



## **Avaya Solution & Interoperability Test Lab**

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# **Application notes for AudioCodes Mediant 3000 with Avaya Communication Server 1000 Release 7.0 – Issue 1.0**

### **Abstract**

These Application Notes describe a solution comprised of Avaya Communication Server 1000 Release 7.0 and the AudioCodes Mediant 3000 Version 6.00A.020.002. During the compliance testing, the Mediant 3000 was able to register, as a SIP gateway endpoint, via SIP trunk to the Communication Server 1000. The Mediant 3000 was able to process SIP trunk messages from physical PSTN interfaces with an STM-1, OC3, DS3, T1, or E1 to the Communication Server 1000 and vice versa. The tests of other telephony features such as call transfer, conference, call forward and DTMF relay were executed.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

# 1. Introduction

These application notes provide detail configurations of Avaya Communication Server 1000 Release 7.0 (here after referred to as CS1000) and AudioCodes Mediant 3000 Version 6.00A.020.002 (here after referred to as Mediant 3000) during the compliance testing session. The Mediant 3000 was tested against the non-SIP and SIP clients of the CS1000 Release 7.0. All the applicable telephony feature test cases of Release 7.0 were executed against with the Mediant 3000, where applicable, to ensure the interoperability with CS1000.

## 1.1. Interoperability Compliance Testing

The focus of this compliance testing is to verify that the Mediant 3000 is able to interoperate with the CS1000 Release 7.0. The following interoperability areas were covered:

- Registration of Mediant 3000 to the CS1000 Release 7.0 system via SIP trunk to Network Routing Service (NRS)/SIP Proxy Server (SPS).
- Calls establishment between CS1000 Release 7.0 with Avaya SIP/non-SIP phones and CS1K emulated PSTN on Mediant 3000 side via SIP trunk.
- SIP trunk telephony features: DTMF transmission, voicemail with MWI notification, busy, hold, transfer, conference.
- Codec negotiation.

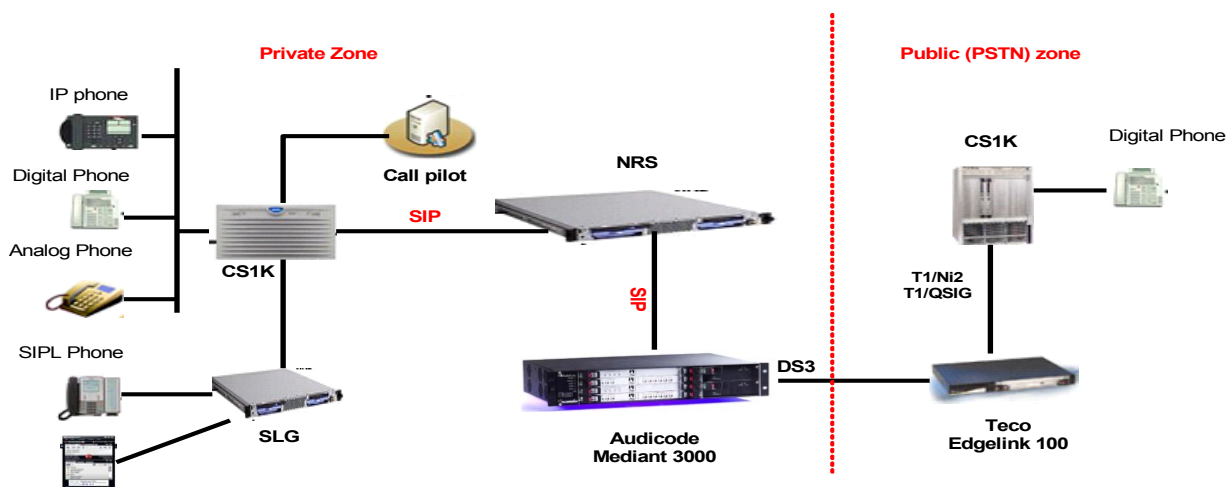
## 1.2. Support

For technical support on AudioCodes Mediant 3000, please contact AudioCodes technical support at:

- Website: <http://www.audiocodes.com>

# 2. Reference Configuration

**Figure 1** illustrates the test configuration used during the compliant testing event between the Avaya CS1000 and the Mediant 3000.



**Figure 1: Lab Diagram**

### 3. Equipment and Software Validated

System	Software/Load Version
Avaya Communication Server 1000	Call Server (CPPM): 7.00Q Signaling Server (CPPM): 7.00.20 SIP Line Gateway (IBM 3550)
Call Pilot	Call Pilot (7.00Q): 05.00.41.20
11xx SIP client (Sigma) SIP soft-phones IP phones	02.02.21.00 SMC3456: v2.6 Build 56941 2004P2:0604DCN,1140:0625C7J
AudioCodes Mediant 3000	6.00A.020.002

### 4. Configure the Avaya CS1000 Release 7.0

This section describes the steps to configure CS1000 Release 7.0 using Element Manager. A command line interface (CLI) option is also available to provision the application on CS1000 Release 7.0 systems. For detailed information, see [2].

#### 4.1. Prerequisite

- CS1000 system has been upgraded to Release 7.0.
- A server which has been
  - o Installed with CS1000 Release 7.0 Linux Base.
  - o Joined CS1000 Release 7.0 Security Domain.
  - o Deployed with Signaling Server Gateway and SIP line Application.
- The following packages are enabled in the key code.

Package Mnemonic	Package Number	Package Description	Package Type (New or Existing or Dependency)	Applicable Market
SIP_LINES	417	SIP Line Service package	New package	Global
FFC	139	Flexible Feature Codes	Existing package	Global
SIPL_NORTEL	415	Nortel SIP Line package	Existing package	--
SIPL_3RDPARTY	416	Third-Party SIP Line Package	Existing package	--

## 4.2. Login to Unified Communications Management (UCM) and Element Manager (EM)

- Use Internet Explorer to launch CS1000 UCM web portal at **http://<IP Address or FQDN>** where **<IP address or FQDN>** is the UCM Framework IP address or FQDN for UCM server.
- Login with the username/password which was defined during the primary security server configuration (not shown). After login, the user will be directed to the Unified Communications Management home page as shown in **Figure 2**.

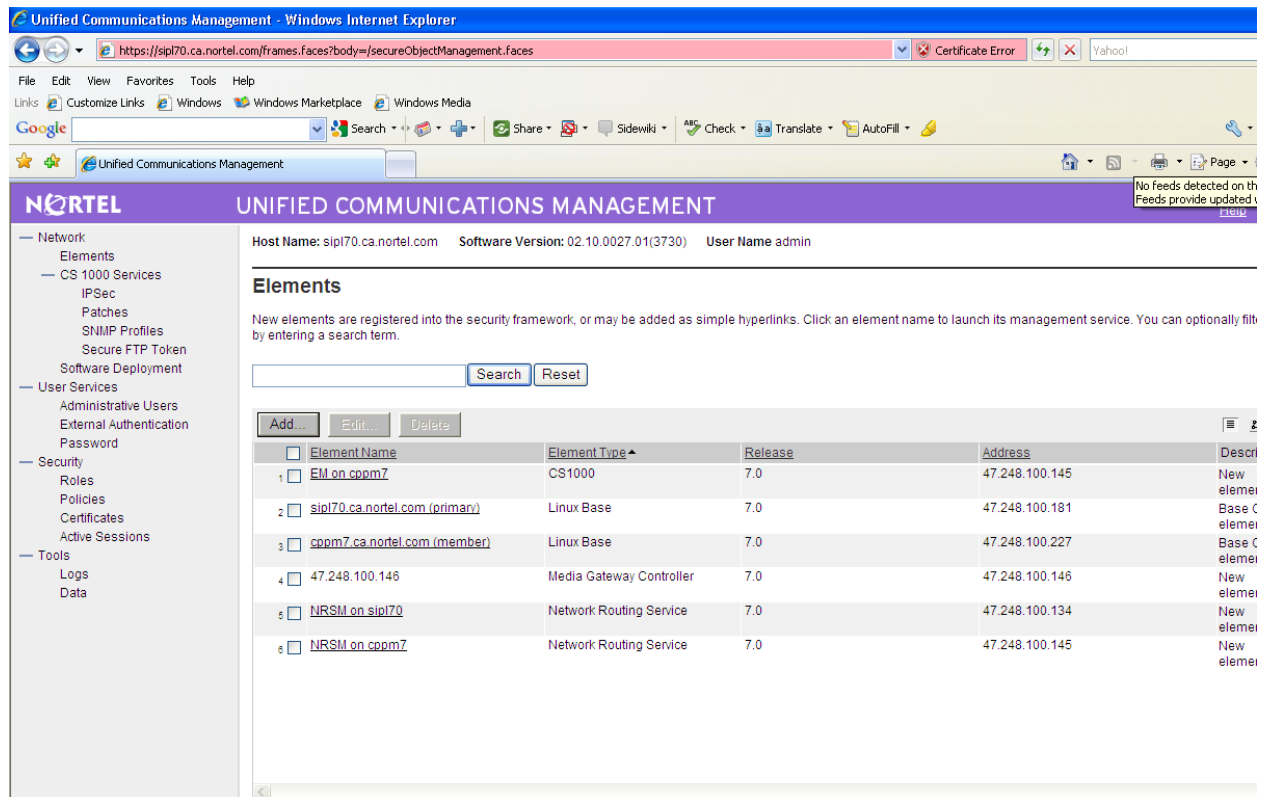
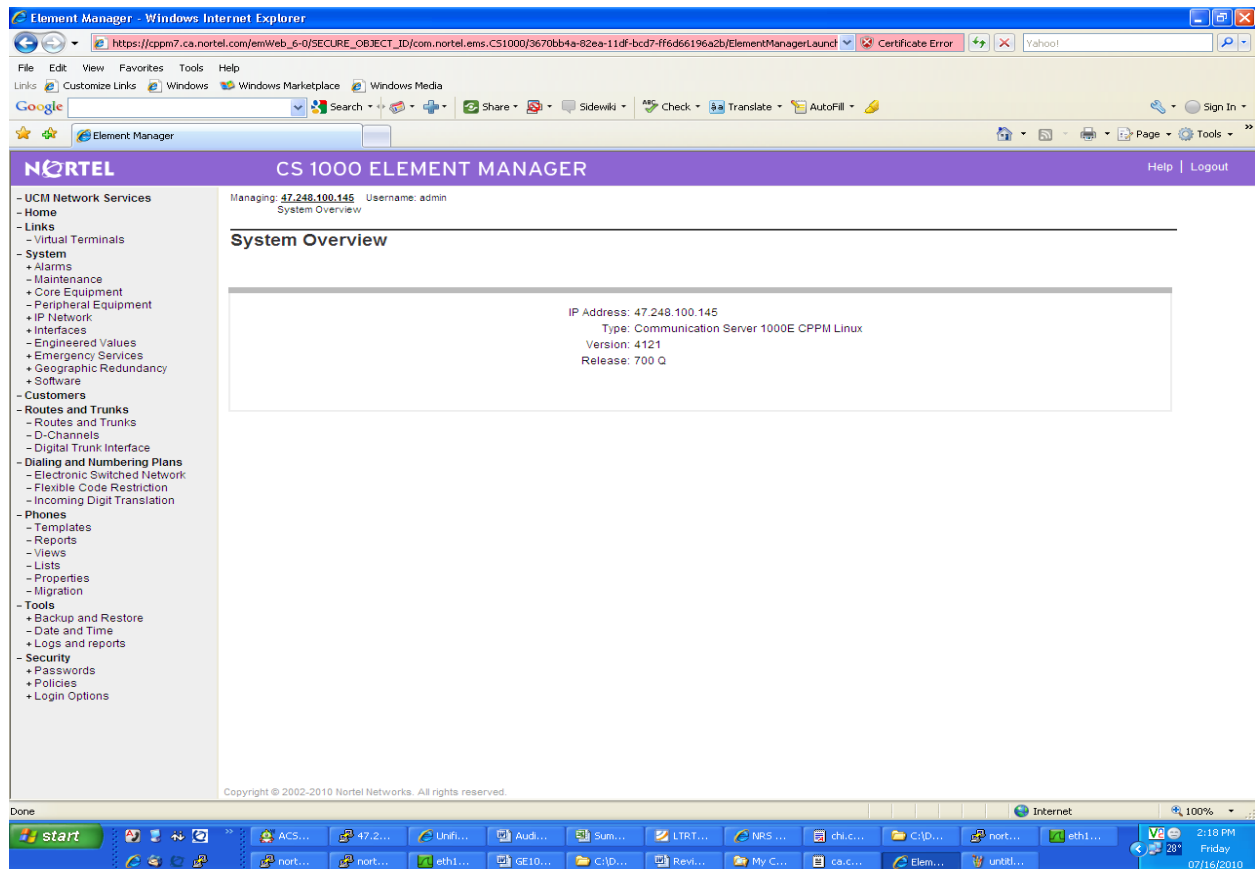


Figure 2: UCM Home Page

- On the Elements page of Unified Communications Management, under the Element Name column, click on the Call Server name, CS1000, to navigate to Element Manager. The CS1000 Element Manager page will appear as shown in **Figure 3** below.

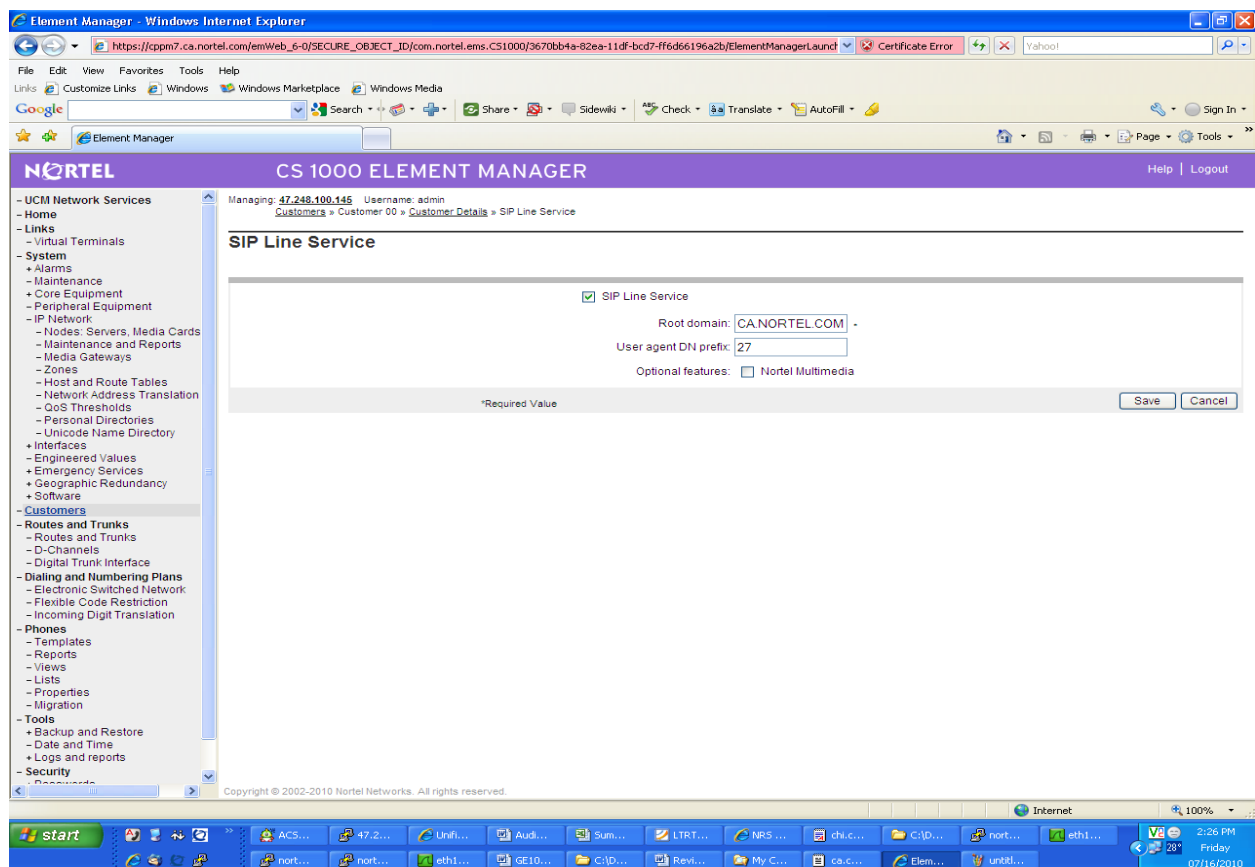


**Figure 3: CS1000 EM Home Page**

### 4.3. Enable Gateway SIPL service and Configure the Root Domain in Customer Data Block (CDB)

The following steps below, as shown in **Figure 4**, describe how to enable SIP line services for a specific customer:

- On the Element Manager (EM) page, on the left menu column, click on **Customers**.
- Select the customer number by clicking the number on the right menu column (not shown).
- Enable SIP Line Service by clicking on **SIP Line Service** link in Customer Details page (not shown) and the **SIP Line Service** check box.
- Enter the SIP Line **Root Domain** name in the **Root Domain** text box.
- Click on **Save** button to complete the set up.



**Figure 4: SIP Line Service in Customers Data Block**

## 4.4. IP Telephony Node Configuration

The following steps below, as shown in **Figure 5**, describe how to create a node on Call Server and its configuration details:

- On the EM page, navigate to **System → IP Network → Nodes: Servers, Media Cards** (not shown).
- Click **Add** on the right menu of this page (not shown) to add a new Signaling server or SIP Line Node to IP Telephony Nodes. To see the Signaling or SIP Line node details, click on the Node ID.
- Enter Node ID in the **Node ID** text box.
- Enter Call Server IP Address in the **Call Server IP Address** text box.
- Enter Node IP Address in the **Node IP Address** text box.
- Enter TLAN Subnet Mask in the **Subnet Mask** text box.
- Enter ELAN Gateway IP Address in the **Gateway IP Address** text box.
- Enter ELAN Subnet Mask in the **Subnet Mask** text box.
- Enable Virtual Trunk Gateway (SIPGw, H323Gw) and SIP Line check box to enable SIPGw and SIP Line.

The screenshot shows the 'New IP Telephony Node' configuration page in the Nortel CS 1000 Element Manager. The page is titled 'New IP Telephony Node' and includes a sidebar with navigation links. The main content area shows configuration fields for Node ID, Call server IP address, TLAN address type, Embedded LAN (ELAN) Gateway IP address and Subnet mask, and Telephony LAN (TLAN) Node IPv4 address and Subnet mask. There are also checkboxes for Applications: SIP Line, UNISIM Line Terminal Proxy Server (LTPS), Virtual Trunk Gateway (SIPGw, H323Gw), Personal Directory (PD), and Presence Publisher. The 'Next' button is visible at the bottom right.

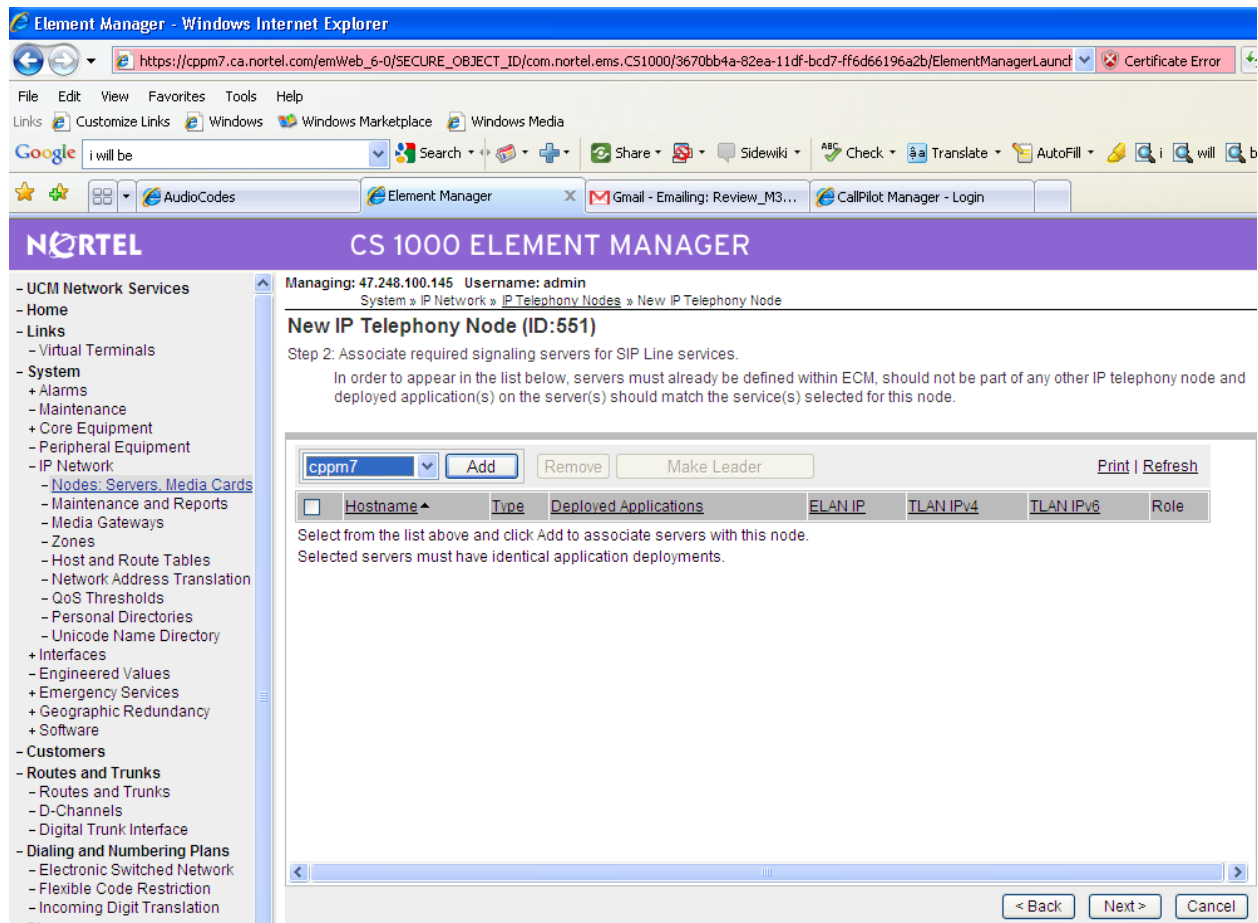
Field	Value
Node ID	551
Call server IP address	47.248.100.145
TLAN address type	IPv4 only
Embedded LAN (ELAN) Gateway IP address	47.248.100.129
Embedded LAN (ELAN) Subnet mask	255.255.255.224
Telephony LAN (TLAN) Node IPv4 address	47.248.100.236
Telephony LAN (TLAN) Subnet mask	255.255.255.224

Applications: ☒ SIP Line, ☐ UNISIM Line Terminal Proxy Server (LTPS), ☒ Virtual Trunk Gateway (SIPGw, H323Gw), ☐ Personal Directory (PD), ☐ Presence Publisher

**Figure 5 – IP Telephony Node**

- Click **Next**. The page to add server to node appears (not shown).

- On Add Server page, from the ***Please Select Server*** list, select the server to add to the node.
- Click ***Add*** (Do not click the Next button).
- Select the check box next to the newly added server, and click ***Make Leader***.
- **Figure 6** shows server “cppm7” selected.



**Figure 6 – IP Telephony Node – Add Server**

- Click ***Next***. The Sip Line gateway Configuration Detail page appears.



- Enter SIP Line domain name in ***SIP Domain name*** text box. This must be the same as the domain name configured in ***Customers, Section 4.3***.
- Click ***Next***.

Element Manager - Windows Internet Explorer

https://cpm7.ca.nortel.com/emWeb\_6-0/SECURE\_OBJECT\_ID/com.nortel.ems.CS1000/3670bb4a-82ea-11df-bcd7-ff6d66196a2b/ElementManagerLaunch Certificate Error

File Edit View Favorites Tools Help

Links Customize Links Windows Windows Marketplace Windows Media

Google i will be Search Share Sidewiki Check Translate AutoFill will b

AudioCodes Element Manager Gmail - Emailing: Review\_M3... CallPilot Manager - Login

**NORTEL** CS 1000 ELEMENT MANAGER

Managing: 47.248.100.145 Username: admin  
System » IP Network » IP Telephony Nodes » New IP Telephony Node

**New IP Telephony Node (ID:551)**  
Step 3: SIP Line Configuration Details.

General | SIP Line Gateway Settings | SIP Line Gateway Service

SIP Line Gateway Application: ☒ Enable gateway service on this node

**General**

SIP domain name:  \*

SLG endpoint name:

SLG Group ID:

SLG Local Sip Port:  (1 - 65535)

SLG Local Tls Port:  (1 - 65535)

**Virtual Trunk Network Health Monitor**

☐ Monitor IP addresses (listed below)  
Information will be captured for the IP addresses listed below.

Monitor IP:  Add

Monitor addresses:  Remove

**SIP Line Gateway Settings**

Security Policy:

Number of byte re-negotiation:

Options: ☐ Client authentication  
☐ v509 Certificate Authentication Enabled

\* Required Value.

< Back Next > Cancel

**Figure 7 – SIP Line Node Details**

- Under the **SIP Line Gateway Services** section, select **MO** from the **SLG Role** list.
- From the **SLG Mode** list, select **S1/S2** (SIP Proxy Server 1 and Server 2).

Managing: 47.248.100.145 Username: admin  
System » IP Network » IP Telephony Nodes » New IP Telephony Node

**New IP Telephony Node (ID:551)**  
Step 3: SIP Line Configuration Details.

General | SIP Line Gateway Settings | SIP Line Gateway Service

SIP Line Gateway Service

Branch / GR Office Settings:

SLG Role: MO  
SLG Mode: S1/S2  
MO SLG IPv4 address: 47.248.100.236  
MO SLG IPv6 address:   
MO SLG Port: 5070 (1 - 65535)  
MO SLG Transport: TCP  
GR SLG IPv4 address: 0.0.0.0  
GR SLG IPv6 address:   
GR SLG Port: 5070 (1 - 65535)  
GR SLG Transport: TCP

IVR Settings:

\* Required Value.

< Back Next > Cancel

**Figure 8 – SIP Line Node Details (cont.)**

- Click **Next**. The **Virtual Trunk Gateway Configuration Details** page appears.
- Enter SIP domain name and Gateway endpoint name.

Managing: 47.248.100.145 Username: admin  
System » IP Network » IP Telephony Nodes » New IP Telephony Node

**New IP Telephony Node (ID:551)**  
Step 4: Virtual Trunk Gateway Configuration Details.

General | SIP Gateway Settings | SIP Gateway Services

Vtrk gateway application: ☒ Enable gateway service on this node

General

Vtrk gateway application: SIP Gateway (SIPGw)  
SIP domain name: ca.nortel.com  
Local SIP port: 5060 (1 - 65535)  
Gateway endpoint name: c ppm7  
Gateway password:   
Application node ID: 551 (0-9999)  
Enable failsafe NRS: ☐  
SIP ANAT: ☐ IPv4 ☐ IPv6

Virtual Trunk Network Health Monitor

☐ Monitor IP addresses (listed below)  
Information will be captured for the IP addresses listed below.  
Monitor IP:   
Add  
Monitor addresses:   
Remove

\* Required Value.

< Back Next > Cancel

**Figure 9 – SIP GW Node Details**

- Enter Primary and Secondary TLAN IP addresses NRS for registration.

**Figure 10 – Proxy or Redirect server**

- Click **Next ->Finish** and wait for the configuration to be saved. The **Node Saved** page then appears as shown in **Figure 11**.

**Figure 11– Transfer Configuration**

- Click **Transfer Now**. The **Synchronize Configuration Files (Node ID 551)** page appears as shown in **Figure 12**.

- Select some or all of the node elements and then click **Start Sync** to transfer the configuration files to the selected servers.

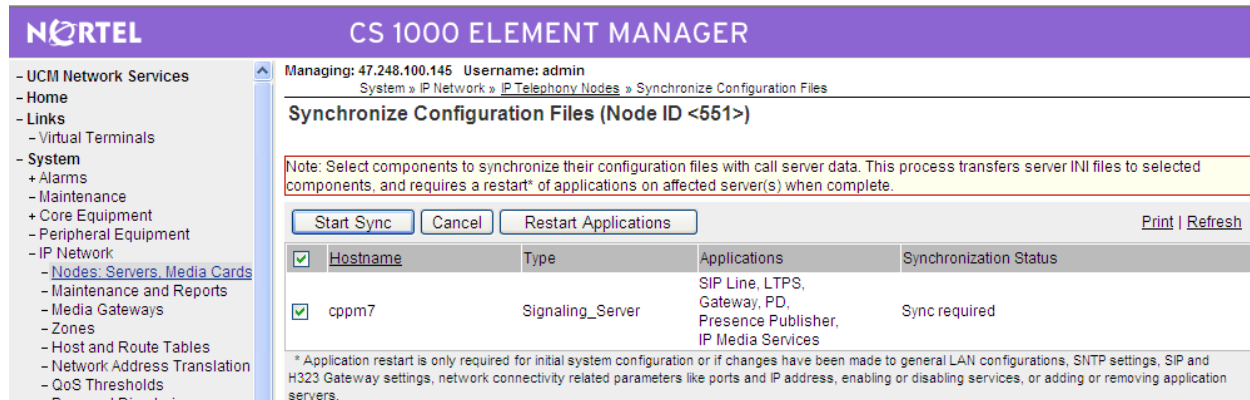


Figure 12 – Synchronize Configuration Files

#### 4.5. D-Channel over IP Configuration

- On the EM page, navigate to **Routes and Trunks** → **D-Channels**.
- Under the **Configuration** section, from the **Choose a D-Channel Number** list, select a D-Channel number (not shown).
- Under the Configuration section, from the **Type** list, select **DCH**. Click **Add** (not shown).
- From the **D channel Card Type (CTYP)** list, select **D-Channels is over IP (DCIP)**. Click **Add** (not shown).

- The **D-Channels xx Property Configuration** page appears, as shown in **Figure 13**.
- From the **Interface type for D-channel (IFC)** list, select **Meridian Meridian1 (SL1)**.
- Click the **Basic options (BSCOPT)** link. The **Basic options (BSCOPT)** list expands.
- Click **Edit** to configure **Remote Capabilities (RCAP)**.

**NORTEL** CS 1000 ELEMENT MANAGER Help | Logout

Managing: 47.248.100.155 Username: admin  
Routes and Trunks > D-Channels > D-Channels 30 Property Configuration

### D-Channels 30 Property Configuration

**- Basic Configuration**

Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D channel Card Type (CTYP)	DCIP
Designator (DES)	SIPLine
Recovery to Primary (RCVP)	<input type="checkbox"/>
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)
Country (CNTY)	ETS 300 =102 basic protocol (ETSI)
D-Channel PRI loop number (DCHL)	
Primary Rate Interface (PRI)	<input type="text"/> <a href="#">more PRI</a>
Secondary PRI2 loops (PRI2)	<input type="text"/>
Meridian 1 node type (SIDE)	Slave to the controller (USR)
Release ID of the switch at the far end (RLS)	5
Central Office switch type (CO_TYPE)	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum (ISLM)	4000 <span style="color: green;">Range: 1 - 4000</span>
Signaling Server Resource Capacity (SSRC)	1800 <span style="color: green;">Range: 0 - 4000</span>
<b>- Basic options (BSCOPT)</b>	
Primary D-channel for a backup DCH (PDCH)	<input type="text"/> <span style="color: green;">Range: 0 - 254</span>
- PINX customer number (PINX_CUST)	<input type="text"/>
- Progress signal (PROG)	<input type="text"/>
- Calling Line Identification (CLID)	<input type="text"/>
- Output request Buffers (OTBF)	32
- D-channel transmission Rate (DRAT)	56 kb/s when LCMT is AMI (56K)
- Channel Negotiation option (CNEG)	No alternative acceptable, exclusive. (1)
- Remote Capabilities (RCAP)	<a href="#">Edit</a>
<b>+ - Change protocol timer value (TIMR)</b>	
- B channel Service messaging. (BSRV)	<input type="checkbox"/>
<b>+ Advanced options (ADVOPT)</b>	
<b>+ Feature Packages</b>	

[Submit](#) [Refresh](#) [Delete](#) [Cancel](#)

**Figure 13 – SIP D-Channel Property Configuration**

- **Figure 14** now appears.
- Select the **Message waiting interworking with DMS-100 (MWI)** check box. This must be enabled to support voice mail notification on SIP endpoints.
- Select the **Network name display method 2 (ND2)** check box. This must be enabled to support name display between SIP endpoints.
- At the bottom of the **Remote Capabilities Configuration** page, click **Return - Remote Capabilities**.
- The **D-Channel xx Property Configuration** page reappears. Click **Submit** button shown at the bottom of the page.

Managing: 47.248.100.155 Username: admin  
Routes and Trunks > D-Channels > D-Channels 30 Property Configuration > - Remote Capabilities Configuration

### - Remote Capabilities Configuration

Input Description	Input Value
Basic rate interface (BRI)	<input type="checkbox"/>
Call completion on busy using integer value (CCBI)	<input type="checkbox"/>
Call completion on busy using object identifier (CCBO)	<input type="checkbox"/>
Call completion on busy for QSIG and EuroSDN BRI (CCBS)	<input type="checkbox"/>
Call completion on no response using integer value (CCNI)	<input type="checkbox"/>
Call completion on no response using object identifier (CCNO)	<input type="checkbox"/>
Call completion to no reply for QSIG and EuroSDN BRI (CCNR)	<input type="checkbox"/>
Network call park (CPK)	<input type="checkbox"/>
Connected line identification presentation (COLP)	<input type="checkbox"/>
Call transfer integer (CTI)	<input type="checkbox"/>
Call transfer object (CTO)	<input type="checkbox"/>
Diversion info. is sent using integer value (DV1I)	<input type="checkbox"/>
Diversion info. is sent using object identifier (DV1O)	<input type="checkbox"/>
Rerouting requests processed using integer value (DV2I)	<input type="checkbox"/>
Rerouting requests processed using object identifier (DV2O)	<input type="checkbox"/>
Diversion info. sent. rerouting requests processed (DV3I)	<input type="checkbox"/>
EuroSDN - div. info sent. rerouting req. processed (DV3O)	<input type="checkbox"/>
Call transfer notification and invocation to EuroSDN (ECTO)	<input type="checkbox"/>
Malicious call identification (MCID)	<input type="checkbox"/>
MCDN QSIG conversion (MQC)	<input type="checkbox"/>
Remote D-channel is on a MSDL card (MSL)	<input type="checkbox"/>
<b>Message waiting interworking with DMS-100 (MWI)</b>	<input checked="" type="checkbox"/>
Network access data (NAC)	<input type="checkbox"/>
Network call trace supported (NCT)	<input type="checkbox"/>
Network name display method 1 (ND1)	<input type="checkbox"/>
<b>Network name display method 2 (ND2)</b>	<input checked="" type="checkbox"/>
Network name display method 3 (ND3)	<input type="checkbox"/>
Name display - integer ID coding (NDI)	<input type="checkbox"/>
Name display - object ID coding (NDO)	<input type="checkbox"/>
Path replacement uses integer values (PRI)	<input type="checkbox"/>
Path replacement uses object identifier (PRO)	<input type="checkbox"/>
Release Link Trunks over IP (RLTI)	<input type="checkbox"/>
Remote virtual queuing (RVQ)	<input type="checkbox"/>
Trunk anti-tromboning operation (TAT)	<input type="checkbox"/>
User to user service 1 (UUS1)	<input type="checkbox"/>
NI-2 name display option. (NDS)	<input type="checkbox"/>
Message waiting indication using integer values (OMWI)	<input type="checkbox"/>
Message waiting indication using object identifier (OMWO)	<input type="checkbox"/>
User to user signalling (UUI)	<input type="checkbox"/>

Return - Remote Capabilities Cancel

**Figure 9 – SIP D-Channel RCAP Configuration Details**



## 4.6. Application Module Link (AML) over Embedded LAN (ELAN) Configuration for SIP Line

- On the EM page, navigate to **System** → **Interfaces** → **Application Module Link**.
- Click **Add** to add an Application Module Link (not shown). **New Application Module Link** page as shown in **Figure 15** appears.
- Enter AML port in the **Port number** text box. The SIP Line Service can use ports 32 to port 127. In this case, SIP Line Service is configured to use port 32.
- Click **Save** to save the configuration.

The screenshot shows the 'New Application Module Link' configuration page in the Nortel CS 1000 Element Manager. The page has a purple header with the Nortel logo and 'CS 1000 ELEMENT MANAGER'. A navigation menu on the left lists various system services, with 'Application Module Link' selected under the 'System' category. The main content area contains the following fields and options:

- Managing:** 47.248.100.155 Username: admin
- Breadcrumb:** System » Interfaces » Application Module Link » New Application Module Link
- Title:** New Application Module Link
- Port number:** 32 (range 16 - 127)
- AML over ELAN:** (checked)
- Description:** SIPLine
- Link control system parameters:** (unchecked)
- Maximum octets:** 512 (per HDLC frame)
- Buttons:** Save, Cancel

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**Figure 15 – Application Module Link Configuration**

## 4.7. Value Added Server (VAS) Configuration for SIP Line

- On the EM page, navigate to *System* → *Interfaces* → *Value Added Server*.
- Click *Add* to add new Value Added Server (not shown). The *Add Value Added Server* page appears.
- Click on the *Ethernet LAN Link*.
- Enter the Ethernet LAN Link number in the *Ethernet LAN Link* text box.
- Ensure that the *Application Security* check box is unchecked.
- Click *Save*.

The screenshot displays the Nortel CS 1000 Element Manager web interface. The top header shows the Nortel logo and 'CS 1000 ELEMENT MANAGER'. A navigation menu on the left lists various system components, with 'Value Added Server' selected under the 'Interfaces' section. The main content area is titled 'Edit Value Added Server 032'. It contains the following configuration fields:

- Ethernet LAN Link:** 032
- ELAN port configured in ADAN:** (checked)
- Application Security:** (unchecked)
- Interval:** 1 (dropdown menu)
- Time interval for checking the link for overload in five second increments:** (label)
- Message Count Threshold:** 9999 (text box) with a multiplier of \* (10 - 9999)

At the bottom right of the configuration area are 'Save' and 'Cancel' buttons. The footer of the page includes the copyright notice: 'Copyright © 2002-2009 Nortel Networks. All rights reserved.'

**Figure 16 – Value Added Service for Application Module Link**



## 4.8. Virtual Trunk Zone Configuration

- On the EM page, navigate to *System → IP Network → Zones*.
- On the *Zones* page, select *Bandwidth Zones* (not shown).
- On the *Bandwidth Zones* page, select a *Bandwidth Zone number* from the list, and click *Add* (not shown).
- On the *Zone Basic Property and Bandwidth Management* page as shown in **Figure 17**, set the zone properties based on bandwidth availability. It is recommended to set the *Zone Strategy* to *Best Quality (BQ)*.
- From the *Zone Intent (ZBRN)* list, select *VTRK (VTRK)*.
- Click *Submit*.

The screenshot shows the Nortel CS 1000 Element Manager interface. The title bar reads 'NORTEL CS 1000 ELEMENT MANAGER' with 'Help | Logout' on the right. The breadcrumb trail is 'System > IP Network > Zones > Bandwidth Zones > Bandwidth Zones 254 > Zone Basic Property and Bandwidth Management'. The main heading is 'Zone Basic Property and Bandwidth Management'. Below this is a table with two columns: 'Input Description' and 'Input Value'. The table contains the following entries:

Input Description	Input Value
Zone Number (ZONE):	254
Intrazone Bandwidth (INTRA_BW):	100000
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	100000
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	VTRK (VTRK)
Description (ZDES):	

At the bottom of the form are buttons for 'Submit', 'Refresh', 'Delete', and 'Cancel'. The footer text reads 'Copyright © 2002-2009 Nortel Networks. All rights reserved.'

Figure 17 – Virtual Trunk Zone Configuration

## 4.9. SIPGW & SIP Line Route Data Block (RDB) Configuration

- On the EM page, navigate to *Routes and Trunks → Routes and Trunks*.
- Click *Add* to select a customer number (not shown).
- On the *Customer xx, New Route Configuration* page, from the *Route number (ROUT)* list, select a route number (not shown).
- From the *Trunk type (TKTP)* list, select *TIE trunk data block (TIE)*.
- When Trunk Type (TKTP) is selected, the following options appear:
  - Trunk type M911P (M911P)
  - The route is for a virtual trunk route (VTRK)
  - Digital trunk route (DTRK)
  - Integrated services digital network option (ISDN)
- From the *Incoming and outgoing trunk (ICOG)* field, select *Incoming and Outgoing (IAO)*.
- In the *Access code for the trunk route (ACOD)* field, enter the access code.
- Select *The route is for virtual trunk route (VTRK)* check box.
- In the *Zone for codec selection and bandwidth management (ZONE)* field, enter the zone number. (Use the same zone as configured in 4.8 “Virtual Trunk Zone Configuration”)
- In the *Node ID of signaling server of this route (NODE)* field, enter the node ID of the SIP Line Gateway.

- From the **Protocol ID for the route (PCID)** list, select **SIP Line (SIPL)**.
- Select the **Integrated services digital network option (ISDN)** check box.
- From the **Mode of operation (MODE)** list, select **Route uses ISDN Signaling Link (ISLD)**.
- In the **D channel number (DCH)** field, enter the D-channel number.
- From the **Interface type for route (IFC)** list, select **Meridian M1 (SL1)**.
- Ensure the **Network calling name allowed (NCNA)** and **Insert ESN Access Code (INAC)** check boxes are selected.
- For the **Basic Route Options, Network Options, General Options, and Advanced Configurations** sections. The default values were used.
- Click **Submit**.

**NORTEL** CS 1000 ELEMENT MANAGER Help | Logout

Managing: 47.240.100.155 Username: admin  
Routes and Trunks » Routes and Trunks » Customer 0, Route 30 Property Configuration

### Customer 0, Route 30 Property Configuration

**- Basic Configuration**

Route data block (RDB) (TYPE)

Customer number (CUST)

Route number (ROUT)

Designator field for trunk (DES)

Trunk type (TKTP)

Incoming and outgoing trunk (ICOG)

Access code for the trunk route (ACOD)

Trunk type M911P (M911P) ☐

The route is for a virtual trunk route (VTRK) ☒

Zone for codec selection and bandwidth management (ZONE)  Range: 0 - 255

Node ID of signaling server of this route (NODE)  Range: 0 - 9999

Protocol ID for the route (PCID)

Integrated services digital network option (ISDN) ☒

Mode of operation (MODE)

D channel number (DCH)  Range: 0 - 254

Interface type for route (IFC)

Private network identifier (PNI)  Range: 0 - 32700

Network calling name allowed (NCNA) ☒

Network call redirection (NCRD) ☐

Trunk route optimization (TRO) ☐

Recognition of DTI2 ABCD FALT signal for ISL (FALT) ☐

Channel type (CHTY)

Call type for outgoing direct dialed TIE route (CTYP)

Insert ESN access code (INAC) ☒

Integrated service access route (ISAR) ☐

Display of access prefix on CLID (DAPC) ☐

Mobile extension route (MBXR) ☐

**+ Basic Route Options**

**+ Network Options**

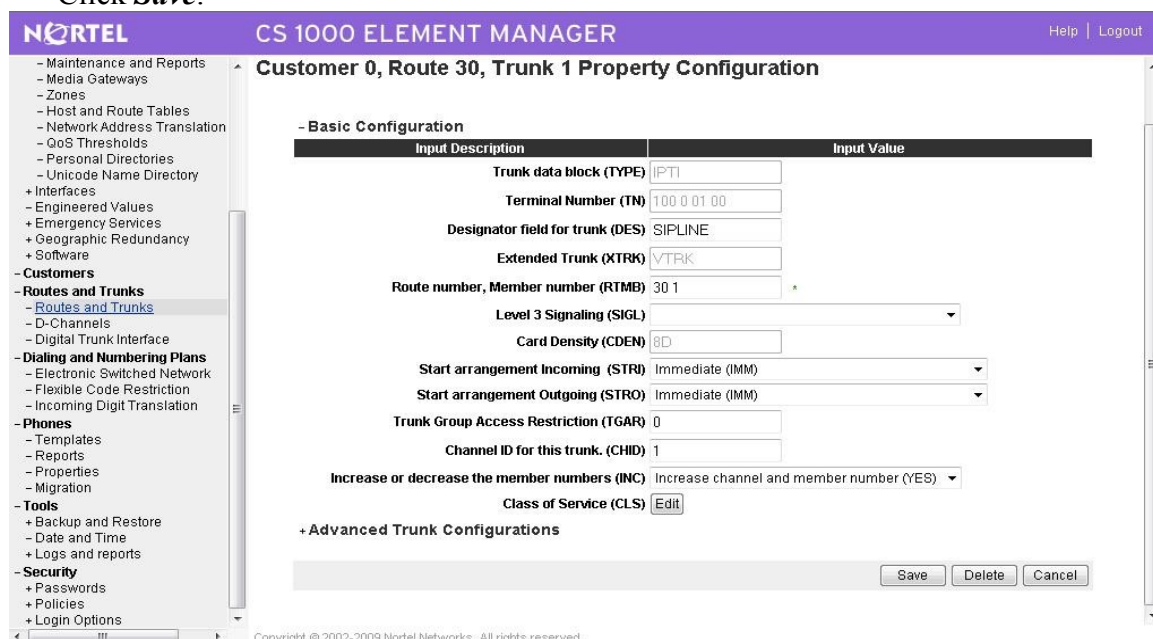
**+ General Options**

**+ Advanced Configurations**

**Figure 18 –SIP & SIP Line Route Configuration**

## 4.10. SIPGW & SIP Line Virtual Trunk Configuration

- On the EM page, navigate to **Routes and Trunks** → **Routes and Trunks**.
- Select the customer for which you are configuring Virtual Trunks.
- Click **Add trunk associated with the route listing** to add new trunk members.
- The **Customer xx, Route yy, New Trunk Configuration** Web page appears.
- Choose **Multiple trunk input number (MTINPUT)** if you are using more than one trunk.
- From the **Trunk data block (TYPE)** list, select **IP Trunk (IPTI)**.
- In the **Terminal Number (TN)** field, enter a TN.
- Enter a **Route number, Member number (RTMB)**.
- Enter a **Trunk Group Access Restriction (TGAR)** value.
- In the **Channel ID for this trunk (CHID)** field, enter a **channel ID** (where the range is 1 to 382).
- To specify a **Class of Service (CLS)** for the trunk, click **Edit**. The **Class of Service Configuration** Web page appears (not shown).
- Select a **Class of Service** (not shown).
- Click **Return Class of Service** to return to the **New Trunk Configuration** Web page (not shown).
- Select **Basic Configuration**. The **Basic Configuration** list expands as shown in **Figure 19**.
- From the **Start arrangement Incoming (STRI)** list, select a value for the start arrangement for incoming calls.
- From the **Start arrangement Outgoing (STRO)** list, select a value for the start arrangement for outgoing calls.
- Select **Advanced Trunk Configurations**. The **Advanced Trunk Configurations** list expands (not shown).
- Configure **Network Class of Service group (NCOS)**.
- Click **Save**.



**NORTEL** CS 1000 ELEMENT MANAGER Help | Logout

Customer 0, Route 30, Trunk 1 Property Configuration

**- Basic Configuration**

Input Description	Input Value
Trunk data block (TYPE)	IPTI
Terminal Number (TN)	100 0 01 00
Designator field for trunk (DES)	SIPLINE
Extended Trunk (XTRK)	VTRK
Route number, Member number (RTMB)	30 1
Level 3 Signaling (SIGL)	
Card Density (CDEN)	8D
Start arrangement Incoming (STRI)	Immediate (IMM)
Start arrangement Outgoing (STRO)	Immediate (IMM)
Trunk Group Access Restriction (TGAR)	0
Channel ID for this trunk (CHID)	1
Increase or decrease the member numbers (INC)	Increase channel and member number (YES)
Class of Service (CLS)	Edit

**+ Advanced Trunk Configurations**

Save Delete Cancel

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Figure 109 –SIPGW & SIP Line Trunk Configuration

## 4.11. Unistim & SIP Line Phones Configuration

Following is a sample configuration for a SIP Line endpoint. Depending on supported features and service access level of the user, this configuration can be adjusted accordingly.

```
>LD 11

REQ: prt
TYPE: tnb
TN 96 0 1 27
DATE
PAGE
DES

DES POLYCO
TN 096 0 01 27 VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL
MCCL YES
SIPN 1 ← Set this to 1 and set SIP3 to 0 if this TN is reserved for Nortel SIP Phones
SIP3 0 ← Set this to 1 and set SIPN to 0 if this TN is reserved for third party SIP Phones
FMCL 0
TLSV 0
SIPU 55573
NDID 551
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID
NHTN
CFG_ZONE 001
CUR_ZONE 001
ERL
ECL 0
FDN 55576 ← If CLS FNA is equipped, call will be forwarded no answer to this number
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 2 ← This field must be set first if call pickup is equipped (CLS PUA)
SCI 0
SSU
XLST
SCPW 1234
SFLT NO
CAC_MFC 0
CLS UNR FBA WTA LPR PUA MTD FNA HTA TDD HFA CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
```

```

POD DSX VMD SLKD CCSD SWD LND CNDA
CFTD SFD MRD DDV CNIA CDCA MSID DAPA BFED RCBF
ICDD CDMD LLCN MCTD CLBD AUTU
GPUA DPUA DNDA CFXA ARHD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCBD RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3 MCBN
FDSD NOVD VOLA VOUD CDMR ICRD MCDD T87D MSNV FRA PKCH
CPND LANG ENG
RCO 0
HUNT 55576 ← If CLS HTA/FBA is equipped, call will be forwarded busy to this
number
LHK 0
PLEV 02
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 55573 0 MARP
      CPND
        CPND LANG ROMAN
          NAME M 55573
          XPLN 13
          DISPLAY_FMT FIRST, LAST
01 HOT U 2655573 MARP 0
02 SCU 0004 ← Speed Call User
03
04 MSB ← This key can be different than key 04 to enable Make Set Busy
(MBS) feature
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16 55574
20 RGA
21 PRK
22 RNP
23
24 PRS

```

25	CHG
26	CPN
27	
28	
29	
30	
31	

#### 4.12. PSTN Trunk Configuration (emulated by CS1000 6.0)

Following is a sample configuration which was used during the compliance test. For more information about PRI Trunk Configuration, see [3].

##### Procedure summary

This procedure is applied for both the CS1000 system under test and CS1000 PSTN simulator.

No.	Overlay	Action
1	LD 17	Adding a PRI card
2	LD 17	Adding a PRI D-Channel
3	LD 15	Defining a PRI customer
4	LD 16	Defining a PRI service route
5	LD 14	Defining service channels and PRI trunks
6	LD 73	Defining system timers and clock controller
7	LD 48	Enable TMDI or PRI MSDI card
8	LD 60	Enable Clock Controller
9	LD 60	Enable Digital trunk loop
10	LD 96	Enable D-channel

### Adding a PRI card

The programming example below shows how to add a PRI card using LD 17. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	CHG	Change data.
TYPE	CFN	Configuration data block.
CEQU	YES	Changes to common equipment.
DLOP	3	Digital Trunk Interface Loop
MG_CARD	12 0 3	MG card assigned to super loop.
MODE	PRI	Mode of operation
TMDI	YES	Card is TMDI card
TRSH	0	Threshold

### Adding a PRI D-channel

The programming example below shows how to add a PRI D-channel using LD 17. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	CHG	Change existing data
TYPE	CFN	Configuration data block.
ADAN	NEW DCH 3	Add a primary D-channel (any unused SDI port.)  xx = 1-9 for Option 11C main cabinet, 11-19 for IP expansion cabinet 1, 21-29 for IP expansion cabinet 2, 31-39 for IP expansion cabinet 3, and 41-49 for IP expansion cabinet 4.  Xx = 11-14, 21-24, 31-34, 41-44 of the first, second, third and fourth Media Gateway, respectively.

CTYP	TMDI	Card type where:  MSDL = The NTB51BA Downloadable D-Channel Daughterboard.  TMDI = TMDI (NTRB21) card.
DES	T1_QSIG	Designator field.
USR	PRI	D-channel is for ISDN PRI only. <b>Note:</b> 2.0 Mb only supports PRI or SHA user
IFC	ISGF	Interface type.
DCHL	3	PRI card number carries the D-channel. Must match entry made for the "CDNO" associated with the "DCHI" prompt above.  Where: xx = 1-9 for Option 11C main cabinet, 11-19 for IP expansion cabinet 1, 21-29 for IP expansion cabinet 2, 31-39 for IP expansion cabinet 3, and 41-49 for IP expansion cabinet 4.  xx = 11-14, 21-24, 31-34, 41-44 of the first, second, third and fourth Media Gateway, respectively.
SIDE	NET	NET = network, the controlling switch (applied for CS1000 PSTN simulator) USR = slave to the controller (applied for CS1000 system under test)
REL.	5	Software rel. of far-end. This is the current software rel. of the far-end. If the far-end has an incompatible rel. of software, it prevents the sending of application messages, for example, 'Network Ring Again.
RCAP	CCBI CCNI PRI DV3I CTI QMWI	Remote Capabilities.
PR_TRIGS	DIV 2 3	Path Replacement Triggers
PR_TRIGS	CNG 2 3	
PR_TRIGS	CON 2 3	
PR_TRIGS	CTR2 2 3	



### Defining a PRI customer

The programming example below shows how to define a PRI customer using LD 15. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CDB	Customer data block.
CUST	0	Customer number.
ISDN	YES	Customer is equipped with ISDN.

### Defining a PRI service route

The programming example below shows how to add a PRI service route using LD 16. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	NEW	Create new data
TYPE	RDB	Route data block
CUST	0	Customer number
ROUT	3	Route number
DES	T1_QSIG	Designator field for trunk
TKTP	TIE	Trunk type
DTRK	YES	Digital trunk route
ISDN	YES	ISDN option
MODE	PRI	Route used for PRI only
PNI	1	Customer private network identifier. Is the same as the CDB PNI at far-end.
IFC	ISGF	Interface type.
ICOG	IAO	Incoming and outgoing
ACOD	8010	Trunk access code

### Defining service channels and PRI trunks

The programming example below shows how to create service channels and PRI trunks using LD 14. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	NEW 23	Create 23 new trunks
TYPE	Tie	Trunk type
TN	3 1	Loop (card) and channel number for digital trunks
PCML	MU	System PCM law.
DES	T1_QSIG	Designator field for trunk
CUST	0	Customer number
RTMB	3 1	Service route number and trunk member number
CLS	UNR DTN	Trunk Class Of Service

### Defining system timers and clock controller parameters

Note: This step is only applied for the CS1000 PSTN simulator system which keeps the clock controller.

The programming example below shows how to define system timers and clock controller parameters using LD 73. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	CHG	Change data.
TYPE	DDB	Digital Data Block
MGCLK	4 0 1	Card slot number for Media Gateway 4 0
PREF		Card number of PRI/DTI/SILC or DTI2/PRI2/SILC containing the primary clock reference.
SREF		Card number of PRI/DTI/SILC or DTI2/PRI2/SILC containing the primary clock reference.

### Enabling T1 QSIG Service

### Enable TMDI card

The example below shows how to enable TMDI card using LD 48.

```
>ld 48
LNK000
.enl tmdi 4 0 1 fdl
OK
```

### Enable Clock Controller

The example below shows how to enable clock controller using LD 60.

```
>ld 60
DTI000
.enl cc 4 0
OK
```

### Enable PRI loop

The example below shows how to enable PRI loop using LD 60.

```
>ld 60
DTI000
.enll 3
OK
.
```

### Enable D-Channel

The D-Channel may not automatically come up. The example below shows how to enable PRI D-channel using LD 96.

```
>ld 96
DCH000
.enl dch 3
.
DCH: 3 EST CONFIRM TIME: 14:31:53 16/07/2010

DCH 3 UIPE_OMSG CC_RESTART_REQ REF 00000000 CH 0 TOD 14:31:53 CK
D7847B49
TYPE: ALL CHANNEL

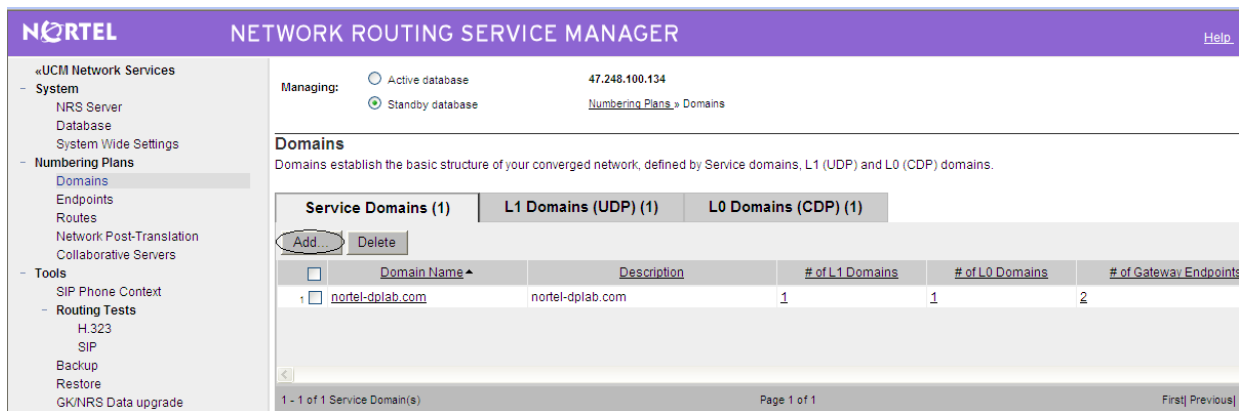
DCH 3 UIPE_IMSG CC_RESTART_CONF REF 00008000 TOD 14:31:53 CK D7847BB3
TYPE: ALL CHANNEL
```

#### 4.13. Define CS1000 and AudioCodes Mediant 3000 on NRS/SPS

This section describes how to create SIP Domain and how to add a SIP gateway endpoint to the domain in NRS. (Do the same in SPS)

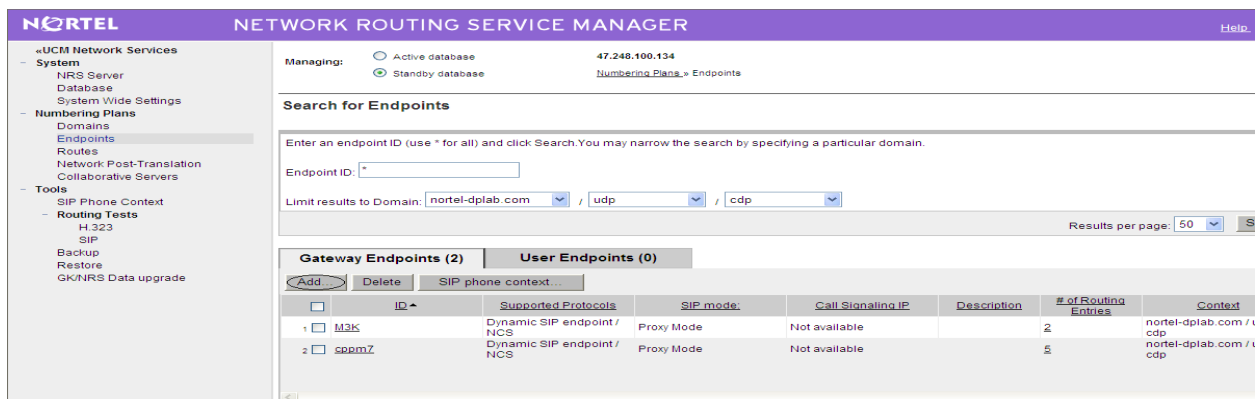
Launch the web GUI of NRS using Internet Explorer by browsing to the TLAN address of NRS. On the left menu column, click on Domains and the domains detail page will appear as shown in **Figure 20**. Choose "Standby database" mode by clicking on radio circle box.

- Choose **Service Domains** tab and click **Add** (**Figure 20** below), input domain name then click **Save** (not shown)
- Choose **L1 Domain (UDP)** tab with appropriate Service Domain in Filter by Domain check box then click **Add**, input L1 Domain name then click **Save** (not shown)
- Choose **L0 Domain (CDP)** tab with appropriate Service Domain and L1 Domain in Filter by Domain check box then click **Add**, input L0 Domain name then click **Save** (not shown)



**Figure 20: Numbering Plans: domains**

Choose "Endpoints" and select appropriate Domain in "limit result to Domain box" then click **Add** (see **Figure 21** below), input End point name, enable SIP support with appropriate transport protocol then save (not shown).



**Figure 21: Numbering Plans: Endpoints**

Go to System-> Database-> Click on 'Cut over' then click on 'Commit' button.

## 5. Configuration for AudioCodes Mediant 3000

The AudioCodes Mediant 3000 was tested with the AudioCodes software load 6.00A.020.002. An example of a Mediant 3000 configuration file used during testing can be found in the Appendix. The configuration of the Mediant 3000 was changed based on the specific requirements of the test cases. For example, the trunk protocol type was changed for specific test cases (T1\_QSIG versus T1\_NI2\_ISDN).

The configuration file below shows an example of a possible Mediant 3000 configuration. This file should not be used to directly configure the Mediant 3000 in the field.

### 5.1. Configure file: BOARD.INI

```
*****
;
** Ini File **
*****

;Board: Mediant 3000
;M3K Board Type: TrunkPack 6310
;Serial Number: 698647
;Slot Number: 1
;Software Version: 6.00A.020.002
;DSP Software Version: 491096AE3 => 600.17
;Board IP Address: 47.248.100.110
;Board Subnet Mask: 255.255.255.224
;Board Default Gateway: 47.248.100.97
;Ram size: 512M Flash size: 32M
;Num of DSP Cores: 126 Num DSP Channels: 2016
;Profile: NONE
;Key features:;Board Type: Mediant 3000;SS7 Links: MTP2=8 MTP3=8 M2UA=8 M3UA=8
;Channel Type: RTP ATM PCI DspCh=2016 ;Security: IPSEC MediaEncryption
StrongEncryption EncryptControlProtocol ;PSTN Protocols: ISDN IUA=63 CAS ;DSP Voice
features: EC128mSec IpmDetector ;IP Media: Conf ;PSTN STM1\SONET Interface
Supported;PSTN T3 Interfaces=3;Coders: G723 G729 G728 NETCODER GSM-FR GSM-EFR
AMR EVRC-QCELP G727 H264 MPEG4 EG711 ;Control Protocols: MGCP MEGACO H323
SIP ;Default features:;Coders: G711 G726;
;-----
```

#### [SYSTEM Params]

```
SyslogServerIP = 47.248.100.57
EnableSyslog = 1
NTPServerIP = 47.248.100.110
ENABLEPARAMETERSMONITORING = 1
ActivityListToLog = 'pvc', 'afl', 'dr', 'fb', 'swu', 'ard', 'naa', 'spc'
TLSVersion = 1
```

[BSP Params]

PCMLawSelect = 3  
TDMBusClockSource = 4  
LocalMediaDefaultGW = 47.248.100.97  
LocalMediaIPAddress = 47.248.100.110  
LocalMediaSubnetMask = 255.255.255.224  
LocalControlIPAddress = 47.248.100.110  
LocalControlSubnetMask = 255.255.255.224  
LocalOAMIPAddress = 47.248.100.110  
LocalOAMSubnetMask = 255.255.255.224  
LocalOAMDefaultGW = 47.248.100.97

[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]

[MEGACO Params]

EP\_Num\_0 = 0  
EP\_Num\_1 = 1  
EP\_Num\_2 = 0  
EP\_Num\_3 = 0  
EP\_Num\_4 = 0

[PSTN Params]

ProtocolType\_0 = 23  
ProtocolType\_1 = 23  
ProtocolType\_2 = 23  
ProtocolType\_3 = 10  
ProtocolType\_4 = 23  
ProtocolType\_5 = 23  
ProtocolType\_6 = 23  
ProtocolType\_7 = 23  
ProtocolType\_8 = 23  
ProtocolType\_9 = 23  
ProtocolType\_10 = 23  
ProtocolType\_11 = 23  
ProtocolType\_12 = 23  
ProtocolType\_13 = 23  
ProtocolType\_14 = 23

ProtocolType\_15 = 23  
ProtocolType\_16 = 23  
ProtocolType\_17 = 23  
ProtocolType\_18 = 23  
ProtocolType\_19 = 23  
ProtocolType\_20 = 23  
ProtocolType\_21 = 23  
ProtocolType\_22 = 23  
ProtocolType\_23 = 23  
ProtocolType\_24 = 23  
ProtocolType\_25 = 23  
ProtocolType\_26 = 23  
ProtocolType\_27 = 23  
ProtocolType\_28 = 23  
ProtocolType\_29 = 23  
ProtocolType\_30 = 23  
ProtocolType\_31 = 23  
ProtocolType\_32 = 23  
ProtocolType\_33 = 23  
ProtocolType\_34 = 23  
ProtocolType\_35 = 23  
ProtocolType\_36 = 23  
ProtocolType\_37 = 23  
ProtocolType\_38 = 23  
ProtocolType\_39 = 23  
ProtocolType\_40 = 23  
ProtocolType\_41 = 23  
ProtocolType\_42 = 23  
ProtocolType\_43 = 23  
ProtocolType\_44 = 23  
ProtocolType\_45 = 23  
ProtocolType\_46 = 23  
ProtocolType\_47 = 23  
ProtocolType\_48 = 23  
ProtocolType\_49 = 23  
ProtocolType\_50 = 23  
ProtocolType\_51 = 23  
ProtocolType\_52 = 23  
ProtocolType\_53 = 23  
ProtocolType\_54 = 23  
ProtocolType\_55 = 23  
ProtocolType\_56 = 23  
ProtocolType\_57 = 23  
ProtocolType\_58 = 23  
ProtocolType\_59 = 23  
ProtocolType\_60 = 23

ProtocolType\_61 = 23  
ProtocolType\_62 = 23  
ProtocolType\_63 = 23  
ProtocolType\_64 = 23  
ProtocolType\_65 = 23  
ProtocolType\_66 = 23  
ProtocolType\_67 = 23  
ProtocolType\_68 = 23  
ProtocolType\_69 = 23  
ProtocolType\_70 = 23  
ProtocolType\_71 = 23  
ProtocolType\_72 = 23  
ProtocolType\_73 = 23  
ProtocolType\_74 = 23  
ProtocolType\_75 = 23  
ProtocolType\_76 = 23  
ProtocolType\_77 = 23  
ProtocolType\_78 = 23  
ProtocolType\_79 = 23  
ProtocolType\_80 = 23  
ProtocolType\_81 = 23  
ProtocolType\_82 = 23  
ProtocolType\_83 = 23  
FramingMethod = D  
TrunkAdministrativeState\_0 = 0  
TrunkAdministrativeState\_1 = 0  
TrunkAdministrativeState\_2 = 2  
TrunkAdministrativeState\_3 = 2  
TrunkAdministrativeState\_4 = 0  
TrunkAdministrativeState\_5 = 0  
TrunkAdministrativeState\_6 = 0  
TrunkAdministrativeState\_7 = 0  
TrunkAdministrativeState\_8 = 0  
TrunkAdministrativeState\_9 = 0  
TrunkAdministrativeState\_10 = 0  
TrunkAdministrativeState\_11 = 0  
TrunkAdministrativeState\_12 = 0  
TrunkAdministrativeState\_13 = 0  
TrunkAdministrativeState\_14 = 0  
TrunkAdministrativeState\_15 = 0  
TrunkAdministrativeState\_16 = 0  
TrunkAdministrativeState\_17 = 0  
TrunkAdministrativeState\_18 = 0  
TrunkAdministrativeState\_19 = 0  
TrunkAdministrativeState\_20 = 0  
TrunkAdministrativeState\_21 = 2



TrunkAdministrativeState\_22 = 2  
TrunkAdministrativeState\_23 = 2  
TrunkAdministrativeState\_24 = 2  
TrunkAdministrativeState\_25 = 2  
TrunkAdministrativeState\_26 = 2  
TrunkAdministrativeState\_27 = 2  
TrunkAdministrativeState\_28 = 2  
TrunkAdministrativeState\_29 = 2  
TrunkAdministrativeState\_30 = 2  
TrunkAdministrativeState\_31 = 2  
TrunkAdministrativeState\_32 = 2  
TrunkAdministrativeState\_33 = 2  
TrunkAdministrativeState\_34 = 2  
TrunkAdministrativeState\_35 = 2  
TrunkAdministrativeState\_36 = 2  
TrunkAdministrativeState\_37 = 2  
TrunkAdministrativeState\_38 = 2  
TrunkAdministrativeState\_39 = 2  
TrunkAdministrativeState\_40 = 2  
TrunkAdministrativeState\_41 = 2  
TrunkAdministrativeState\_42 = 2  
TrunkAdministrativeState\_43 = 2  
TrunkAdministrativeState\_44 = 2  
TrunkAdministrativeState\_45 = 2  
TrunkAdministrativeState\_46 = 2  
TrunkAdministrativeState\_47 = 2  
TrunkAdministrativeState\_48 = 2  
TrunkAdministrativeState\_49 = 2  
TrunkAdministrativeState\_50 = 2  
TrunkAdministrativeState\_51 = 2  
TrunkAdministrativeState\_52 = 2  
TrunkAdministrativeState\_53 = 2  
TrunkAdministrativeState\_54 = 2  
TrunkAdministrativeState\_55 = 2  
TrunkAdministrativeState\_56 = 2  
TrunkAdministrativeState\_57 = 2  
TrunkAdministrativeState\_58 = 2  
TrunkAdministrativeState\_59 = 2  
TrunkAdministrativeState\_60 = 2  
TrunkAdministrativeState\_61 = 2  
TrunkAdministrativeState\_62 = 2  
TrunkAdministrativeState\_63 = 2  
TrunkAdministrativeState\_64 = 2  
TrunkAdministrativeState\_65 = 2  
TrunkAdministrativeState\_66 = 2  
TrunkAdministrativeState\_67 = 2

TrunkAdministrativeState\_68 = 2  
TrunkAdministrativeState\_69 = 2  
TrunkAdministrativeState\_70 = 2  
TrunkAdministrativeState\_71 = 2  
TrunkAdministrativeState\_72 = 2  
TrunkAdministrativeState\_73 = 2  
TrunkAdministrativeState\_74 = 2  
TrunkAdministrativeState\_75 = 2  
TrunkAdministrativeState\_76 = 2  
TrunkAdministrativeState\_77 = 2  
TrunkAdministrativeState\_78 = 2  
TrunkAdministrativeState\_79 = 2  
TrunkAdministrativeState\_80 = 2  
TrunkAdministrativeState\_81 = 2  
TrunkAdministrativeState\_82 = 2  
TrunkAdministrativeState\_83 = 2  
PSTNTransmissionType = 2

#### [SS7 Params]

#### [Voice Engine Params]

CallProgressTonesFilename = 'M2k\_usa\_tones.dat'  
V21ModemTransportType = 2  
DTMFDetectorSensitivity = 1

#### [WEB Params]

LogoWidth = '145'  
HTTPSCipherString = 'ALL'  
ScenarioFileName = 'SCENARIO.dat'

#### [SIP Params]

PLAYRBTONE2IP = 1  
REGISTRATIONTIME = 3600  
ISPROXYUSED = 1  
ISREGISTERNEEDED = 1  
AUTHENTICATIONMODE = 1  
SIPDESTINATIONPORT = 5060  
PLAYRBTONE2TEL = 1  
ROUTEMODEIP2TEL = 1  
ROUTEMODETEL2IP = 1  
RADDEBLEVEL = 2  
CHANNELSELECTMODE = 1

RADLOGOUTPUT = 1  
GWDEBUGLEVEL = 5  
ENABLEPROXYKEEPALIVE = 2  
ENABLEEARLYMEDIA = 1  
SIPSESSIONEXPIRES = 3600  
PROXYNAME = 'nortel-dplab.com'  
SIPGATEWAYNAME = 'nortel-dplab.com'  
USERNAME = 'M3K'  
PASSWORD = '0000'  
ALWAYSSENDTOPROXY = 1  
PROXYREDUNDANCYMODE = 1  
USEGATEWAYNAMEFOROPTIONS = 1  
PREFERROUTETABLE = 1  
ISFAXUSED = 2  
SIPTRANSPORTTYPE = 0  
GWREGISTRATIONNAME = 'M3K'  
REGISTRARNAME = "  
PLAYBUSYTONE2ISDN = 2  
LOCALISDNRBSOURCE\_0 = 0  
LOCALISDNRBSOURCE\_1 = 0  
LOCALISDNRBSOURCE\_2 = 1  
LOCALISDNRBSOURCE\_3 = 1  
LOCALISDNRBSOURCE\_4 = 0  
LOCALISDNRBSOURCE\_5 = 0  
LOCALISDNRBSOURCE\_6 = 0  
LOCALISDNRBSOURCE\_7 = 0  
LOCALISDNRBSOURCE\_8 = 0  
LOCALISDNRBSOURCE\_9 = 0  
LOCALISDNRBSOURCE\_10 = 0  
LOCALISDNRBSOURCE\_11 = 0  
LOCALISDNRBSOURCE\_12 = 0  
LOCALISDNRBSOURCE\_13 = 0  
LOCALISDNRBSOURCE\_14 = 0  
LOCALISDNRBSOURCE\_15 = 0  
LOCALISDNRBSOURCE\_16 = 0  
LOCALISDNRBSOURCE\_17 = 0  
LOCALISDNRBSOURCE\_18 = 0  
LOCALISDNRBSOURCE\_19 = 0  
LOCALISDNRBSOURCE\_20 = 0  
LOCALISDNRBSOURCE\_21 = 0  
LOCALISDNRBSOURCE\_22 = 0  
LOCALISDNRBSOURCE\_23 = 0  
LOCALISDNRBSOURCE\_24 = 0  
LOCALISDNRBSOURCE\_25 = 0  
LOCALISDNRBSOURCE\_26 = 0  
LOCALISDNRBSOURCE\_27 = 0

LOCALISDNRBSOURCE\_28 = 0  
LOCALISDNRBSOURCE\_29 = 0  
LOCALISDNRBSOURCE\_30 = 0  
LOCALISDNRBSOURCE\_31 = 0  
LOCALISDNRBSOURCE\_32 = 0  
LOCALISDNRBSOURCE\_33 = 0  
LOCALISDNRBSOURCE\_34 = 0  
LOCALISDNRBSOURCE\_35 = 0  
LOCALISDNRBSOURCE\_36 = 0  
LOCALISDNRBSOURCE\_37 = 0  
LOCALISDNRBSOURCE\_38 = 0  
LOCALISDNRBSOURCE\_39 = 0  
LOCALISDNRBSOURCE\_40 = 0  
LOCALISDNRBSOURCE\_41 = 0  
LOCALISDNRBSOURCE\_42 = 0  
LOCALISDNRBSOURCE\_43 = 0  
LOCALISDNRBSOURCE\_44 = 0  
LOCALISDNRBSOURCE\_45 = 0  
LOCALISDNRBSOURCE\_46 = 0  
LOCALISDNRBSOURCE\_47 = 0  
LOCALISDNRBSOURCE\_48 = 0  
LOCALISDNRBSOURCE\_49 = 0  
LOCALISDNRBSOURCE\_50 = 0  
LOCALISDNRBSOURCE\_51 = 0  
LOCALISDNRBSOURCE\_52 = 0  
LOCALISDNRBSOURCE\_53 = 0  
LOCALISDNRBSOURCE\_54 = 0  
LOCALISDNRBSOURCE\_55 = 0  
LOCALISDNRBSOURCE\_56 = 0  
LOCALISDNRBSOURCE\_57 = 0  
LOCALISDNRBSOURCE\_58 = 0  
LOCALISDNRBSOURCE\_59 = 0  
LOCALISDNRBSOURCE\_60 = 0  
LOCALISDNRBSOURCE\_61 = 0  
LOCALISDNRBSOURCE\_62 = 0  
LOCALISDNRBSOURCE\_63 = 0  
LOCALISDNRBSOURCE\_64 = 0  
LOCALISDNRBSOURCE\_65 = 0  
LOCALISDNRBSOURCE\_66 = 0  
LOCALISDNRBSOURCE\_67 = 0  
LOCALISDNRBSOURCE\_68 = 0  
LOCALISDNRBSOURCE\_69 = 0  
LOCALISDNRBSOURCE\_70 = 0  
LOCALISDNRBSOURCE\_71 = 0  
LOCALISDNRBSOURCE\_72 = 0  
LOCALISDNRBSOURCE\_73 = 0

LOCALISDNRBSOURCE\_74 = 0  
LOCALISDNRBSOURCE\_75 = 0  
LOCALISDNRBSOURCE\_76 = 0  
LOCALISDNRBSOURCE\_77 = 0  
LOCALISDNRBSOURCE\_78 = 0  
LOCALISDNRBSOURCE\_79 = 0  
LOCALISDNRBSOURCE\_80 = 0  
LOCALISDNRBSOURCE\_81 = 0  
LOCALISDNRBSOURCE\_82 = 0  
LOCALISDNRBSOURCE\_83 = 0  
PLAYRBTONE2TRUNK\_0 = -1  
PLAYRBTONE2TRUNK\_1 = -1  
PLAYRBTONE2TRUNK\_2 = 1  
PLAYRBTONE2TRUNK\_3 = 1  
PLAYRBTONE2TRUNK\_4 = -1  
PLAYRBTONE2TRUNK\_5 = -1  
PLAYRBTONE2TRUNK\_6 = -1  
PLAYRBTONE2TRUNK\_7 = -1  
PLAYRBTONE2TRUNK\_8 = -1  
PLAYRBTONE2TRUNK\_9 = -1  
PLAYRBTONE2TRUNK\_10 = -1  
PLAYRBTONE2TRUNK\_11 = -1  
PLAYRBTONE2TRUNK\_12 = -1  
PLAYRBTONE2TRUNK\_13 = -1  
PLAYRBTONE2TRUNK\_14 = -1  
PLAYRBTONE2TRUNK\_15 = -1  
PLAYRBTONE2TRUNK\_16 = -1  
PLAYRBTONE2TRUNK\_17 = -1  
PLAYRBTONE2TRUNK\_18 = -1  
PLAYRBTONE2TRUNK\_19 = -1  
PLAYRBTONE2TRUNK\_20 = -1  
PLAYRBTONE2TRUNK\_21 = -1  
PLAYRBTONE2TRUNK\_22 = -1  
PLAYRBTONE2TRUNK\_23 = -1  
PLAYRBTONE2TRUNK\_24 = -1  
PLAYRBTONE2TRUNK\_25 = -1  
PLAYRBTONE2TRUNK\_26 = -1  
PLAYRBTONE2TRUNK\_27 = -1  
PLAYRBTONE2TRUNK\_28 = -1  
PLAYRBTONE2TRUNK\_29 = -1  
PLAYRBTONE2TRUNK\_30 = -1  
PLAYRBTONE2TRUNK\_31 = -1  
PLAYRBTONE2TRUNK\_32 = -1  
PLAYRBTONE2TRUNK\_33 = -1  
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PLAYRBTONE2TRUNK\_36 = -1  
PLAYRBTONE2TRUNK\_37 = -1  
PLAYRBTONE2TRUNK\_38 = -1  
PLAYRBTONE2TRUNK\_39 = -1  
PLAYRBTONE2TRUNK\_40 = -1  
PLAYRBTONE2TRUNK\_41 = -1  
PLAYRBTONE2TRUNK\_42 = -1  
PLAYRBTONE2TRUNK\_43 = -1  
PLAYRBTONE2TRUNK\_44 = -1  
PLAYRBTONE2TRUNK\_45 = -1  
PLAYRBTONE2TRUNK\_46 = -1  
PLAYRBTONE2TRUNK\_47 = -1  
PLAYRBTONE2TRUNK\_48 = -1  
PLAYRBTONE2TRUNK\_49 = -1  
PLAYRBTONE2TRUNK\_50 = -1  
PLAYRBTONE2TRUNK\_51 = -1  
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PLAYRBTONE2TRUNK\_61 = -1  
PLAYRBTONE2TRUNK\_62 = -1  
PLAYRBTONE2TRUNK\_63 = -1  
PLAYRBTONE2TRUNK\_64 = -1  
PLAYRBTONE2TRUNK\_65 = -1  
PLAYRBTONE2TRUNK\_66 = -1  
PLAYRBTONE2TRUNK\_67 = -1  
PLAYRBTONE2TRUNK\_68 = -1  
PLAYRBTONE2TRUNK\_69 = -1  
PLAYRBTONE2TRUNK\_70 = -1  
PLAYRBTONE2TRUNK\_71 = -1  
PLAYRBTONE2TRUNK\_72 = -1  
PLAYRBTONE2TRUNK\_73 = -1  
PLAYRBTONE2TRUNK\_74 = -1  
PLAYRBTONE2TRUNK\_75 = -1  
PLAYRBTONE2TRUNK\_76 = -1  
PLAYRBTONE2TRUNK\_77 = -1  
PLAYRBTONE2TRUNK\_78 = -1  
PLAYRBTONE2TRUNK\_79 = -1  
PLAYRBTONE2TRUNK\_80 = -1  
PLAYRBTONE2TRUNK\_81 = -1

PLAYRBTONE2TRUNK\_82 = -1  
PLAYRBTONE2TRUNK\_83 = -1  
ENABLEHISTORYINFO = 1  
ADDPHONECONTEXTASPREFIX = 1  
REMOVECALLINGNAME = 1  
FAXCNGMODE = 1  
SIPREROUTINGMODE = 1  
SOURCEIPADDRESSINPUT = 0

[SCTP Params]

[VXML Params]

[IPsec Params]

[Audio Staging Params]

[SNMP Params]

SNMPManagerIsUsed\_0 = 1  
SNMPManagerIsUsed\_1 = 0  
SNMPManagerIsUsed\_2 = 0  
SNMPManagerIsUsed\_3 = 0  
SNMPManagerIsUsed\_4 = 0  
SNMPManagerTableIP\_0 = 47.248.100.42  
SNMPManagerTableIP\_1 = 0.0.0.0  
SNMPManagerTableIP\_2 = 0.0.0.0  
SNMPManagerTableIP\_3 = 0.0.0.0  
SNMPManagerTableIP\_4 = 0.0.0.0  
SNMPTrapManagerHostName = '47.248.100.57'

;  
.  
.  
\*\*\* TABLE DS3CONFIG \*\*\*  
.  
.  
.

[ DS3CONFIG ]

FORMAT DS3CONFIG\_Index = DS3CONFIG\_FramingMethod, DS3CONFIG\_ClockSource,  
DS3CONFIG\_LineBuildOut, DS3CONFIG\_CircuitIdentifier, DS3CONFIG\_TrapEnable,  
DS3CONFIG\_PmOnOff, DS3CONFIG\_TappingEnable, DS3CONFIG\_AdminState;  
DS3CONFIG 0 = 1, 0, 4, , 1, 1, 0, 1;  
DS3CONFIG 1 = 1, 0, 4, , 1, 1, 0, 1;

DS3CONFIG 2 = 1, 1, 4, , 1, 1, 0, 1;

[ \DS3CONFIG ]

```
;
;
; *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts
;
```

```
;
;
; *** TABLE PREFIX ***
;
;
;
```

[ PREFIX ]

FORMAT PREFIX\_Index = PREFIX\_DestinationPrefix, PREFIX\_DestAddress,  
PREFIX\_SourcePrefix, PREFIX\_ProfileId, PREFIX\_MeteringCode, PREFIX\_DestPort,  
PREFIX\_SrcIPGroupID, PREFIX\_DestHostPrefix, PREFIX\_DestIPGroupID,  
PREFIX\_SrcHostPrefix, PREFIX\_TransportType, PREFIX\_SrcTrunkGroupID;  
PREFIX 0 = \*, nortel-dplab.com, \*, 0, 255, 0, -1, , -1, , -1, -1;

[ \PREFIX ]

```
;
;
; *** TABLE CoderName ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts
;
```

```
;
;
; *** TABLE TrunkGroup ***
;
;
;
```

[ TrunkGroup ]

FORMAT TrunkGroup\_Index = TrunkGroup\_TrunkGroupNum, TrunkGroup\_FirstTrunkId,  
TrunkGroup\_FirstBChannel, TrunkGroup\_LastBChannel, TrunkGroup\_FirstPhoneNumber,  
TrunkGroup\_ProfileId, TrunkGroup\_LastTrunkId, TrunkGroup\_Module;  
TrunkGroup 0 = 1, 0, 1, 23, , 0, 0, 255;  
TrunkGroup 1 = 3, 2, 1, 23, , 0, 2, 255;  
TrunkGroup 2 = 4, 3, 1, 23, , 0, 3, 255;

[ \TrunkGroup ]

```
;
```



```
; *** TABLE NumberMapIp2Tel ***  
;  
;  
;
```

```
[ NumberMapIp2Tel ]  
FORMAT NumberMapIp2Tel_Index = NumberMapIp2Tel_DestinationPrefix,  
NumberMapIp2Tel_SourcePrefix, NumberMapIp2Tel_SourceAddress,  
NumberMapIp2Tel_NumberType, NumberMapIp2Tel_NumberPlan,  
NumberMapIp2Tel_RemoveFromLeft, NumberMapIp2Tel_RemoveFromRight,  
NumberMapIp2Tel_LeaveFromRight, NumberMapIp2Tel_Prefix2Add,  
NumberMapIp2Tel_Suffix2Add, NumberMapIp2Tel_IsPresentationRestricted,  
NumberMapIp2Tel_SrcTrunkGroupID, NumberMapIp2Tel_SrcIPGroupID;  
NumberMapIp2Tel 0 = 224, *, *, 4, 1, 0, 0, 7, , , 255, -1, -1;  
NumberMapIp2Tel 1 = 41, *, *, 4, 9, 0, 0, 7, , , 255, -1, -1;  
NumberMapIp2Tel 2 = 43, *, *, 4, 9, 0, 0, 7, , , 255, -1, -1;  
NumberMapIp2Tel 3 = *, *, *, 4, 9, 0, 0, 7, , , 255, -1, -1;
```

```
[ \NumberMapIp2Tel ]
```

```
;  
;  
; *** TABLE NumberMapTel2Ip ***  
;  
;  
;
```

```
[ NumberMapTel2Ip ]  
; ** NOTE: Changes were made to active configuration.  
; ** The data below is different from current values.  
FORMAT NumberMapTel2Ip_Index = NumberMapTel2Ip_DestinationPrefix,  
NumberMapTel2Ip_SourcePrefix, NumberMapTel2Ip_SourceAddress,  
NumberMapTel2Ip_NumberType, NumberMapTel2Ip_NumberPlan,  
NumberMapTel2Ip_RemoveFromLeft, NumberMapTel2Ip_RemoveFromRight,  
NumberMapTel2Ip_LeaveFromRight, NumberMapTel2Ip_Prefix2Add,  
NumberMapTel2Ip_Suffix2Add, NumberMapTel2Ip_IsPresentationRestricted,  
NumberMapTel2Ip_SrcTrunkGroupID, NumberMapTel2Ip_SrcIPGroupID;  
NumberMapTel2Ip 0 = 210, *, *, 255, 255, 0, 0, 7, , 4, 255, -1, -1;  
NumberMapTel2Ip 1 = 40, *, *, 255, 255, 0, 0, 7, , , 255, -1, -1;  
NumberMapTel2Ip 2 = 42, *, *, 255, 255, 0, 0, 7, , , 255, -1, -1;  
NumberMapTel2Ip 46 = 46, *, *, 255, 255, 0, 0, 7, , , 255, -1, -1;  
NumberMapTel2Ip 47 = *, *, *, 255, 255, 0, 0, 7, , , 255, -1, -1;
```

```
[ \NumberMapTel2Ip ]
```

```
;  
;  
; *** TABLE SourceNumberMapIp2Tel ***  
;  
;  
;
```

```
[ SourceNumberMapIp2Tel ]
FORMAT SourceNumberMapIp2Tel_Index = SourceNumberMapIp2Tel_DestinationPrefix,
SourceNumberMapIp2Tel_SourcePrefix, SourceNumberMapIp2Tel_SourceAddress,
SourceNumberMapIp2Tel_NumberType, SourceNumberMapIp2Tel_NumberPlan,
SourceNumberMapIp2Tel_RemoveFromLeft, SourceNumberMapIp2Tel_RemoveFromRight,
SourceNumberMapIp2Tel_LeaveFromRight, SourceNumberMapIp2Tel_Prefix2Add,
SourceNumberMapIp2Tel_Suffix2Add, SourceNumberMapIp2Tel_IsPresentationRestricted,
SourceNumberMapIp2Tel_SrcTrunkGroupID, SourceNumberMapIp2Tel_SrcIPGroupID;
SourceNumberMapIp2Tel 0 = *, *, *, 4, 9, 0, 0, 7, , , 0, -1, -1;
SourceNumberMapIp2Tel 1 = *, *, *, 4, 1, 0, 0, 7, , , 0, -1, -1;
SourceNumberMapIp2Tel 2 = *, *, *, 2, 1, 0, 0, 7, , , 0, -1, -1;
SourceNumberMapIp2Tel 3 = *, *, *, 1, 1, 0, 0, 7, , , 0, -1, -1;
SourceNumberMapIp2Tel 4 = *, *, *, 2, 9, 0, 0, 255, , , 0, -1, -1;
SourceNumberMapIp2Tel 5 = *, *, *, 3, 1, 0, 0, 7, , , 0, -1, -1;
```

```
[ \SourceNumberMapIp2Tel ]
;
; *** TABLE SourceNumberMapTel2Ip ***
;
;
;
```

```
[ SourceNumberMapTel2Ip ]
FORMAT SourceNumberMapTel2Ip_Index = SourceNumberMapTel2Ip_DestinationPrefix,
SourceNumberMapTel2Ip_SourcePrefix, SourceNumberMapTel2Ip_SourceAddress,
SourceNumberMapTel2Ip_NumberType, SourceNumberMapTel2Ip_NumberPlan,
SourceNumberMapTel2Ip_RemoveFromLeft, SourceNumberMapTel2Ip_RemoveFromRight,
SourceNumberMapTel2Ip_LeaveFromRight, SourceNumberMapTel2Ip_Prefix2Add,
SourceNumberMapTel2Ip_Suffix2Add, SourceNumberMapTel2Ip_IsPresentationRestricted,
SourceNumberMapTel2Ip_SrcTrunkGroupID, SourceNumberMapTel2Ip_SrcIPGroupID;
SourceNumberMapTel2Ip 1 = *, *, *, 255, 255, 0, 0, 7, , , 255, -1, -1;
```

```
[ \SourceNumberMapTel2Ip ]
;
;
; *** TABLE PstnPrefix ***
;
;
;
```

```
[ PstnPrefix ]
; ** NOTE: Changes were made to active configuration.
; ** The data below is different from current values.
FORMAT PstnPrefix_Index = PstnPrefix_DestPrefix, PstnPrefix_TrunkGroupId,
PstnPrefix_SourcePrefix, PstnPrefix_SourceAddress, PstnPrefix_ProfileId,
PstnPrefix_SrcIPGroupID, PstnPrefix_DestHostPrefix, PstnPrefix_SrcHostPrefix;
PstnPrefix 0 = 41, 3, *, *, 0, -1, *, *;
PstnPrefix 1 = 43, 4, *, *, 0, -1, *, *;
```

[ \PstnPrefix ]

```
.  
;  
; *** TABLE Dns2Ip ***  
;  
;  
;
```

[ Dns2Ip ]  
FORMAT Dns2Ip\_Index = Dns2Ip\_DomainName, Dns2Ip\_FirstIpAddress,  
Dns2Ip\_SecondIpAddress, Dns2Ip\_ThirdIpAddress, Dns2Ip\_FourthIpAddress;  
Dns2Ip 0 = nortel-dplab.com, 47.248.100.181, 0.0.0.0, 0.0.0.0, 0.0.0.0;  
Dns2Ip 1 = nortel-dplab.com, 47.248.100.227, 0.0.0.0, 0.0.0.0, 0.0.0.0;

[ \Dns2Ip ]

```
.  
;  
; *** TABLE ProxyIp ***  
;  
;  
;
```

[ ProxyIp ]  
FORMAT ProxyIp\_Index = ProxyIp\_IpAddress, ProxyIp\_TransportType, ProxyIp\_ProxySetId;  
ProxyIp 0 = 47.248.100.181, 0, 0;  
ProxyIp 1 = 47.248.100.227, 0, 0;

[ \ProxyIp ]

```
.  
;  
; *** TABLE TxDtmfOption ***  
;  
;  
;
```

[ TxDtmfOption ]  
FORMAT TxDtmfOption\_Index = TxDtmfOption\_Type;  
TxDtmfOption 0 = 4;

[ \TxDtmfOption ]

```
.  
;  
; *** TABLE TrunkGroupSettings ***  
;  
;  
;
```

[ TrunkGroupSettings ]

```

FORMAT TrunkGroupSettings_Index = TrunkGroupSettings_TrunkGroupId,
TrunkGroupSettings_ChannelSelectMode, TrunkGroupSettings_RegistrationMode,
TrunkGroupSettings_GatewayName, TrunkGroupSettings_ContactUser,
TrunkGroupSettings_ServingIPGroup, TrunkGroupSettings_MWIInterrogationType;
TrunkGroupSettings 0 = 1, 1, 255, , , -1, 255;

```

```

[ \TrunkGroupSettings ]

```

```

;
;
; *** TABLE PhoneContext ***
;
;
;

```

```

[ PhoneContext ]
; ** NOTE: Changes were made to active configuration.
; ** The data below is different from current values.
FORMAT PhoneContext_Index = PhoneContext_Npi, PhoneContext_Ton,
PhoneContext_Context;
PhoneContext 0 = 0, 0, udp;
PhoneContext 1 = 9, 2, udp;
PhoneContext 2 = 9, 4, cdp.udp;

```

```

[ \PhoneContext ]

```

```

;
;
; *** TABLE ProxySet ***
;
;
;

```

```

[ ProxySet ]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive,
ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod,
ProxySet_IsProxyHotSwap, ProxySet_SRD, ProxySet_ClassificationInput;
ProxySet 0 = 2, 15, 0, 1, 0, 0;

```

```

[ \ProxySet ]

```

```

;
;
; *** TABLE CodersGroup0 ***
;
;
;

```

```

[ CodersGroup0 ]
FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce;
CodersGroup0 0 = g711Alaw64k, 10, 0, -1, 0;

```

CodersGroup0 1 = g711Alaw64k, 20, 0, -1, 0;  
 CodersGroup0 2 = g711Alaw64k, 30, 0, -1, 0;  
 CodersGroup0 3 = g711Ul原因64k, 10, 0, -1, 0;  
 CodersGroup0 4 = g711Ul原因64k, 20, 0, -1, 0;  
 CodersGroup0 5 = g711Ul原因64k, 30, 0, -1, 0;  
 CodersGroup0 6 = g729, 10, 0, -1, 0;  
 CodersGroup0 7 = g729, 20, 0, -1, 0;  
 CodersGroup0 8 = g729, 30, 0, -1, 0;  
 CodersGroup0 9 = g7231, 30, 0, 4, 0;

[ \CodersGroup0 ]

## 5.2. Screen shots

- Configuring Proxy and Registration Parameters: The 'Proxy & Registration' page allows you to configure parameters that are associated with Proxy and Registration.
- Gateway name in 'Proxy & Registration' page should match with domain name in NRS. In this case Gateway name on Mediant 3000 is: nortel-dplab.com (see **Figure 20**).
- User name and Password in 'Proxy & Registration page' should match with Gateway Endpoint name and Authentication password defined on NRS (see **Figure 21**).

For example in this case:

On Mediant 3000: username: M3K, password: 0000

On NRS: Gateway Endpoint name: M3K, Authentication password: 0000

Basic Parameter List	
Use Default Proxy	Yes
Proxy Set Table	
Proxy Name	nortel-dplab.com
Redundancy Mode	Homing
Proxy IP List Refresh Time	60
Enable fallback to Routing Table	Disable
Prefer Routing Table	Yes
Use Routing Table for Host Names and Profiles	Disable
Always Use Proxy	Enable
Redundant Routing Mode	Routing Table
SIP ReRouting Mode	Send to Proxy
Enable Registration	Enable
Registrar Name	
Registrar IP Address	
Registrar Transport Type	Not Configured
Registration Time	3600
Re-registration Timing [%]	50
Registration Retry Time	30
Registration Time Threshold	0
Re-register On INVITE Failure	Disable
ReRegister On Connection Failure	Disable

**Figure 22: Proxy & Registration**

- Configuring the Proxy Sets table: the 'Proxy Sets Table' page as shown in **Figure 23** allows you to define Proxy Sets. A Proxy Set is a group of Proxy servers defined by IP address or fully qualified domain name (FQDN). You can define up to ten Proxy Sets. For each Proxy server address you can define the transport type.

The screenshot shows the 'Proxy Sets Table' configuration page. The left sidebar contains a tree view with categories like PSTN Settings, SS7 Configuration, Sigtran Configuration, Security Settings, Protocol Configuration, Media Realm Configuration, Trunk Group, Protocol Definition, SIP General Parameters, DTMF & Dialing, Application Network Setting, and Proxies, Registration, IP Groups. The 'Proxy Sets Table' is selected under the 'Proxies, Registration, IP Groups' category.

The main content area has a 'Proxy Set ID' dropdown set to 0. Below it is a table with 5 rows for defining proxy sets:

	Proxy Address	Transport Type
1	47.248.100.181	UDP
2	47.248.100.227	UDP
3		
4		
5		

Below the table is a section for additional settings:

Enable Proxy Keep Alive	Using Register
Proxy Keep Alive Time	15
Proxy Load Balancing Method	Disable
Is Proxy Hot Swap	Yes
SRD Index	0

**Figure 23: Proxy set tables**

- Configure outbound IP routing rules: the 'Outbound IP Routing Table' page provides a table for configuring up to 180 out bound IP call routing rules. The device uses these rules to route calls (Tel or IP) to IP destinations (When a proxy server is not used for routing).

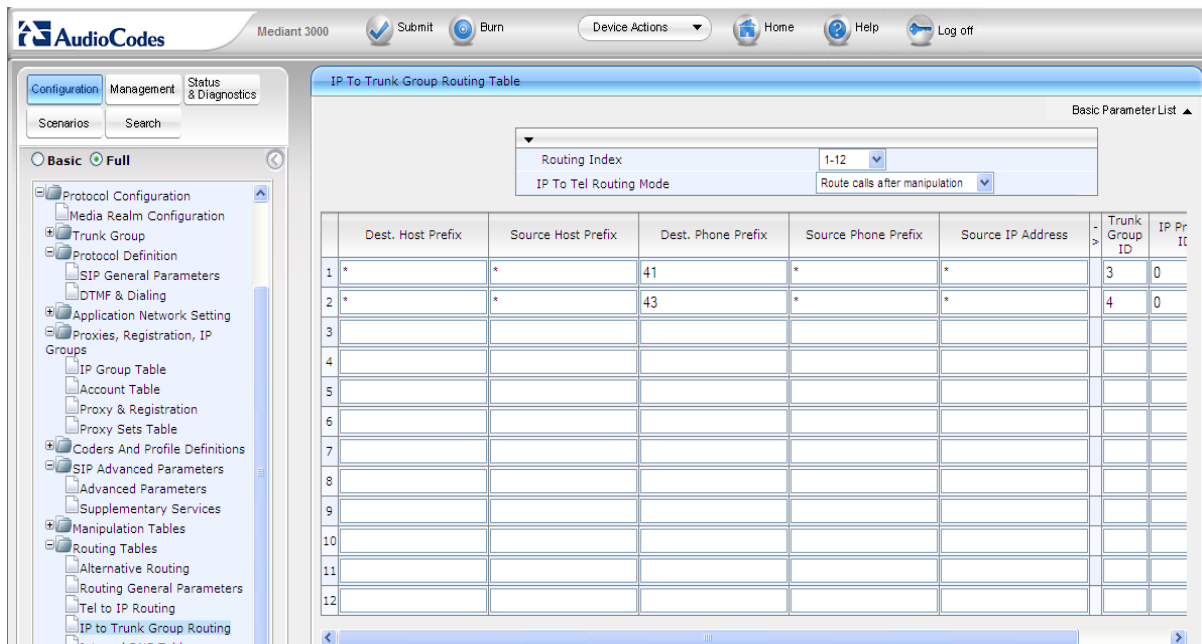
The screenshot shows the 'Tel to IP Routing' configuration page. The left sidebar is similar to Figure 23, with 'Tel to IP Routing' selected under the 'Routing Tables' category.

The main content area has a 'Routing Index' dropdown set to 1-10 and a 'Tel To IP Routing Mode' dropdown set to 'Route calls after manipulation'. Below this is a table for configuring routing rules:

	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Port	Transport Type	Dest. IP Group ID
1	*	*	*	nor-el-dplab.com		Not Configured	0
2						Not Configured	
3						Not Configured	
4						Not Configured	
5						Not Configured	
6						Not Configured	
7						Not Configured	
8						Not Configured	
9						Not Configured	
10						Not Configured	

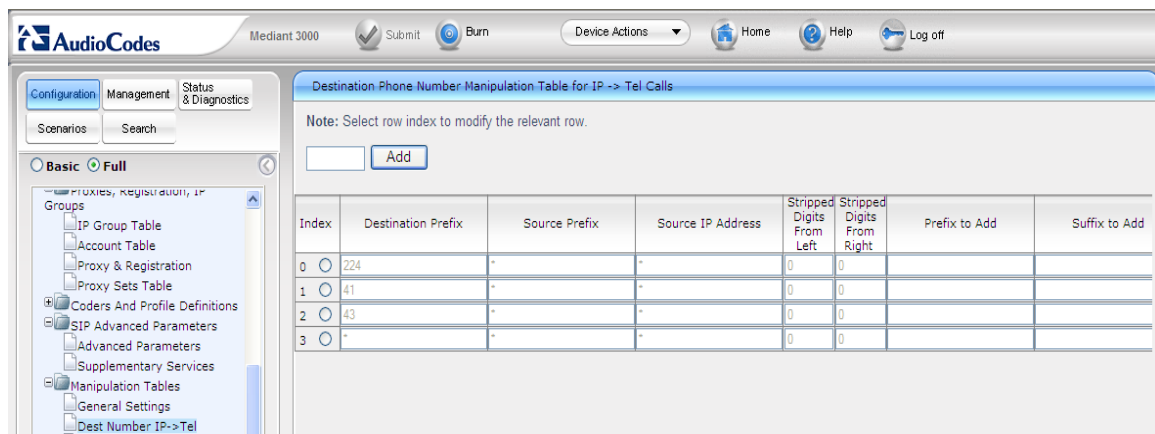
**Figure 24: Tel to IP Routing**

- Configuring the Inbound IP routing table: the 'Inbound IP Routing Table' page allows you to configure up to 120 in bound call routing rules. The device uses these rules for IP to IP routing and IP to Tel routing.



**Figure 25: IP to Trunk Group Routing**

- Manipulation Tables: the Manipulation Tables submenu allows you to configure number manipulation and mapping of NPI/TON to SIP messages. This submenu includes the following items:
  - o Dest number IP->Tel as shown in **Figure 26**.



**Figure 26: Dest Number IP->Tel**

- Dest number Tel->IP: Manipulate number received from PSTN for the outgoing SIP Diversion, Resource-Priority, or History-Info header that is sent to IP. Shown below in **Figure 27**.

The screenshot shows the AudioCodes Mediant 3000 web interface. The left sidebar contains a tree view with categories like 'Configuration', 'Management', and 'Status & Diagnostics'. Under 'Configuration', there are sub-items like 'Proxies, Registration, IP', 'Groups', 'IP Group Table', 'Account Table', 'Proxy & Registration', 'Proxy Sets Table', 'Coders And Profile Definitions', 'SIP Advanced Parameters', 'Advanced Parameters', 'Supplementary Services', 'Manipulation Tables', 'General Settings', 'Dest Number IP->Tel', and 'Dest Number Tel->IP'. The main content area is titled 'Destination Phone Number Manipulation Table for Tel -> IP Calls'. It includes a note: 'Note: Select row index to modify the relevant row.' and an 'Add' button. Below this is a table with the following columns: Index, Source Trunk Group, Destination Prefix, Source Prefix, Stripped Digits From Left, Stripped Digits From Right, Prefix to Add, Suffix to Add, and Number Length. The table contains five rows of data.

Index	Source Trunk Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number Length
0	-1	210	*	0	0		4	7
1	-1	40	*	0	0			7
2	-1	42	*	0	0			7
46	-1	46	*	0	0			7
47	-1	*	*	0	0			7

**Figure 27: Dest number Tel->IP**

- Source Number IP->Tel as shown in **Figure 28**.

The screenshot shows the AudioCodes Mediant 3000 web interface. The left sidebar contains a tree view with categories like 'Configuration', 'Management', and 'Status & Diagnostics'. Under 'Configuration', there are sub-items like 'Proxies, Registration, IP', 'Groups', 'IP Group Table', 'Account Table', 'Proxy & Registration', 'Proxy Sets Table', 'Coders And Profile Definitions', 'SIP Advanced Parameters', 'Advanced Parameters', 'Supplementary Services', 'Manipulation Tables', 'General Settings', 'Dest Number IP->Tel', 'Dest Number Tel->IP', and 'Source Number IP->Tel'. The main content area is titled 'Source Phone Number Manipulation Table for IP -> Tel Calls'. It includes a note: 'Note: Select row index to modify the relevant row.' and an 'Add' button. Below this is a table with the following columns: Index, Destination Prefix, Source Prefix, Source IP Address, Stripped Digits From Left, Stripped Digits From Right, Prefix to Add, and Suffix to Add. The table contains six rows of data.

Index	Destination Prefix	Source Prefix	Source IP Address	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add
0	*	*	*	0	0		
1	*	*	*	0	0		
2	*	*	*	0	0		
3	*	*	*	0	0		
4	*	*	*	0	0		
5	*	*	*	0	0		

**Figure: 28: Source Number IP->Tel**



- Source Number Tel->IP as shown in **Figure 29**.

The screenshot shows the AudioCodes Mediant 3000 web interface. The left sidebar contains a tree view with categories like 'Configuration', 'Management', and 'Status & Diagnostics'. Under 'Configuration', there are sub-items like 'Scenarios' and 'Search'. The main content area is titled 'Source Phone Number Manipulation Table for Tel -> IP Calls'. It includes a note: 'Note: Select row index to modify the relevant row.' and a 'Basic Parameter List' link. Below this is a table with the following columns: Index, Source Trunk Group, Source IP Group, Destination Prefix, Source Prefix, Stripped Digits From Left, Stripped Digits From Right, Prefix to Add, and Suffix to Add. The table has one row with index 1, Source Trunk Group -1, Source IP Group -1, Destination Prefix \*, Source Prefix \*, Stripped Digits From Left 0, Stripped Digits From Right 0, Prefix to Add, and Suffix to Add. There is an 'Add' button next to the index field.

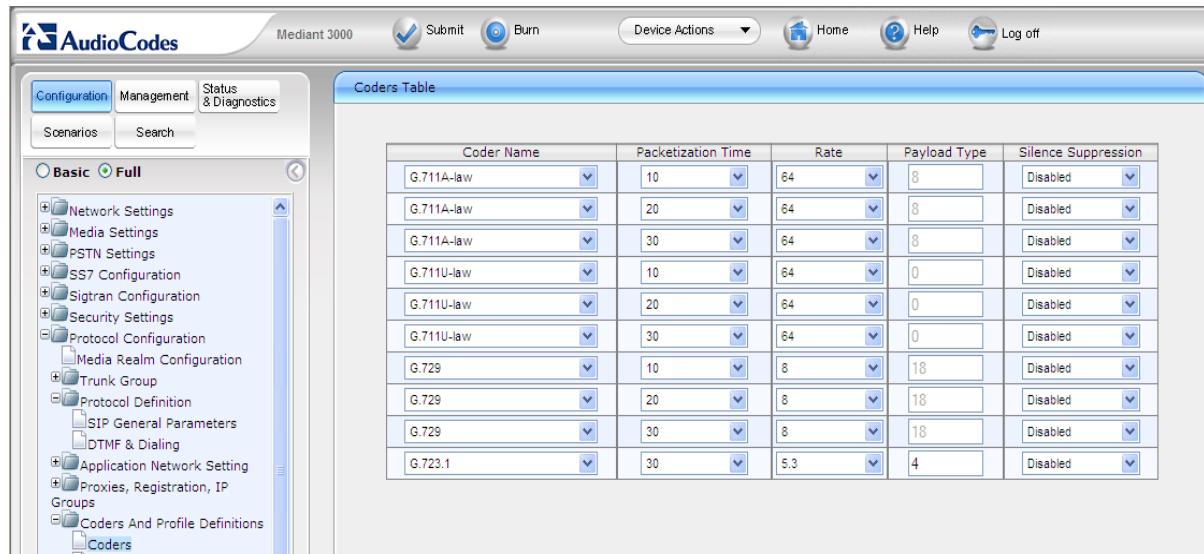
**Figure 29: Source Number Tel->IP**

- Mapping NPI/TON to SIP Phone-Context: the 'Phone-Context Table' is used to map Numbering Plan Indication (NPI) and Type of Number (TON) to the SIP Phone-Context parameter, as shown below in **Figure 30**.

The screenshot shows the AudioCodes Mediant 3000 web interface. The left sidebar is the same as in Figure 29. The main content area is titled 'Phone Context Table'. It includes a dropdown menu for 'Add Phone Context As Prefix' and a dropdown for 'Phone Context Index' with values '1-10'. Below this is a table with three columns: NPI, TON, and Phone Context. The table has 10 rows. The first three rows are pre-filled: Row 1 has NPI 'Unknown', TON 'Unknown', and Phone Context 'udp'; Row 2 has NPI 'Private', TON 'Level 1 Regional', and Phone Context 'udp'; Row 3 has NPI 'Private', TON 'Level 0 Regional(Lo', and Phone Context 'cdp,udp'. The remaining rows have empty fields for NPI, TON, and Phone Context.

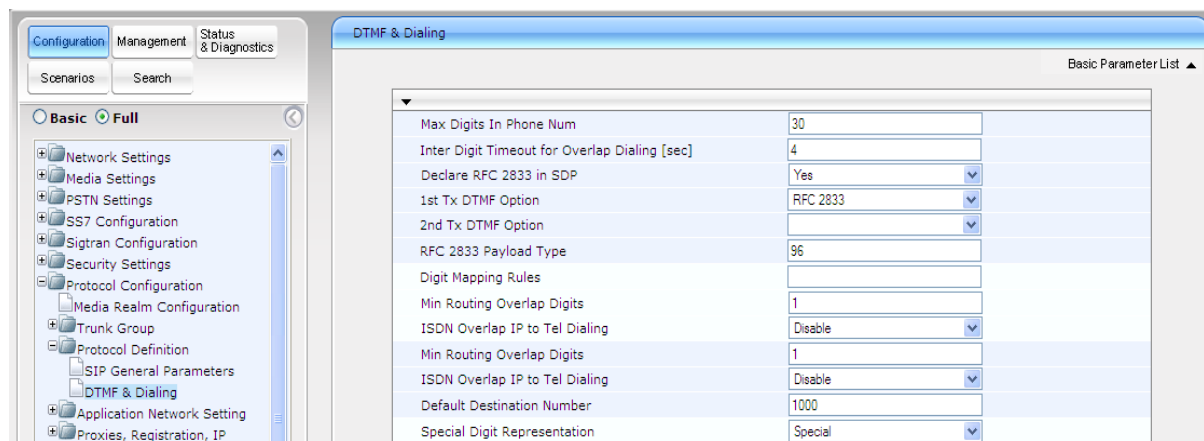
**Figure 30: Phone-context**

- Configuring Coders: the 'Coders' page allows you to configure up to ten coders (and their attributes) for the device. The First coder in the list has the highest priority and is used by the device whenever possible. If the far end cannot use the first coder, then device attempts to use the next coder in the list, and so on.



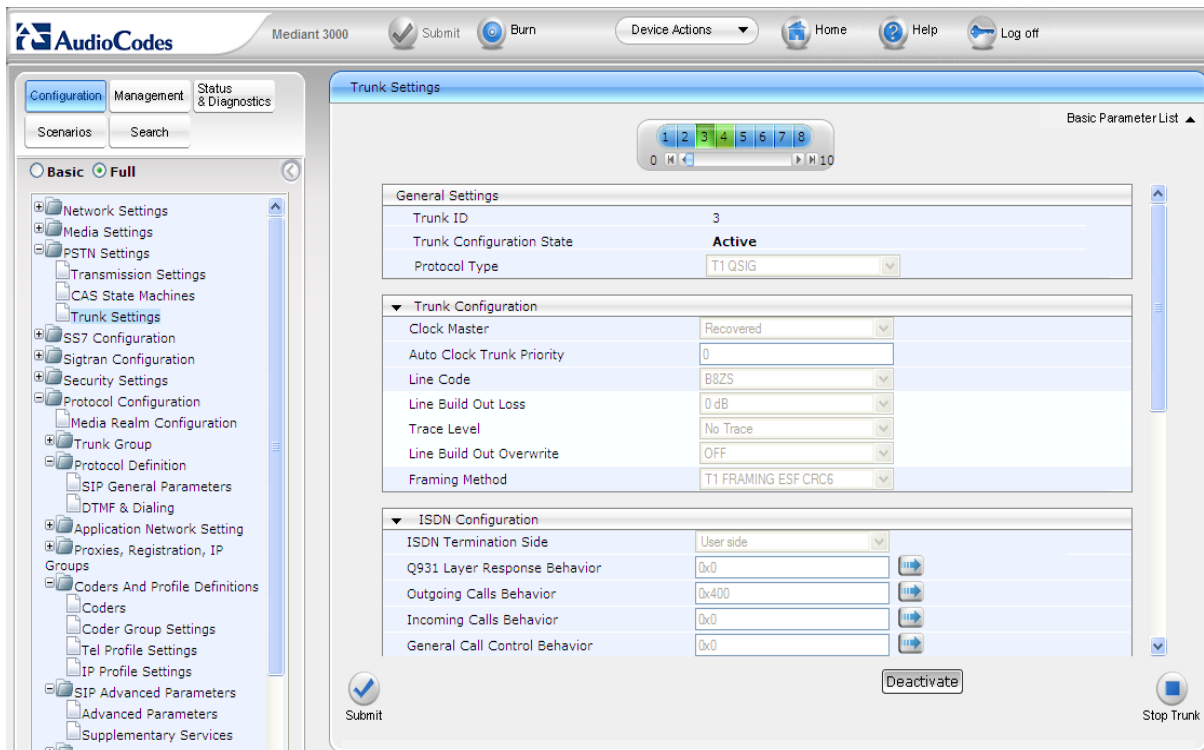
**Figure 31: Coders**

- Configuring DTMF and Dialing parameters: the 'DTMF & Dialing' page is used to configure parameters associated with dual-tone multi-frequency (DTMF) and dialing, as shown below in **Figure 32**.



**Figure 32: DTMF & Dialling**

- Configuring the Trunk Settings: the ‘Trunk Settings’ page as seen in **Figure 33** below allows you to configure device trunks. This includes selecting the PSTN protocol and configuring related parameters.
  - + Protocol Type: Configure the protocol type for a specific Trunk.
  - + T1\_QSIG = ECMA 143 QSIG over T1



**Figure 33: T1\_QSIG**

+T1\_NI2\_ISDN = National ISDN 2 PRI protocol

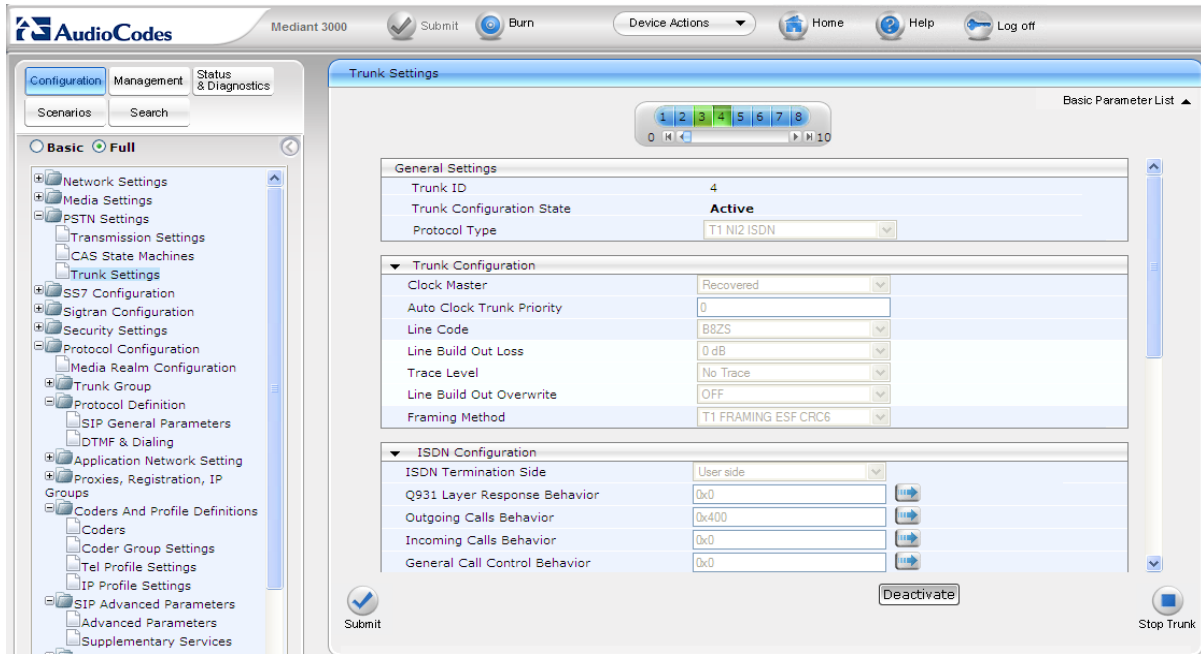


Figure 34: T1\_NI2\_ISDN

- Configuring the Transmission Settings: The 'Transmission Settings' page as seen in **Figure 35** allows you to define the PSTN transmission type (i.e., T3/DS3 or SONET/SDH).

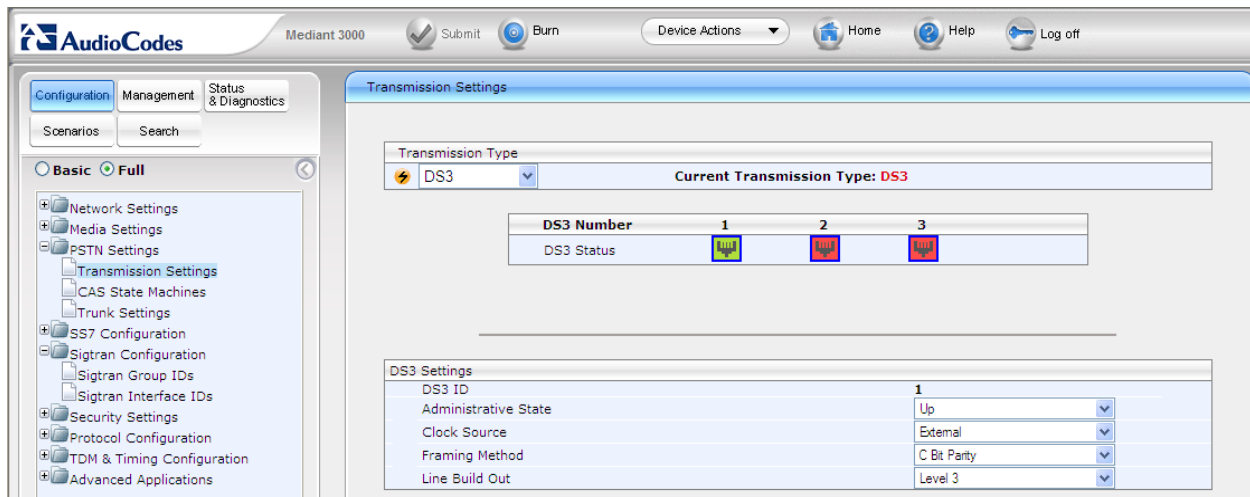
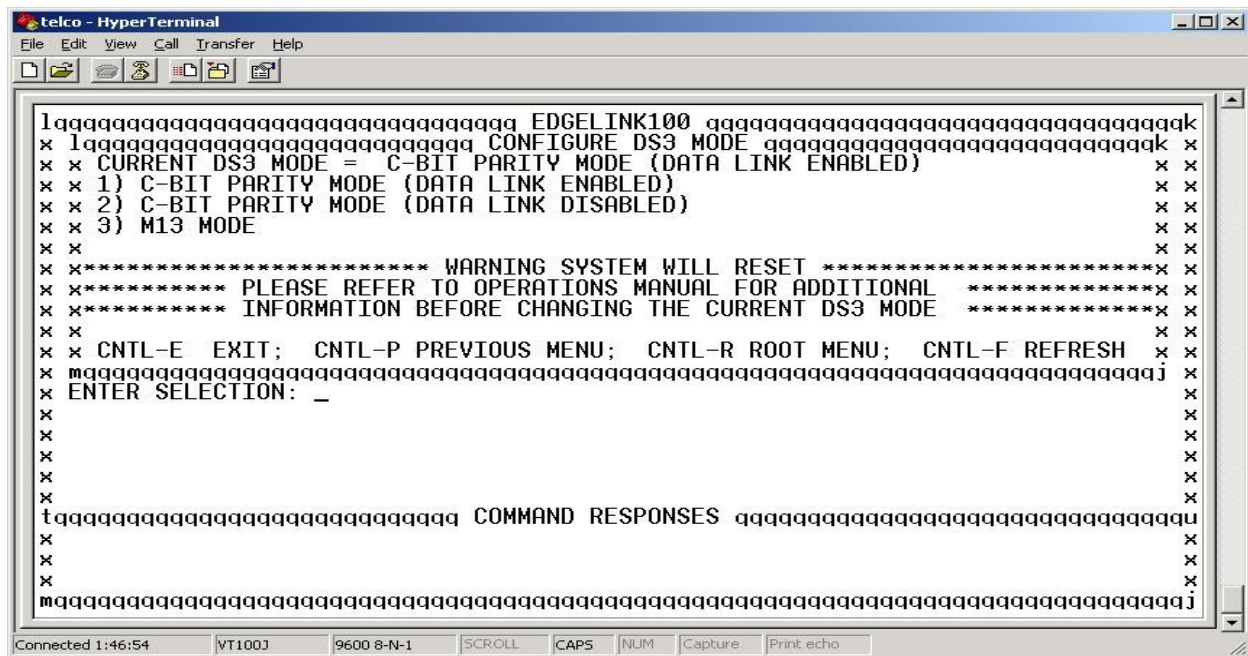


Figure 35: DS3 Transmission Settings

### 5.3 Configuration for Telco Edgelink 100

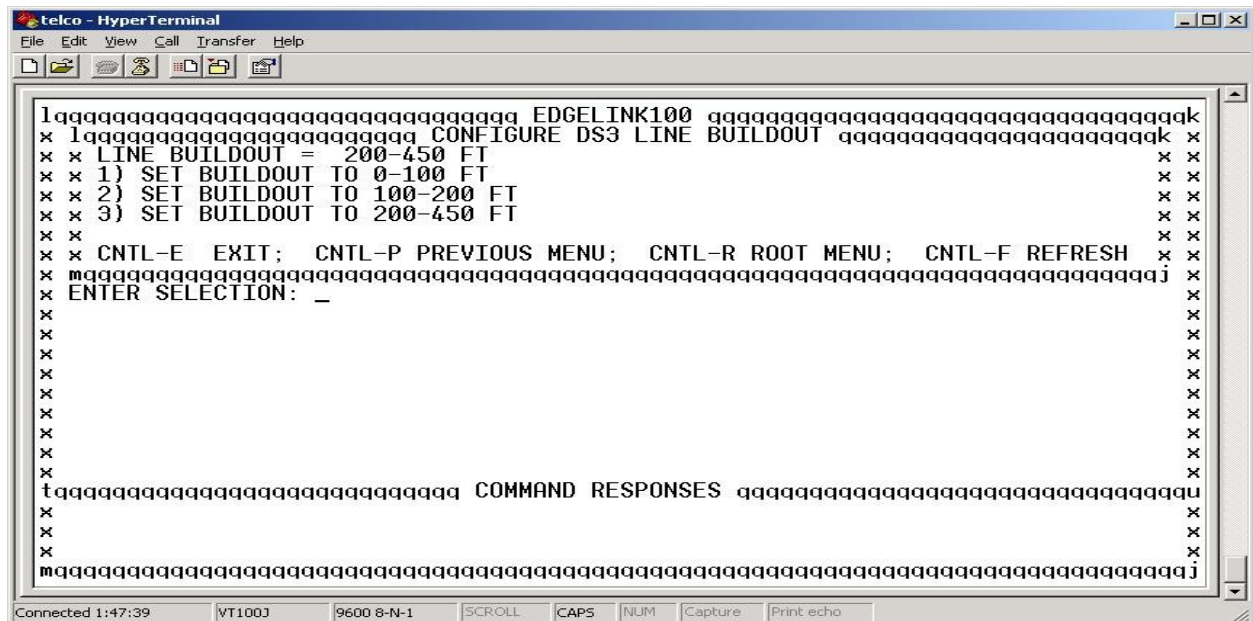
In this test set up/configuration, the Telco Edgelink 100 is used as MUX device to do conversion of the DS3 interface on the AudioCodes Mediant 3000 to the DS1 interface which is connecting to the PSTN via PRI QSIG protocol. This Telco Edgelink can be replaced by any other third party MUX conversion device for the same purpose. The following setting parameters are used for DS3, DS1, cable length, clocking source, and frame format. Make sure that these parameters are set properly on the Mediant 3000 and Telco Edgelink 100 device.

### 5.3.1. DS3 Mode



### Figure 36: DS3 mode

## - DS3 line buildout



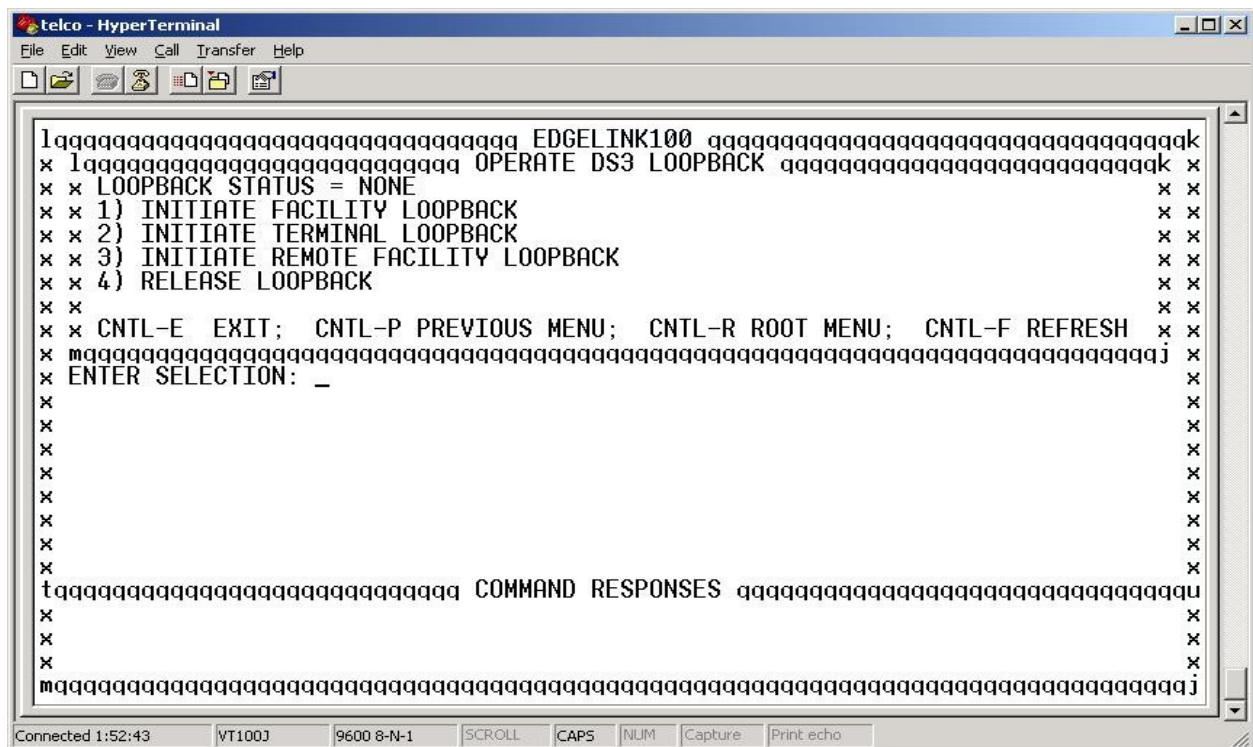
The screenshot shows a HyperTerminal window titled "telco - HyperTerminal". The menu displayed is for "EDGE LINK100" and "CONFIGURE DS3 LINE BUILDOUT". The menu options are:

- LINE BUILDOUT = 200-450 FT
- 1) SET BUILDOUT TO 0-100 FT
- 2) SET BUILDOUT TO 100-200 FT
- 3) SET BUILDOUT TO 200-450 FT

Navigation instructions: CNTL-E EXIT; CNTL-P PREVIOUS MENU; CNTL-R ROOT MENU; CNTL-F REFRESH. The prompt "ENTER SELECTION: \_" is shown. The status bar at the bottom indicates "Connected 1:47:39", "VT100J", "9600 8-N-1", and various control buttons like SCROLL, CAPS, NUM, Capture, and Print echo.

Figure 37: DS3 line buildout

## - DS3 Loopback



The screenshot shows a HyperTerminal window titled "telco - HyperTerminal". The menu displayed is for "EDGE LINK100" and "OPERATE DS3 LOOPBACK". The menu options are:

- LOOPBACK STATUS = NONE
- 1) INITIATE FACILITY LOOPBACK
- 2) INITIATE TERMINAL LOOPBACK
- 3) INITIATE REMOTE FACILITY LOOPBACK
- 4) RELEASE LOOPBACK

Navigation instructions: CNTL-E EXIT; CNTL-P PREVIOUS MENU; CNTL-R ROOT MENU; CNTL-F REFRESH. The prompt "ENTER SELECTION: \_" is shown. The status bar at the bottom indicates "Connected 1:52:43", "VT100J", "9600 8-N-1", and various control buttons like SCROLL, CAPS, NUM, Capture, and Print echo.

Figure 38: DS3 loop back



[illegible][illegible]

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The screenshot shows a HyperTerminal window titled "telco - HyperTerminal". The menu bar includes File, Edit, View, Call, Transfer, and Help. Below the menu is a toolbar with icons for file operations and communication. The main text area displays a series of commands and responses from a device named "EDGELINK100".

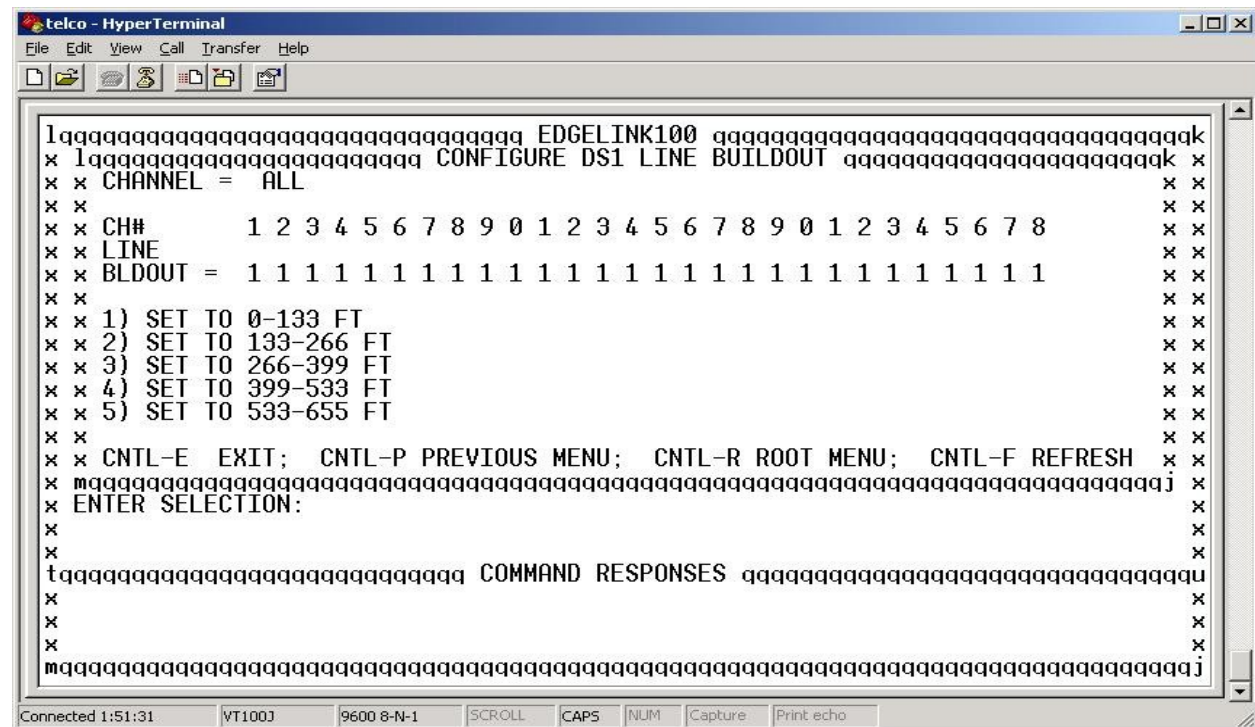
```

lgggggggggggggggggggggggggggggggg EDGELINK100 ggggggggggggggggggggggggggggggggk
x x lggggggggggggggggggggggggggggggg CONFIGURE DS1 LINE CODE ggggggggggggggggggggggggggggggg x
x x CHANNEL = ALL x x
x x
x x CH#      1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 x x
x x LINE x x
x x CODE =   B B B B B B B B B B B B B B B B B B B B B B B B B B B B x x
x x
x x 1) SET TO (A)MI x x
x x 2) SET TO (B)8ZS x x
x x
x x CNTL-E EXIT; CNTL-P PREVIOUS MENU; CNTL-R ROOT MENU; CNTL-F REFRESH x x
x mggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggj x
x ENTER SELECTION: _ x
x
x
x
x
x
x
x tgggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggg x
x ILLEGAL OPERATION - WRONG SERVICE MODE x
x
x mgggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggggjj x

```

At the bottom of the window, a status bar shows connection details: Connected 1:50:59, VT1003, 9600 8-N-1, SCROLL, CAPS, NUM, Capture, and Print echo.

## - DS1 Line Buildout

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[illegible]

### Figure 43: DS1 Loopback Mode

## 6. General Test Approach and Test Results

The focus of this interoperability compliance testing was primarily to verify the SIP trunk signaling establishment between AudioCodes Mediant 3000 and Avaya Communication Server 1000.

### 6.1. General test approach

The general test approach was to have the CS1000 clients/users placing calls to Mediant 3000 from the CS1000 and vice versa. The main objectives were to verify the Mediant 3000 successfully performed the following:

- Register to CS1000 Release 7.0 NRS/SPS domain as a SIP gateway endpoint.
- Call establishment between PSTN phones and Avaya CS1000 SIP and non SIP phone/clients.
- Basic call operation: DTMF transmission, voicemail with MWI notification, busy, hold.
- Advance CS1000 Call Server features: Call forward busy/no answer, call redirection and conference: Avaya phones as a transferor for blind/consultative transfers and as a moderator for the 3 ways conference call.
- Codec negotiations.

### 6.2. Test Results

The objectives outlined in **Section 6.1** were verified and met.

## 7. Verification Steps

This section includes some steps that can be followed to verify the configuration.

- Place a call from and to the PSTN phone (emulate by CS1000 Release 6.0) and verify that the call is established with 2 way speech path.
- During the call, use syslog tool, pcap (ethereal) at the Mediant 3000 Gateway and clients to make sure that all SIP request/response messages are correct.

## 8. Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 6.1**. The AudioCodes Mediant 3000 version 6.00A.020.002 has successfully passed compliance testing with Avaya Communication Server 1000 Release 7.0.

## 9. Additional References

Product documentation for Avaya products may be found at:

<http://support.nortel.com/go/main.jsp>

[1] *Communication Server 1000 SIP Line Fundamental, Rel. 7.0, Revision 01.08, February 2010, Document Number NN43001-508*

[2] *Communication Server 1000E Maintenance, Rel. 7.0, Revision 03.16, January 2010, Document Number NN43041-700*

[3] *Communication Server 1000 ISDN Primary Rate Interface Installation and Commissioning, Revision 01.03, August 2007, Document Number NN43001-301*

[4] *Troubleshooting Guide for Distributors, Rel. 7.0, Revision 02.02, December 2009, Document Number NN43001-730*

[5] *Communication Server 1000E Installation and Commissioning, Rel. 7.0, Revision 03.06, February 2010, Document Number NN43041-310*

[6] *Communication Server 1000E Software Upgrades, Revision 03.12, February 2010, Document Number NN43041-458*

[7] *Communication Server 1000E Linux Platform Base and Applications Installation and Commissioning, Revision 03.10, February 09, 2009, Document Number NN43001-315*

[8] *Communication Server 1000 Unified Communications Management Common Services Fundamentals, Revision: 03.04, September 28, 2009, Document Number NN43001-116*

Product information for AudioCodes Mediant 3000 products can be found at:

<http://www.audiocodes.com>

## 10. Appendix - Physical connection between Mediant 3000-Edgeline100 and Edgeline100-CS1000

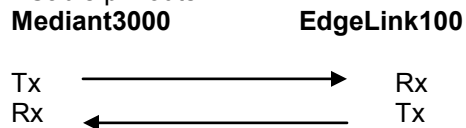
### + DS3 connection: Mediant3000 and EdgeLink100

- Diagram: [refer lab diagram](#).

- Cable type: Copper Coaxial

- Cable length: 200-450 ft

- Cable pin outs:



### + DS1 connection: EdgeLink100 and CS1000

- Diagram: [refer lab diagram](#).

- Cable type: Copper Twisted Pair

- Cable length: <655 ft

- Cable pin outs: This is the illustration for 1 DS1 connection using the 1<sup>st</sup> pair of EdgeLink100 64 pins connector and CS1000 RJ21 50 pins connector.

#### Edgeline100 (64 pins)

- DSX1- IN

Rx Ring – Pin 1

Rx Tip - Pin 33

- DSX1- OUT

Tx Ring – Pin 1

Tx Tip - Pin 33

#### CS1000 (RJ21-50 pins)

23 Tx RingRing - Pin 23

48 Tx Tip – Pin 48

24 Rx Ring – Pin 24

49 Rx Tip – Pin 49

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