Avaya Solution & Interoperability Test Lab

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 vise 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.

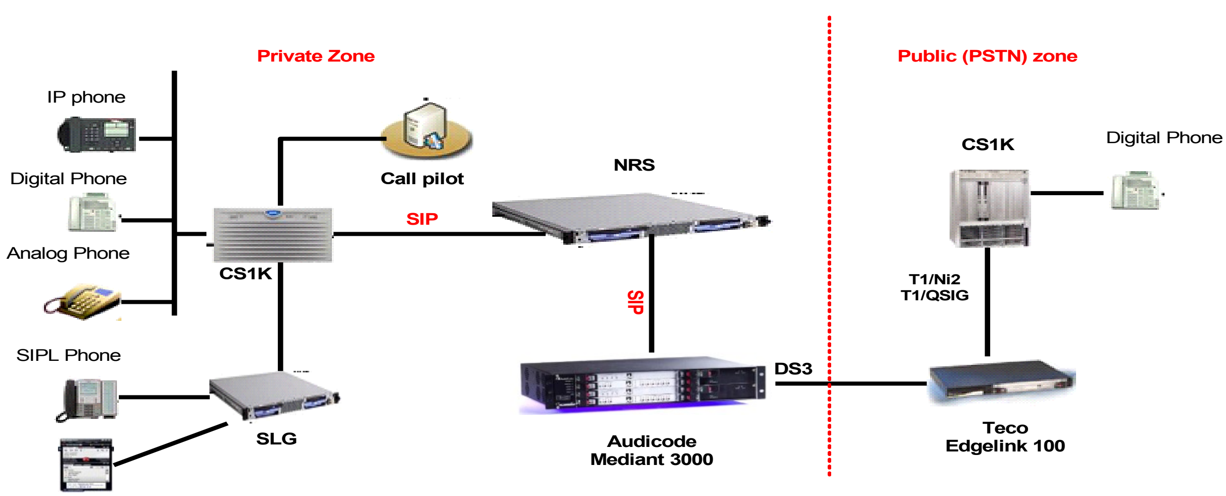

Figure 1: Lab Diagram

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
    - Installed with CS1000 Release 7.0 Linux Base.
    - Joined CS1000 Release 7.0 Security Domain.
    - 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**.

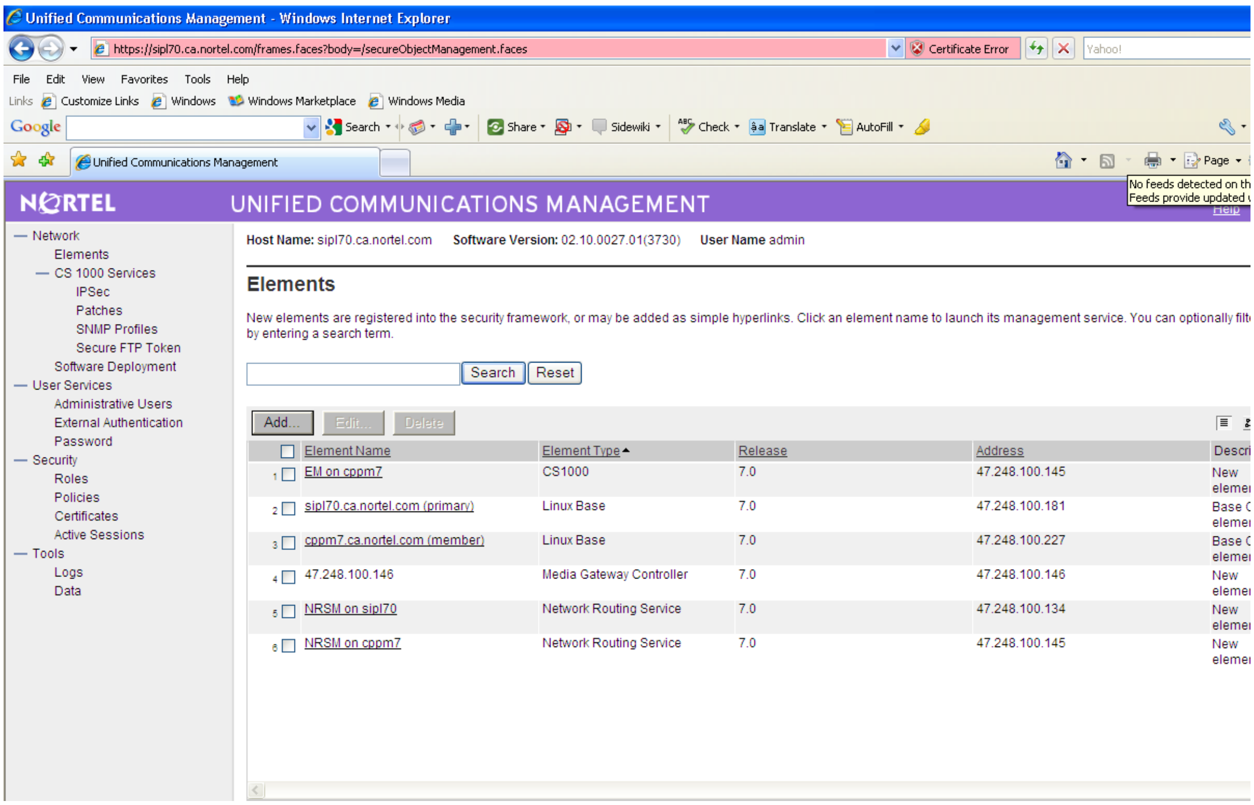

Figure 2: UCM Home Page

- 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.

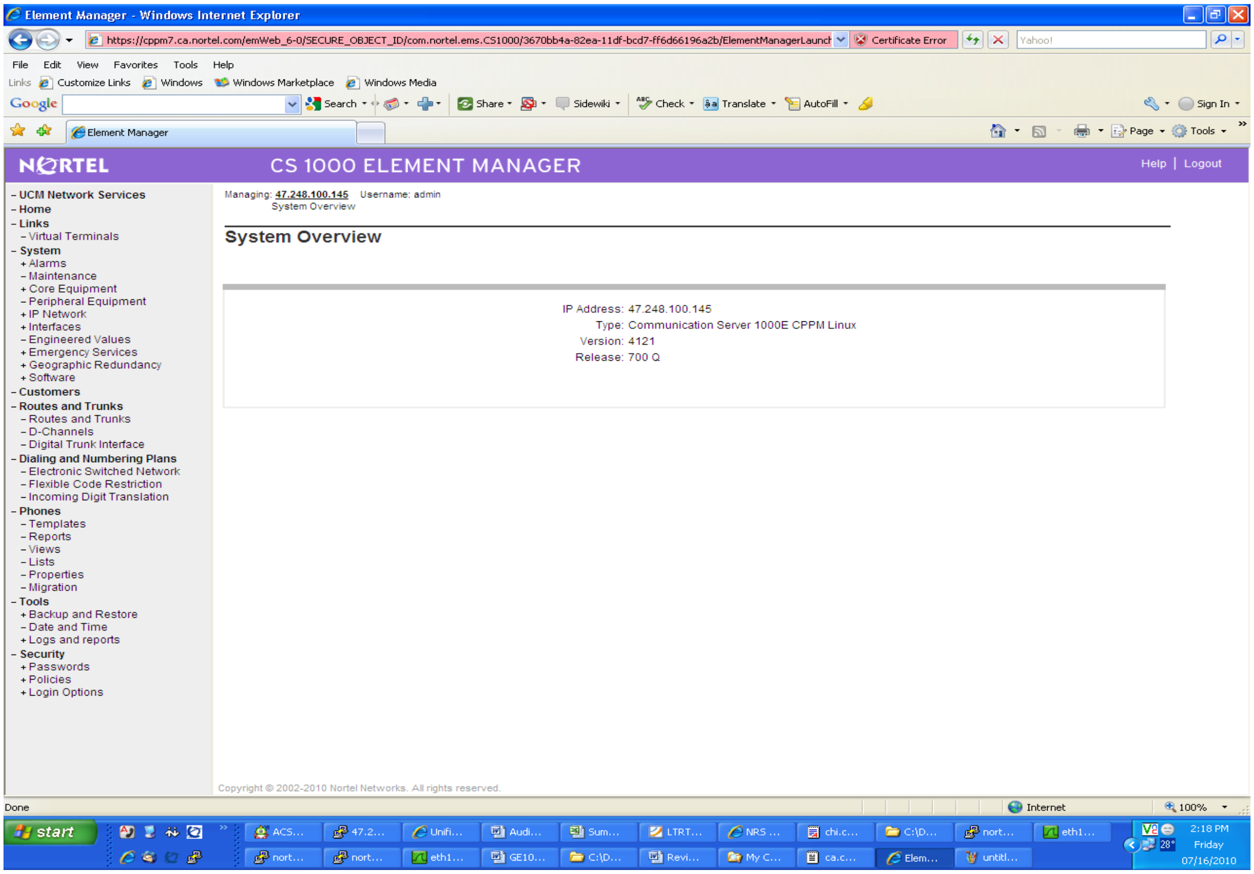

Figure 3: CS1000 EM Home Page

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.

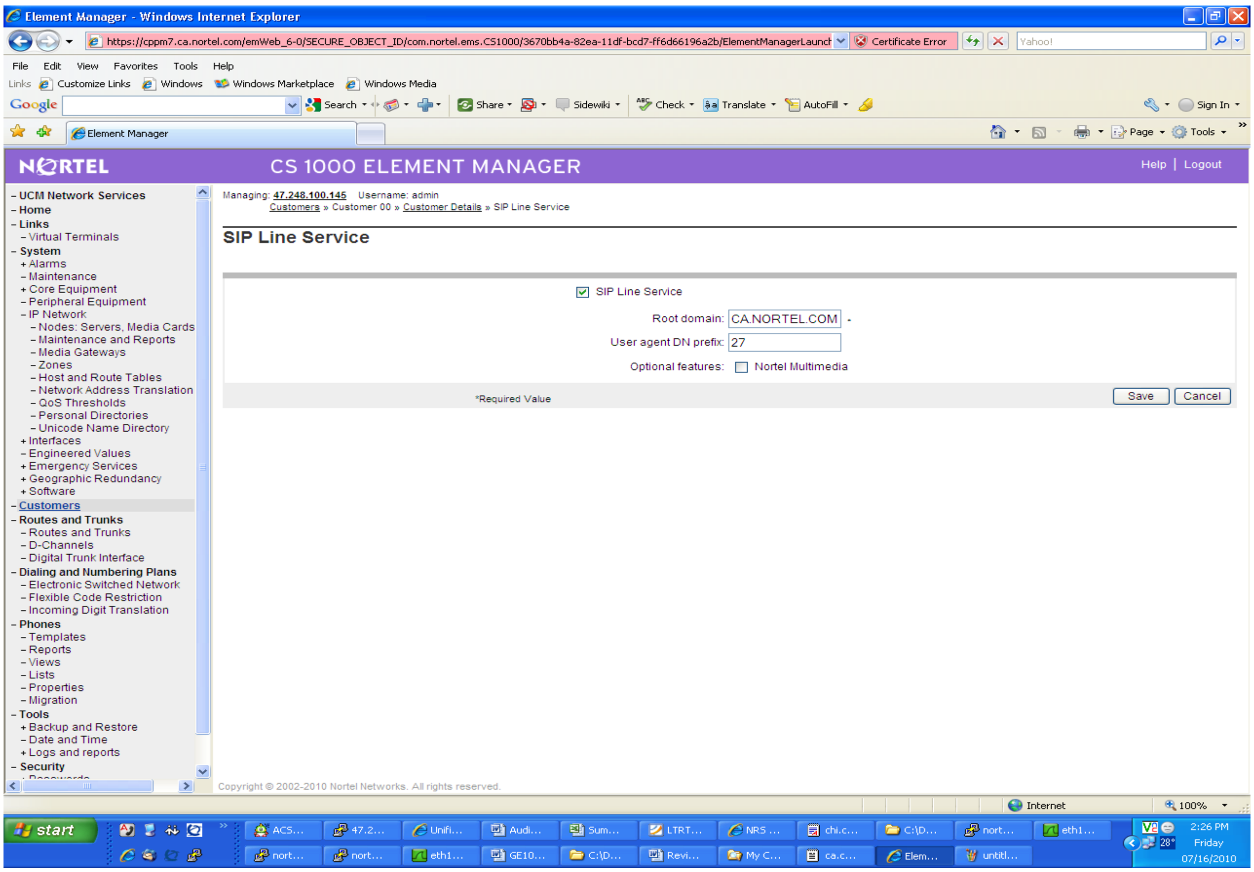

Figure 4: SIP Line Service in Customers Data Block

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.

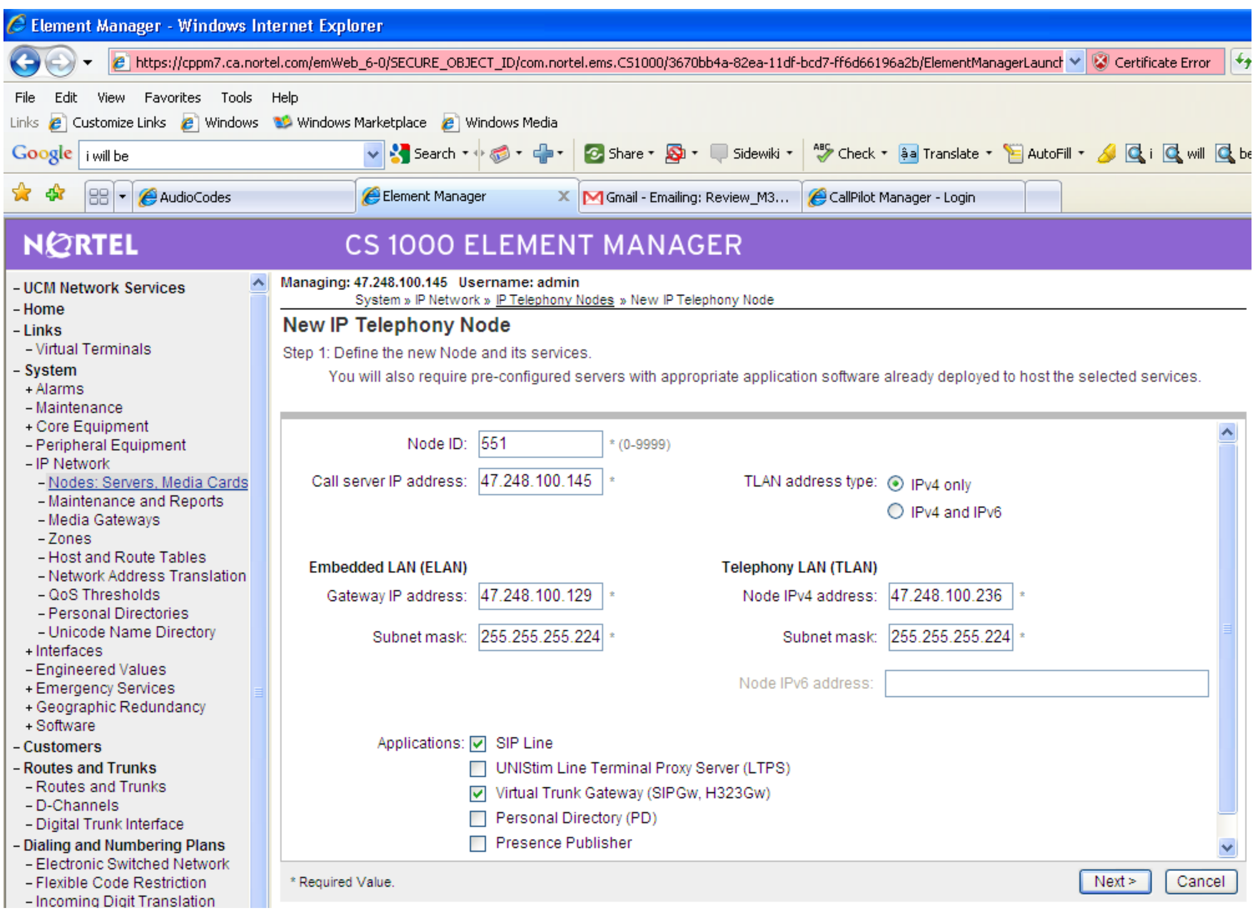

Figure 5 – IP Telephony Node

- 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.

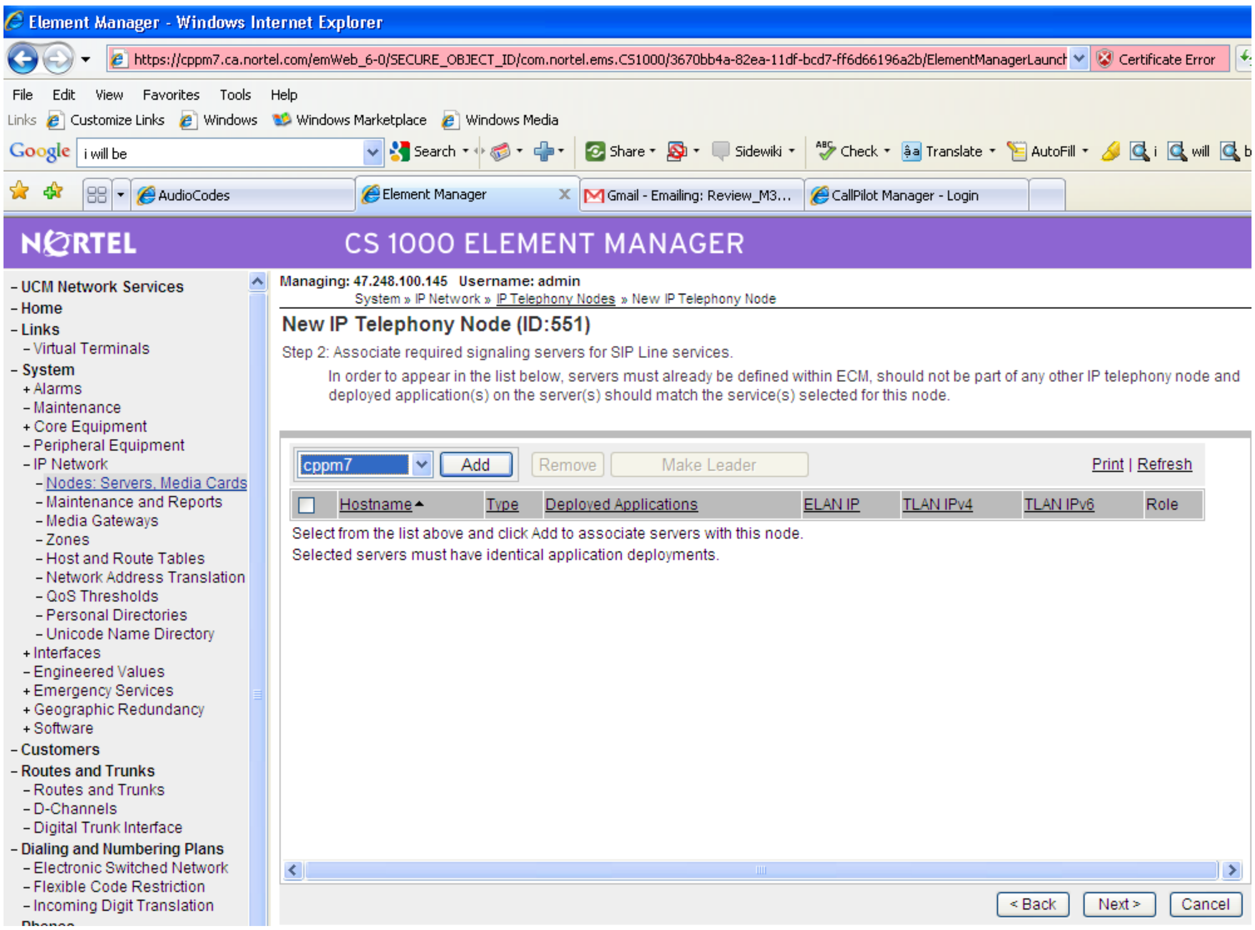

Figure 6 – IP Telephony Node – Add Server

- 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.***

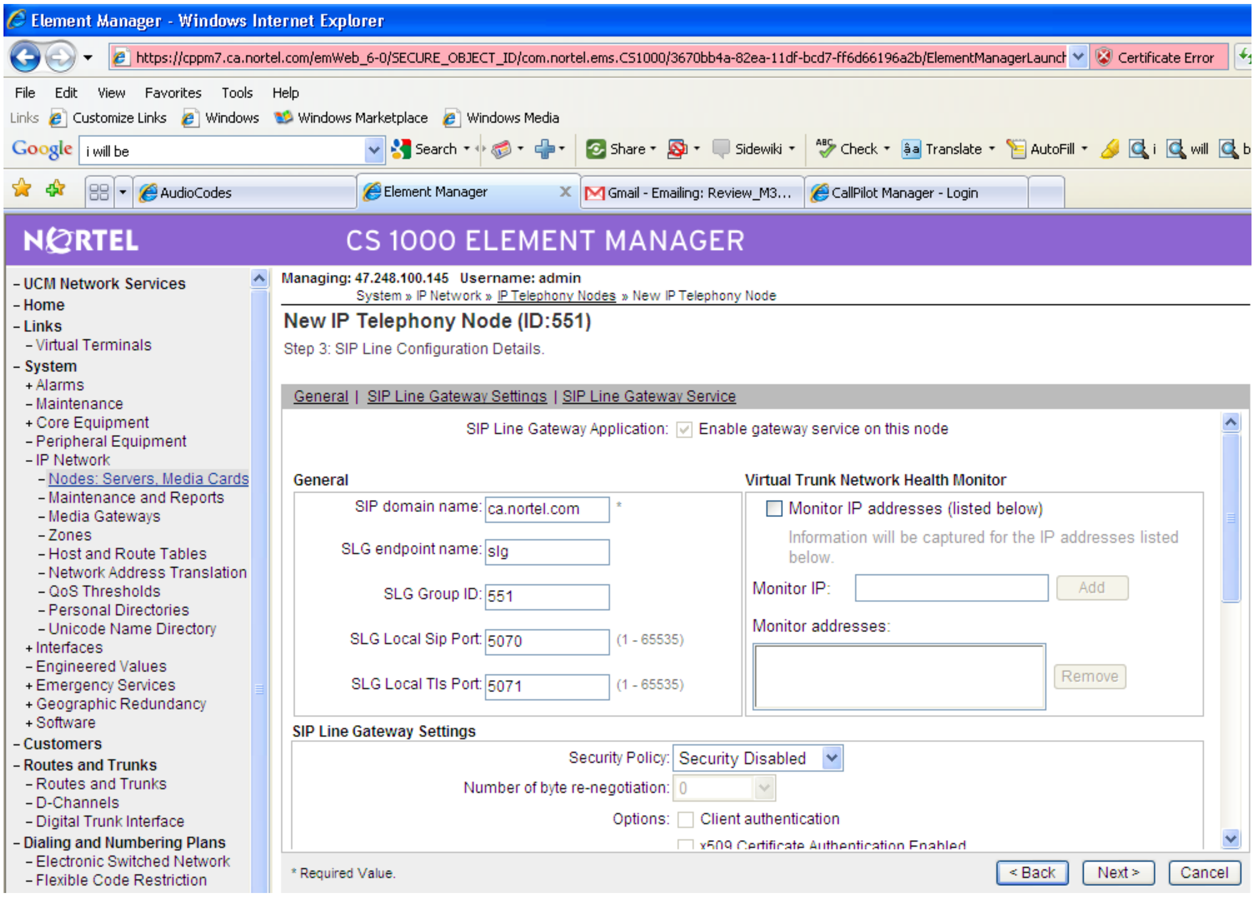

Figure 7 – SIP Line Node Details

- 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).

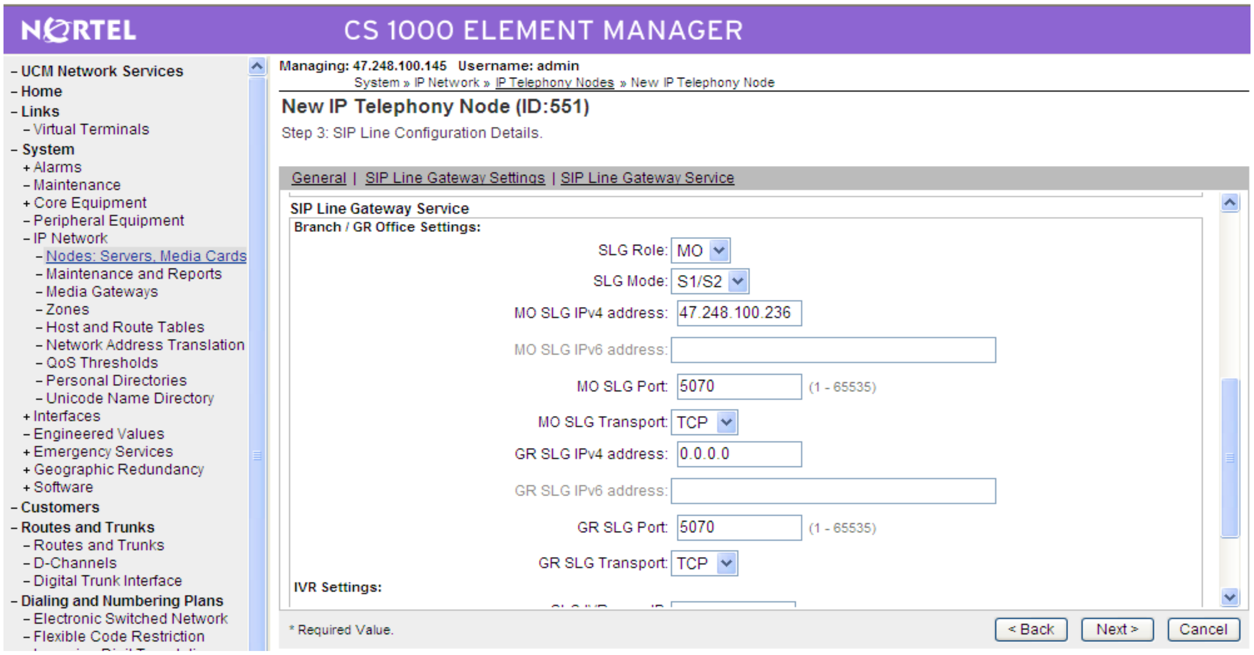

Figure 8 – SIP Line Node Details (cont.)

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

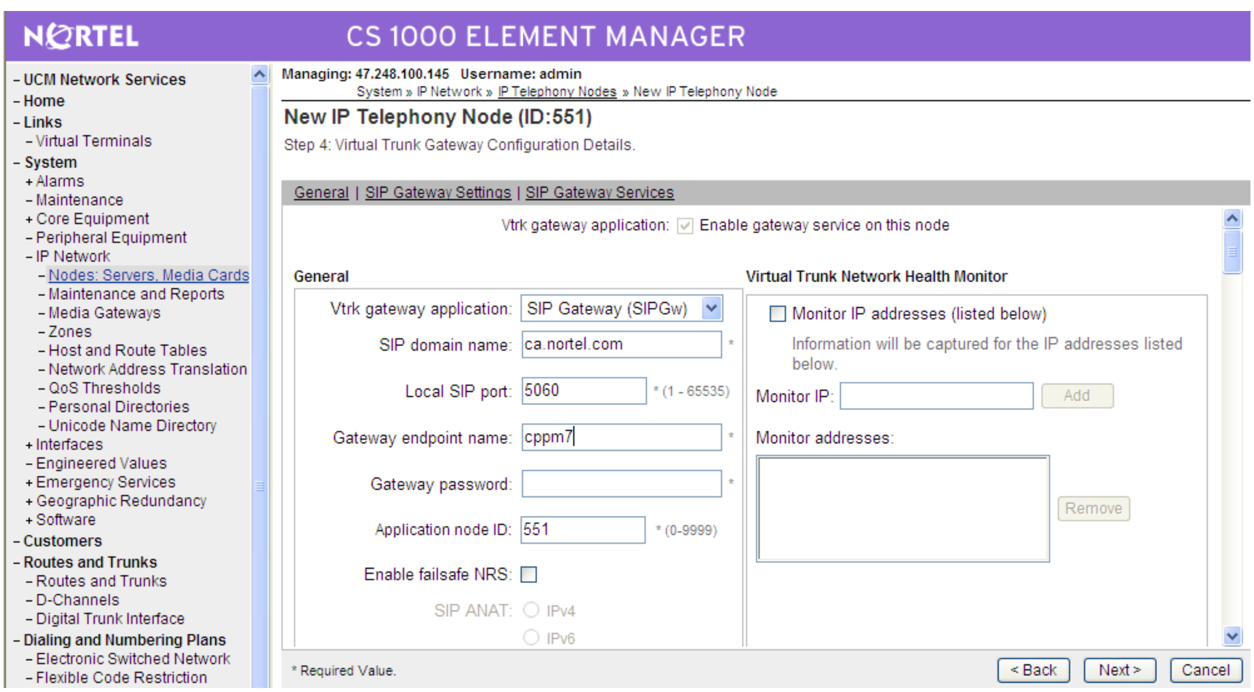

Figure 9 – SIP GW Node Details

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

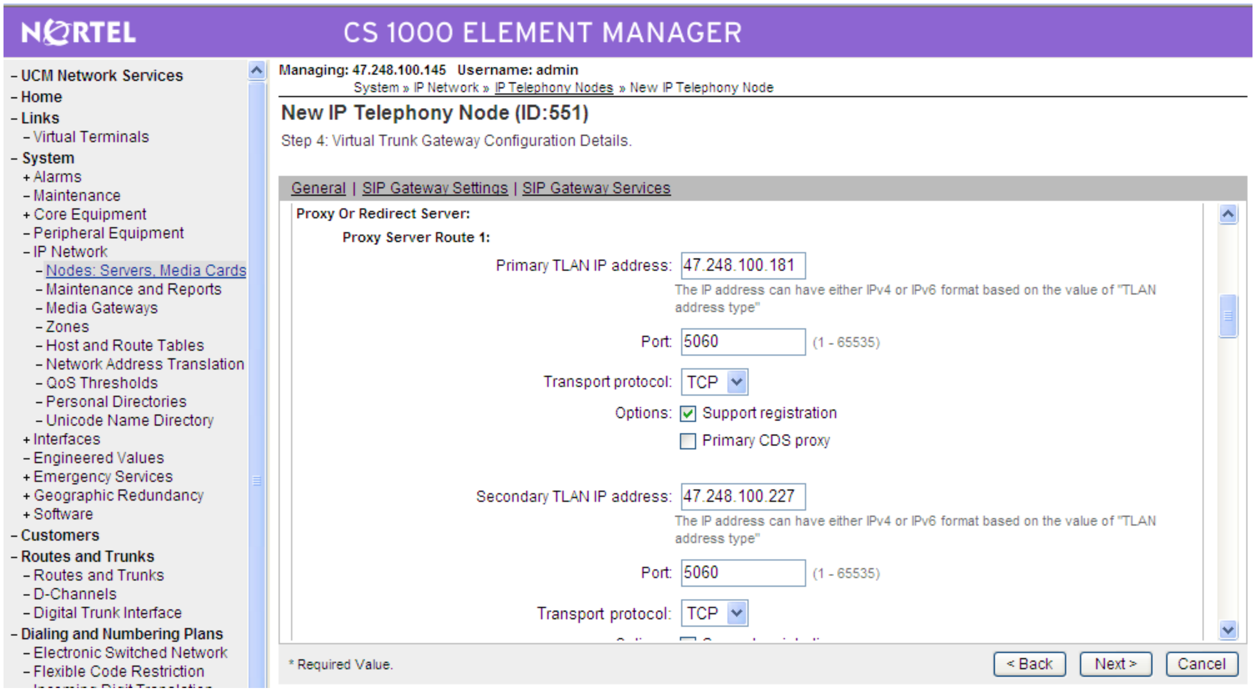

Figure 10 – Proxy or Redirect server

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

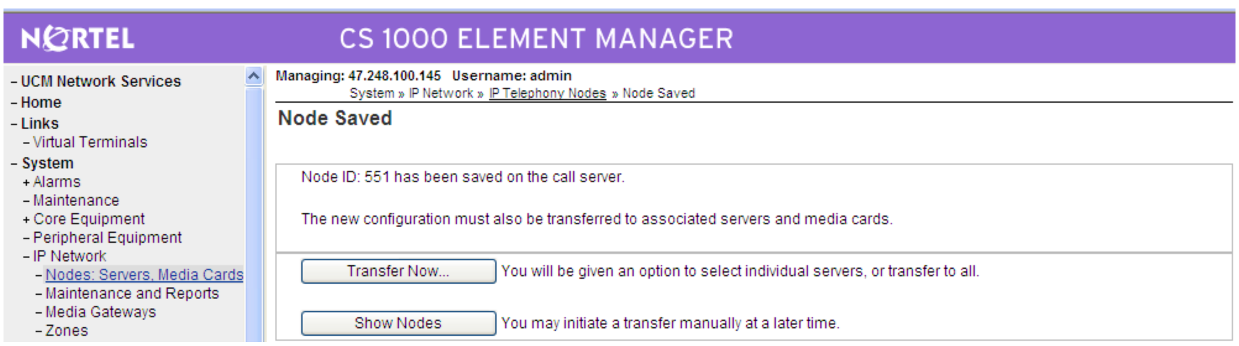

Figure 11– Transfer Configuration

- 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.

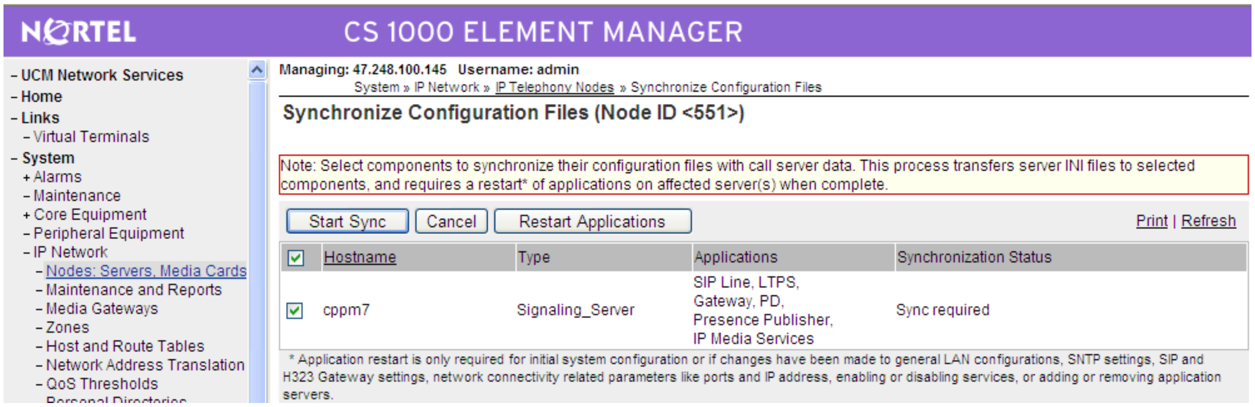

Figure 12 – Synchronize Configuration Files

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).***

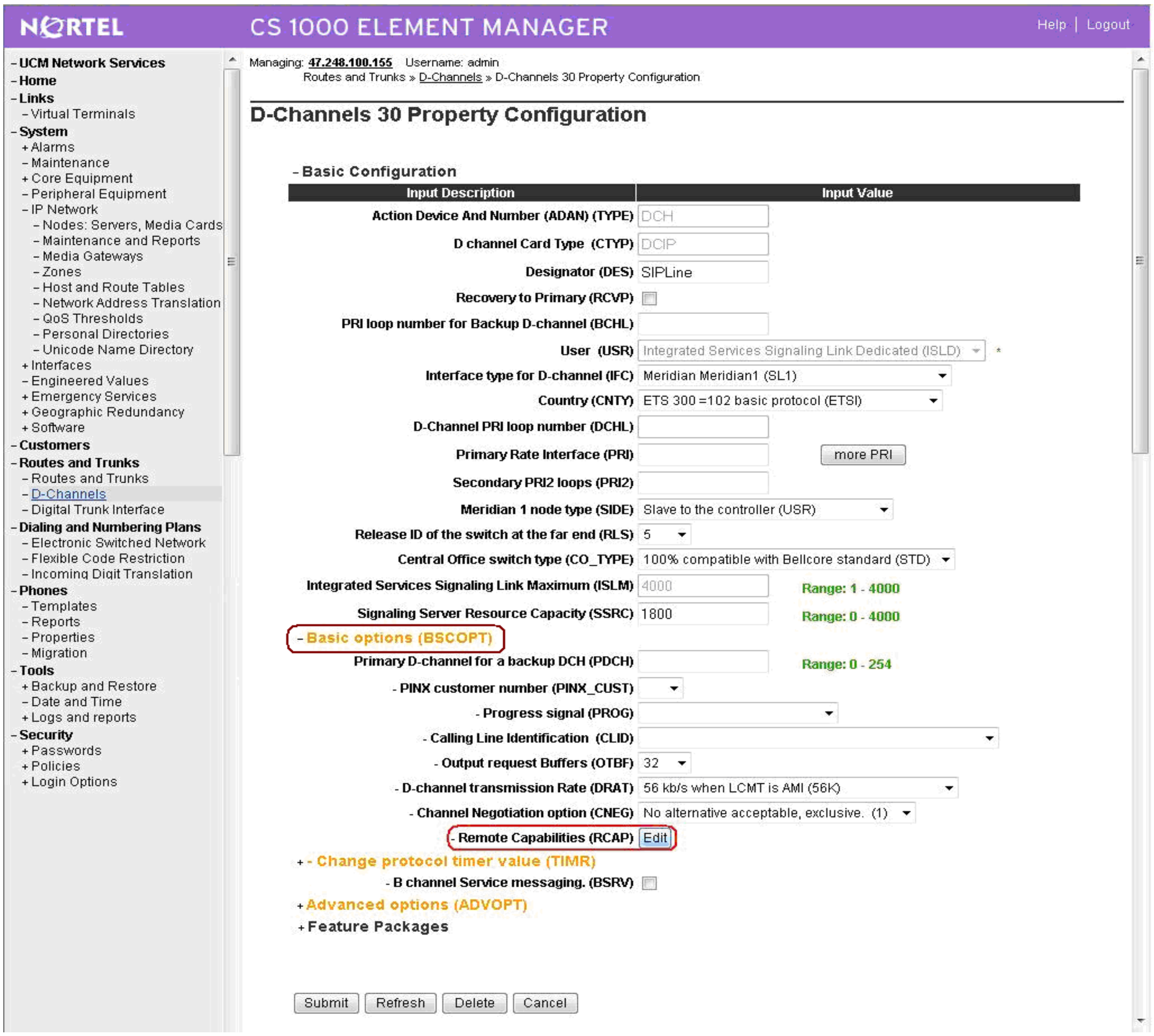

Figure 13 – SIP D-Channel Property Configuration

- **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.

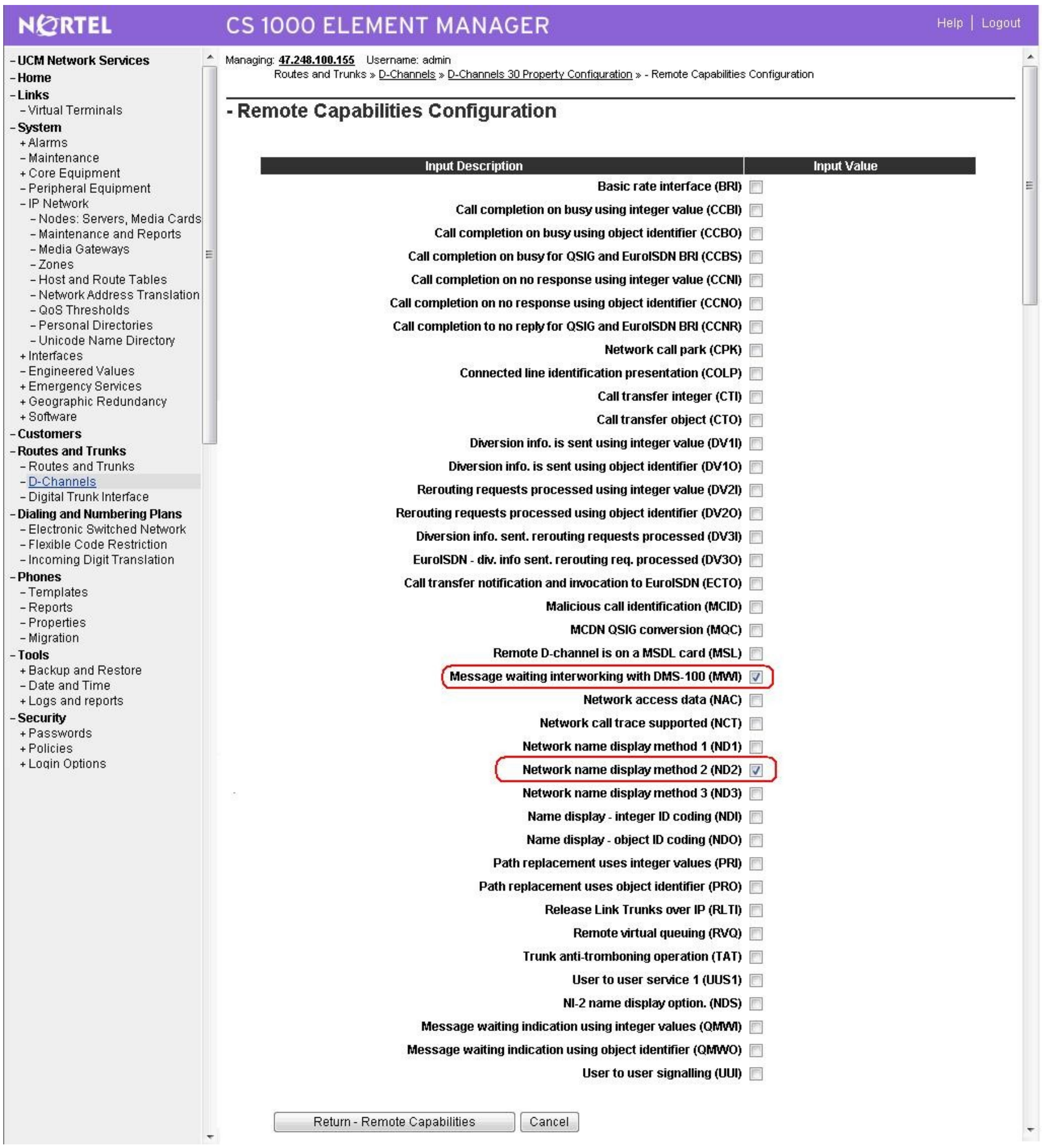

Figure 9 – SIP D-Channel RCAP Configuration Details

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.

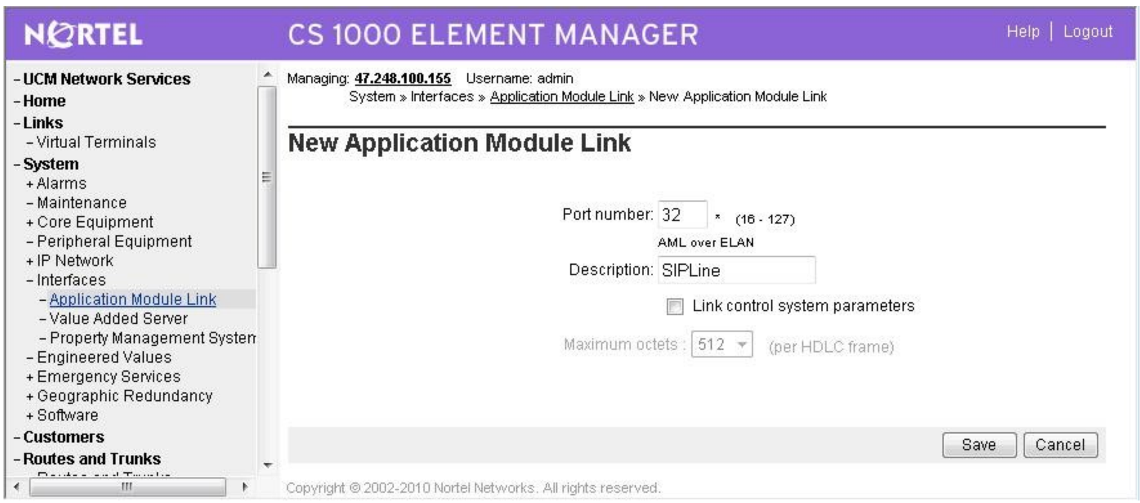

Figure 15 – Application Module Link Configuration

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**.

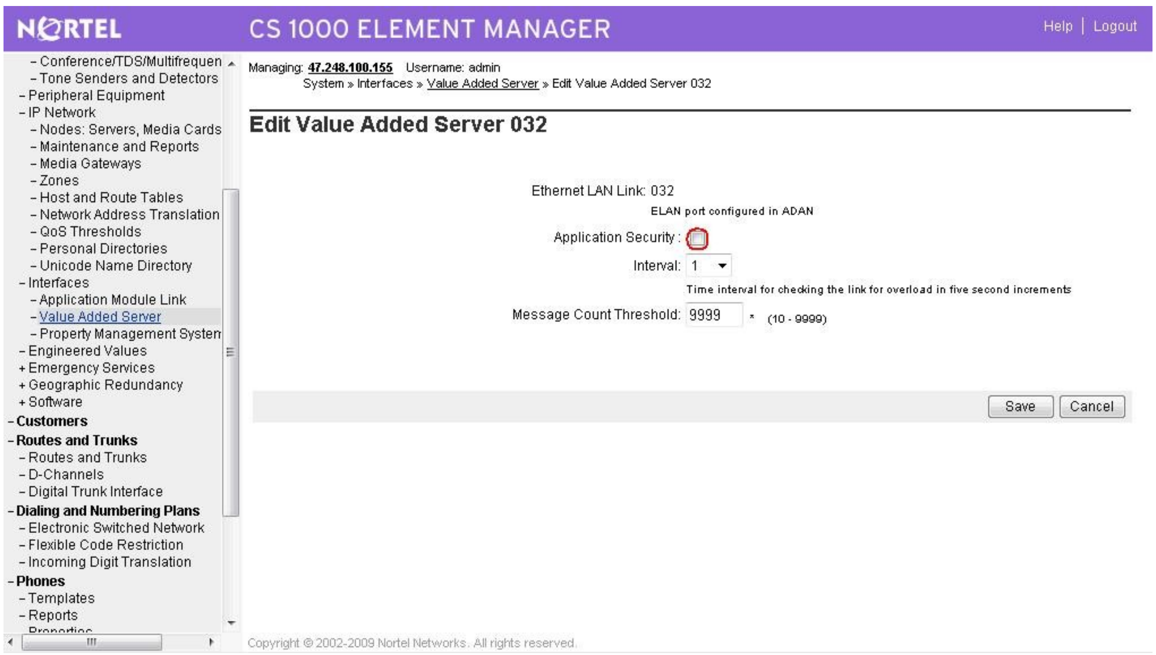

Figure 16 – Value Added Service for Application Module Link

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***.

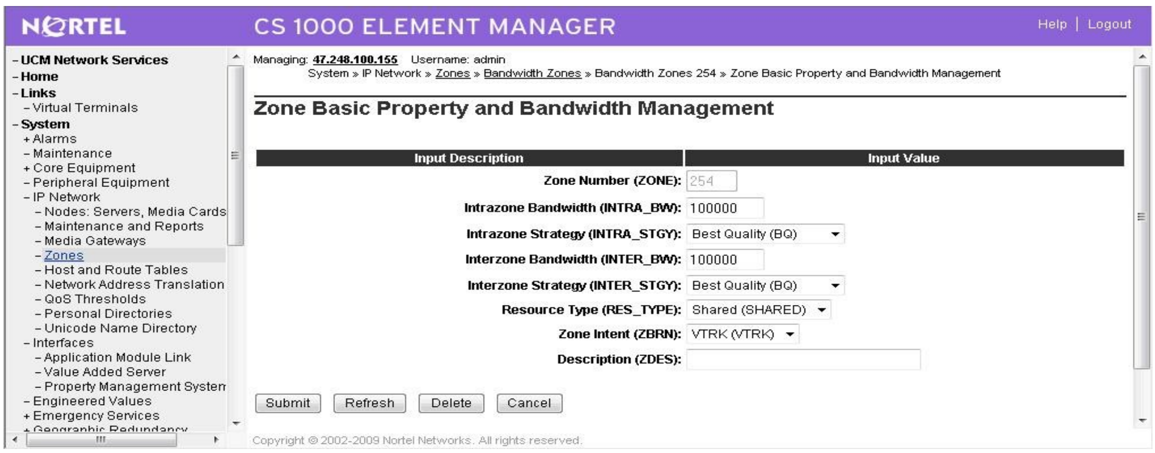

Figure 17 – Virtual Trunk Zone Configuration

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***.

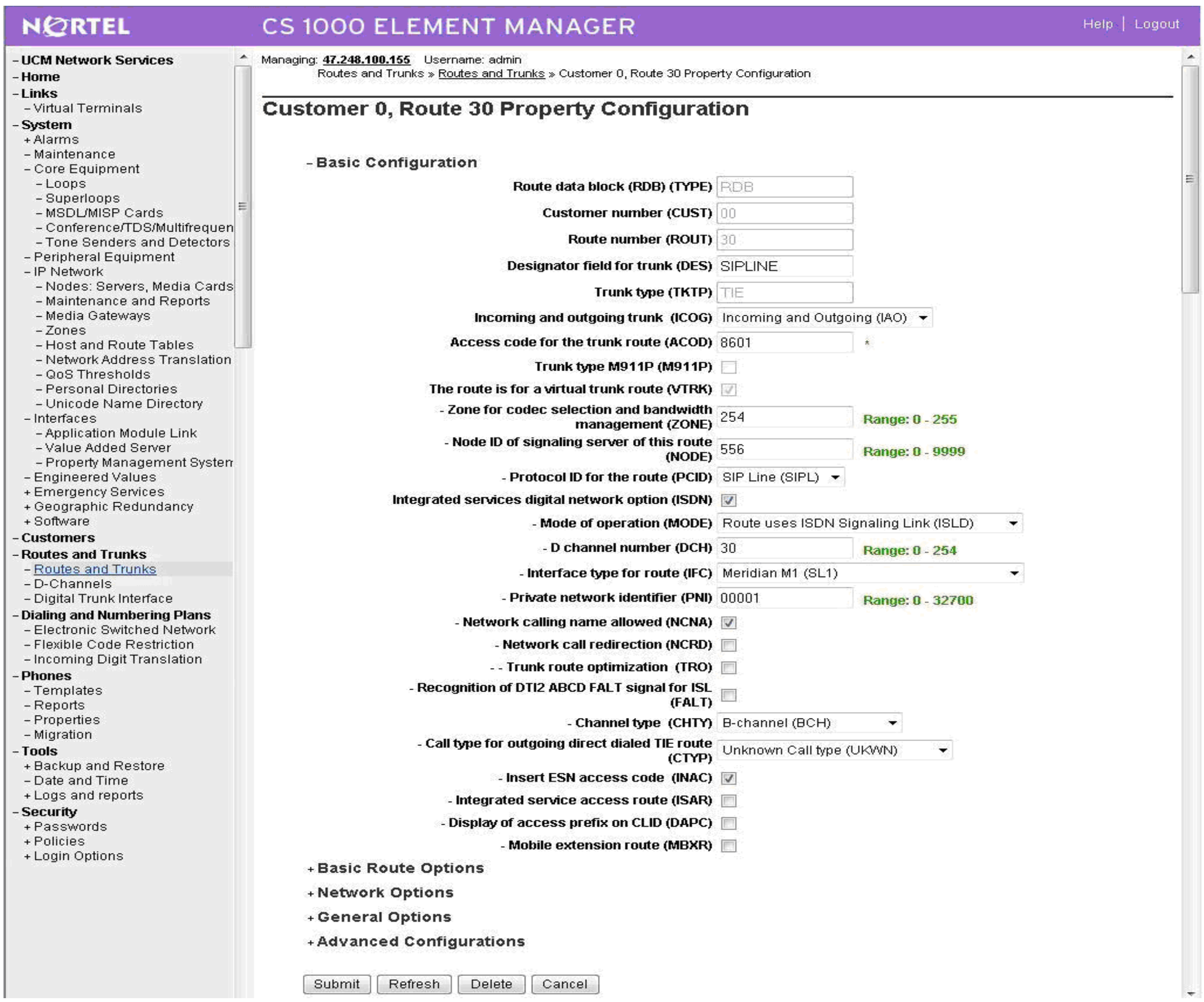

Figure 18 –SIP & SIP Line Route Configuration

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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***.

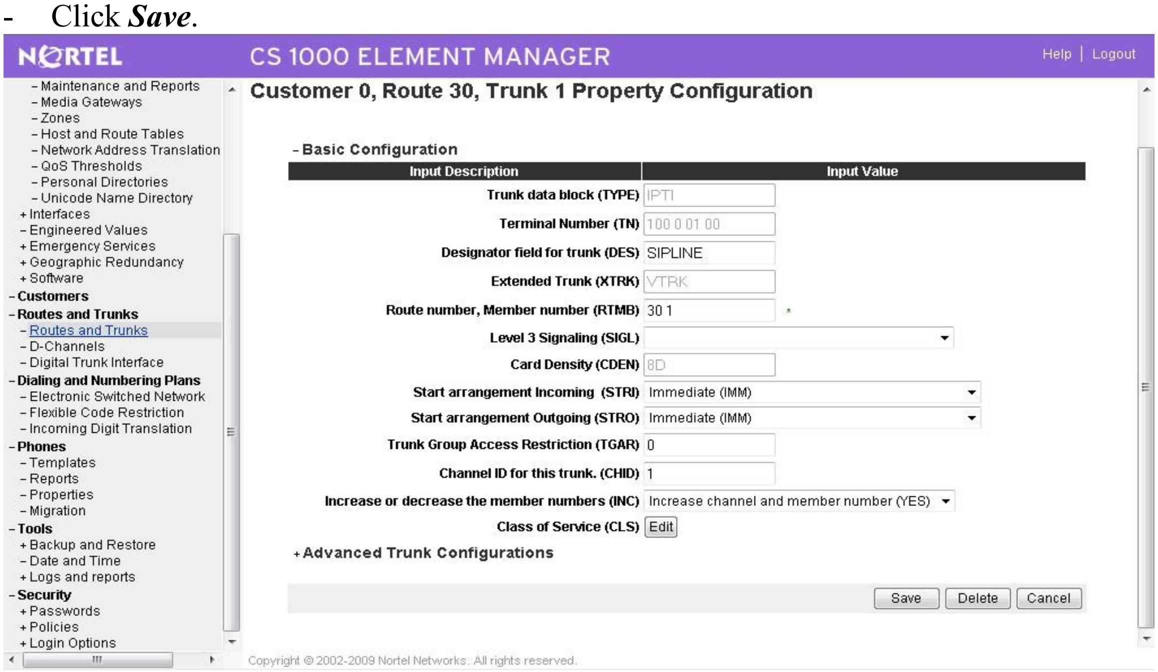

Figure 109 –SIPGW & SIP Line Trunk Configuration

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 RCBD
     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 MSDL 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 NTBK51BA Downloadable D-Channel Daughterboard. TMDI = TMDI (NTRB21) card.
DES	T1_QSIG	Designator field.
USR	PRI	D-channel is for ISDN PRI only. ***Note:*** 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 PR_TRIGS PR_TRIGS PR_TRIGS	DIV 2 3 CNG 2 3 CON 2 3 CTR2 2 3	Path Replacement Triggers

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)

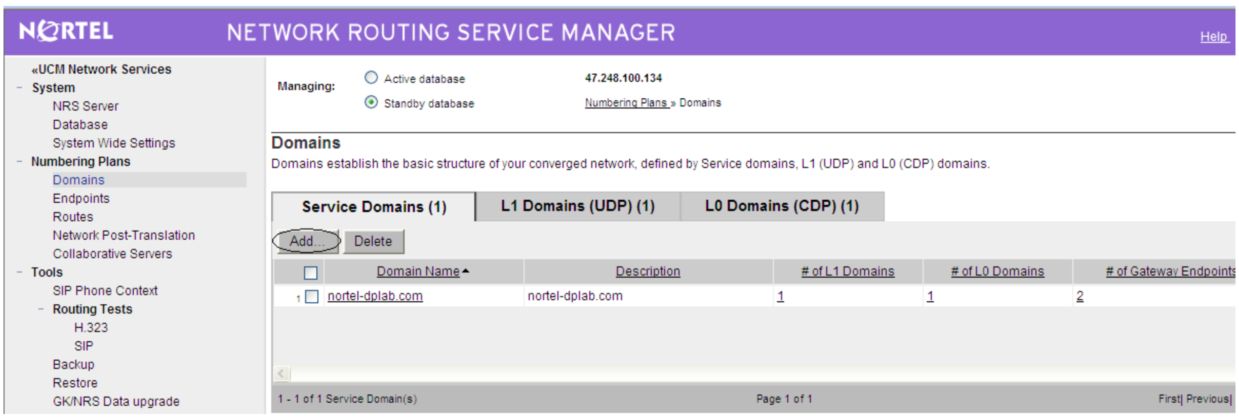

Figure 20: Numbering Plans: domains

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).

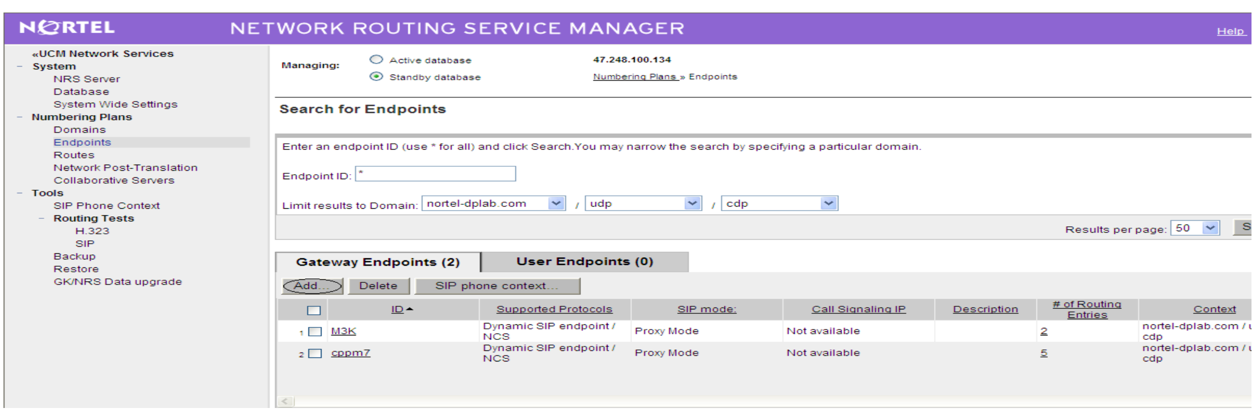

Figure 21: Numbering Plans: Endpoints

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
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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
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LOCALISDNRBSOURCE_32 = 0
LOCALISDNRBSOURCE_33 = 0
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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
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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
PLAYRBTONE2TRUNK_34 = -1
PLAYRBTONE2TRUNK_35 = -1

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
PLAYRBTONE2TRUNK_52 = -1
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PLAYRBTONE2TRUNK_57 = -1
PLAYRBTONE2TRUNK_58 = -1
PLAYRBTONE2TRUNK_59 = -1
PLAYRBTONE2TRUNK_60 = -1
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 = g711Ulaw64k, 10, 0, -1, 0;
CodersGroup0 4 = g711Ulaw64k, 20, 0, -1, 0;
CodersGroup0 5 = g711Ulaw64k, 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

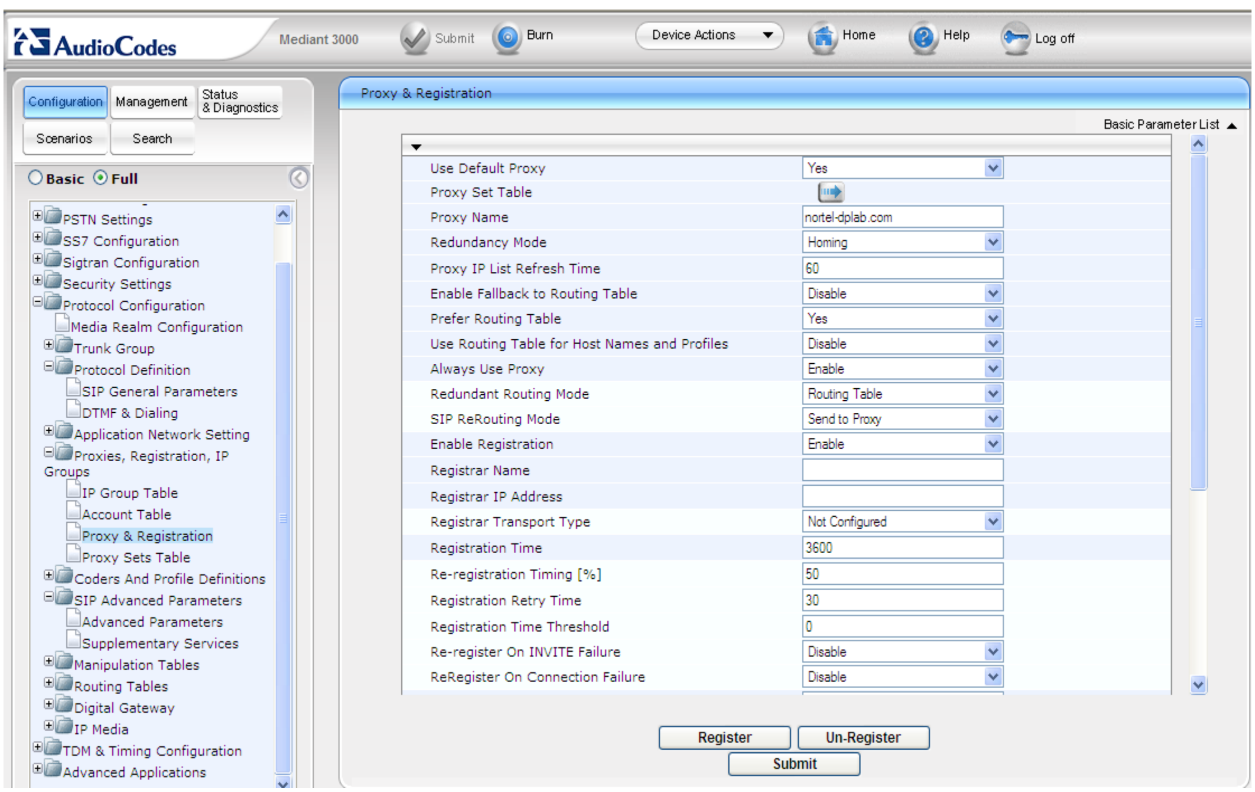


Figure 22: Proxy & Registration

- 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.

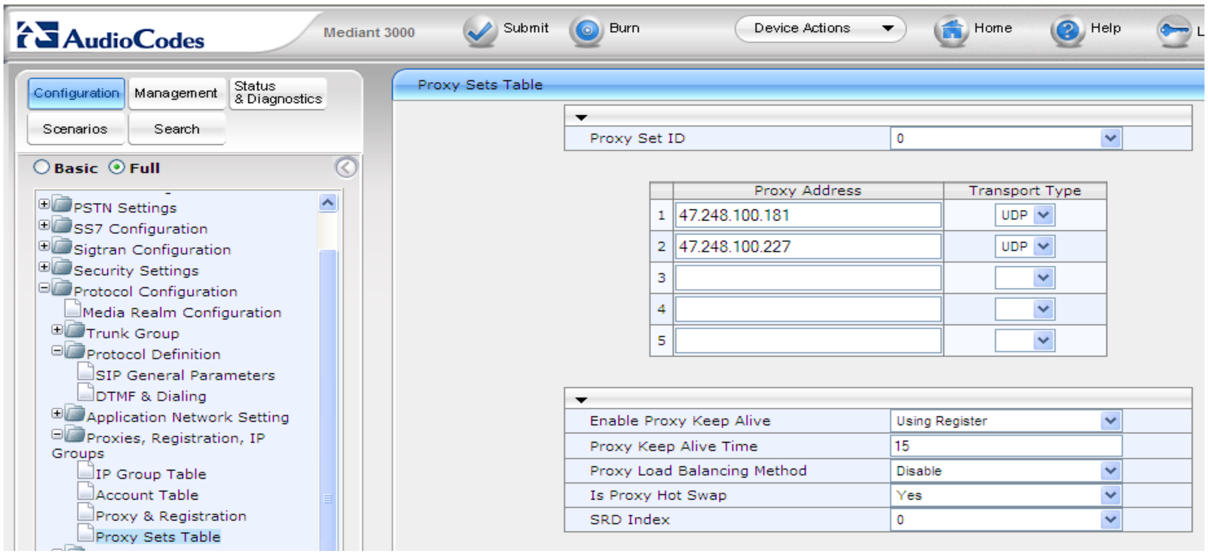


Figure 23: Proxy set tables

- 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).

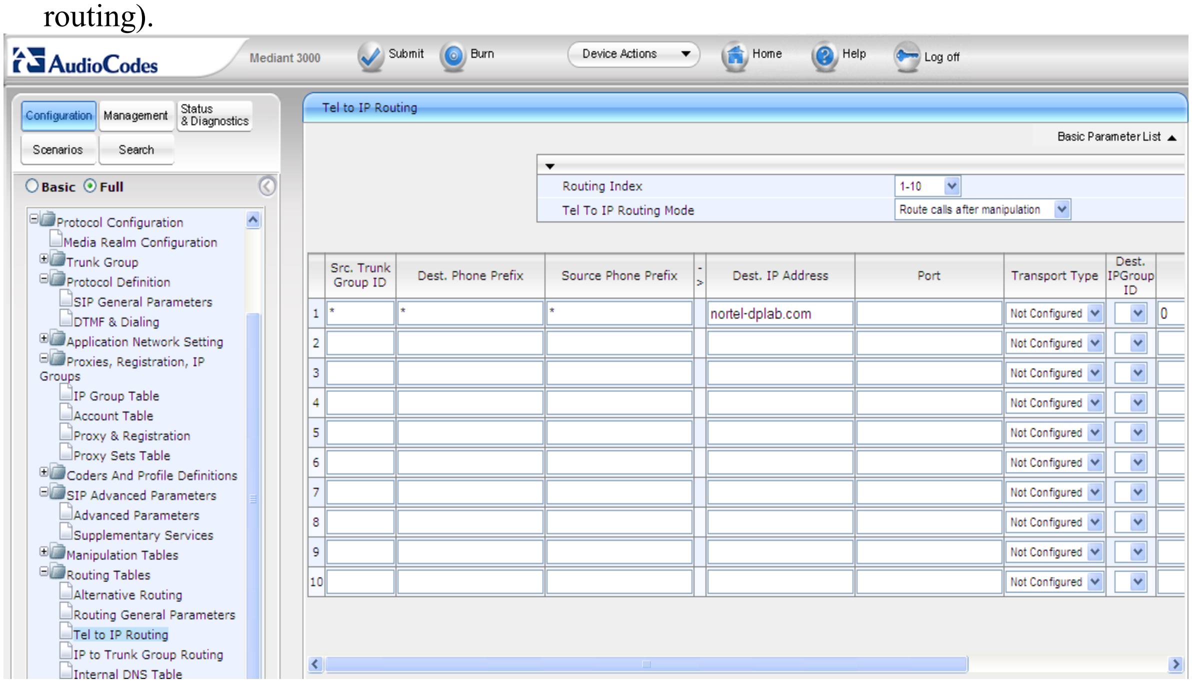


Figure 24: Tel to IP Routing

- 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.

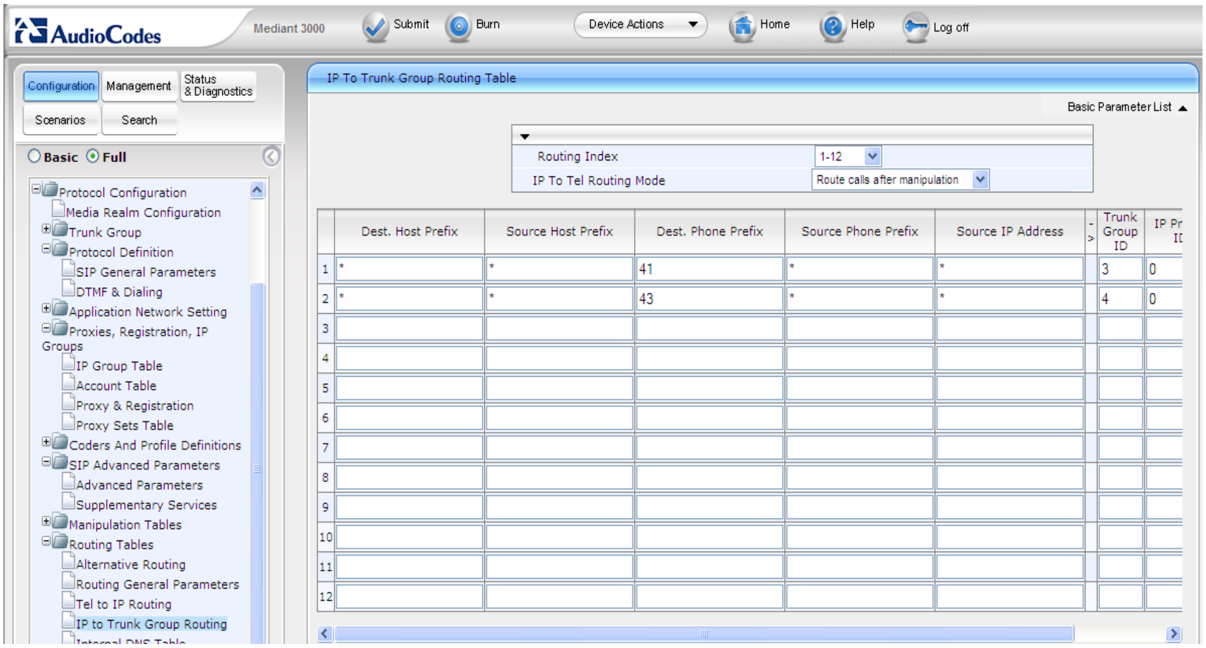


Figure 25: IP to Trunk Group Routing

- 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**.

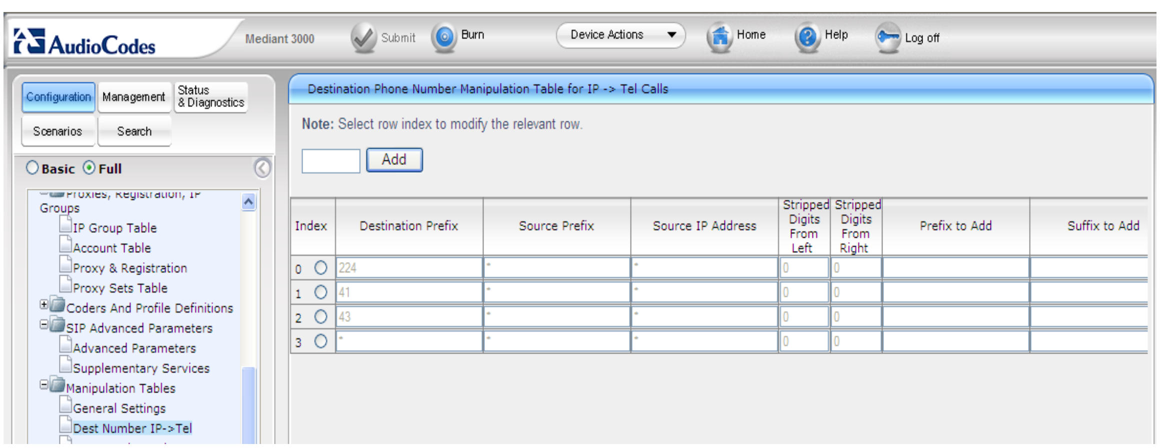
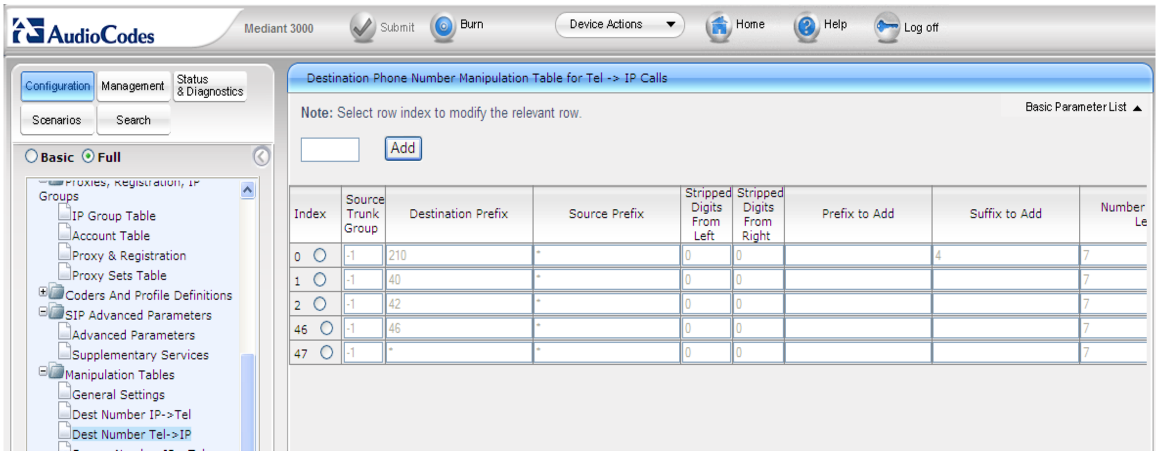


Figure 26: Dest Number IP->Tel

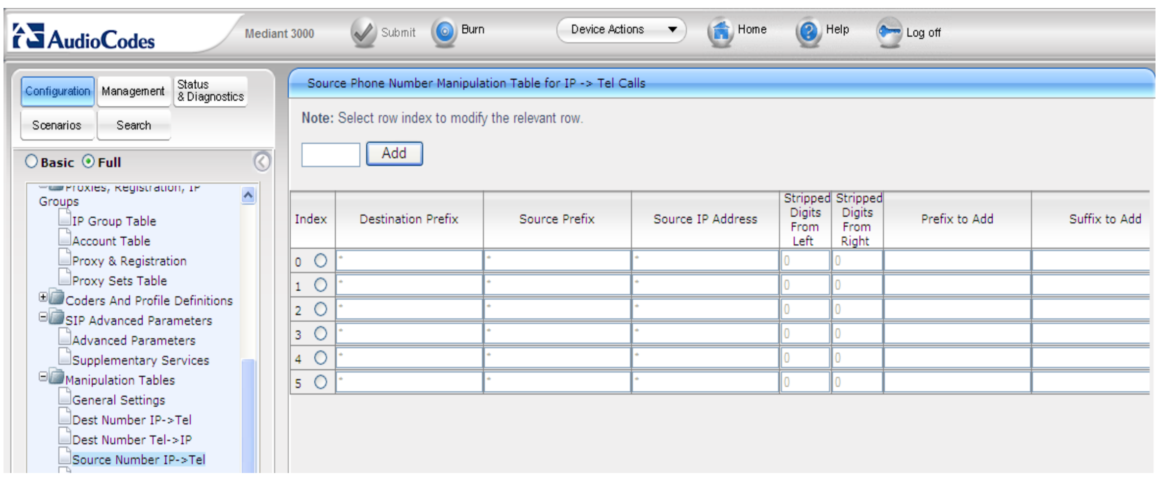
- Dest number Tel->IP: Manipulate number received from PSTN for the outgoing SIP Diversion, Resource-Priority, or History-Info header that is sent to IP. Shown below in **Figure 27**.



Index	Source Trunk Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number Le
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**.



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**.

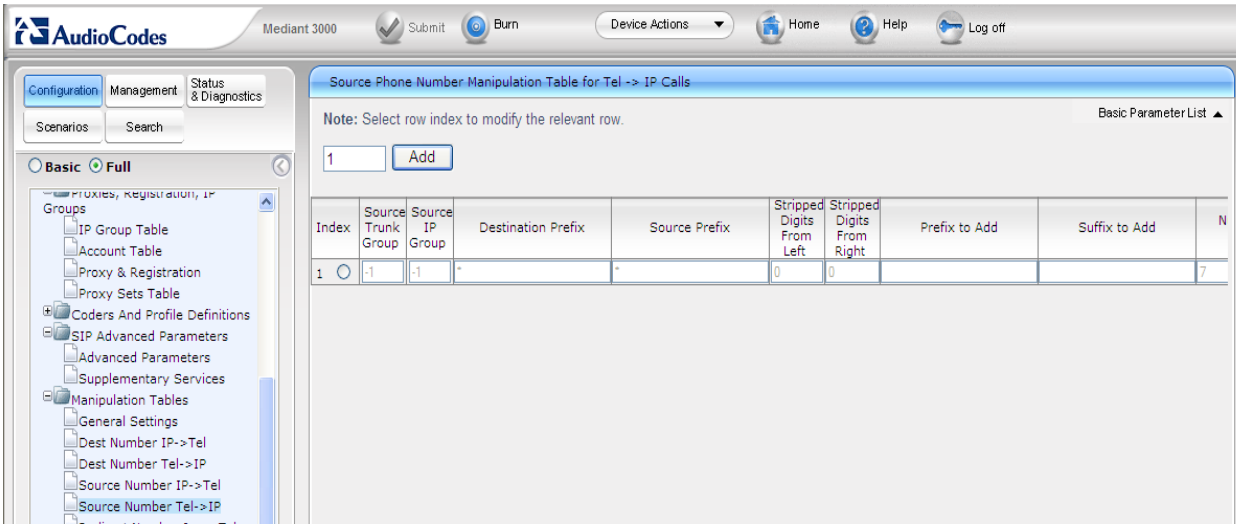


Figure 29: Source Number Tel->IP

- 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**.

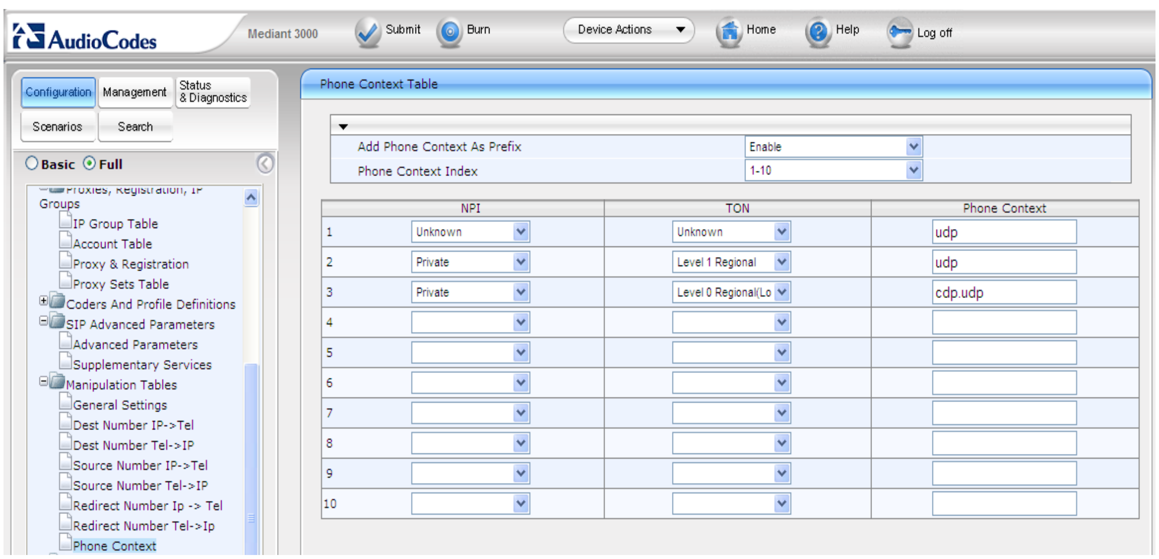


Figure 30: Phone-context

- 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.

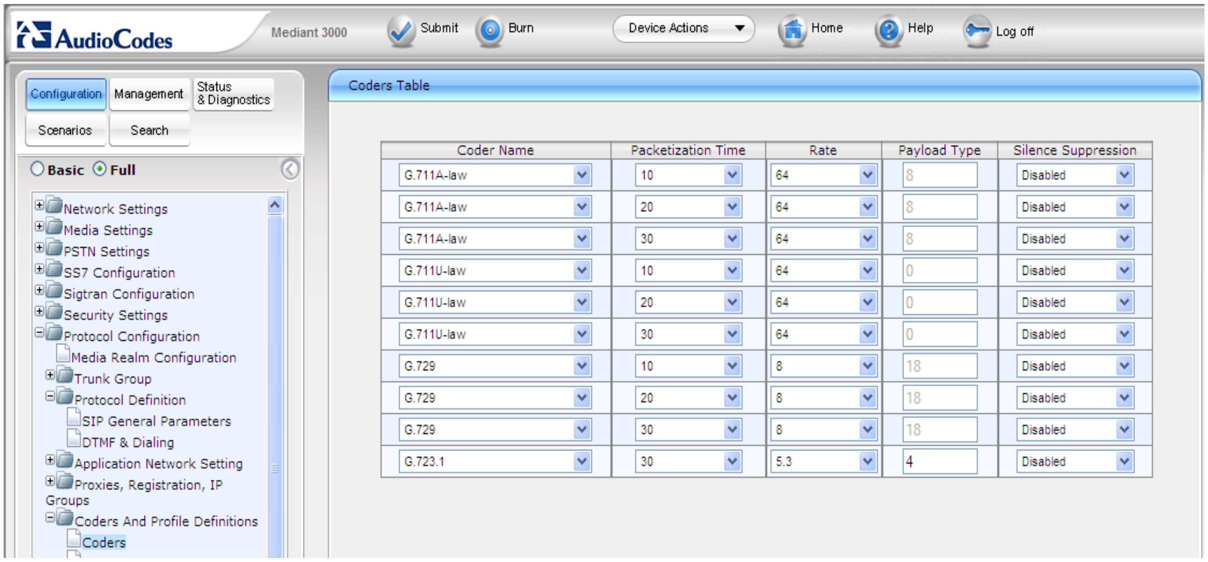


Figure 31: Coders

- 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**.

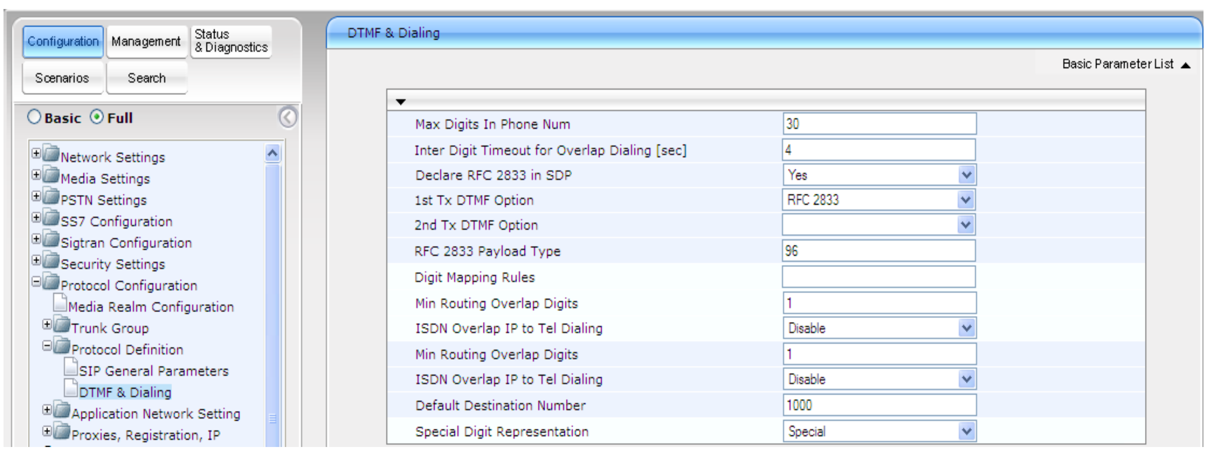


Figure 32: DTMF & Dialling

- 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

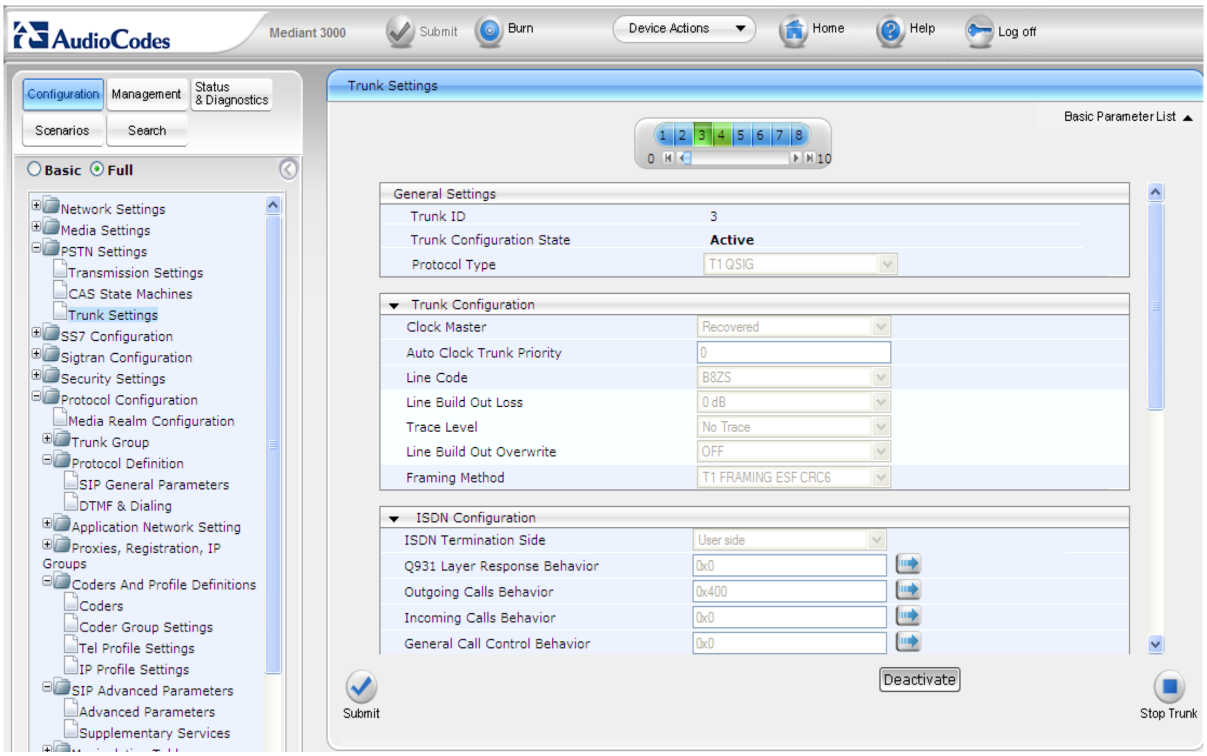


Figure 33: T1_QSIG

+T1_NI2_ISDN = National ISDN 2 PRI protocol

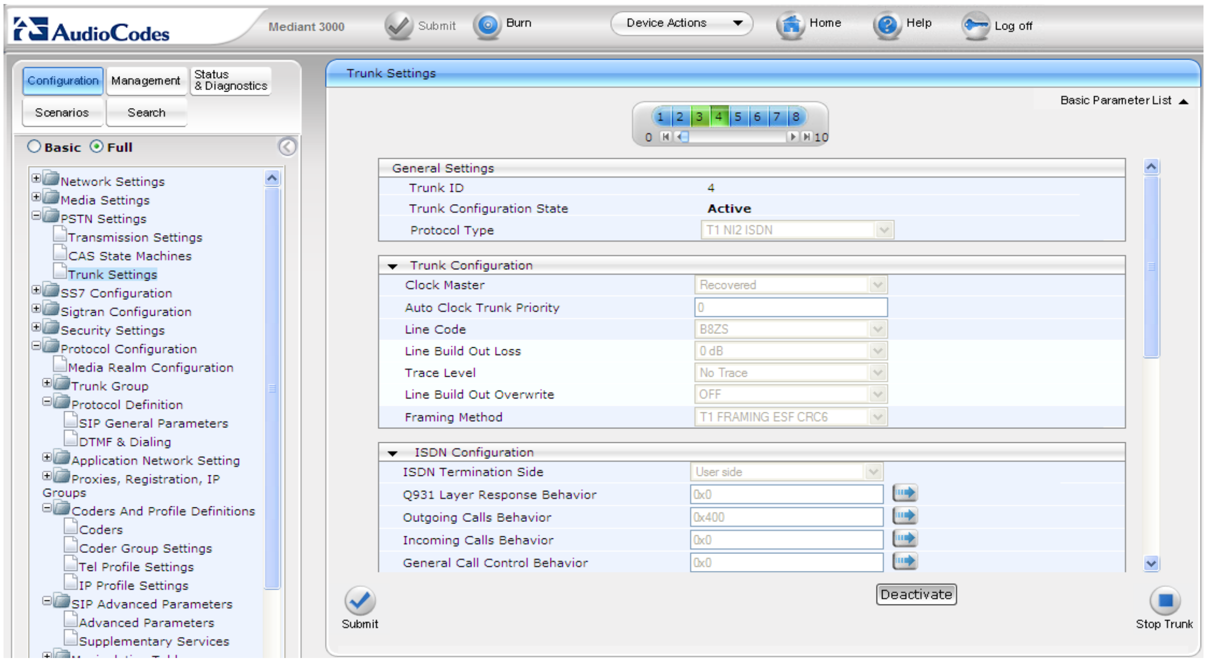


Figure 34: T1_NI2_ISDN

- 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).

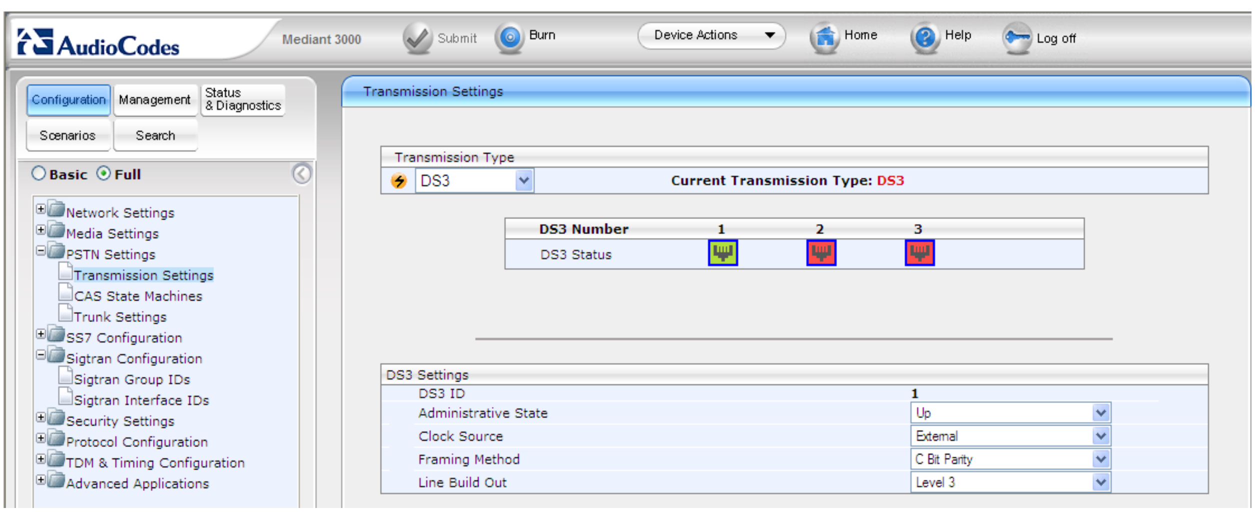


Figure 35: DS3 Transmission Settings

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

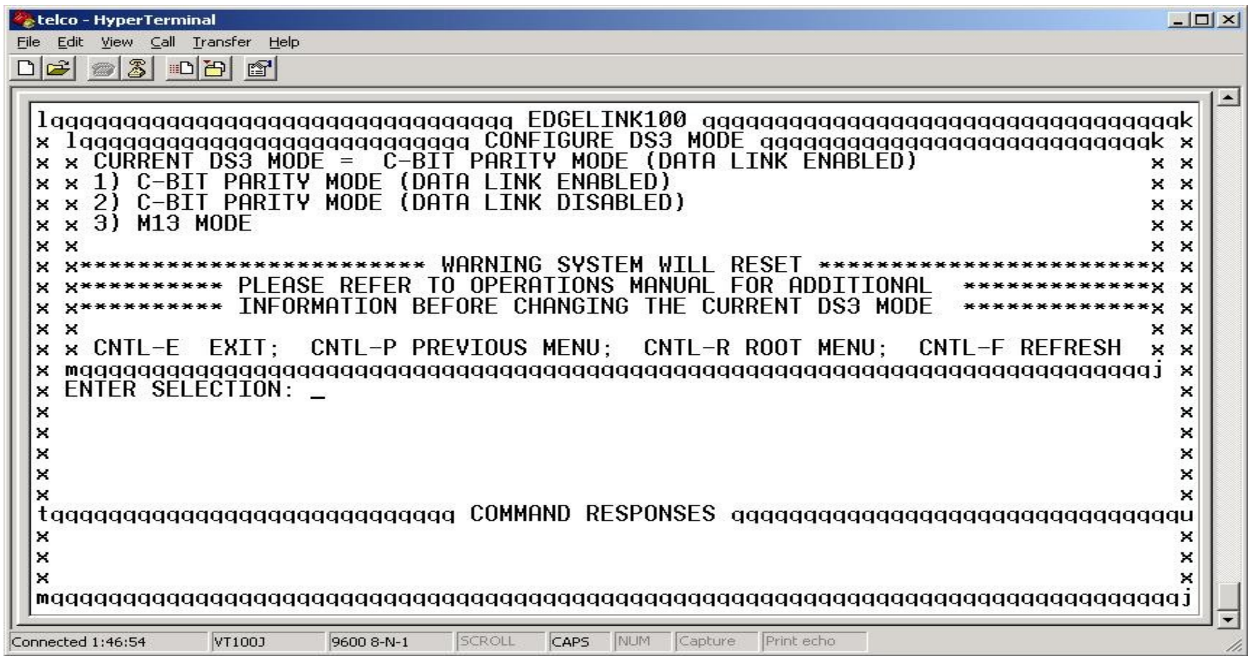


Figure 36: DS3 mode

- DS3 line buildout

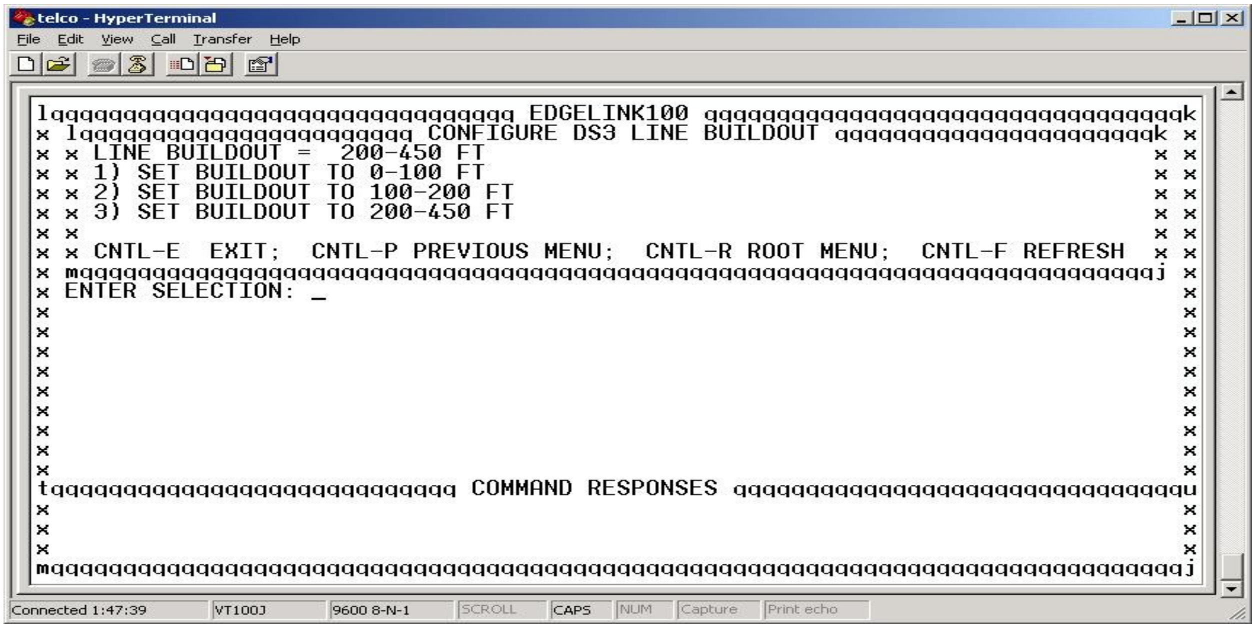


Figure 37: DS3 line buildout

- DS3 Loopback

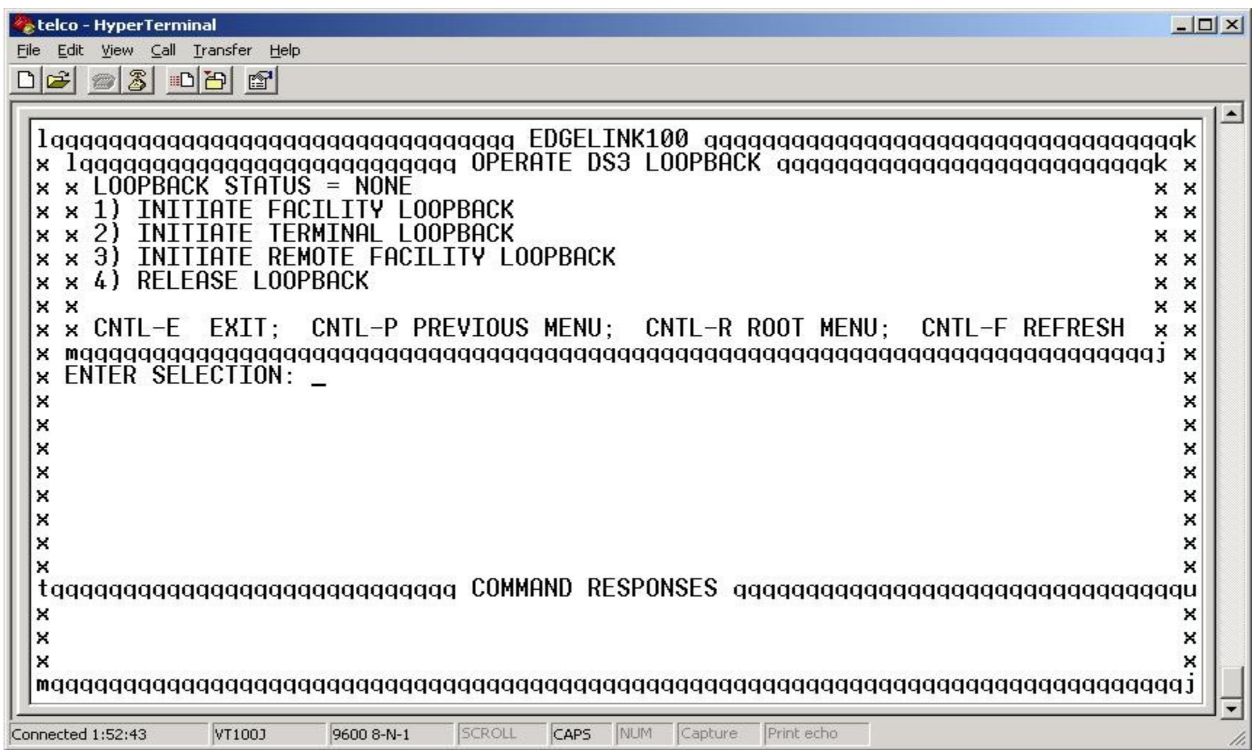


Figure 38: DS3 loop back

- **DS3 BER**

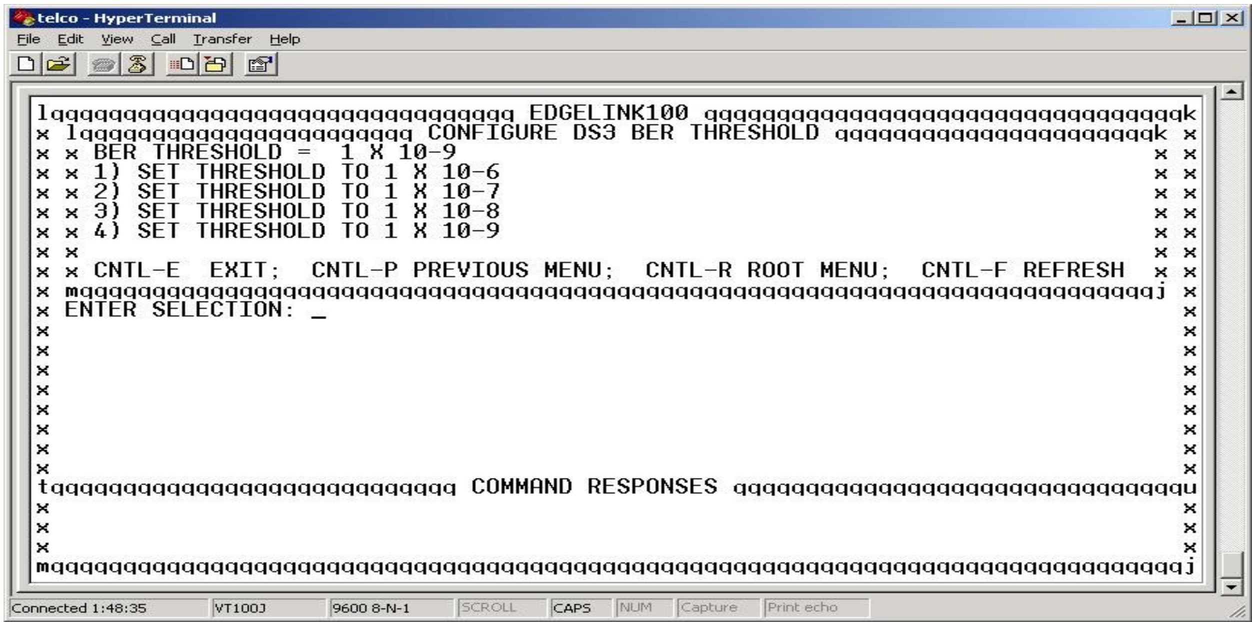


Figure 39: DS3 BER

- **DS3 Timing**

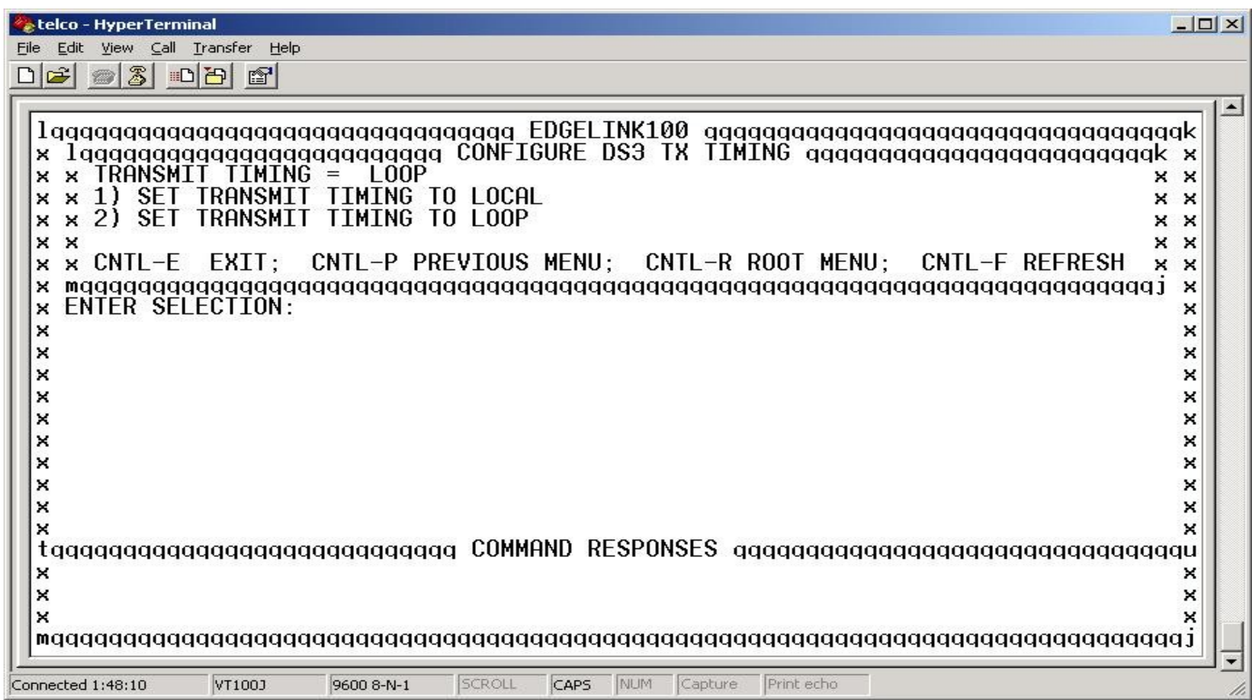


Figure 40: DS3 Timing

5.3.2. DS1 Line Code

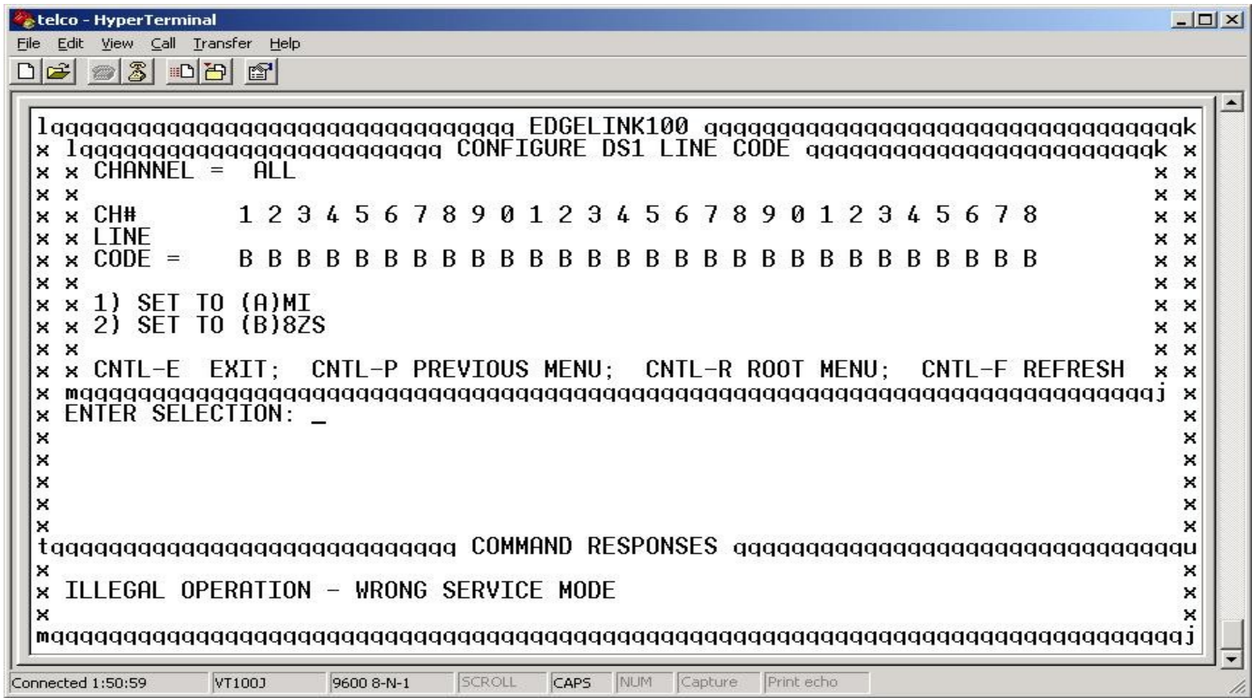


Figure 41: DS1 Line Code

- DS1 Line Buildout

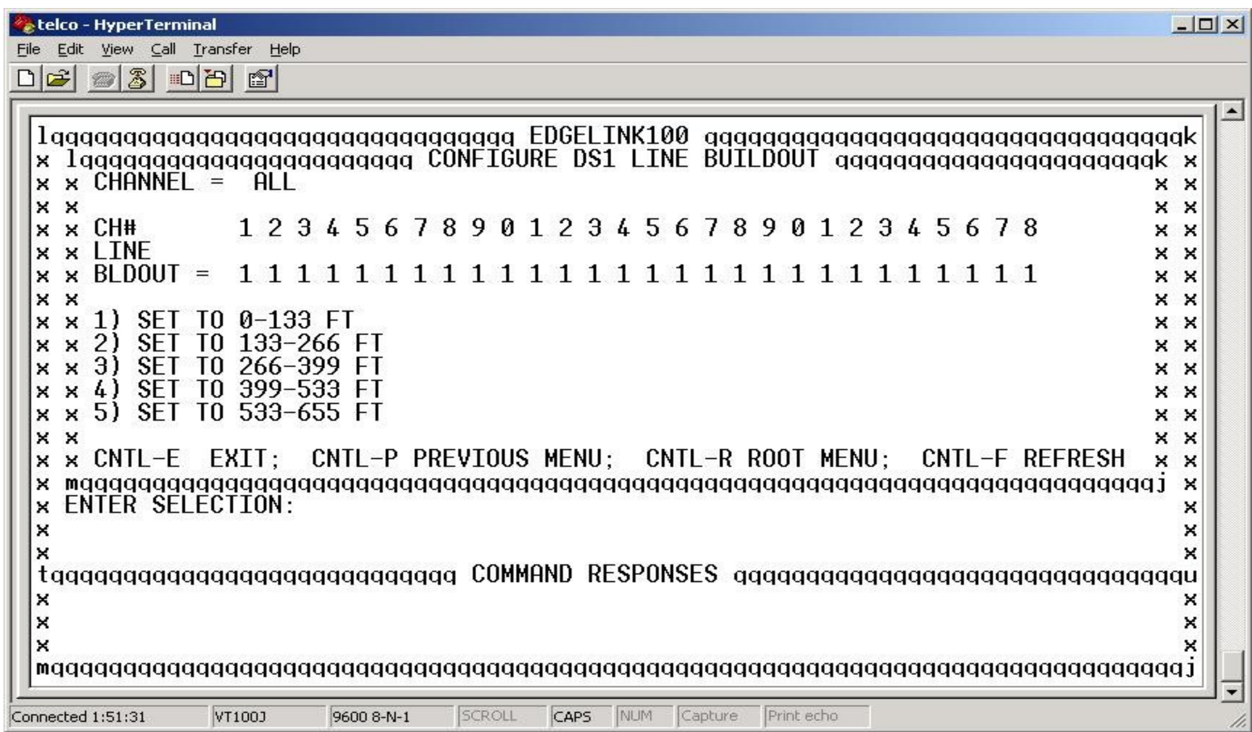


Figure 42: DS1 Line Buildout

- DS1 Loopback Mode

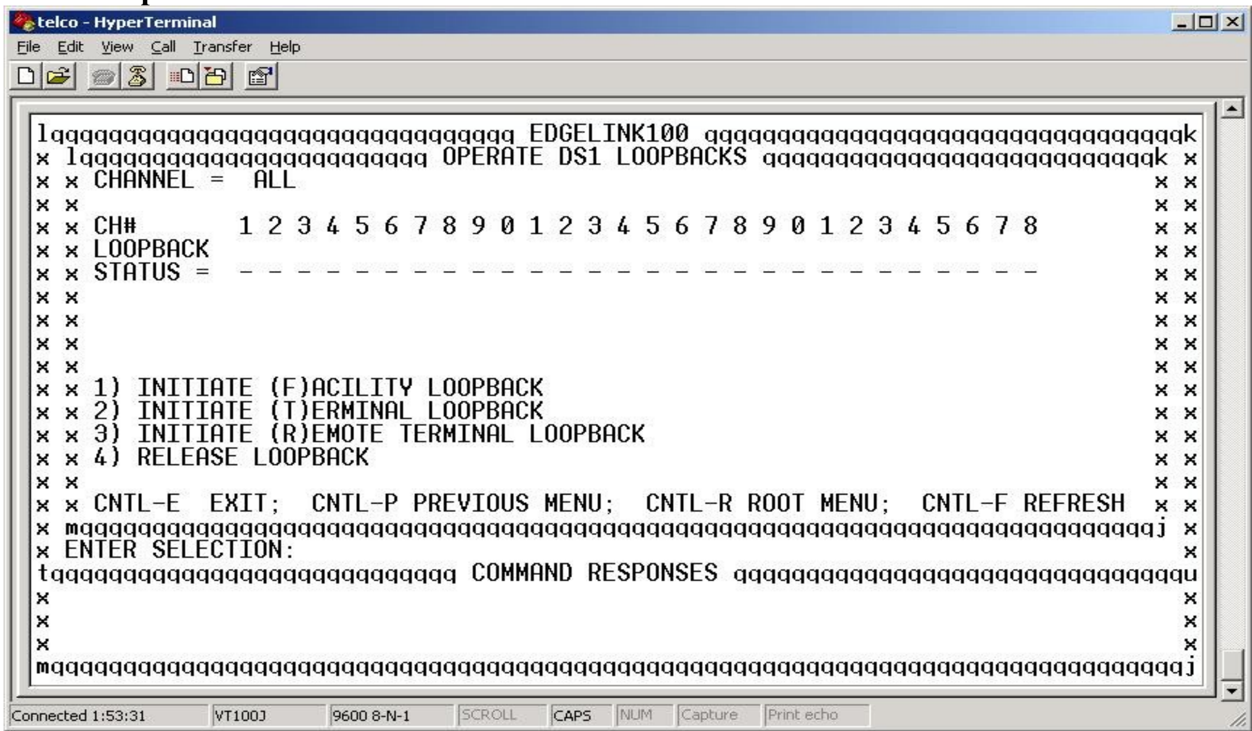


Figure 43: DS1 Loopback Mode

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-Edgelink100 and Edgelink100-CS1000

+ DS3 connection: Mediant3000 and EdgeLink100

- Diagram: refer lab diagram.

- Cable type: Copper Coaxial

- Cable length: 200-450 ft

- Cable pin outs:

Mediant3000		**EdgeLink100**
Tx	→	Rx
Rx	←	Tx

+ 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 1st pair of EdgeLink100 64 pins connector and CS1000 RJ21 50 pins connector.

Edgelink100 (64 pins)		**CS1000 (RJ21-50 pins)**
- DSX1- IN		
Rx Ring – Pin 1	←	23 Tx RingRing - Pin 23
Rx Tip - Pin 33	←	48 Tx Tip – Pin 48
- DSX1- OUT		
Tx Ring – Pin 1	→	24 Rx Ring – Pin 24
Tx Tip - Pin 33	→	49 Rx Tip – Pin 49

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.