the Modify MGCP System screen, as shown in Figure 3, to change the default. If you change the MGCP signaling port, all gateways attempting to connect to the Compact Switch must also be reprovisioned to send MGCP messages to that port.

Figure 3. Modify MGCP System – Configuration Tab



The PM Stats tab displays the performance statistics for an MGCP port. The number of attributes can be extensive. See Figure 4 for a sample of the PM Stats tab on the Modify MGCP System screen. For a list of potential attributes, refer to the *PlexView EMS User Guide*.

Figure 4. Modify MGCP System – PM Stats Tab

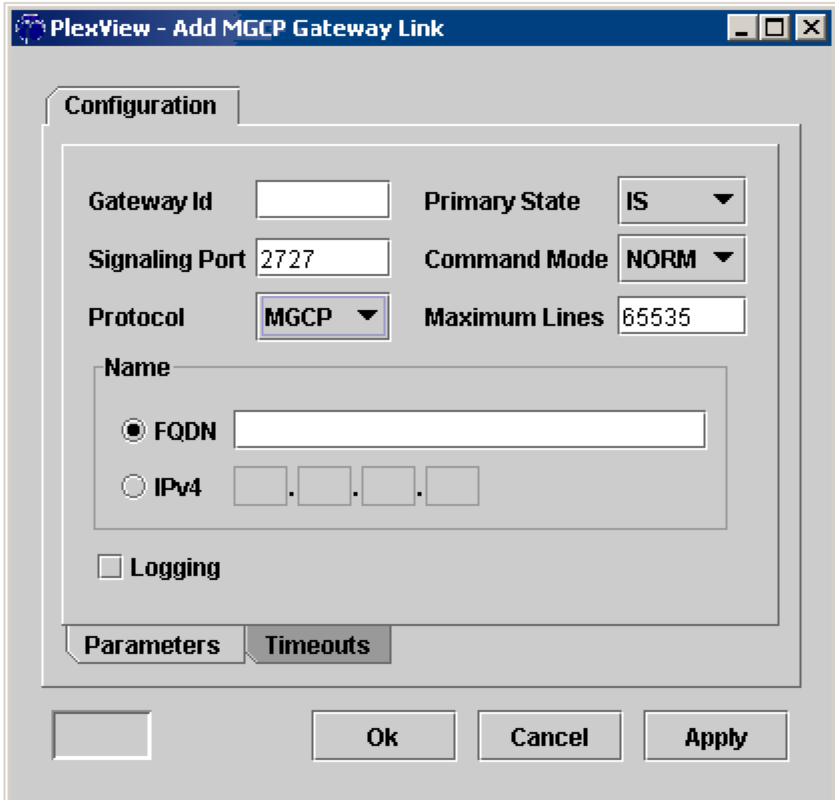


Right-clicking on a signaling port (listed on the Navigator under MGCP → System) enables you to select Init Register, which clears all attributes from the PM Stats tab on the Modify MGCP System screen.

Adding an MGCP Gateway Link

When using the PlexView EMS to provision for MGCP, you must first provision MGCP gateways. To do so, use the Add MGCP Gateway Link screen, as shown in Figure 5. You must always provision at least one gateway link when provisioning for MGCP. This is done on the Parameters tab of the Configuration tab on the Add MGCP Gateway Link screen. On this tab, you specify the gateway IP address, which is comprised of number, or the domain name of the gateway, such as lucent.com.

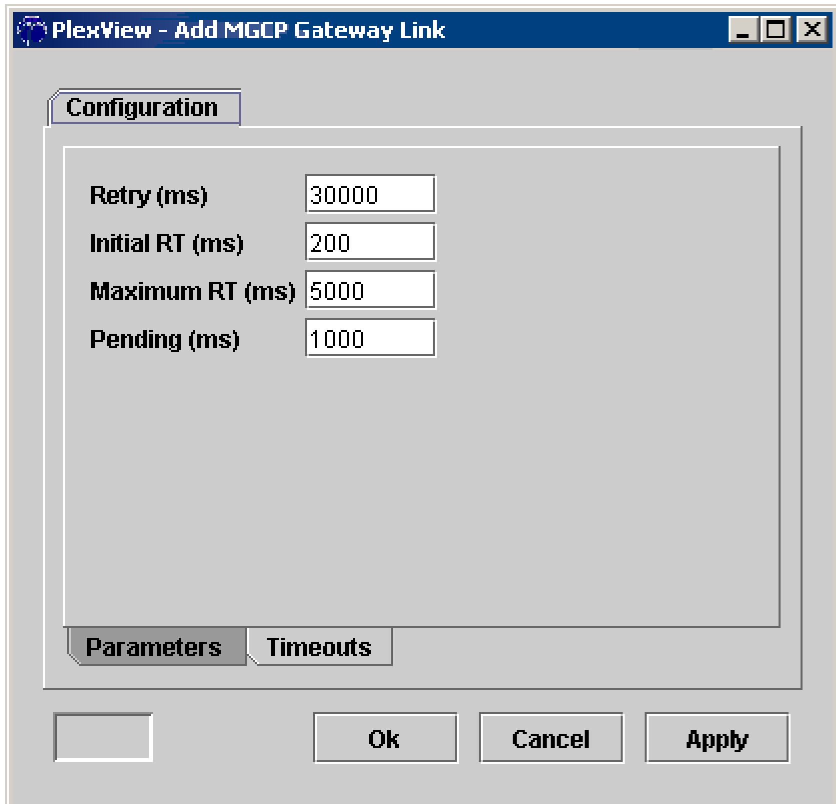
Figure 5. Add MGCP Gateway Link, Configuration – Parameters Tab



On the Parameters tab displayed in Figure 5, you can specify the protocol to be assigned to the gateway. The options are MGCP and NCS. NCS, or Network Call Signaling, is a protocol used specifically for PacketCable environments.

The Timeouts tab, as shown in Figure 6, is also available from the Configuration tab on the Add MGCP Gateway Link screen.

Figure 6. Add MGCP Gateway Link, Configuration – Timeouts Tab

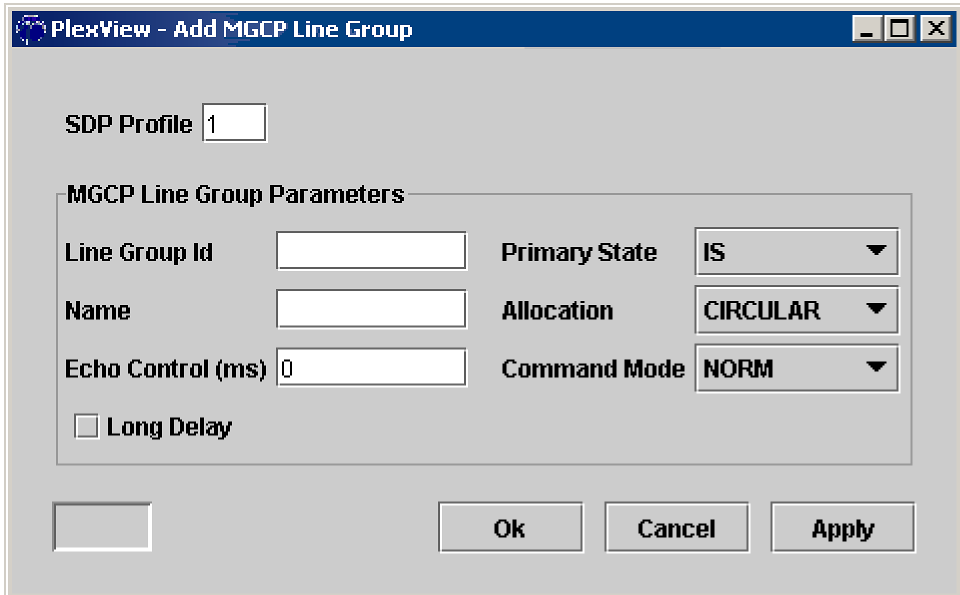


Adding a Line Group

Once you add a gateway link, you then add a line group by using the Add MGCP Line Group screen, as shown in Figure 7. On this screen you must select an SDP profile, which can be any number from 1 to 255.

Note: The SDP number is configured using the Add IMT SDP Profile screen. This screen is used not only to configure SDPs for MGCP, but also for IMT trunks.

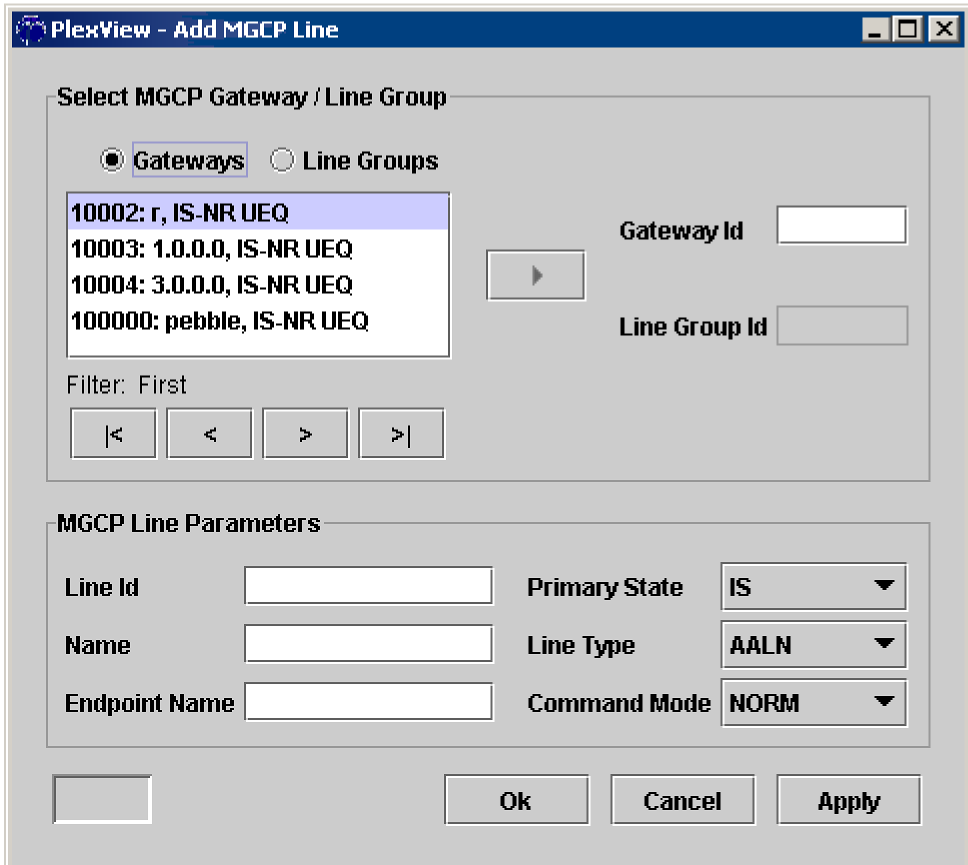
Figure 7. Add MGCP Line Group



Adding MGCP Lines to the MGCP Gateway and Line Group

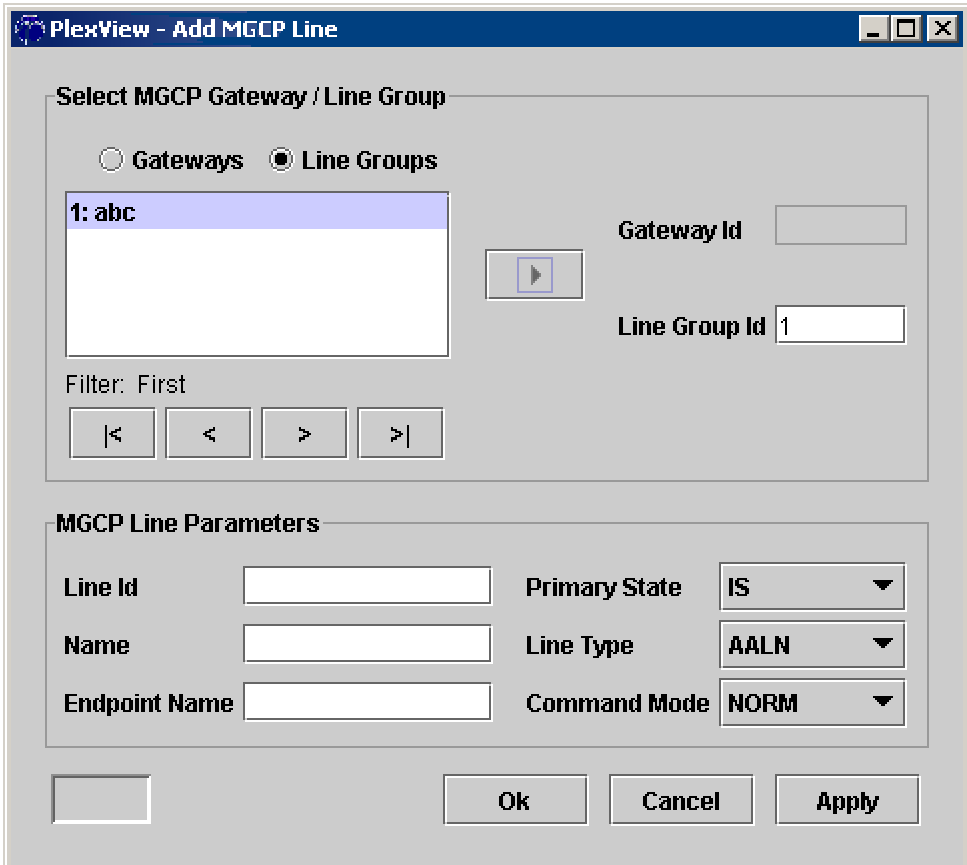
The last step in using the EMS to provision for MGCP is to provision a line for the gateway and line group. This is done on the Add MGCP Line screens shown in Figure 8 and Figure 9.

Figure 8. Add MGCP Line to Gateway



To assign a line to a gateway and line group, you first select Gateways and then click the arrow to add the Gateway ID. You then select Line Groups and choose a line group from the list, clicking the arrow to add the line to the Line Group ID. When this is finished, you can assign the Line ID and other line information. Your entry in the Endpoint Name field must match exactly the endpoint name as configured in the gateway and which will appear in all MGCP protocol messages for that endpoint.

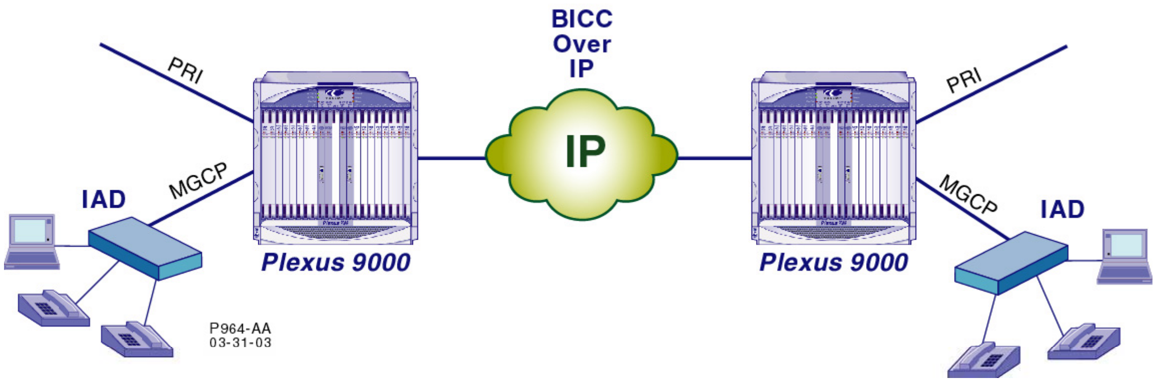
Figure 9. Add MGCP Line to Line Group



BICC Over IP Support

The Compact Switch supports Bearer Independent Call Control (BICC) signaling over SS7 as well as BICC signaling over IP for multi-chassis Compact Switch configurations. The BICC protocol is similar to ISUP, however, calls are assigned a call reference instead of a Circuit Identification Code (CIC). The call reference points to resources used at the Bearer Plane to provide the data path, which may include ATM VCCs or IP paths as well as TDM trunks. A simplified example of a multi-chassis configuration supporting BICC over IP is shown in Figure 10.

Figure 10. BICC Over IP for Multi-chassis Signaling



The ability to transport SS7 traffic from one Compact Switch Point Code to another without using TDM-MTP2 is supported. This eliminates the need to set up a physical mesh of T1 connections between the Compact Switch. M3UA over SCTP over IP supplies this feature (i.e., equivalent to SS7 F-links over IP). M3UA and additional CODEC negotiation are supported, thus allowing VoIP trunking between two or more Compact Switch.

M3UA

M3UA is used as the transport for BICC over IP. M3UA delivers reliable transport of ISUP and other SS7 protocols over IP. Therefore, M3UA must be configured on each Compact Switch so that one Compact Switch can signal another Compact Switch. Using M3UA enables the following for BICC support:

- Configurable upper interface to connect to any of the following potential users: ISUP, TCAP, or BICC
- Routing on a Point Code
- 'Point Code Pause' to the applications, in the same fashion that MTP3 does, so that an application server (an M3UA instance on a remote MGC) results in blocking calls/transactions to that Point Code. These pauses are due to either SCTP or inter-chassis heartbeat loss of connectivity.
- Alarms for Point Code Pause. These events will require changes in the FAM because the source entity will be M3UA instead of MTP3 PC pause.
- SCTP communication fault (remote node) alarms. The AID for the alarm is PC-SPA/B. The preference is to only generate this alarm for the protection SCTP association.
- VoIP over BICC. BICC also supports MGCP-to-MGCP calls between two Compact Switches, but without the bearer path traversing the Compact Switch. BICC CODEC Negotiation

The use of the BICC protocol in release 3.8 and higher supports negotiation of these CODECs:

- G711-aLaw (PCMA on the SDP Profile screen in the EMS)
- G711-uLaw (PCMU on the SDP Profile screen in the EMS)
- G726
- G729a
- G729b (with silence suppression)

For detailed information about CODECs, refer to the CODEC Support section.

Calls can be originated from an MGCP or SIP subscriber to and from the Compact Switch-supported BICC protocol, including the following Packet Interfaces/Protocols:

- MGCP to/from BICC – off-net IP call (IP subscriber calling a subscriber (either TDM or IP) that is attached to a remote Compact Switch).
- SIP (Trunk) to/from BICC – feature server subscriber calling a subscriber (IP or TDM) attached to a remote Compact Switch.

For detailed information about CODECs, refer to the CODEC Support section.

Configuring for BICC Using the EMS

To place BICC over IP calls on the Compact Switch, you must configure specific information on the Compact Switch using the PlexView Element Management System (EMS). For instance, the first step you should take on each chassis is to configure BICC to use IP instead of MTP3. In doing so, you must establish a unique OWN Point Code per chassis. You use the Trunk Group Params tab of the Add BICC Trunk Group screen on the EMS to define a unique OWN Point Code for a Compact Switch. Figure 11 provides a sample of the screen.

For each Compact Switch, you also need to enter the IP addresses and Point Code of the other chassis. You use the Session Info tab on the Add BICC Trunk Group screen to assign IP addresses to a trunk group. See Figure 12 for a sample.

Figure 11. Add BICC Trunk Group, Trunk Group Params Tab

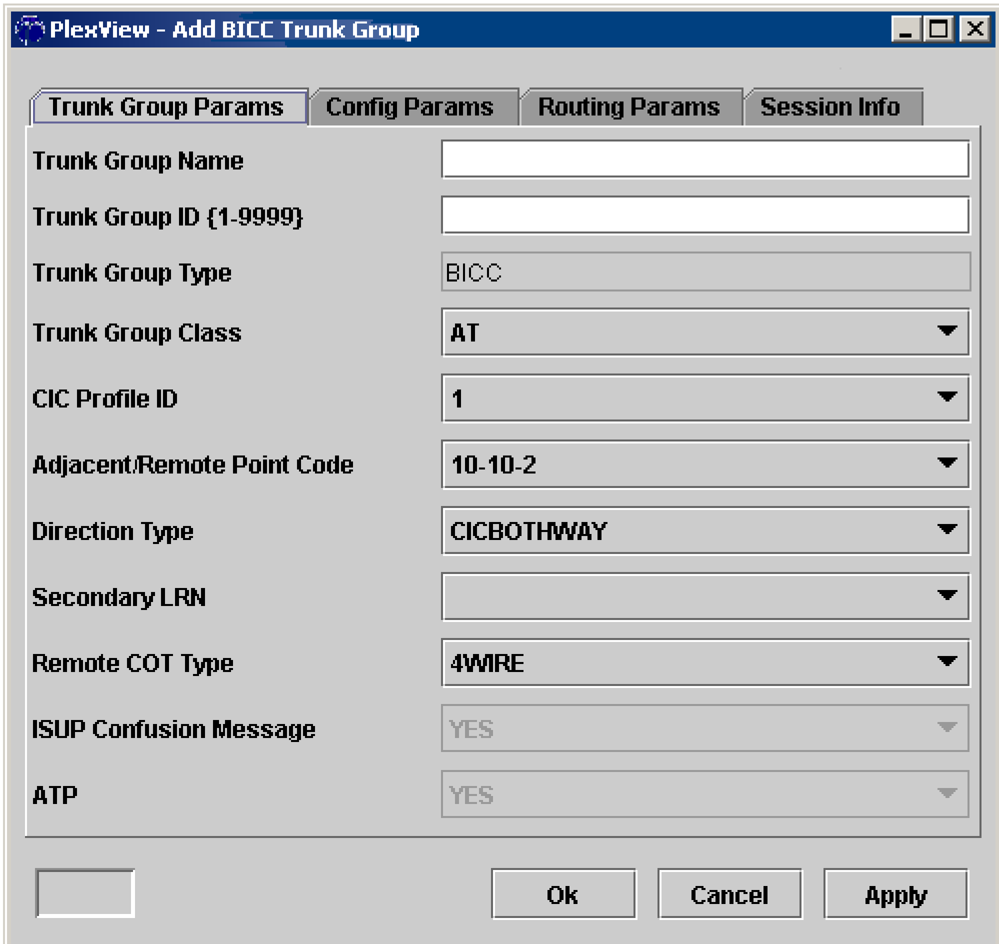
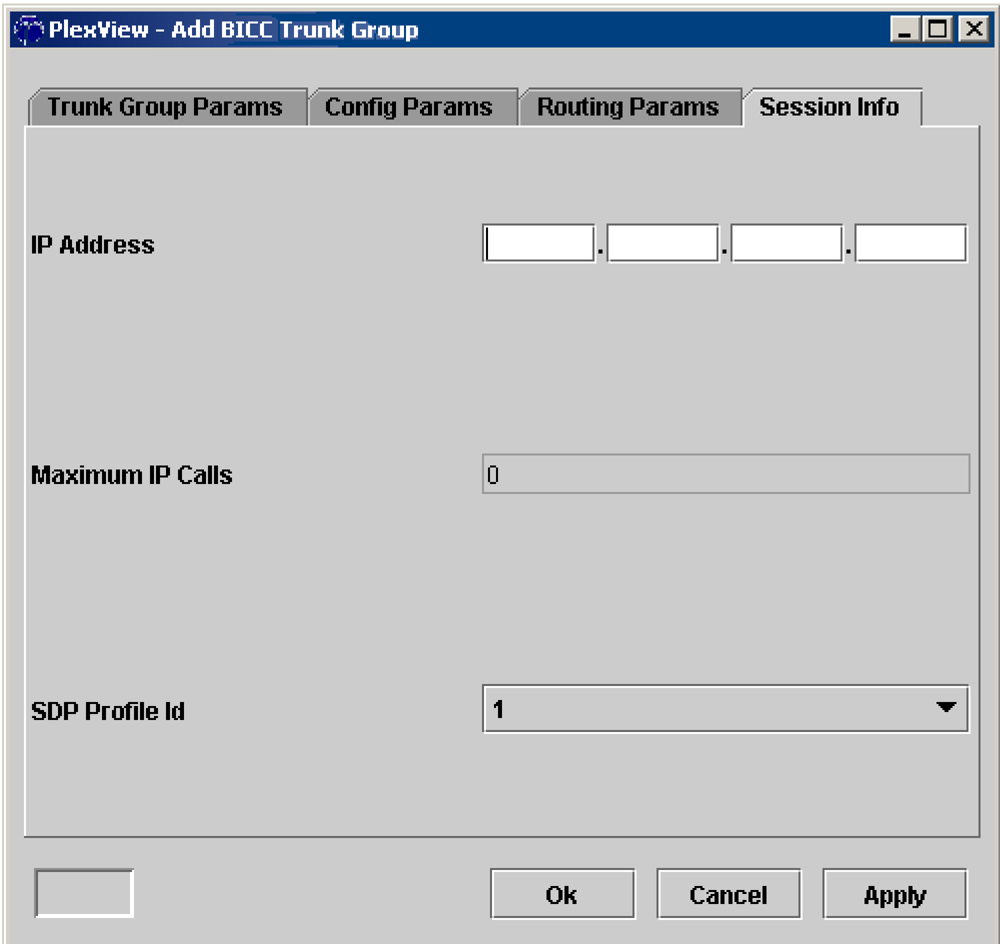


Figure 12. Add BICC Trunk Group, Session Info Tab



Once this configuration is performed, communication is established and enables the M3UA to "unpause" the Point Code. If each Compact Switch is not configured to support communication, then alarms will be generated until the Compact Switch on each end of a communication association (1 to 2 and 2 to 1) is configured. You can then finish BICC trunk provisioning.

Note: Limiting the number of call references can be used to limit the number of calls between chassis.

After provisioning BICC trunks, you can then configure the VoIP equipment (VPS modules and GigE modules, for example) to support the trunk groups. Use the screens listed under the VOIP object on the EMS Navigator to provision the VoIP IOMs. Once this is done you can provision the TDM interfaces or access interfaces, and the routing to trunk group relationships. Calls can then be placed after all of this provisioning occurs.

CODEC Support

The Compact Switch supports these compression/decompression (CODEC) technologies:

- G.711 µLaw (mu-Law)
- G.711 aLaw
- G.723.1
- G.726
- G.729a/b

Additionally, the Compact Switch supports these CODEC-related technologies:

- Clear-Channel Support Over Packet (X-CCD, CISCO CLEAR CHANNEL, and CLEARMODE): Clear-Channel data support lets the Compact Switch negotiate how to send restricted/unrestricted digital data over an RTP stream. Digital data in this case is potentially HDLC data received from ISDN devices.
- DTMF Relay (RFC2833): RFC-2833 describes how to send DTMF tones "out-of-band" over RTP. Support for RFC-2833 is required when the peer end-point does not have DTMF tone detectors. It requires the Compact Switch to transmit the presence of DTMF digits in the audio stream out-of-band.

These CODECs are available on each installed VSM (Voice Server Module). The Compact Switch performs CODEC negotiation during call establishment between VoIP end-points, per RFC-3264. To establish a VoIP call, the peer end-points must have at least one common CODEC assigned to the call establishment end-point.

In the Compact Switch, CODECS are specified in preference (priority) order in Session Descriptor Protocol (SDP) profiles referenced by SIP trunks. When the Compact Switch originates a call on a SIP trunk, the first CODEC listed in the SDP profile for the trunk is the preferred CODEC that the Compact Switch wishes to use for the call. If the remote peer cannot support the preferred CODEC, due to resource limitations, but can support another CODEC in the preference list, then the remote peer responds to the Compact Switch with the next highest CODEC in the preference list that it can support and the call is established. The preference (priority) list has an effect only when the Compact Switch originates the call.

The Compact Switch supports CODEC negotiation in these situations:

- On-net (IP to IP calls), when the Compact Switch is not involved in the bearer path. The Compact Switch does not limit the CODEC negotiation, meaning that the CODEC will be negotiated between the two endpoints. In this case, the CODECs provisioned in the Compact Switch SDP profile are immaterial to the negotiation. For instance, a CODEC not supported at all by the Compact Switch could be selected by two endpoints.
- Off-net (IP to TDM or TDM to IP calls), when the Compact Switch terminates the IP flow. The Compact Switch supports CODEC priority lists on a per-trunk or per-line group basis.

Note: This priority only has effect when the Compact Switch originates the call.

The following parameters are passed as part of CODEC negotiation; however, only the CODEC is negotiated and could result in a call rejection due to unsupported CODECs. The Compact Switch complies with the requested silence suppression and packetization time.

- CODEC: G.711-aLaw, G.711-µLaw, G.723.1, G.726, G.729a and G.729b (which is G.729a with Silence Suppression)
- Packetization Time (msec): 10, 20, 30, 40 (VoIP); 60 for G.723.1 only
- Silence Suppression: on, off

Note: These silence suppression parameters are not supported: silence timer, silence preference, silence insertion descriptors, and silence fixed noise level.

- DTMF Relay (RFC2833) enable/disable: yes, no

Note*:* SDP parameters clock rate, bandwidth, and echo cancellation enable are not negotiated.

Clear-Channel Support Over Packet

The Compact Switch supports Clear-Channel for transporting data (64kbps unrestricted) traffic over IP (RTP). This consists of 8-bit samples with a sampling rate of 8000 Hz with no echo cancellation or silence suppression.

The typical application is for IP trunking (SIP) of ISDN or switched 56 data traffic. This consists of 8-bit samples with a sampling rate of 8000 Hz with no echo cancellation or silence suppression.

RFC2833

The Compact Switch supports tone relay as defined by the IETF RFC2833 standard (Events 0-15). DTMF tones can be transported directly over the voice stream within RTP or they can be extracted out to the voice stream and encoded as RTP-named telephone events. The applications for RFC2833 are as follows:

- In a feature server environment, the Compact Switch needs to connect to a media server to provide announcements and digit collection. Some low cost media servers (e.g., software-based) do not have DTMF tone detection functions. This environment requires the Compact Switch to detect the DTMF tones and send it to the media server via RTP-named telephone events.
- In general, compressed low bit rate CODECs (G.723, G.726, G.729a) cannot transport DTMF tones reliably within the voice traffic. RFC2833 can be used to relay the DTMF tones. The alternative of up-speeding to G.711 for passing the DTMF tones is not practical due to the overhead of switching back and forth.
- In an environment where the IP network is of low quality with large jitter, long delay or high packet loss, the G.711 or G.726 CODEC may not be able to reliably transport DTMF tones. Therefore, RFC2833 may be needed to relay the DTMF tones.

Only RFC events 0-15 (DTMF tones 0-9, *, #, A-D) are supported with RFC2833. Other tones defined in RFC2833 (e.g., FAX, Modem, MF) are not supported.

How CODEC Negotiation Takes Place on the Compact Switch

In general, CODEC negotiation involves both the Compact Switch and the far end switch suggesting the CODEC to use, then the Compact Switch attempting to achieve a match. For incoming IP calls, the Compact Switch accepts the first CODEC on the received SDP list that matches a configured CODEC in the locally configured SDP profile for that incoming trunk group. The Compact Switch rejects the call if no CODEC matches. The Compact Switch accepts silence suppression and DTMF tone relay options when they also are configured in the local SDP profile of the incoming trunk group. The Compact Switch enables DTMF tone relay when the far end requests it.

The following three tables demonstrate CODEC options between the Compact Switch and a remote endpoint. In the tables, “Signaled” identifies what either switch puts into its signaling, whereas “TX” and “RX” identify what is actually sent and what is expected to be received. Violet fill signifies the Compact Switch; gray fill signifies the endpoint.

Table A shows the various combinations of CODECs and packet times possible when the Compact Switch originates a call. In this scenario, the Compact Switch is the “ingress” that processes a local descriptor generated from the SDP profile – there is no remote descriptor to work against.

Table A. CODEC Combinations when Originating a Call

<table>
<tr><th colspan="4">Compact Switch</th><th colspan="4">Endpoint</th><th colspan="4">Result</th></tr>
<tr><th colspan="2">Profile</th><th colspan="2">Signaled</th><th colspan="2">Profile</th><th colspan="2">Signaled</th><th>TX</th><th>RX</th><th>TX</th><th>RX</th></tr>
<tr><td rowspan="3">10ms</td><td>G.711</td><td rowspan="3">ptime=10</td><td>G.711</td><td rowspan="3">10ms</td><td>G.711</td><td rowspan="3">ptime=10</td><td rowspan="3">G.711</td><td rowspan="3">G.711</td><td rowspan="3">G.711</td><td rowspan="3">G.711</td><td rowspan="3">G.711</td></tr>
<tr><td>G.729</td><td>G.729</td><td>G.729</td></tr>
<tr><td>G.723</td><td></td><td>G.723</td></tr>
</table>

Compact Switch →				Endpoint				Result			
Profile		Signaled		Profile		Signaled		TX	RX	TX	RX
10ms	G.711 G.729 G.723	ptime=10	G.711 G.729	20ms	G.711 G.729 G.723	ptime=10	G.711	G.711	G.711	G.711	G.711
10ms	G.711 G.729 G.723	ptime=10	G.711 G.729	30ms	G.711 G.729 G.723	ptime=10	G.711	G.711	G.711	G.711	G.711
10ms	G.711 G.729 G.723	ptime=10	G.711 G.729	10ms	G.729 G.711	ptime=10	G.711	G.711	G.711	G.711	G.711
10ms	G.711 G.729 G.723	ptime=10	G.711 G.729	30ms	G.723 G.711	ptime=10	G.711	G.711	G.711	G.711	G.711
10ms	G.711 G.729 G.723	ptime=10	G.711 G.729	30ms	G.723 G.729 G.711	ptime=10	G.711	G.711	G.711	G.711	G.711
20ms	G.729 G.711	ptime=20	G.729 G.711	10ms	G.711 G.729 G.723	ptime=20	G.729	G.729	G.729	G.729	G.729
20ms	G.729 G.711	ptime=20	G.729 G.711	20ms	G.711 G.729 G.723	ptime=20	G.729	G.729	G.729	G.729	G.729
20ms	G.729 G.711	ptime=20	G.729 G.711	30ms	G.711 G.729 G.723	ptime=20	G.729	G.729	G.729	G.729	G.729
20ms	G.729 G.711	ptime=20	G.729 G.711	10ms	G.729 G.711	ptime=20	G.729	G.729	G.729	G.729	G.729
20ms	G.729 G.711	ptime=20	G.729 G.711	30ms	G.723 G.711	ptime=20	G.711	G.711	G.711	G.711	G.711
20ms	G.729 G.711	ptime=20	G.729 G.711	30ms	G.723 G.729 G.711	ptime=20	G.729	G.729	G.729	G.729	G.729

Table B shows the various combinations of CODECs and packet times possible when the Compact Switch receives a call. In this scenario, the Compact Switch is the "egress" that processes a local descriptor generated from the SDP profile against the remote descriptor received from the endpoint.

Table B. CODEC Combinations when Receiving a Call

Endpoint →				Compact Switch				Result			
Profile		Signaled		Profile		Signaled		TX	RX	TX	RX
10ms	G.711 G.729 G.723	ptime=10	G.711 G.729 G.723	10ms	G.711 G.729 G.723	ptime=10	G.711	G.711 (10ms)	G.711 (10ms)	G.711 (10ms)	G.711 (10ms)

Endpoint				Compact Switch				Result			
Profile		Signaled		Profile		Signaled		TX	RX	TX	RX
20ms	G.711 G.729 G.723	ptime=20	G.711 G.729 G.723	10ms	G.711 G.729 G.723	ptime=20	G.711	G.711 (20ms)	G.711 (20ms)	G.711 (20ms)	G.711 (20ms)
30ms	G.711 G.729 G.723	ptime=30	G.711 G.729 G.723	10ms	G.711 G.729 G.723	ptime=30	G.711	G.711 (30ms)	G.711 (30ms)	G.711 (30ms)	G.711 (30ms)
10ms	G.729 G.711	ptime=10	G.729 G.711	10ms	G.711 G.729 G.723	ptime=10	G.729	G.729 (10ms)	G.729 (10ms)	G.729 (10ms)	G.729 (10ms)
30ms	G.723 G.711	ptime=30	G.723 G.711	10ms	G.711 G.729 G.723	ptime=30	G.723	G.723 (30ms)	G.723 (30ms)	G.723 (30ms)	G.723 (30ms)
30ms	G.723 G.729 G.711	ptime=30	G.723 G.729 G.711	10ms	G.711 G.729 G.723	ptime=30	G.723	G.723 (30ms)	G.723 (30ms)	G.723 (30ms)	G.723 (30ms)
10ms	G.711 G.729 G.723	ptime=10	G.711 G.729 G.723	20ms	G.729 G.711	ptime=10	G.711	G.711 (10ms)	G.711 (10ms)	G.711 (10ms)	G.729 (10ms)
20ms	G.711 G.729 G.723	ptime=20	G.711 G.729 G.723	20ms	G.729 G.711	ptime=20	G.711	G.729 (20ms)	G.729 (20ms)	G.729 (20ms)	G.729 (20ms)
30ms	G.711 G.729 G.723	ptime=30	G.711 G.729 G.723	20ms	G.729 G.711	ptime=30	G.711	G.729 (30ms)	G.729 (30ms)	G.729 (30ms)	G.729 (30ms)
10ms	G.729 G.711	ptime=10	G.729 G.711	20ms	G.729 G.711	ptime=10	G.729	G.729 (10ms)	G.729 (10ms)	G.729 (10ms)	G.729 (10ms)
30ms	G.723 G.711	ptime=30	G.723 G.711	20ms	G.729 G.711	ptime=30	G.711	G.711 (30ms)	G.711 (30ms)	G.711 (30ms)	G.711 (30ms)
30ms	G.723 G.729 G.711	ptime=30	G.723 G.729 G.711	20ms	G.729	ptime=30	G.729	G.729 (30ms)	G.729 (30ms)	G.729 (30ms)	G.729 (30ms)

Table C shows the various combinations of CODECs and packet times possible when the Compact Switch receives a call using the remote CODEC list for preference. In this scenario, the Compact Switch is the “egress” that processes the remote descriptor received from the endpoint.

Table C. CODEC Combinations when Receiving a Call Using Remote CODEC List for Preference

Endpoint				Compact Switch				Result			
Profile		Signaled		Profile		Signaled		TX	RX	TX	RX
10ms	G.711	ptime=10	G.711	10ms	G.711	ptime=10	G.711	G.711	G.711	G.711	G.711

Endpoint				Compact Switch				Result			
Profile		**Signaled**		**Profile**		**Signaled**		**TX**	**RX**	**TX**	**RX**
	G.729 G.723		G.729 G.723		G.729 G.723			(10ms)	(10ms)	(10ms)	(10ms)
20ms	G.711 G.729 G.723	ptime=20	G.711 G.729 G.723	10ms	G.711 G.729 G.723	ptime=20	G.711	G.711 (20ms)	G.711 (20ms)	G.711 (20ms)	G.711 (20ms)
30ms	G.711 G.729 G.723	ptime=30	G.711 G.729 G.723	10ms	G.711 G.729 G.723	ptime=30	G.711	G.711 (30ms)	G.711 (30ms)	G.711 (30ms)	G.711 (30ms)
10ms	G.729 G.711	ptime=10	G.729 G.711	10ms	G.711 G.729 G.723	ptime=10	G.729	G.729 (10ms)	G.729 (10ms)	G.729 (10ms)	G.729 (10ms)
30ms	G.723 G.711	ptime=30	G.723 G.711	10ms	G.711 G.729 G.723	ptime=30	G.723	G.723 (30ms)	G.723 (30ms)	G.723 (30ms)	G.723 (30ms)
30ms	G.723 G.729 G.711	ptime=30	G.723 G.729 G.711	10ms	G.711 G.729 G.723	ptime=30	G.723	G.723 (30ms)	G.723 (30ms)	G.723 (30ms)	G.723 (30ms)
10ms	G.711 G.729 G.723	ptime=10	G.711 G.729 G.723	20ms	G.729 G.711	ptime=10	G.711	G.711 (10ms)	G.711 (10ms)	G.711 (10ms)	G.711 (10ms)
20ms	G.711 G.729 G.723	ptime=20	G.711 G.729 G.723	20ms	G.729 G.711	ptime=20	G.711	G.711 (20ms)	G.711 (20ms)	G.711 (20ms)	G.711 (20ms)
30ms	G.711 G.729 G.723	ptime=30	G.711 G.729 G.723	20ms	G.729 G.711	ptime=30	G.711	G.711 (30ms)	G.711 (30ms)	G.711 (30ms)	G.711 (30ms)
10ms	G.729 G.711	ptime=10	G.729 G.711	20ms	G.729 G.711	ptime=10	G.729	G.729 (10ms)	G.729 (10ms)	G.729 (10ms)	G.729 (10ms)
30ms	G.723 G.711	ptime=30	G.723 G.711	20ms	G.729 G.711	ptime=30	G.711	G.711 (30ms)	G.711 (30ms)	G.711 (30ms)	G.711 (30ms)
30ms	G.723 G.729 G.711	ptime=30	G.723 G.729 G.711	20ms	G.729	ptime=30	G.729	G.729 (30ms)	G.729 (30ms)	G.729 (30ms)	G.729 (30ms)

The following four tables show silence suppression options between the Compact Switch and a remote endpoint for CODECs G.711 and G.729; silence suppression is unavailable for G.723. In the tables, “Signaled” identifies what either switch puts into its signaling, whereas “TX” and “RX” identify what is actually sent and what is expected to be received. Violet fill signifies the Compact Switch; gray fill signifies the endpoint. Note that the Compact Switch properly processes incoming SID packets regardless of whether silence suppression is enabled; thus the “RX” side for the Compact Switch is always true.

Table D shows silence suppression options for G.711 when the Compact Switch originates a call.

Table D. Silence Suppression Options for G.711 when Originating

Compact Switch		Endpoint		Result			
Profile	Signaled	Profile	Signaled	TX	RX	TX	RX
N	N	N	N	N	Y	N	N
N	N	Y	Y	N	Y	N	N
Y	Y	Y	Y	Y	Y	Y	Y
Y	Y	N	N	N	Y	N	N

Table E shows silence suppression options for G.711 when the Compact Switch receives a call.

Table E. Silence Suppression Options for G.711 when Receiving

Endpoint		Compact Switch		Result			
Profile	Signaled	Profile	Signaled	TX	RX	TX	RX
N	N	N	N	N	N	N	N
N	N	Y	N	N	N	N	N
Y	Y	Y	Y	Y	Y	Y	Y
Y	Y	N	N	N	N	N	N

Table F shows silence suppression options for G.729 when the Compact Switch originates a call.

Table F. Silence Suppression Options for G.729 when Originating

Compact Switch		Endpoint		Result			
Profile	Signaled	Profile	Signaled	TX	RX	TX	RX
N	N	N	N	N	N	N	N
N	N	Y	Y	N	N	N	N
Y	Y	Y	Y	Y	Y	Y	Y
Y	Y	N	N	N	N	N	N

Table G shows silence suppression options for G.729 when the Compact Switch receives a call.

Table G. Silence Suppression Options for G.729 when Receiving

Endpoint		Compact Switch		Result			
Profile	Signaled	Profile	Signaled	TX	RX	TX	RX
N	N	N	N	N	N	N	N
N	N	Y	N	N	N	N	N
Y	Y	Y	Y	Y	Y	Y	Y
Y	Y	N	N	N	N	N	N

The following two tables demonstrate DTMF tone relay options between the Compact Switch and a remote endpoint. In the tables, violet fill signifies the Compact Switch; gray fill signifies the endpoint. "Signaled" identifies what either switch puts into its signaling, whereas "TX" and "RX" identify what is actually sent and what is expected to be received. "Inband" DTMF tone relay is voice-encoded; "Outband" DTMF tone relay is in RFC2833 tone relay RTP packets.

Table H shows DTMF tone relay options when the Compact Switch originates a call.

Table H. DTMF Tone Relay Options when Originating

Compact Switch →		Endpoint		Result			
Profile	**Signaled**	**Profile**	**Signaled**	**TX**	**RX**	**TX**	**RX**
Outband	Outband	Inband	Inband	Inband	Inband	Inband	Inband
Outband	Outband	Outband	Outband	Outband	Outband	Outband	Outband
Inband	Inband	Outband	Outband	Outband	Outband	Outband	Outband
Inband	Inband	Inband	Inband	Inband	Inband	Inband	Inband

Table I shows DTMF tone relay options when the Compact Switch receives a call.

Table I. DTMF Tone Relay Options when Receiving

Endpoint →		Compact Switch		Result			
Profile	**Signaled**	**Profile**	**Signaled**	**TX**	**RX**	**TX**	**RX**
Inband	Intband	Outband	Inband	Inband	Inband	Inband	Inband
Outband	Outband	Outband	Outband	Outband	Outband	Outband	Outband
Outband	Outband	Intband	Outband	Outband	Outband	Outband	Outband
Inband	Inband	Inband	Inband	Inband	Inband	Inband	Inband

TDM-VOIP Outgoing Call

For an outgoing call, the SIP (INVITE), message generated by the Compact Switch has an SDP media entry containing the offered CODECs, in preference order, with specified packet time and silence suppression state.

The SDP media entry in the corresponding Answer message received for SIP contains the CODEC selected by the remote peer for the call. The Compact Switch complies with the selection by the remote peer for CODEC, packet time, silence suppression, and Tone Relay payload type.

If the remote peer supports none of the CODECs in the initial outgoing call request, the call cannot be established and is disconnected.

TDM-VOIP Incoming Call

For an incoming call, the Compact Switch receives a proper SIP (INVITE) message with an SDP media entry containing a list of CODECs in preference order from the remote peer, as well as packet time, silence suppression and Tone Relay options.

The Compact Switch compares the received CODEC preference list against the CODEC list specified in the SDP Profile assigned to the trunk group. The Compact Switch sets up the call using the most-preferred CODEC in the preference list from the remote peer that is also common to the local SDP profile in the Compact Switch and for which the Compact Switch has sufficient resources for support. It then sends a proper SIP protocol message indicating its selection in the media entry of the SDP profile.

The Compact Switch also uses the remote peer's choice for packet time, silence suppression, and Tone Relay options in the incoming call request, if supported by the Compact Switch. For example, the Compact Switch does not currently support G.723.1 silence suppression as specified in Annex A. If the selected CODEC is G.723.1 and the remote peer requested enabling silence suppression, the Compact Switch would respond within the SDP that Annex A cannot be supported.

The previously provisioned SDP profiles (see Figure 14) for SIP calls are used to transfer the CODEC list. You assign the SDP to the SIP trunk using the screen illustrated in Figure 14.

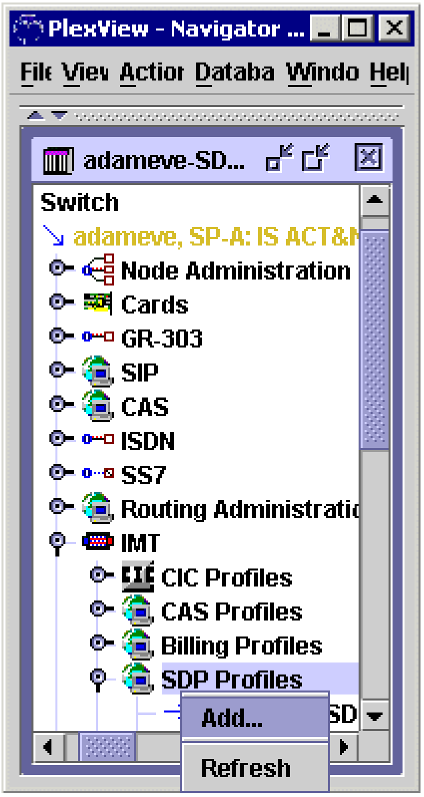
Assigning CODECs to SDP Profiles Using the EMS

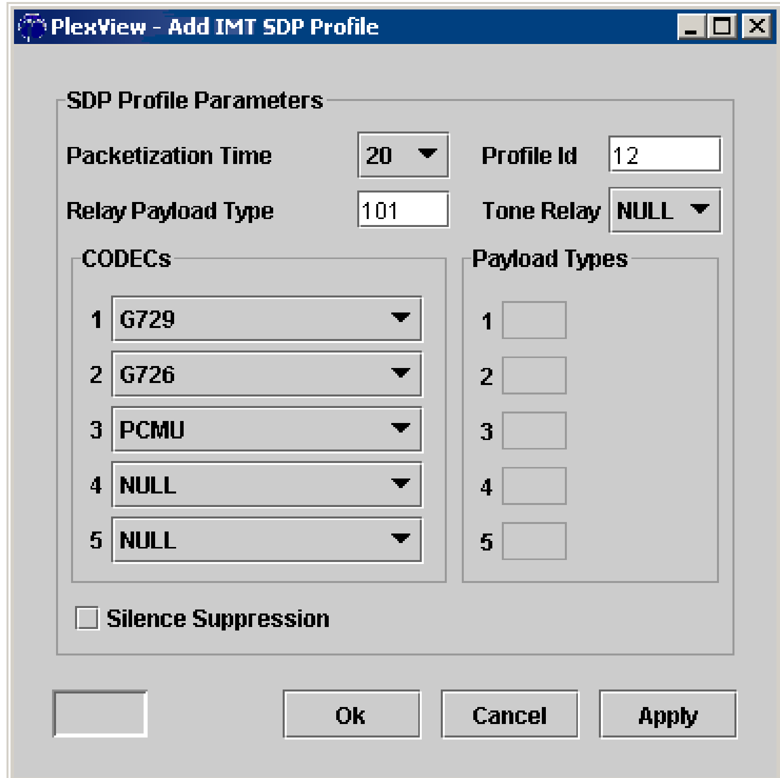
To assign CODECs to SDP profiles, which are then assigned to SIP, SIP-T, ISUP, and CAS trunks, you use the PlexView EMS to first provision the SDP profile, and then to assign the profile to each trunk or line group. You use the Add IMT SDP Profile screen to add an SDP profile.

When provisioning an SDP profile, you must specify the packetization time (compression time), whether DTMF tone relay is enabled and the corresponding relay payload type, whether silence suppression is enabled, and up to five selected CODECs per SDP profile. The configurable packetization time can be set from 10 to 40 ms.

CODEC choices are PCMU (G.711-µLaw for US), PCMA (G.711-aLaw for Europe), G.723.1 (5.3k or 6.3k selectable), G.726 (MSB or LSB first, depending on the communicating device), G.729a, X-CCD (clear channel), CISCO CLEAR CHANNEL, and CLEARMODE. There are also packet type mappings for the different clear mode types. G.729b (G.729a with silence suppression), Clear Channel Support Over Packet and DTMF Relay. See Figure 13 for a sample of the EMS screen used to provision an SDP profile.

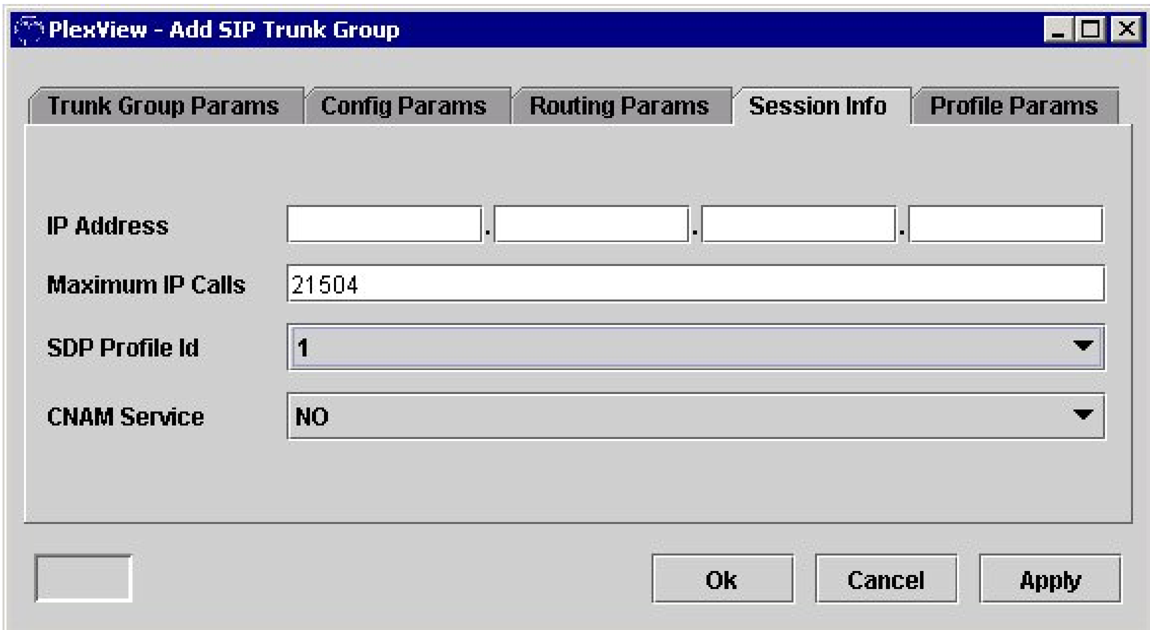
Figure 13. Add IMT SDP Profile





When provisioning trunks groups, you then assign the already provisioned SDP profile to the trunk group. For instance, to assign an SDP to a SIP trunk group, use the screen illustrated in Figure 14.

Figure 14. Add SIP or SIP-T Trunk Group, Session Info Tab, SDP Profile



SIP CODEC Negotiation

The following CODEC limitations exist for support of SIP INFO:

- The long # or double # can be detected on the packet voice stream as long as it is G.711 or G.726. It cannot be detected for G.729a or RFC2833-named telephony events.
- It is only used for transporting of the long # or double #. No other DTMF tones are supported (e.g., *).
- The Compact Switch does not support RFC2833-named telephony events; therefore, a way in which to support SIP INFO is to use the SIP SUBSCRIBE/NOTIFY messages to pass the RFC2833-named telephony events. The usage of SUBSCRIBE/NOTIFY for DTMF tone detection is also supported to eliminate the need for a Media Server to do the detection (i.e., the information is passed via the signaling path rather than the bearer).

For more information, refer to the CODEC Support section.

VoIP Diagnostics

The Compact Switch supports IP traceroute, which is a diagnostic tool used for providing a trace of the list of routers traversed to a specific destination, along with the number of hops. Traceroute is a standard IP diagnostic utility.

Related Documents

- *TL1 Commands Reference Guide*
- *Installation and Operations Manual, Vol. I*
- *Element Management System User's Guide*

Notes: