AT&T VOIP
Nortel CS 1000 (Release 5.00W / 5.50J) SIP Configuration Guide
For Use with AT&T VoEVPN Services

NN10000-108
Issue 2.1
02/04/2010

CS 1000

TABLE OF CONTENTS

TABLE OF FIGURES

1 Introduction

This document provides a configuration guide to assist Nortel Networks CS 1000 administrators in connecting to AT&T Voice over Enhanced Virtual Private Network (VoEVPN) service.

This configuration guide pertains specifically to the AT&T Voice over EVPN service and is not intended for configuring a CS 1000 system for new installation.

1.1 Pre-IP PBX Configuration Activity

This guide assumes that the administrator is knowledgeable in CS 1000 IP PBX programming and operations.

An important tool that the administrators should have at their disposal prior to testing their IP PBX with Voice over EVPN service is a network protocol analyzer. Such software can be used to run traces on problem calls so the information can be shared with equipment and network engineers. There is a free version of such software that can be obtained at http://www.wireshark.org/.
A second alternative that customers may use is TCPDUMP which can be found on most UNIX and Linux systems. To use this software the customer should have Wireshark or TCPDUMP loaded on a server that is connected to a LAN switch or hub that can monitor both the signaling and media packets on any calls between the customer PBX and the Voice over EVPN managed router. Please note, however, that AT&T does not offer, warrant, or support this software, and any use of the Wireshark or TCPDUMP software is entirely at the customer's own risk.

1.2 Customer Questions

Section 4 of this guide provides screen shots and instructions for the configuration of your IP PBX. Should you have questions regarding these instructions, please call Brian Stegemoller at +1 (972) 685-6629; ((972) 745-5139 after 2/22/2010). When calling this number please have the following information available:

- Company name
- Company location
- Administrator name and phone number
- IP PBX name and software version
- Customer Configuration Guide – Issue number and date

1.3 Trouble Reporting

Nortel and AT&T will make every effort to quickly resolve reported troubles. The time required for trouble shooting can be reduced if the customer has the necessary detailed information available when reporting a problem. Prior to reporting a problem please provide a Wireshark or TCPDUMP trace of the failed call.

1.4 Document Feedback

IP PBX administrators who would like to provide feedback on the contents of this document should send it to Brian Stegemoller at (briantegemo@avaya.com) with a copy to Al Chee (alchee@avaya.com) and Steven Chen (stevenchen@avaya.com).

1.5 Document Change History

Draft 0.1	12/1/2008; Initial draft
Draft 0.2	12/11/2008; ensured that VoEVPN acronym is used throughout document.; added notation for failed CallPilot voice mail retrieval and added emergency 911 limitation in Section 2; inserted Universal Trunk Card in equipment list and changed G.729 codec supported, from G.729AB to G.729 30 ms payload size in Section 3.
Issue 1.0	12/17/2008; General availability for issue 1.0 of CCG guide.
Issue 2.0	02/04/2009; Corrected Media Gateway Controller (MGC) Codec from G.729B to G.729A and VAD value unchecked.
Issue 2.1	02/04/2010 Updated Contact Information to Reflect AVAYA Merger

2 Special Notes

VoEVPN Managed Access Router Information
All the VoEVPN access router fixes associated with the CS1000 are included in Cisco release IOS 12.4.(15)T.6 and later.

Voice Mail Retrieval from IOS Cisco Gateway to Nortel CallPilot Fails.
When compressed RTP is enabled on access routers, on-net voice mail retrieval to the Nortel CallPilot does not work properly. It is recommended to deploy larger bandwidth circuits if remote CallPilot voice mail retrieval is desired. Alternatively, the PSTN can also be used to access CallPilot voice mail systems.

E**mergency 911/E911 Service Limitations**
All emergency 911 calls should be routed through PSTN trunks at each CS1000 IP PBX locations.

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3 Overview

This section provides a service overview of the Nortel Networks Communication Server 1000 Release 5.5 (CS1000 R5.5) IP PBX integration with AT&T Voice over Enhanced Virtual Private Network (VoEVPN) service. This will enable the CS1000 to place VoIP Gateway to Gateway, on-net and off-net PSTN (hop-off only) calls using the SIP (Session Initiation Protocol) protocol.

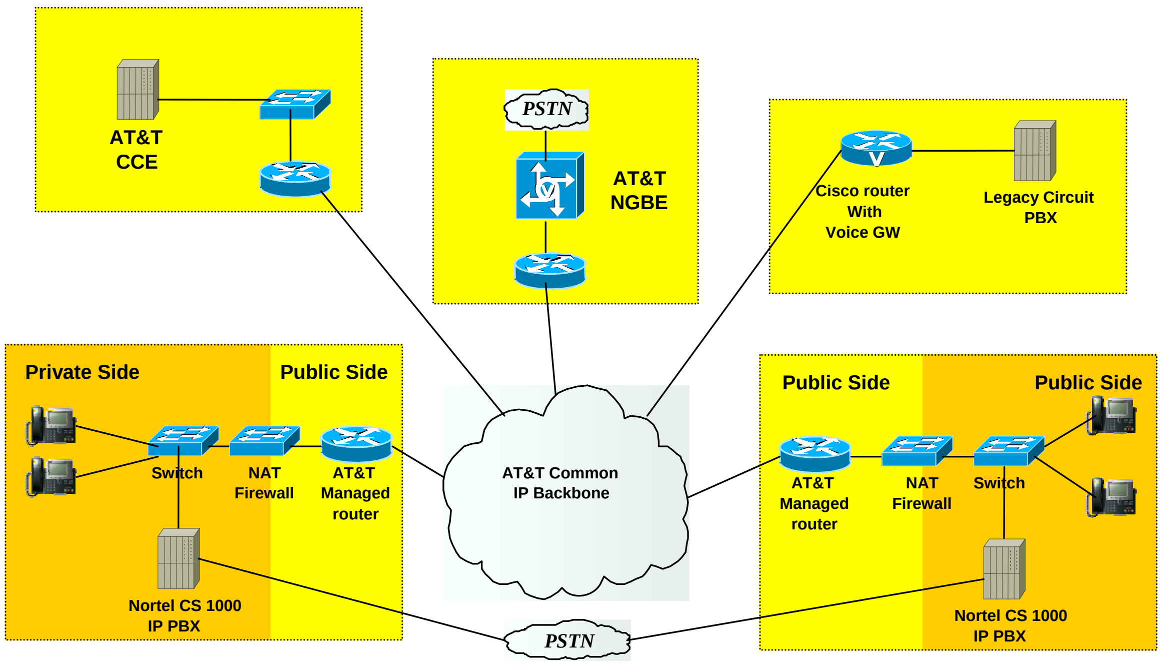

Figure 1: AT&T VoEVPN Architecture

The Nortel CS 1000 customer premises site shall consist of the following components:

- Nortel IP Phone 2000 series, 1100 series, IP Softphone 2050 (Release 3.00.0197 and up)* – These phones use the Nortel UNIStim signaling protocol to communicate to the Nortel CS 1000 IP PBX for call feature and routing support. These phones can be connected to a Nortel Ethernet switch (ES 470, ERS 5520, etc.) that supplies in-line power (IEEE 802.3af) to the phones.
- Nortel CS 1000 IP PBX – This unit consists of the following:
    - Dedicated Signaling Server (can be COTS, ISP 1100, or CP-PM)
    - Call Server (can be CP-PIV or CP-PM cards)
    - Media Gateway Controller (MGC) card to provide Digital Signaling Processor (DSP) resources for connecting IP and Time Division Multiplexing (TDM) devices together and for advanced applications such as conferences and voicemail access
    - CallPilot voicemail system (optional)

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- o Digital Line Card (DLC) for Meridian digital sets (optional)
- o Analog Message Waiting Line Card (AM/WLC) for connection to fax machines and analog sets (optional)
- o 8- Port Universal Trunk Card (Analog trunks to PSTN)
- o TMDI card as a PRI/T1 trunk to the PSTN (optional)

IMPORTANT A dedicated Signaling Server will be needed for connectivity to AT&T VoEVPN service. Private MCDN features would require an additional Signaling Server. See figure below:

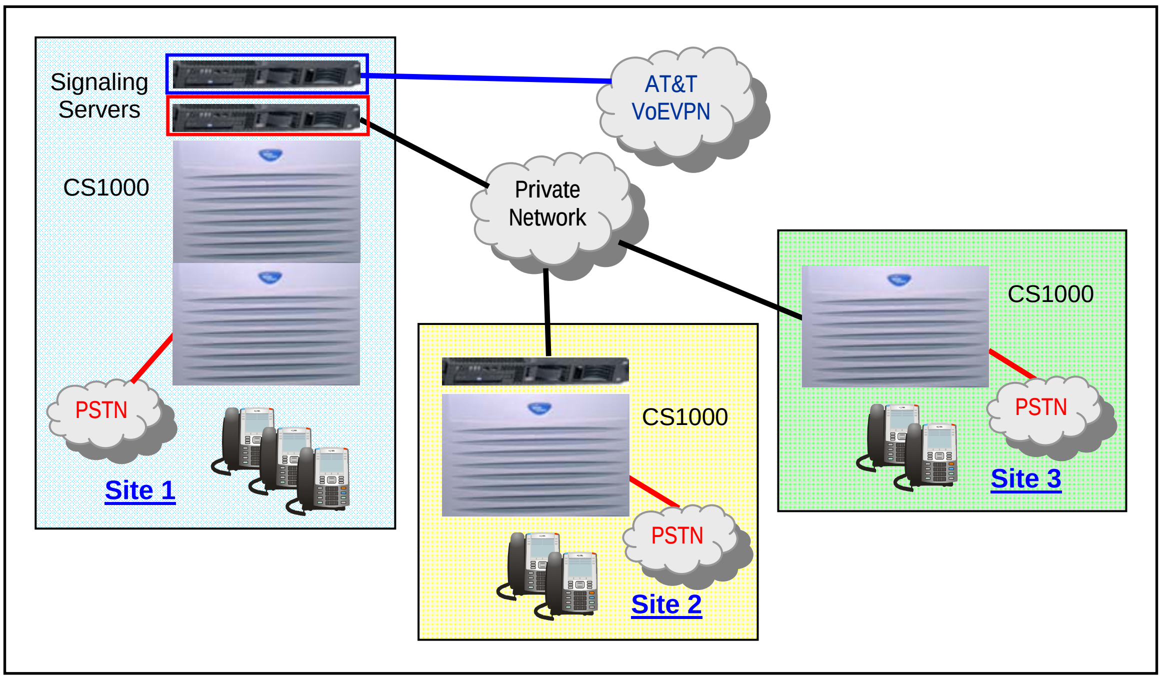

Figure 2: AT&T VoEVPN and private networking

Please note that this guide <u>does not</u> describe the procedures to configure private MCDN functionality with the CS 1000. This guide only pertains to the Signaling Server connected to AT&T Voice over EVPN.

The following routing scenarios are supported by the Nortel CS 1000 IP PBX and **DO NOT** use the AT&T Call Control:

- Local Nortel CS 1000 phone to local Nortel CS 1000 phone
- Inbound PSTN to Nortel Networks CS1000 phone
- Outbound local PSTN calls from the Nortel Networks CS1000 phones.
- Outbound local N11 (i.e. 411, 911) calls from the Nortel Networks CS1000 phones

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The following routing scenarios are supported by the CS 1000 IP PBX and **DO** use the AT&T Call Control. For voice calls, the G.729A codec and 30 ms payload shall be used:

- Nortel CS 1000 phones to PSTN (domestic US and international)
- Nortel CS 1000 phones to legacy PBX site with Cisco gateway
- Legacy PBX site with Cisco gateway to Nortel CS 1000 phones
- Nortel CS 1000 phones at one Nortel CS 1000 IP PBX site to Nortel CS 1000 phones at another Nortel CS 1000 IP PBX site.

Fax was tested and is supported on the CS 1000 using the T.38 fax protocol through the AT&T VoEVPN network to/from (except PSTN) the following:

- PSTN (Outbound only)
- Legacy PBX site with Cisco gateway
- Another CS 1000 IP PBX site

* The new release of the Nortel IP Softphone 2050 (Release 3.00.0197) is supported with CS 1000 and AT&T VoEVPN.

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4 Configuration Guide

This configuration guide specifies the Nortel CS 1000 screens that must be configured and updated to support the AT&T VoEVPN service.

4.1 Nortel CS 1000 Version

The Nortel CS 1000 IP PBX software version 5.5 is required.

4.1.1 Nortel CS 1000 Release 5.5

The Nortel Networks CS 1000 Call Server must be running release **5.50J**. You can check the version of CS 1000 by viewing the following screen on the Home page:

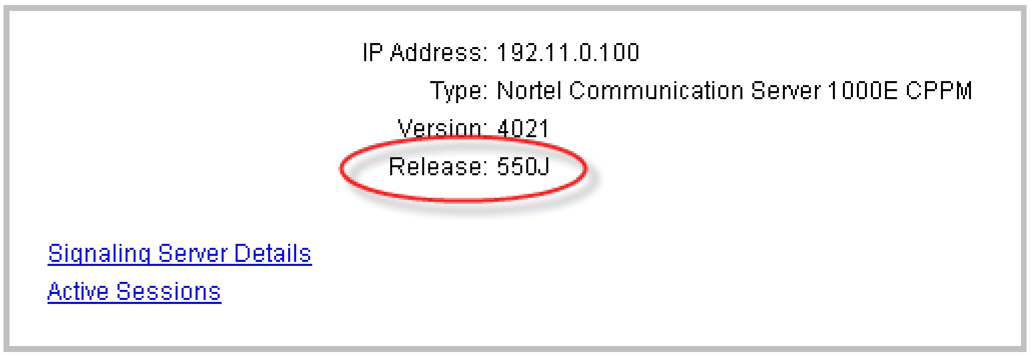

Figure 3: CS 1000 Release 5.5 Call Server software

The CS 1000 Signaling Server must be running release **sse-5.50.12**:

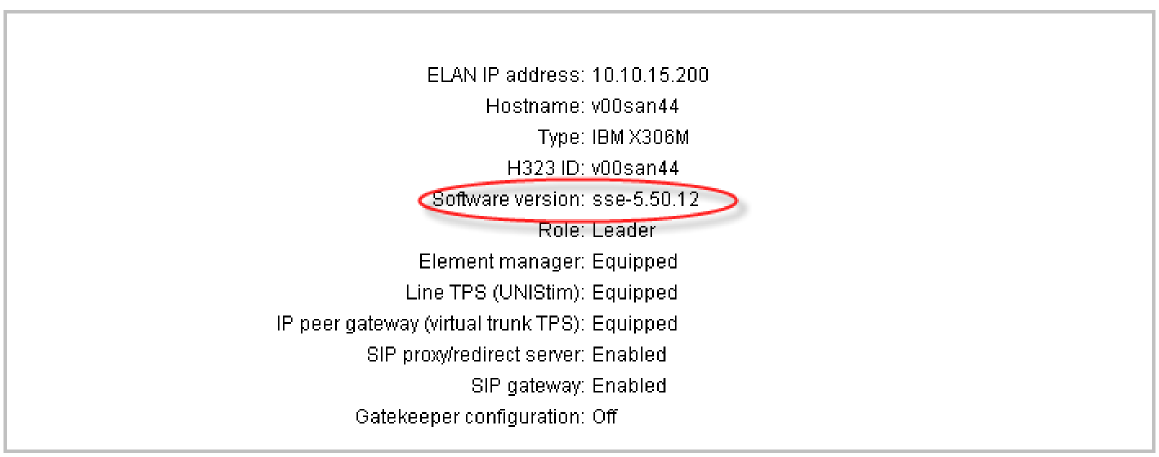

Figure 4: CS 1000 Release 5.5 Signaling Server software

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4.2 Nortel CS 1000 Patches

To verify installed/applied patches on the Signaling Server, go to
System » Software » Servers and Media Cards, select the appropriate Element (Signaling Server) and Platform types, and click on the “PSTAT” button.

To download CS 1000 patches, please refer to Section 7, Appendix A: Downloading CS 1000 Patches.

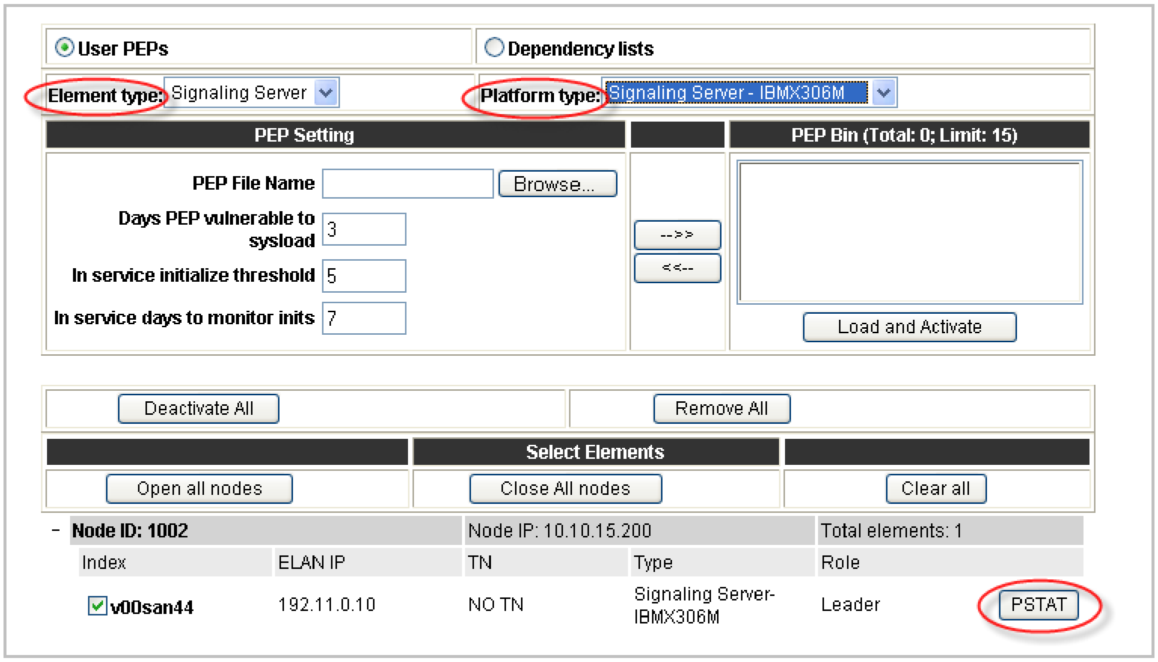

Figure 5: Viewing CS 1000 Signaling Server patches

4.2.1 Patches for Nortel CS 1000 Release 5.5

The following patches are needed for the Signaling Server:

Patch ID	Description
MPLR22452	Don't attach MCDN message for outgoing INVITE
MPLR22968	TO/FROM URI Incorrect - Replace domain population in the FROM field to the Node IP Address of the Signaling Server. Updated version 2 corrected PRACK issue.
MPLR24785	If the privacy value is set to none, then privacy header is removed ** IMPORTANT: Ensure that calling line privacy feature is disabled on the CS 1000 system as well as on the individual sets. **
MPLR25982	No speech path if 183 session in progress comes before 180 Ringing (need this for outbound calls to call prompters and AT&T wireless cell phones)
MPLR23632	Null values should be allowed for Public E.164/National or

	Subscriber fields
MPLR27236	New functionality for handling "maxptime" for outgoing calls. The CS1000 will now use the configured codec payload value e.g. 20ms, 30ms, etc.

To verify installed/applied patches on the Call Server, go to **System » Software » Call Server**, select "PEP Status (PSTAT)" for the command, and click "Submit."

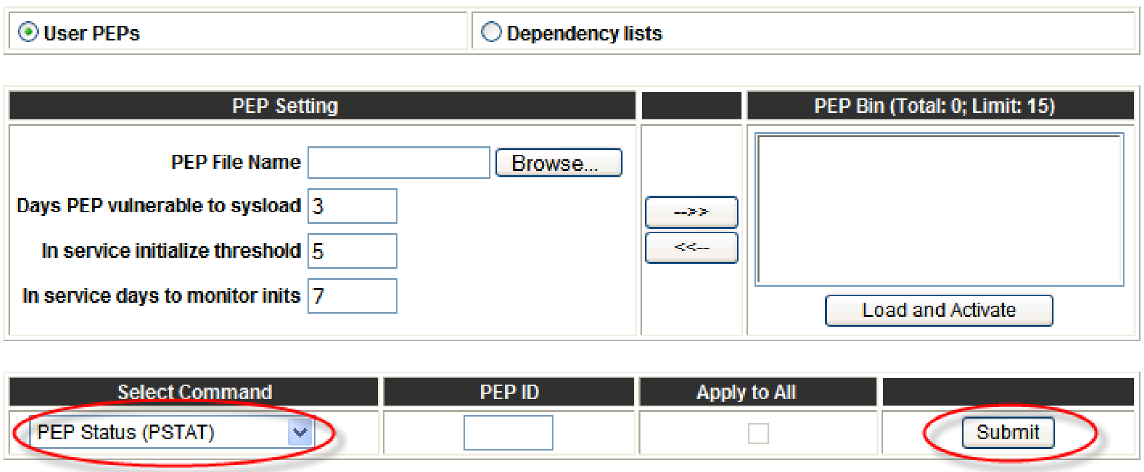

Figure 6: Viewing CS 1000 Call Server patches

Currently no patches are required for the Call Server:

Patch ID	Description
None	

NOTE	In order to view and install CS 1000 Call Server patches, a level 2 user with PDT2 privileges must be logged in.

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4.3 Node Configuration

Add or edit a node in the **System » IP Network » Node Configuration** menu with the following configuration as noted in the below screenshots:

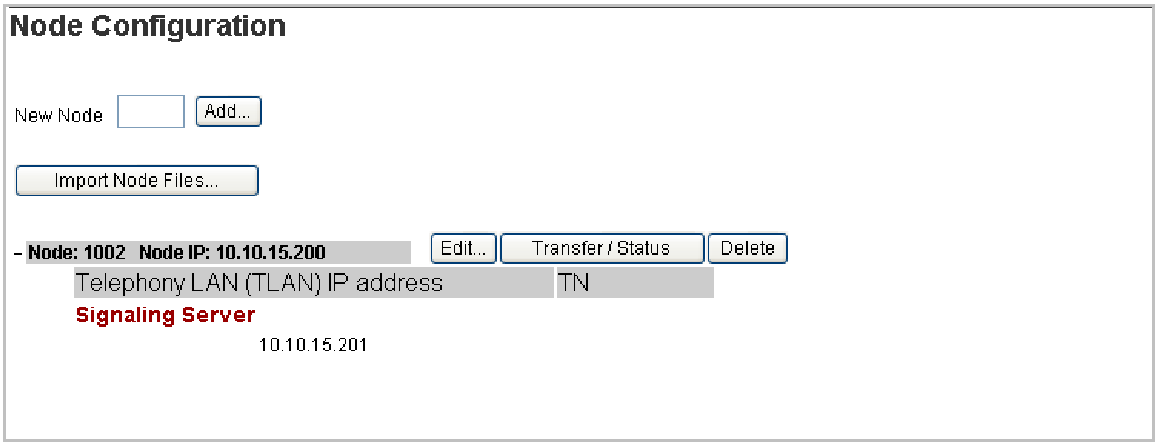

Figure 7: Node configuration screen

4.3.1 IP Codecs

The following codecs should be enabled:

- **G711** with **Voice payload size** to 30 ms/frame
- **G729A** with **Voice payload size** to 30 ms/frame, **VAD** is unchecked
- **T38 FAX**

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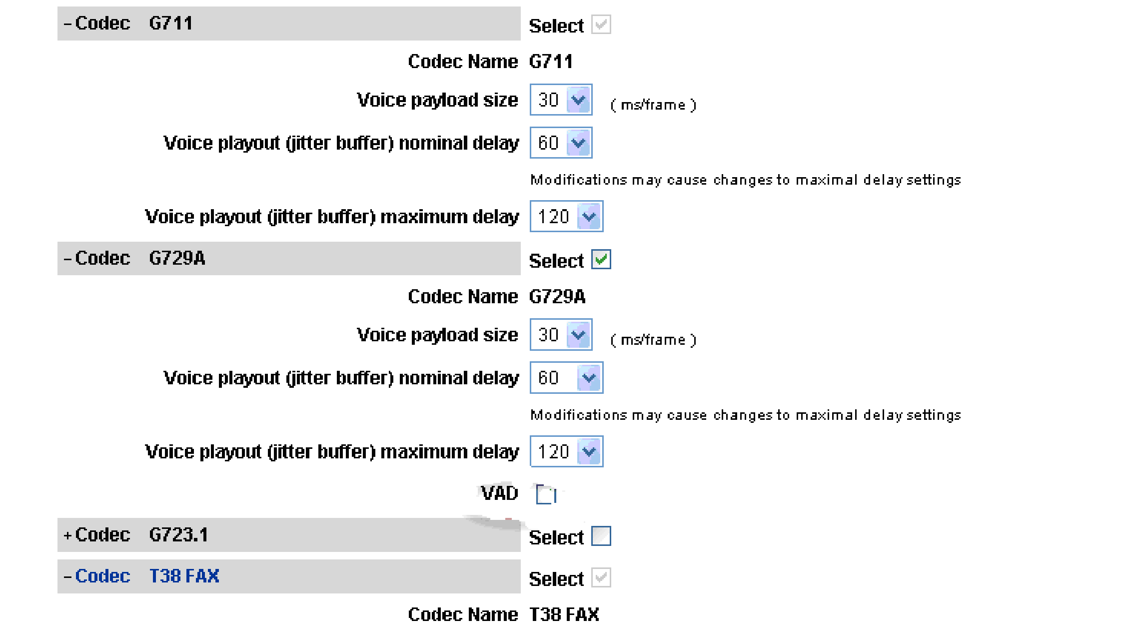


Figure 8: IP codecs

SIP (VTRK) trunks should be in a ZONE that uses “**Best Bandwidth (BB)**.” This will ensure that G.729 codec is offered first.

Ensure that the IP codec settings in this section are also configured on the Media Gateway Controller (MGC) card under the VGW and IP phone codec profile section in the **System » IP Network » Media Gateways » IPMG x x Property Configuration » IPMG x x Media Gateway Controller (MGC) Configuration** menu. For interoperability with AT&T Voice over EVPN, use G.729A (under Codec G729A sub-section, ensure that **VAD** is **unchecked**).

4.3.2 LAN Configuration

All IP sets use the same port ranges for media. This is specified in the LAN configuration section, under the Telephony LAN (TLAN) configuration sub-section. In the figure below, the starting port 29100 was entered. Note: for VoEVPN access routers requiring compression, select a starting port from 16384 through 32768.

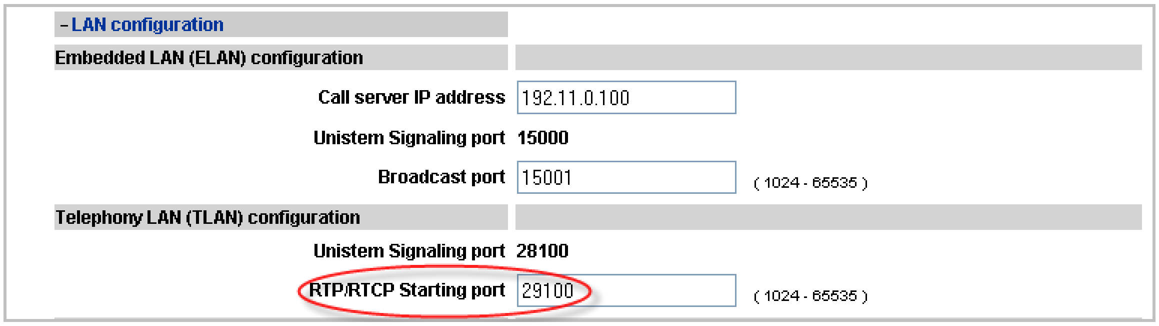


Figure 9: RTP/RTCP port for IP sets

4.3.3 SIP Gateway Settings

AT&T VoEVPN service does not support TLS Security, thus the **Security Policy** should be set to "Security Disabled." Enter the Signaling Server's TLAN IP address for the **Primary Proxy or Redirect (TLAN) IP address**, ensure the **Port** used is 5060, and the **Transport Protocol** used is UDP.

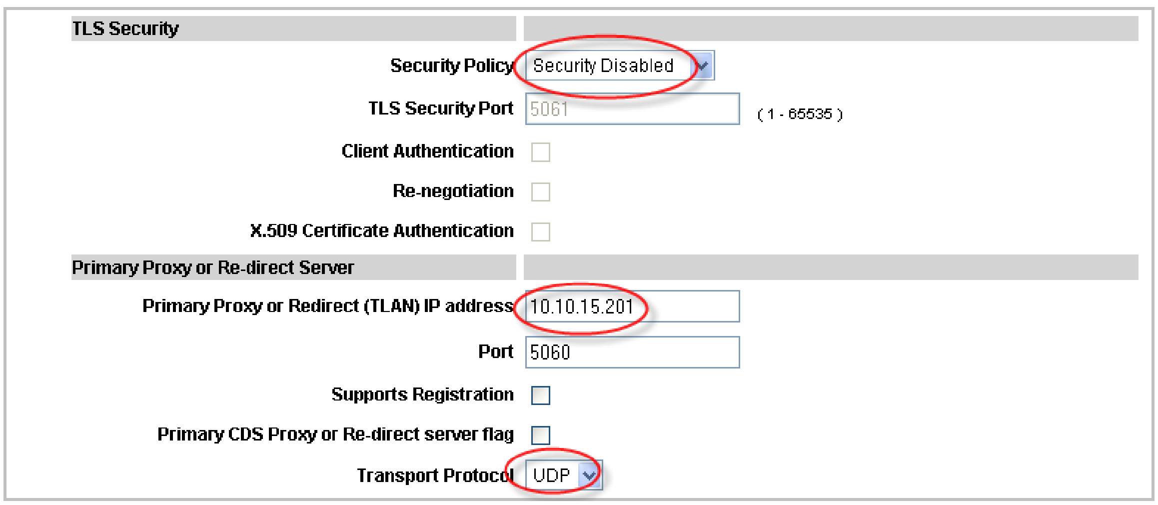


Figure 10: SIP GW settings

4.3.4 SIP URI Map

Ensure that all parameters for the SIP URI Map are left blank; otherwise, calls to AT&T VoEVPN service will fail. This can be done with MPLR23632 loaded.

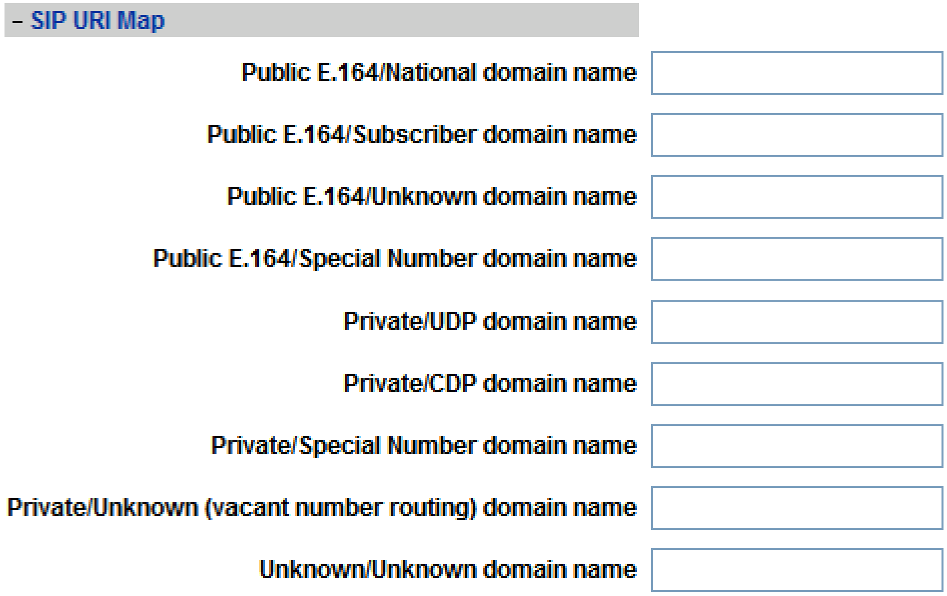


Figure 11: SIP URI map

4.3.5 Signaling Server Properties

For the Signaling Server that will be used to peer with the AT&T VoEVPN network, configure the **Virtual Trunk TPS** to “SIP only”, ensure that the **SIP Proxy/Redirect Server** option is checked, **Local SIP TCP/UDP Port to Listen to** is 5060, **SIP Domain name** is the IP address of the AT&T SIP Proxy Server, and **Enable Gatekeeper** option is unchecked.

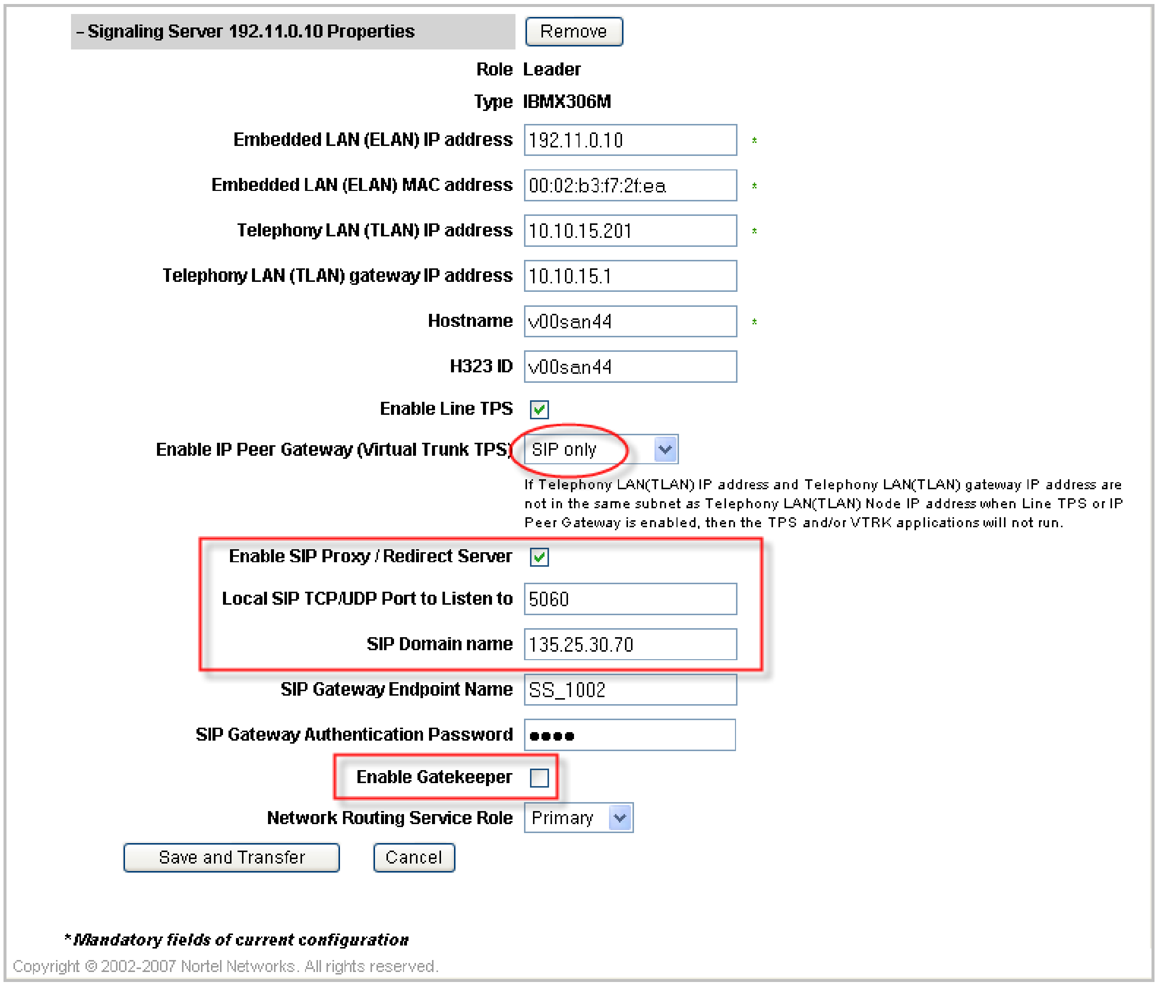


Figure 12: Signaling Server properties

4.4 VoIP Trunking

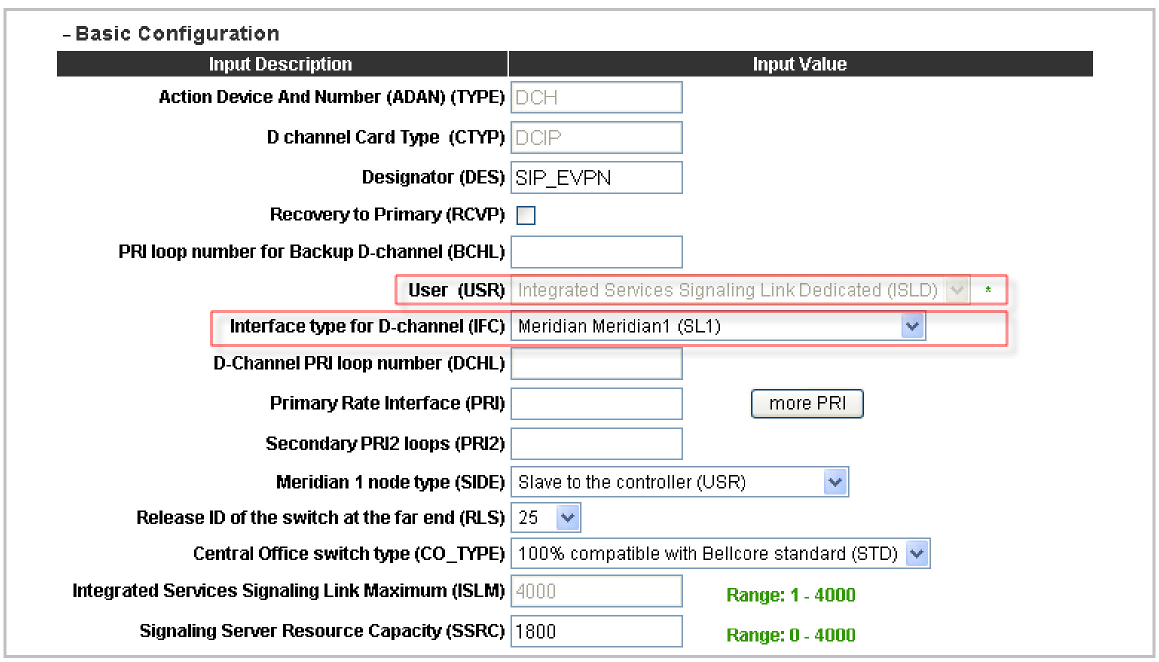
Voice over IP (VoIP) lines, are signaling channels that simulate how CO lines work. However, VoIP lines transmit data to the IP network over a LAN or IP network rather than over physical lines.

To create VoIP trunks on the CS 1000 to the AT&T VoEVPN service, the following steps need to be executed:

- Creating D-Channels
- Creating Incoming Digit Conversion (IDC) and Calling Line Identification (CLID) trees
- Creating Routes
- Adding Trunks to the specific Route
- Creating a Digit Manipulation Index (DMI)
- Creating Route List Block Indices (RLI)
- Allowing/Restricting NPA Codes
- Configuring Special Numbers

4.4.1 Creating D-Channels

Call signaling on the CS 1000 resides on the D-channels. A route will be mapped to this D-channel, in which the Signaling Server will send call signaling to the AT&T VoEVPN service using the specified D-channel. Under the **Routes and Trunks » D-Channels** menu, add a D-channel with the following configuration:



- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D channel Card Type (CTYP)	DCIP
Designator (DES)	SIP_EVPN
Recovery to Primary (RCVP)	
PRI loop number for Backup D-channel (BCHL)	
User (USR)	Integrated Services Signaling Link Dedicated (ISLD) *
Interface type for D-channel (IFC)	Meridian Meridian1 (SL1)
D-Channel PRI loop number (DCHL)	
Primary Rate Interface (PRI)	more PRI
Secondary PRI2 loops (PRI2)	
Meridian 1 node type (SIDE)	Slave to the controller (USR)
Release ID of the switch at the far end (RLS)	25
Central Office switch type (CO_TYPE)	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum (ISLM)	4000 Range: 1 - 4000
Signaling Server Resource Capacity (SSRC)	1800 Range: 0 - 4000

Figure 13: D-channel basic configuration

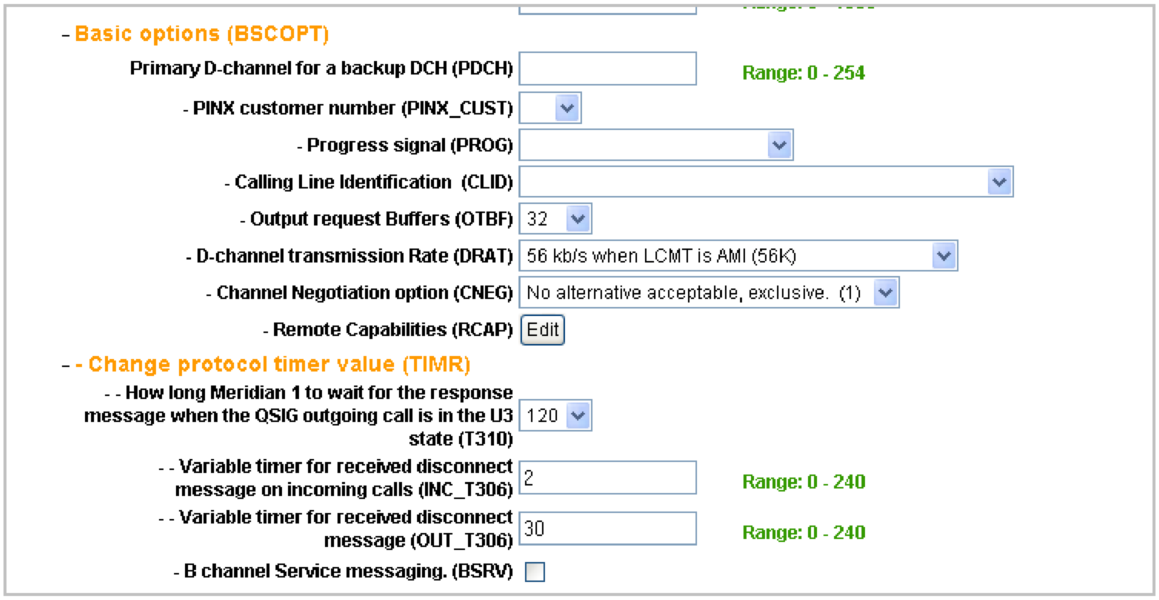


Figure 14: D-channel basic options

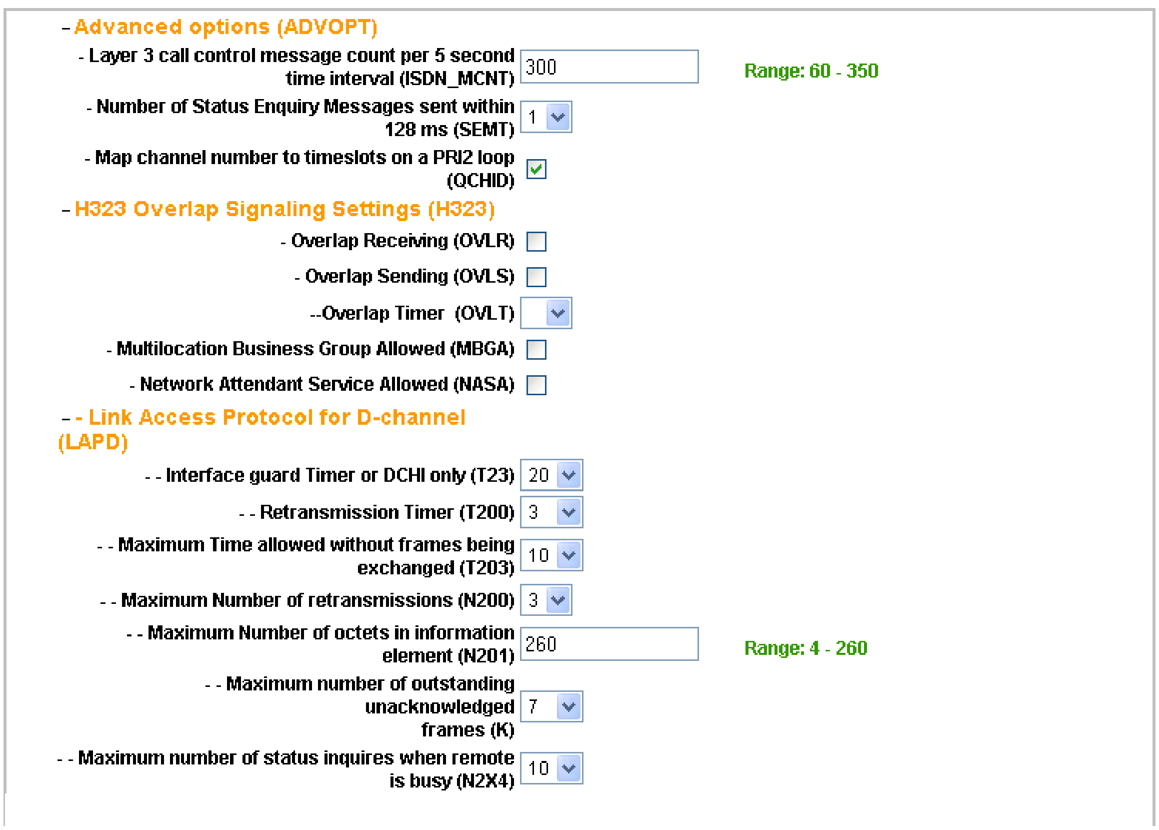


Figure 15: D-channel advanced options

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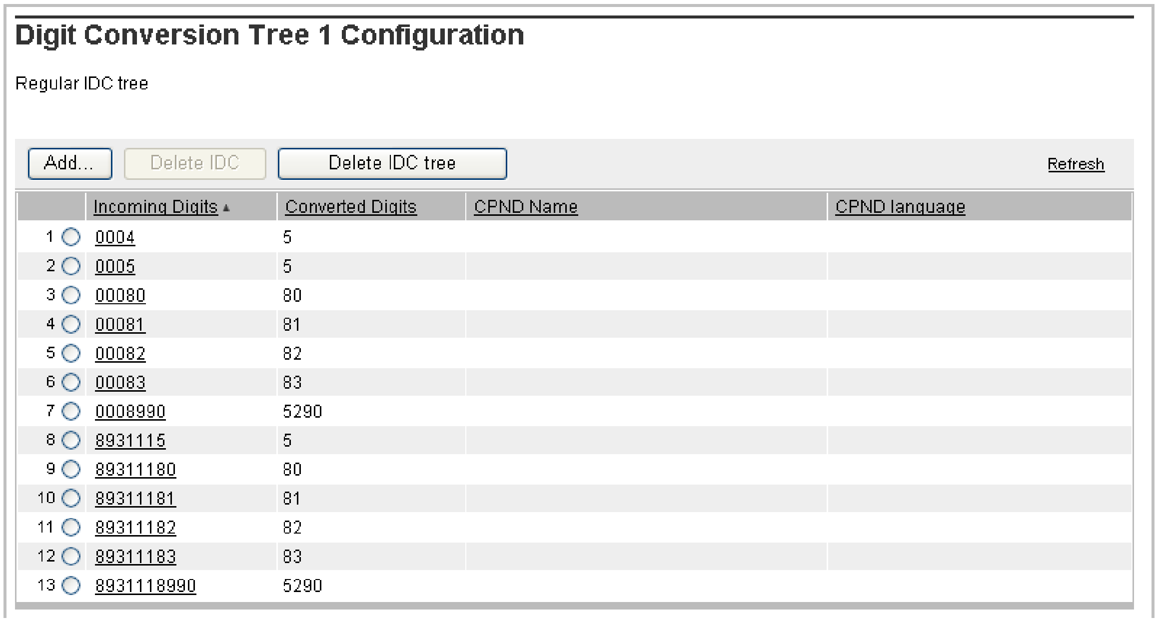
4.4.2 Incoming Digit Conversion and CLID Trees

Recommended best practices for planning DNs with DIDs is to map the last four digits of the Direct Inward Dialing (DID) number to the CS 1000 DNs. However, there will be scenarios where DNs may not match with the AT&T-provided DID number extensions. In this case, the Incoming Digit Conversion (IDC) and Calling Line Identification (CLID) trees will be used.

4.4.2.1 Creating IDC Trees

The IDC tree will allow incoming digits to be converted to specified local extensions. This is done in **Dialing and Numbering Plans » Incoming Digit Translation » Customer 00 » Digit Conversion Tree x Configuration** menu, clicking "Add," then entering incoming and converted digits; see figure below:


Digit Conversion Tree 1 Configuration

Regular IDC tree

Add... | Delete IDC | Delete IDC tree | Refresh

	Incoming Digits	Converted Digits	CPND Name	CPND language
1	0004	5		
2	0005	5		
3	00080	80		
4	00081	81		
5	00082	82		
6	00083	83		
7	0008990	5290		
8	8931115	5		
9	89311180	80		
10	89311181	81		
11	89311182	82		
12	89311183	83		
13	8931118990	5290		

Figure 16: Incoming digit conversion tree

NOTE	Up to 8 digits can be entered in for the Incoming Digits field in Element Manager. To enter more, read below.

If AT&T VoEVPN service sends 732-216-2779 to the CS 1000, the CS 1000 will use entry 2 to strip 732-216-2, convert the digits to 2, append the remainder of the digits and ring the set with DN 2779.

In the case where the full 10 digit number needs to be converted to a DN, the administrator will have to use the Call Server CLI. For example, to convert 732-368-0430 to 2001, go to **ld 49**:

```
>ld 49
DGT000
MEM AVAIL: (U/P): 99201863    USED U P: 5027918 40040    TOT: 104269821
DISK SPACE NEEDED: 41 KBYTES
REQ  chg
TYPE idc
CUST 0                     <customer number>
DCNO 1                     <digit conversion tree number>
IDGT 7323680430            <incoming digit>

7323680432 2001            <converted digit>
IDGT
```

Figure 17: Converting incoming 10 digits to DN

4.4.2.2 Creating CLID Trees

The same concept is applied to Direct Outward Dialing (DOD) numbers for outbound calling numbers. This is done in the Call Server CLI, **ld 15**. For creating CLID tables where the extensions of the phones match the DID/DOD, see Section 4.4.2.2.1; for non-matching extensions, see Section 4.4.2.2.2.

4.4.2.2.1 Matching Extensions

For the example of using DN 2779 to 732-216-2779 (last four digits of extension match DN):

```
>ld 15

MEM AVAIL: (U/P): 99201833    USED U P: 5027918 40070    TOT: 104269821
DISK SPACE NEEDED: 41 KBYTES
REQ: chg
TYPE: net
CUST 0
OPT
AC2
FNP
CLID yes
  SIZE
  INTL
  ENTRY <enter #>
    HNTN 732
    ESA_HLCL
    ESA_INHN
    ESA_APDN
    HLCL 216
    DIDN YES                 * use when DN matches DOD extension
    HLOC
    LSC
    CLASS_FMT DN
   ENTRY # SAVED!
```

Figure 18: Creating CLID table for DODs matching DN

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Be sure when you configure the keys for the IP set in ld 11 with the above configuration, use the appropriate CLID table entry #. For example, `KEY 0 scr 2779 #` will allow the CS 1000 to send calling party number of 732-216-2779 out to AT&T IP Flexible Reach service.

4.4.2.2.2 Non-matching Extensions

For the example where DN 2001 needs to be converted to DOD 732-368-0430:

```
>ld 15

MEM AVAIL: (U/P): 99201833    USED U P: 5027918 40070    TOT: 104269821
DISK SPACE NEEDED: 41 KBYTES
REQ: chg
TYPE: net
CUST 0
OPT
AC2
FNP
CLID yes
  SIZE
  INTL
  ENTRY <enter #>
    HNTN 732
    ESA_HLCL
    ESA_INHN
    ESA_APDN
    HLCL 3680430
    DIDN NO                * use when DN DOES NOT match DOD extension
    HLOC
    LSC
    CLASS_FMT DN
  ENTRY # SAVED!
```

Figure 19: Creating CLID table for DODs not matching DN

Be sure when you configure the keys for the IP set in ld 11 with the above configuration, use the appropriate CLID table entry #.

4.4.3 Creating Routes

Once the D-Channel is created, routes will be created to map to the D-Channel and the IDC trees, if applicable. The route will be configured with SIP trunking to the AT&T IP Flexible Reach service. At the **Routes and Trunks » Routes and Trunks** screen, add a route:

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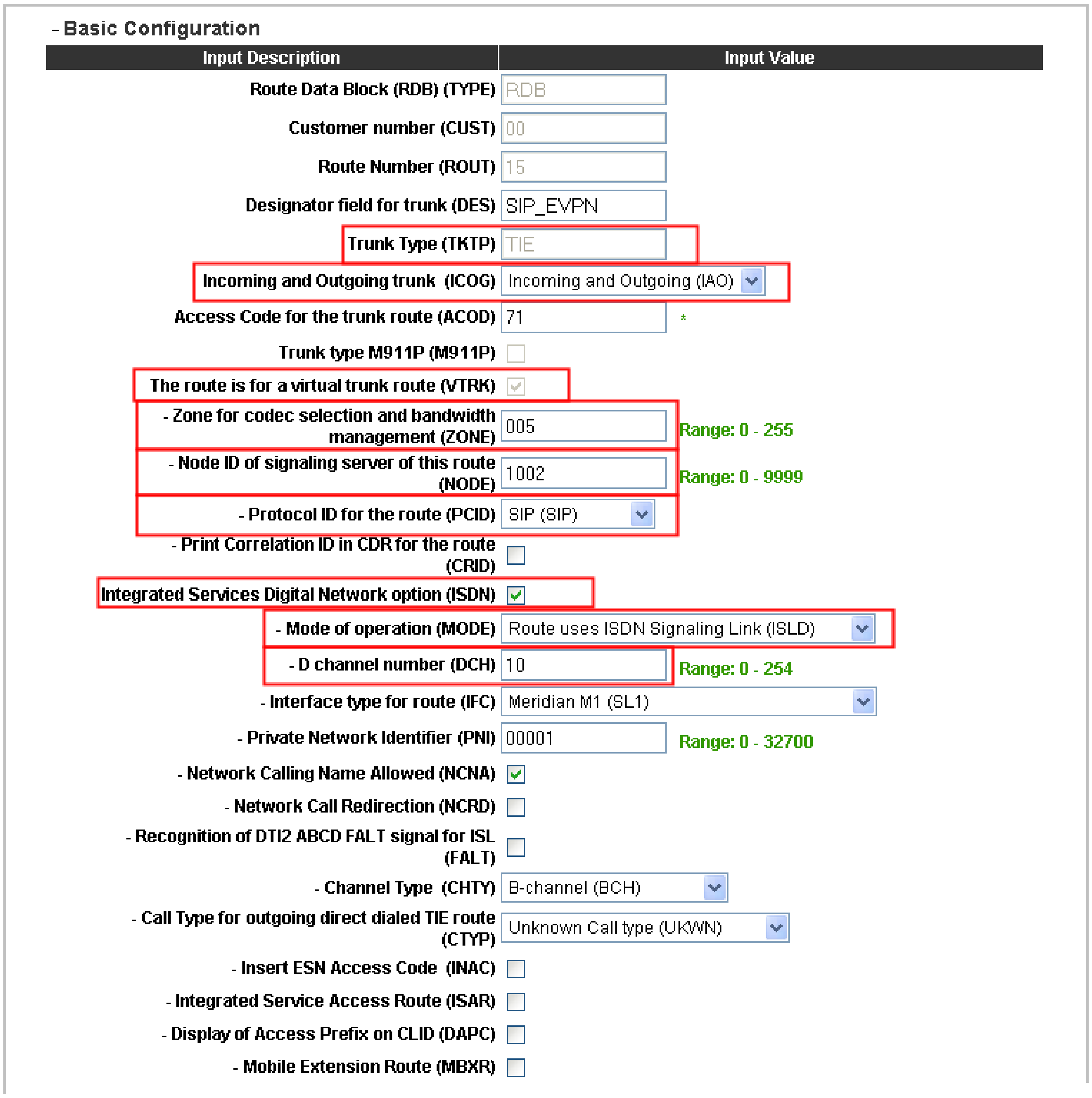

Figure 20: Route configuration

Ensure of the following settings:

- **Trunk Type** is TIE
- **Incoming and Outgoing trunk** is Incoming and Outgoing (IAO)
- **The route is a virtual trunk route** is checked
- **Node ID of signaling server** is set to Signaling Server peering with AT&T Voice Over EVPN service
- **Zone for codec selection and bandwidth management** is set to a zone for the virtual trunks
- **Protocol ID for the route** is SIP
- **ISDN option** is checked
- **Mode of operation** is ISLD
- **D channel number** is D-channel created in Section 4.4.1

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Furthermore, under the same screen, in **Basic Route Options**, ensure that IDC is checked and enter in IDC tree numbers for both Day and Night IDC trees.

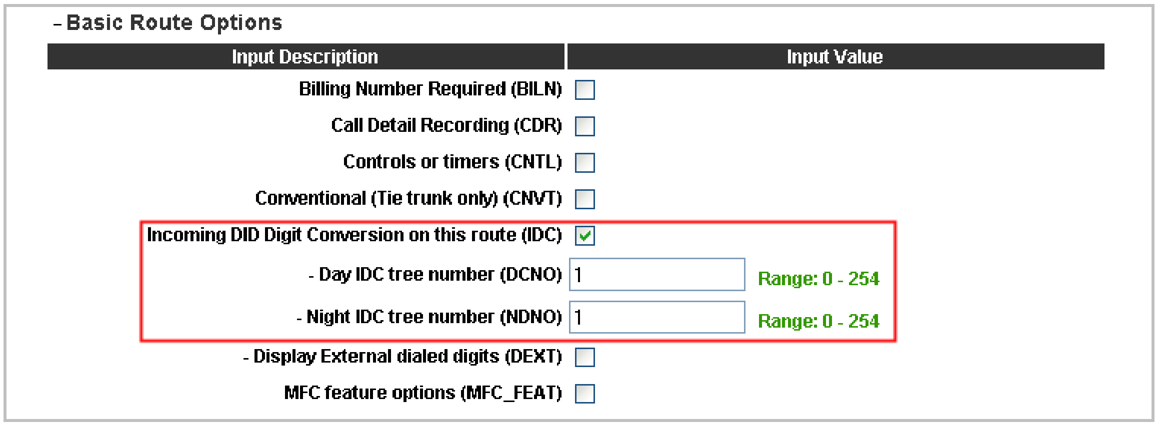

Figure 21: IDC for route

4.4.4 Adding Trunks to the Specific Route

Trunks can now be added to associate with a given route. Back to the **Routes and Trunks » Routes and Trunks** screen, for the route created in Section 4.4.3, click "Add trunk."

NOTE: To add multiple trunks, select the number of trunks to add from the drop-down menu in **MTINPUT**.

Figure 22: Adding multiple trunks

Under the **Basic Configuration**, ensure the following:

- **TYPE** is IPTI
- Enter **TN** for the trunks
- **XTRK** is VTRK
- **RTMB** in following notation <route number in 4.4.3> <number>
- **CDEN** is 8D
- **STRI**, **STRO** is IMM
- **CHID** is a different number for each trunk
- **INC** set to YES

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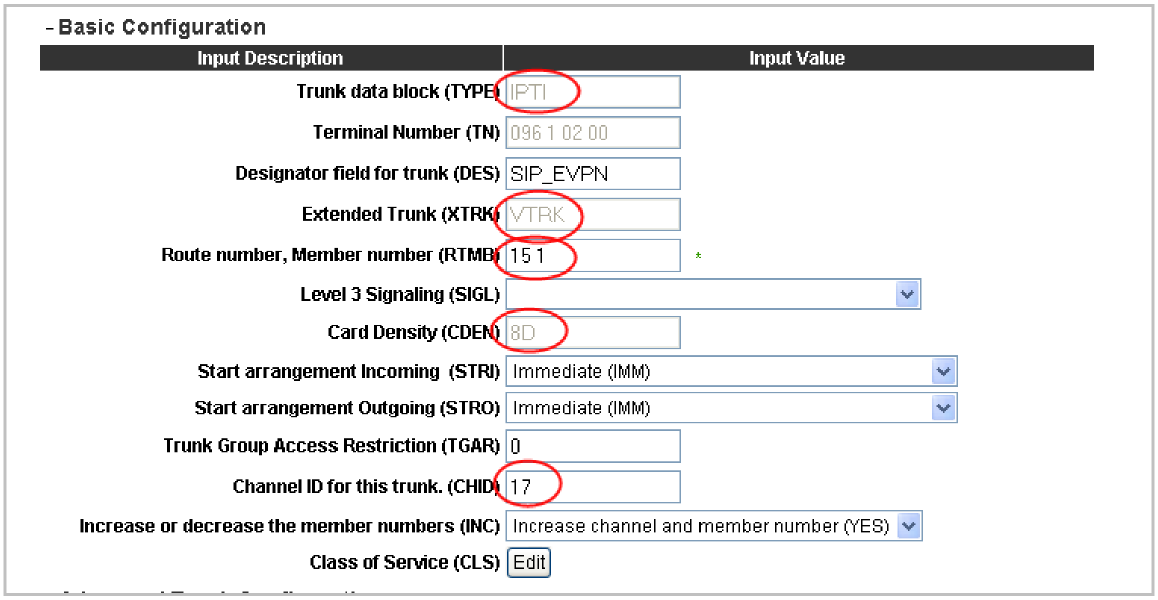

Figure 23: Trunk basic configuration

Under the **Advanced Trunk Configurations**, ensure that SUPN and STYP are checked and PIP, respectively:

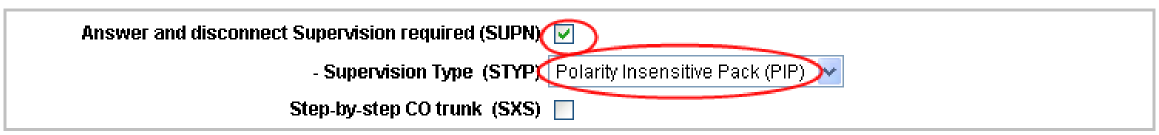

Figure 24: Trunk advanced configuration

4.4.4.1 Disabling Media Security on Virtual Trunks

In conference and transfer call scenarios where the CS 1000 sends a SIP re-INVITE to an endpoint via the virtual trunks to AT&T, if "Media Security" is enabled on these trunks, calls will fail. In order to fully interoperate with AT&T IP Flexible Reach service, "Media Security" must be set to "Media Security Never (MSNV)" under the **Routes and Trunks » Routes and Trunks » Customer 0, Route x, Trunk x Property Configuration » Class of Service Configuration** for each virtual trunk peering with AT&T IP Flexible Reach service.

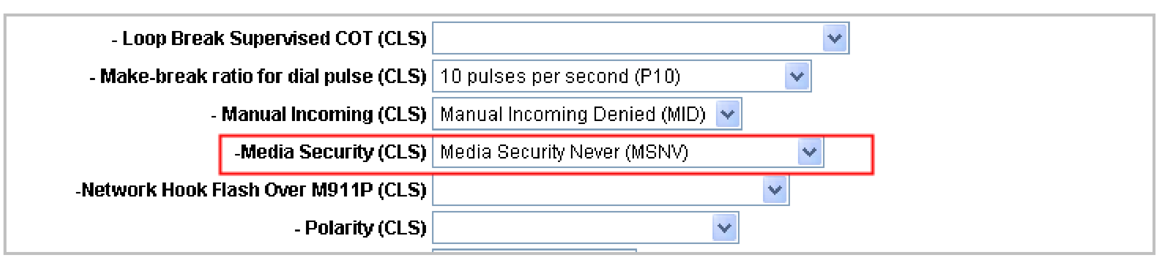

Figure 25: Disabling media security on the trunk

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4.4.5 Creating a Digit Manipulation Index (DMI)

In order for outbound calls (more specifically, special numbers) to send the correct information to the AT&T network, ensure that a Digit Manipulation Index (DMI) is created in **Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Digit Manipulation Block List » Digit Manipulation Block** with the following settings (in this example, DMI 15 is used):

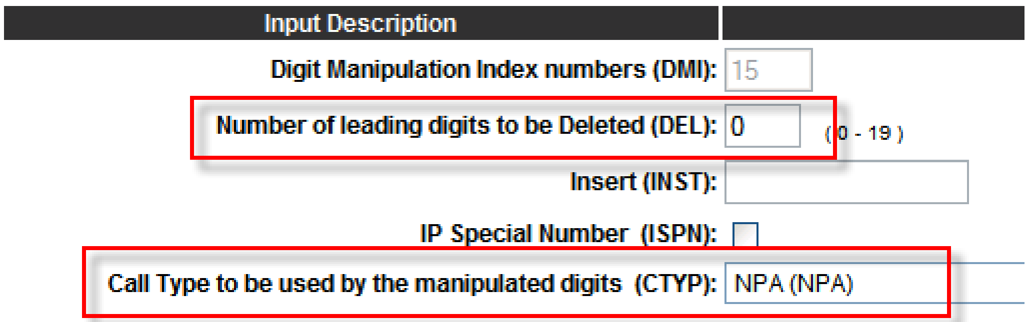


Figure 26: Digit Manipulation Index

This will ensure that the proper calling line ID is sent to the AT&T network, and thus allowing calls to special numbers to complete.

4.4.6 Creating Route List Block Indices (RLI)

When an outbound call is made, based on the number dialed, a route list block index is used to determine the route and D-channel used for call signaling.

To create a RLI, go to **Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Route List Blocks**, and add a route list index.

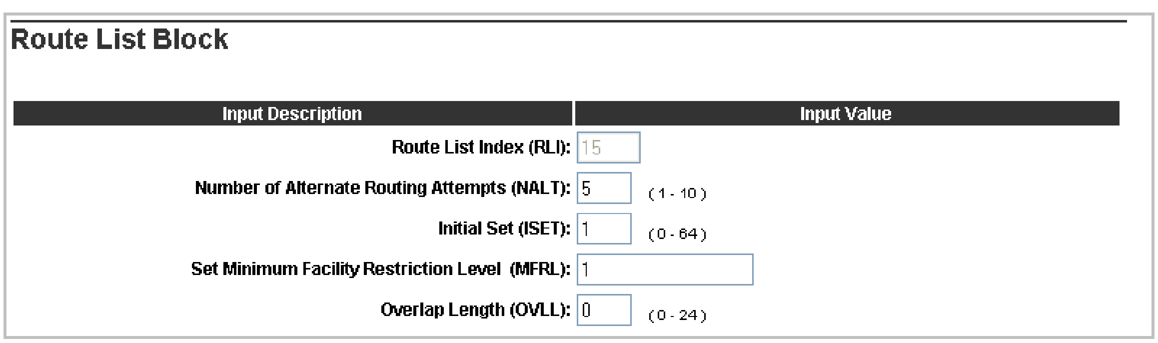


Figure 27: Route list block parameters

Add a “Data Entry Index” and ensure that **ROUT** is set to the route number created in 4.4.3 and **DMI** is set to the DMI created in 4.4.5:

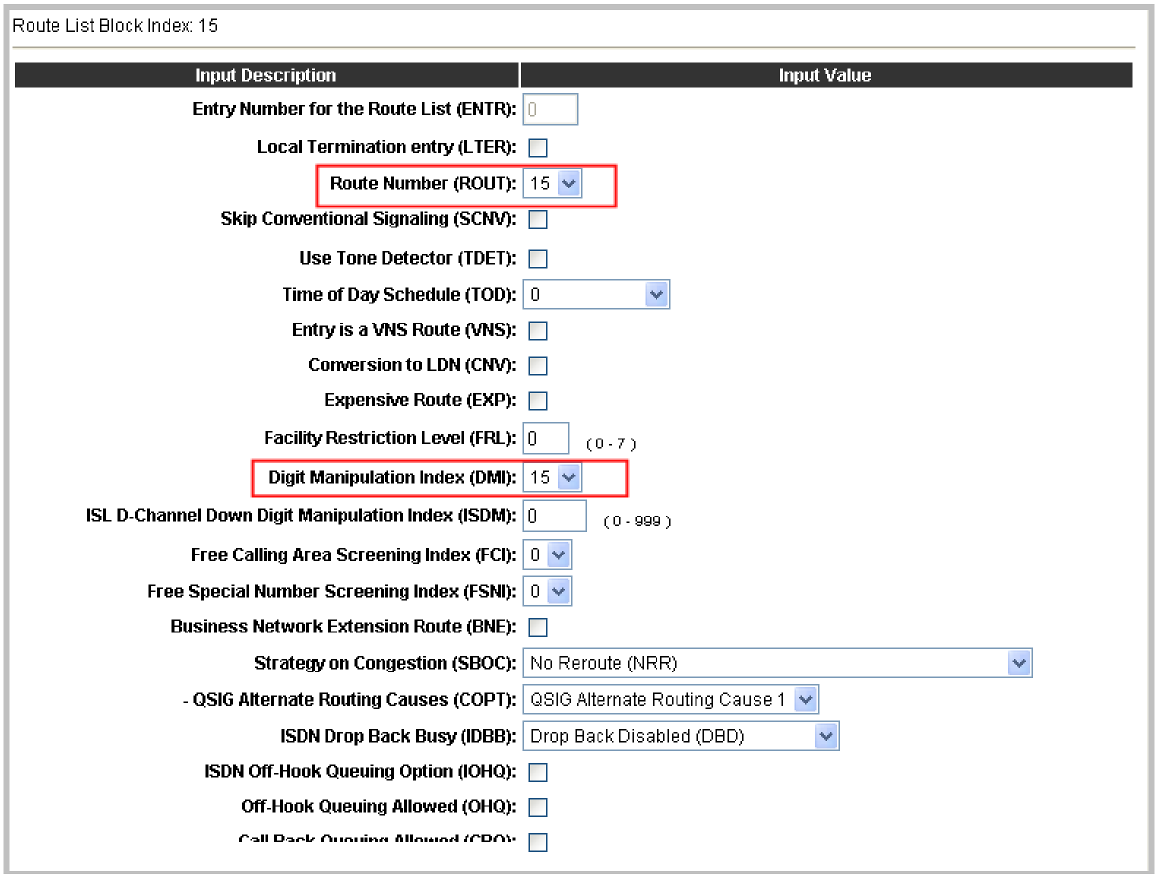


Figure 28: Route list block, data entry index

4.4.7 Allowing/Restricting Numbering Plan Area (NPA) Codes

Add the allowed NPA codes for the CS 1000 in the **Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Numbering Plan (NET) > Access Code 1 » Numbering Plan Area Code List** menu, and map the Route List Index to the RLI configured in the previous section. Below is an example for area code 732:

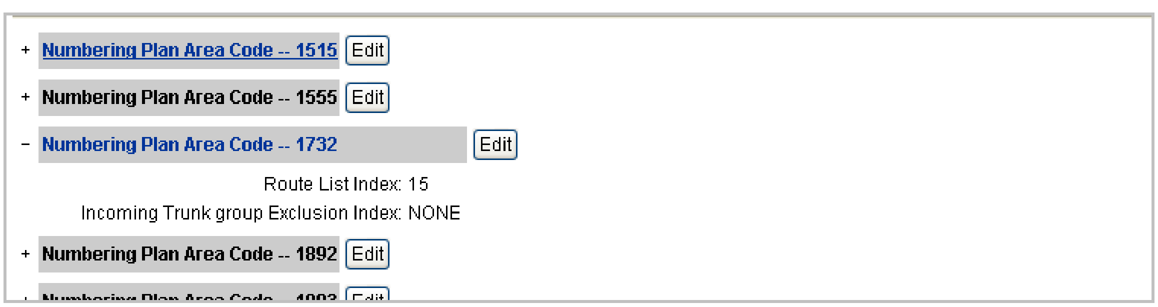


Figure 29: NPA codes

4.4.8 Configuring Special Numbers

Special numbers are numbers that do not follow the NPA dial plans. Examples are 1-800, N11 (411, 911, etc.), and international calls. To configure these, go to **Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Numbering Plan (NET) > Access Code 1 » Special Number List**, and add allowed special numbers. Map these numbers to the RLI in Section 4.4.6; below is an example for 411:

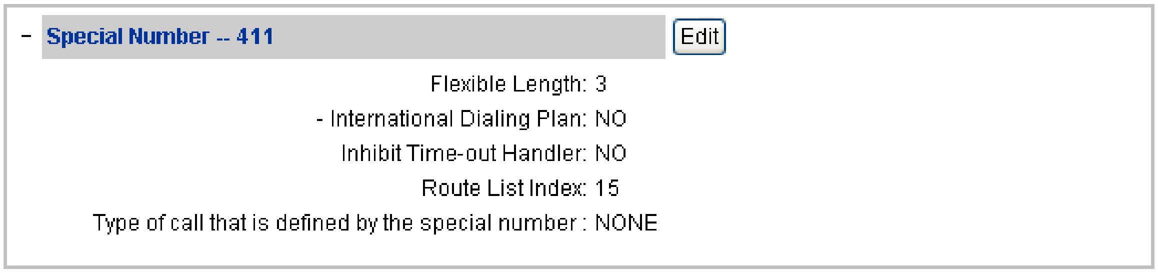


Figure 30: Special numbers

5 Basic Monitoring and Call Tracing

The following procedures below can be used to monitor and trace calls on the CS 1000. The CS 1000 has an extensive suite of diagnostic procedures, and is out of the scope of this document. For more information on advanced diagnostics, please refer to the Nortel CS 1000 Release 5 Technical Documentation.

5.1 Viewing Registered Sets on CS 1000

The following Call Server CLI command in **ld 96** can be used to display registered sets:

```
=> ecnt node
Node: 1001
  Number of Registered Ethersets   : 5
```

Figure 31: Displaying registered sets to a specific node

```
=> ecnt ss

Signaling Server: SS_1001  IP: 192.12.0.10
  Number of Registered Ethersets   : 5
```

Figure 32: Displaying registered sets to a specific Signaling Server

```
=> ecnt modl
2004P2: IP Phone 2004 Phase 2
  Number of IP phones: 2
1140E: IP Phone 1140E
  Number of IP phones: 3
```

Figure 33: Displaying registered sets based on phone models

5.2 Active Call Information

Administrators can view active call information for the specific DN while on a call. Go to **ld 80** in the Call Server CLI and enter the following command: `trac <customer #> <DN>`. Information such as IP addresses, codecs, calling/called party numbers, etc. is displayed:

```
.trac 0 2001

ACTIVE  VTN 096 0 01 01

ORIG   VTN 096 1 02 00   VTRK IPTI  RMBR  16 1 INCOMING VOIP GW CALL
  FAR-END SIP SIGNALLING IP: 135.25.29.135
  FAR-END MEDIA ENDPOINT IP: 135.25.29.70  PORT: 16390
  FAR-END VendorID: Cisco-SIPGateway/IOS-12.x
TERM   VTN 096 0 01 01  KEY 0  SCR MARP  CUST 0  DN 2001  TYPE 1140
  MEDIA ENDPOINT IP: 172.16.6.101  PORT: 28802
MEDIA PROFILE: CODEC G.729A NO-LAW  PAYLOAD 20 ms  VAD OFF
RFC2833:  RXPT  96   TXPT  96   DIAL DN 2001
MAIN_PM  ESTD
TALKSLOT  ORIG  66   TERM  2
EES_DATA:
NONE
QUEU  NONE
CALL ID 0 18822


----  ISDN ISL CALL (ORIG) ----
CALL REF # =  400
BEARER CAP =  VOICE
HLC =
CALL STATE =  10     ACTIVE
CALLING NO =  17323681000  NUM_PLAN:E164    TON:INTERNATIONAL  ESN:UNKNOWN
CALLED NO  =  7323680430  NUM_PLAN:UNKNOWN     TON:UNKNOWN   ESN:UNKNOWN
```

Figure 34: Active call information

5.3 SIP Call Tracing

When a call is being placed to/from the CS 1000, the SIP messages can be outputted to the screen when logged into the Signaling Server CLI. The `SIPCallTrace` command can be turned on to view the SIP messages. The `SIPTraceLevel` command can be set to either 0 or 1; 0 for less-detailed output, 1 for more details.

NOTE	Nortel strongly recommends that this command be left **off** during normal operations; the command to disable is as follows: `SIPCallTrace off`.

Below is a sample output:

```
oam> SIPTraceLevel 1
oam> SIPCallTrace on
oam>
oam> 02/10/2007 16:48:13 LOG0003 SIPNPM: sipNpmPrivacyHdrBuild: : PATCH -
Privac
y Value was equal to NONE, so header not added PATCH
02/10/2007 16:48:13 LOG0006 SIPNPM: SIPCallTrace: This is Outgoing Message
02/10/2007 16:48:13 LOG0006 SIPNPM: SIPCallTrace:
Message: Outgoing method INVITE(0)  chid: 27 Called num: 17324208823 Far End
Sig
naling IP: 172.16.6.111:5060 Transport:UDP  CSeq: 1 INVITE
From: "Al Chee"<sip:7323680430@172.16.6.110;user=phone>
02/10/2007 16:48:13 LOG0006 SIPNPM: SIPCallTrace:
To: <sip:17324208823@135.25.29.135;user=phone>
User-Agent: Nortel CS1000 SIP GW release_5.0 version_sse-5.00.31
02/10/2007 16:48:13 LOG0006 SIPNPM: SIPCallTrace:
Media Info: 172.16.6.101 Codecs:  G729(18) G711 U-Law(0) G711 A-Law(8)
Dynamic(9
6) Dynamic(111) Payload: 20 ms Media State: SIPNPM_MEDIA_SENDRECV

02/10/2007 16:48:13 LOG0006 SIPNPM: SIPCallTrace: This is Incoming Message
02/10/2007 16:48:13 LOG0006 SIPNPM: SIPCallTrace:
Message: Incoming response 302 Moved Temporarily chid: 27 Called num:
1732420882
3 Far End Signaling IP: 135.25.29.135:65535 Transport:UDP  CSeq: 1 INVITE
From: "Al Chee"<sip:7323680430@172.16.6.110;user=phone>
02/10/2007 16:48:13 LOG0006 SIPNPM: SIPCallTrace:
To: <sip:17324208823@135.25.29.135;user=phone>
Contact: <sip:17324208823@135.25.29.135;user=phone>

02/10/2007 16:48:13 LOG0006 SIPNPM: SIPCallTrace: This is Outgoing Message
02/10/2007 16:48:13 LOG0006 SIPNPM: SIPCallTrace:
Message: Outgoing method ACK(1)  chid: 27 Called num: 17324208823 Far End
Signal
ing IP: 172.16.6.111:5060 Transport:UDP  CSeq: 1 ACK
From: "Al Chee"<sip:7323680430@172.16.6.110;user=phone>
02/10/2007 16:48:13 LOG0006 SIPNPM: SIPCallTrace:
To: <sip:17324208823@135.25.29.135;user=phone>
User-Agent: Nortel CS1000 SIP GW release_5.0 version_sse-5.00.31
```

Figure 35: Sample SIPCallTrace output

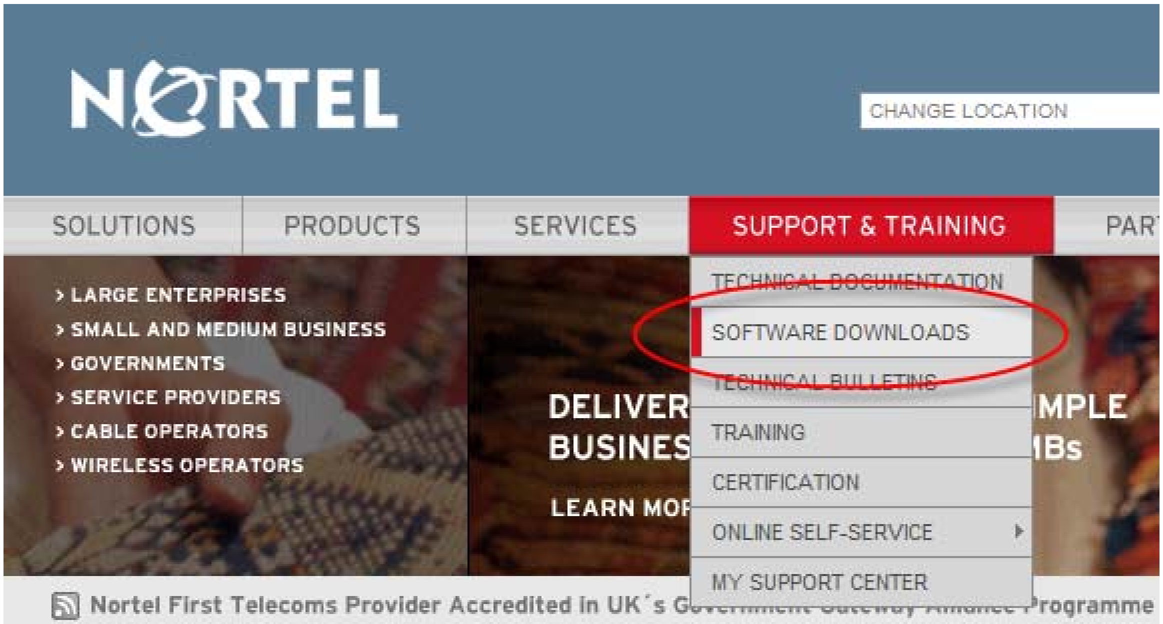
6 Table of Acronyms

Acronym	Definition
AM/WLC	Analog Message Waiting Line Card: provides talk battery and signaling for regular 2-wire common battery 500-type (rotary dial) and 2500-type (Digitone dial) telephones and key telephone equipment. This card also supports message waiting indication for sets equipped with the message waiting feature.
COTS	Commercial Off-The-Shelf: standard server hardware manufactured by a third-party; the CS 1000 supports the following COTS servers – IBM xSeries 306m (types 8848, 8491) and HP ProLiant DL320 G4 servers.
CP-PIV	Common Process Pentium IV: processor for the Call/Signaling Server on a legacy large system (CS 1000M), is also supported on CS 1000E systems.
CP-PM	Common Processor Pentium Mobile: the main processor for the Call Server, controlling all call processing and telephony services. It also provides the system memory required to store operating software and customer data. This is the default processor for the CS 1000E. **Note:** the Signaling Server can also be a CP-PM card.
CS 1000	Communications Server 1000
DLC	Digital Line Card: provides a multiplexed voice, data, and signaling path to and from a digital terminal (telephone) over a 2-wire full duplex 512 kHz Time Compression Multiplexed (TCM) digital link.
ISP 1100	Legacy Nortel Signaling Server running VxWorks real-time operating system. In order for compatibility with Release 5.0, the ISP 1100 must contain at least 1 GB of memory.
MCDN	Meridian Customer Defined Network: private voice networking functionalities/features in a Nortel IP PBX environment
MGC	Media Gateway Controller: provides Digital Signaling

	Processor (DSP) resources for connecting IP and Time Division Multiplexing (TDM) devices together and for advanced applications such as conferences and voicemail access
TLAN	Telephony Local Area Network: network interfacing with AT&T IP Flexible Reach service
TMDI	PRI circuit card to interface with PSTN (if needed)

7 Appendix A: Downloading CS 1000 Patches

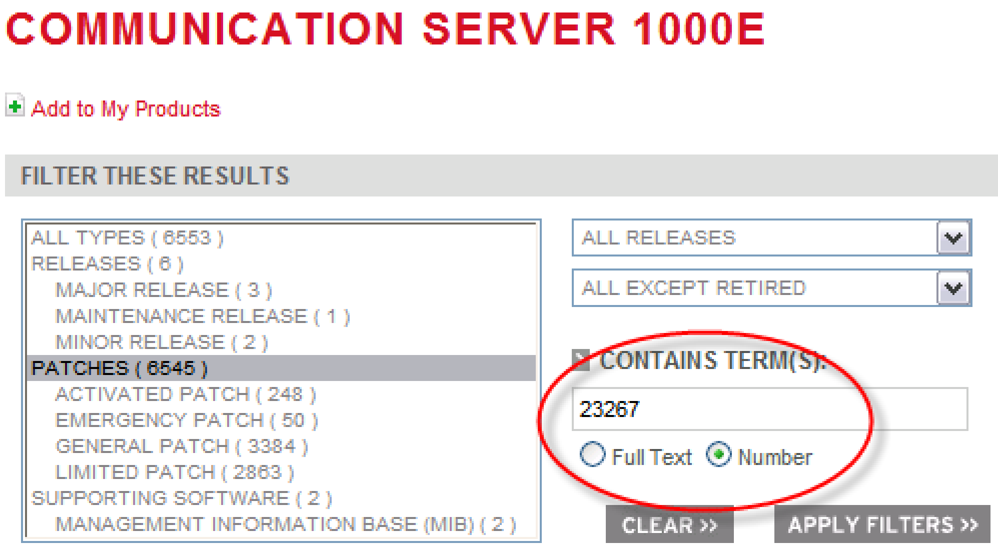
Go to the Nortel website at http://www.nortel.com. Under "Support & Training," select "Software Downloads."



This will take you to the Nortel Technical Support page. Under "Documentation, Software, and Bulletins," select "Voice, Multimedia & Unified Communications."

Under "Communication & Application Services," go to the "Communication Server 1000E" link, which will load the product page. Under the "Software" section, select the "Patches" link.

To search for a specific patch, enter the patch number (i.e. 23267 was entered), and select the "Number" option.



When the search is complete, select the appropriate patch by clicking on the link, which will load the download page.

BE SURE TO DOWNLOAD THE CORRECT PATCH FOR THE APPROPRIATE SOFTWARE LOAD AND HARDWARE TYPE.

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- End of Document -