Avaya Solution & Interoperability Test Lab

Front-Ending Avaya Communication Server 1000 R6.0 with an AudioCodes Mediant 1000 Modular Media Gateway to Support SIP Trunks to Avaya Meeting Exchange Enterprise Edition 5.2 – Issue 1.0

Abstract

These Application Notes present a sample configuration that uses an AudioCodes Mediant 1000 Modular Media Gateway as an E1 PRI-QSIG/SIP gateway to connect Avaya Communication Server 1000 (formerly known as Nortel Communication Server 1000) with Avaya Meeting Exchange Enterprise Edition 5.2.

For the sample configuration, Avaya Communication Server 1000E runs on a co-resident CPPM card while Avaya Meeting Exchange Application and Media servers run on two separate Avaya S8510 servers.

1 Introduction

There are many installations of Avaya Communication Server 1000 which are not SIP or IP capable, or where the software release may not have been SIP interoperability tested, but the customer wishes to deploy Avaya Meeting Exchange Enterprise Edition. In this case an effective solution is to front-end the Avaya Communication Server 1000 with a PRI-QSIG/SIP gateway, which then signals on SIP trunks to Avaya Meeting Exchange Enterprise Edition. This configuration supports basic and advanced conference features. **Figure 1** shows a sample configuration that uses an AudioCodes Mediant 1000 Modular Media Gateway to front-end the Avaya Communication Server 1000 via an E1/PRI QSIG connection. The Mediant 1000 supports a SIP trunk to the Avaya Meeting Exchange Enterprise Edition Application Server. All outbound calls from the Telephones to the conference bridge are routed via this trunk. The SIP trunk uses TCP for transporting the SIP signalling messages while DTMF is transmitted within the RTP stream using RFC2833 compliant messages.

The Avaya 2050 IP Softphone (UNIStim), Avaya M3820 Digital Telephone, Avaya 1120E UNIStim IP Telephone and Avaya 1140E SIP Telephone are supported by Avaya Communication Server 1000E. An analogue phone is attached to a PSTN simulator which is connected with an E1 PRI trunk to the sample telephony network. This configuration enables PSTN users to participate in a conference with other enterprise users on the private network. The Telephones are configured in the 3xxx extension range, while the conference access number (DNIS) on the Meeting Exchange is set to 66666.

For the sample configuration, Avaya Communication Server 1000E runs on a co-resident CPPM card while Avaya Meeting Exchange Application and Media servers run on two separate Avaya S8510 servers.

These Application Notes will focus on configuration of the QSIG and SIP trunks, dial plan, call routing, and conference bridge. Detailed administration of the telephones will not be described (see the appropriate documentation listed in **Section 9**).

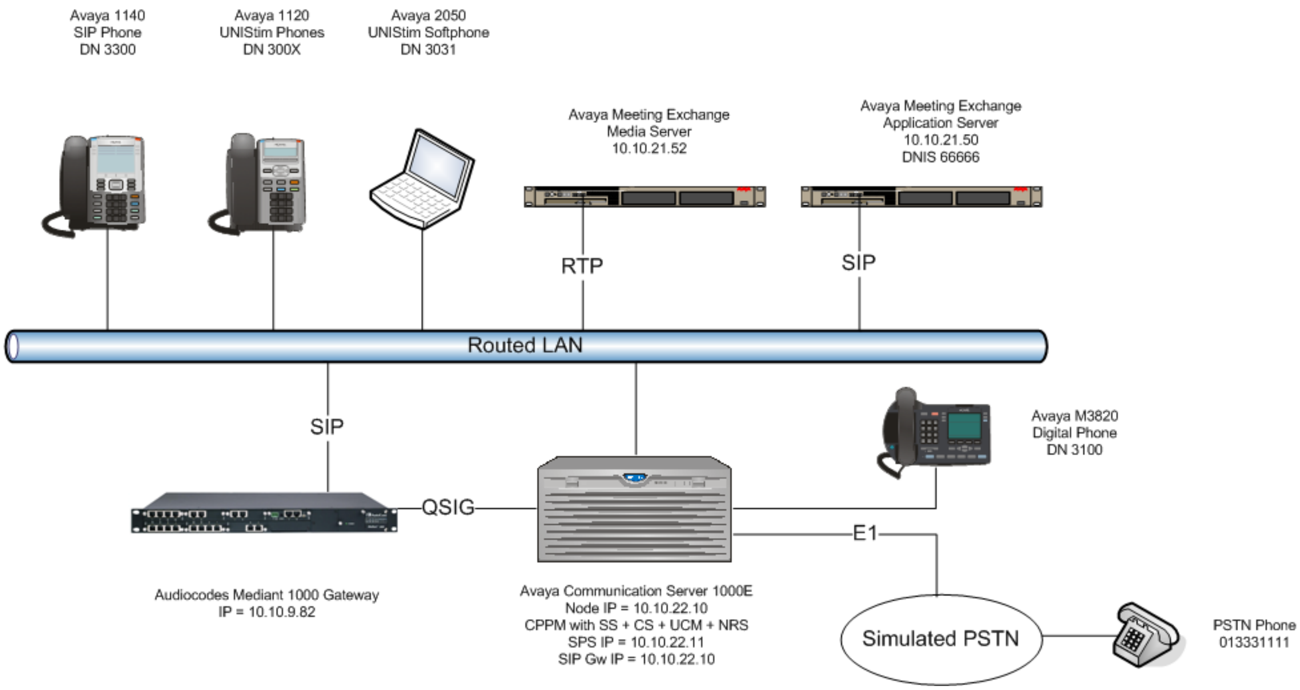


Figure 1 – CS1000 with AudioCodes and Meeting Exchange Enterprise

2 Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software Version
Avaya Communication Server 1000E • NTBK50AAE5 – E1 PRI card	Release 600R, Version 4121 N/A
Avaya 1140 IP Phone	SIP 02.02.21.00
Avaya 1120E IP Phone	UNIStim 0624C60
Avaya IP SoftPhone 2050	UNIStim 3.04.0003
Avaya M3820 Digital Phone	N/A
Avaya S8510 Server	Avaya Meeting Exchange Enterprise Edition Application Server S6200 R5.2 Build 5.2.1.0.4 (GA)
Avaya S8510 Server	Avaya Meeting Exchange Enterprise Edition Media Server S6200 R5.2 Build 5.2.1.0.4 (GA)
AudioCodes Mediant 1000 Modular Media Gateway	5.80A.033

Table 1: Version Numbers of Equipment and Software

3 Configure Avaya Communication Server 1000

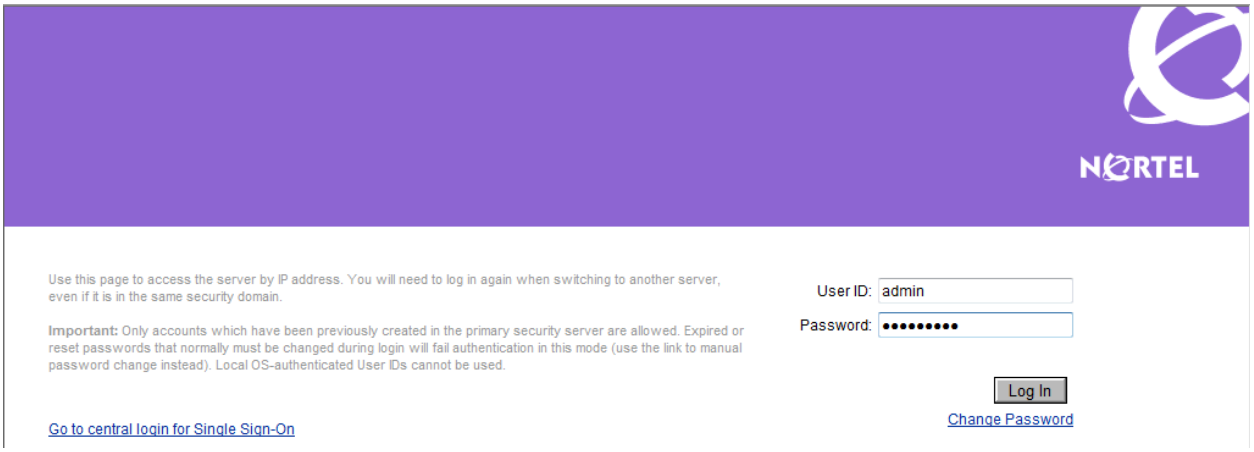
This section focuses on configuring the E1 QSIG trunks on Avaya Communication Server 1000 to reach the AudioCodes M1000. The NTBK50AAE5 E1 2Mbps ISDN trunk card is installed in slot 2 of the Avaya MG 1000E Media Gateway. These Application Notes assume that the telephones are installed and configured and ISDN PRI is not being configured for the first time, so error detection thresholds and clock synchronization control are assumed to be in place. If not, refer to the ISDN Primary Rate Interface document in **Section 8** for detailed descriptions. Furthermore, these Application Notes used the Coordinated Dial Plan (CDP) feature to route calls from the Avaya Communication Server 1000, over the E1 QSIG trunks to Avaya Communication Manager. The CDP feature is assumed to be already enabled on Avaya Communication Server 1000, and therefore will not be described in detail.

The procedures below describe the details of configuring Avaya Communication Server 1000:

- Log in to the Unified Communications Management GUI
- Launch Element Manager
- Verify Equipped Feature Packages
- Administer E1 card
- Administer D-Channel
- Administer routes and trunks
- Administer route list block
- Administer distant steering code
- Enable E1 card
- Enable D-channel automatic establishment

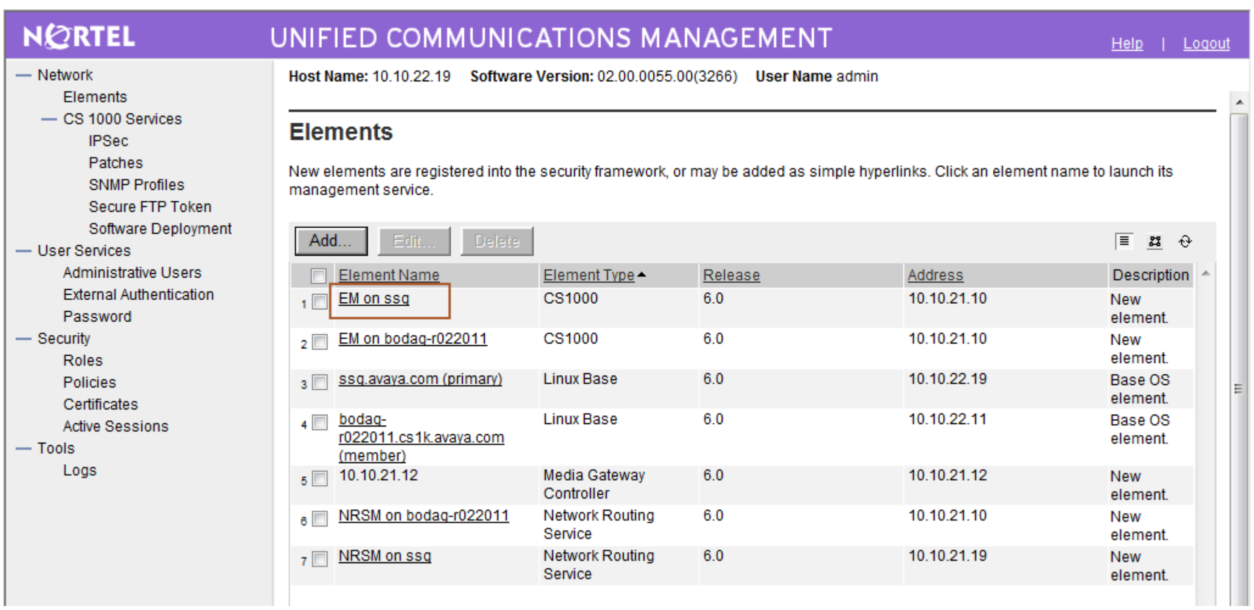
3.1 Log in to the Unified Communications Management GUI

Open an instance of a web browser and connect to the UCM GUI at the following address: **http://<node IP address>.** Log in using an appropriate Username and Password.



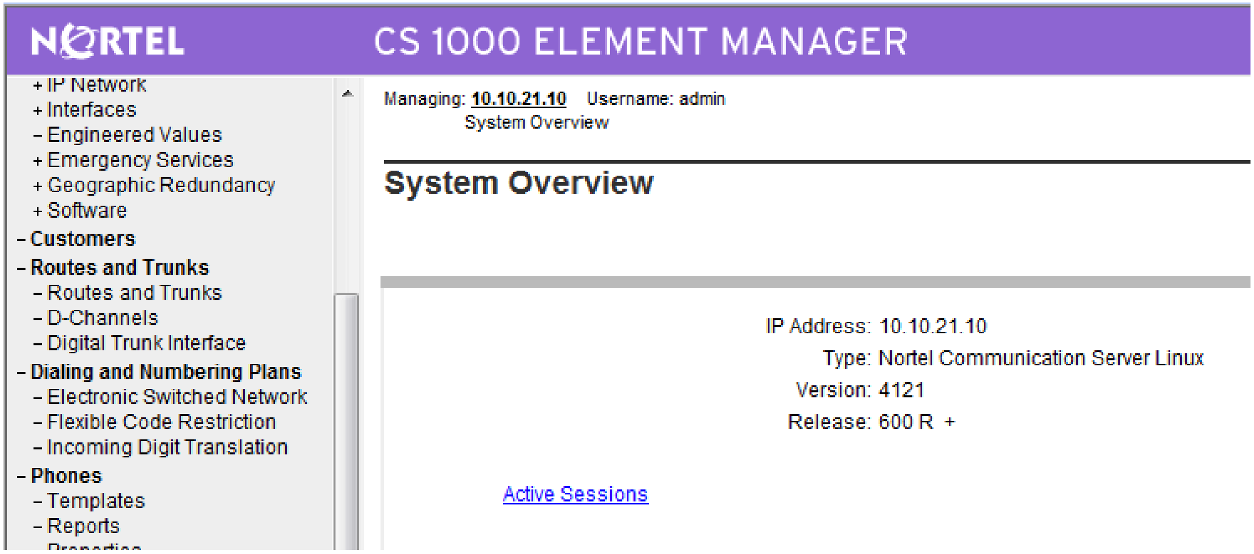
3.2 Launch Element Manager

The **Elements** screen is displayed. In the **Element Name** column click on the active node link. In the sample configuration "**EM on ssg**".



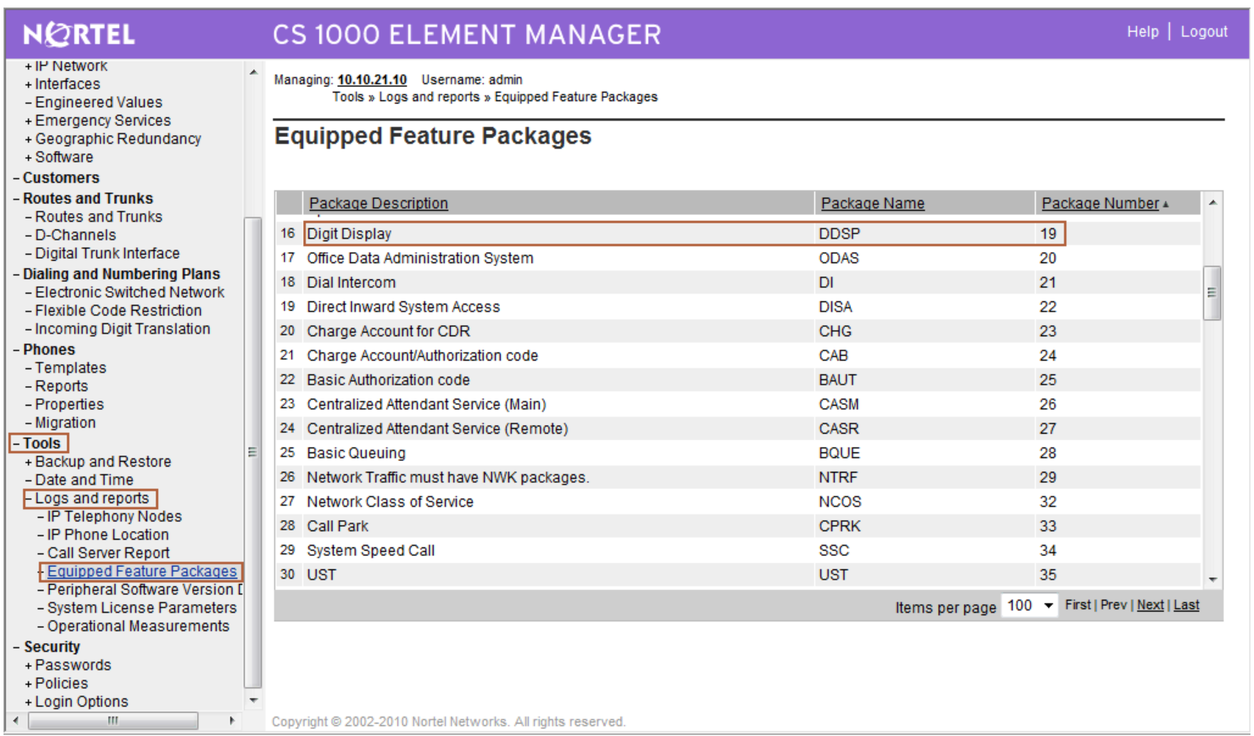
3.3 Verify Equipped Feature Packages

The **System Overview** screen is displayed.



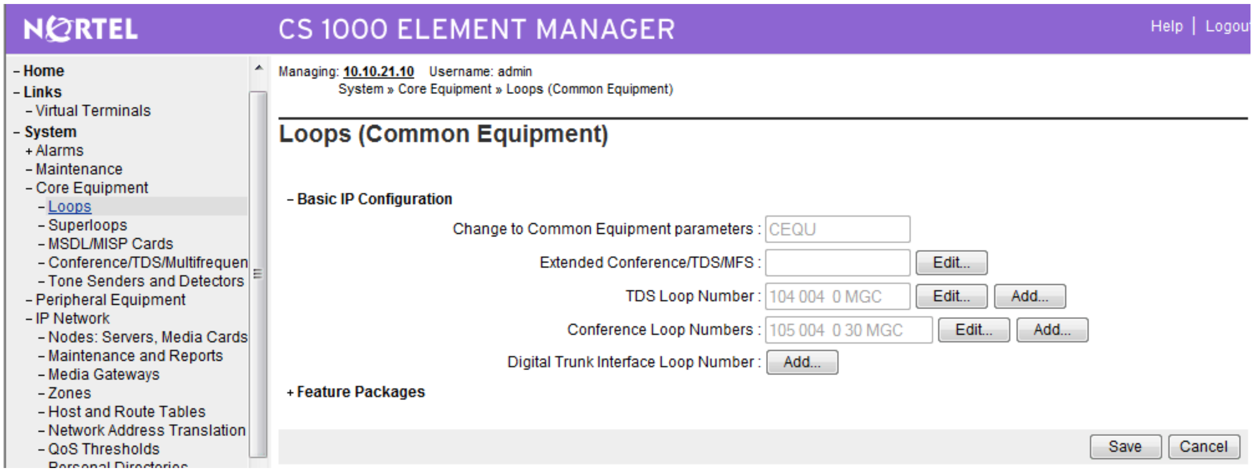
Select **Tools → Logs and reports →Equipped Feature Packages.** The **Equipped Feature Packages** screen is displayed next and shows a listing of the licensed feature packages in sequential order by package number. Scroll down the right pane as necessary to verify that the following feature packages are equipped:

- **19 Digit Display (DDSP)**
- **59 Coordinated Dialing Plan (CDP)**
- **95 Calling Party Name Display (CPND)**
- **145 Integrated Services Digital Network (ISDN)**
- **146 Primary Rate Access (CO) (PRA)**
- **154 2.0 Mb/s Primary Rate Interface (PRI2)**
- **184 Overlap Signaling (M1 to M1 and M1 to 1TR6 CO) (OVLP)**
- **202 International Primary Rate Access (CO) (IPRA)**
- **263 QSIG Interface (QSIG)**
- **305 QSIG Generic Functional protocol (QSIGGF)**
- **316 QSIG Supplementary service (QSIG-SS)**

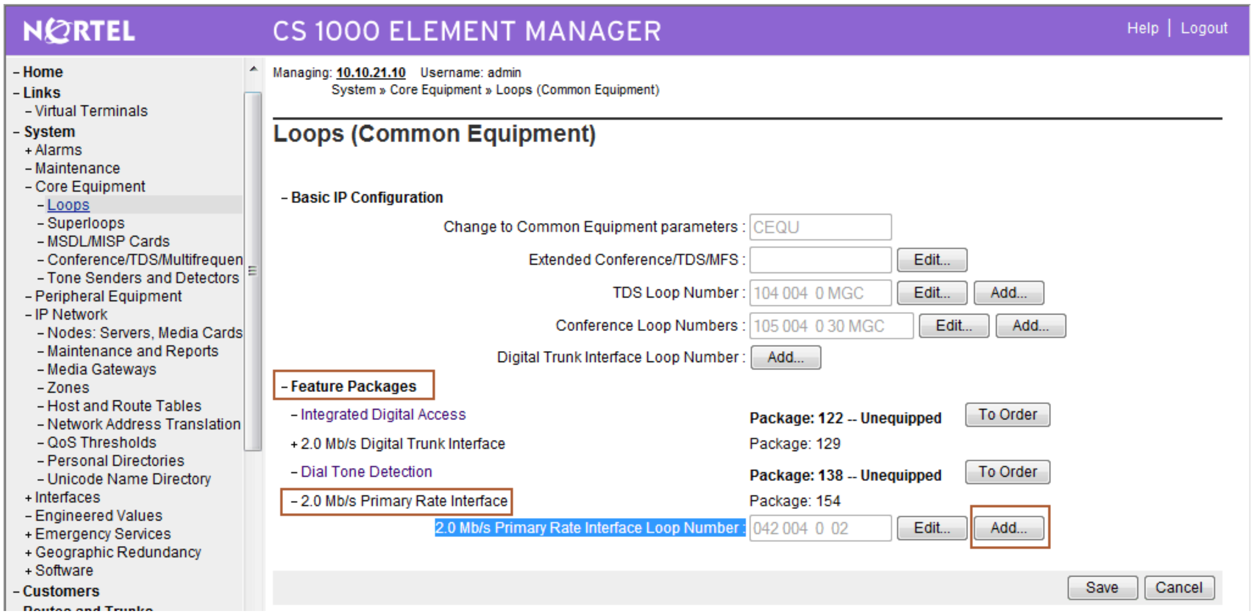


3.4 Administer E1 Card

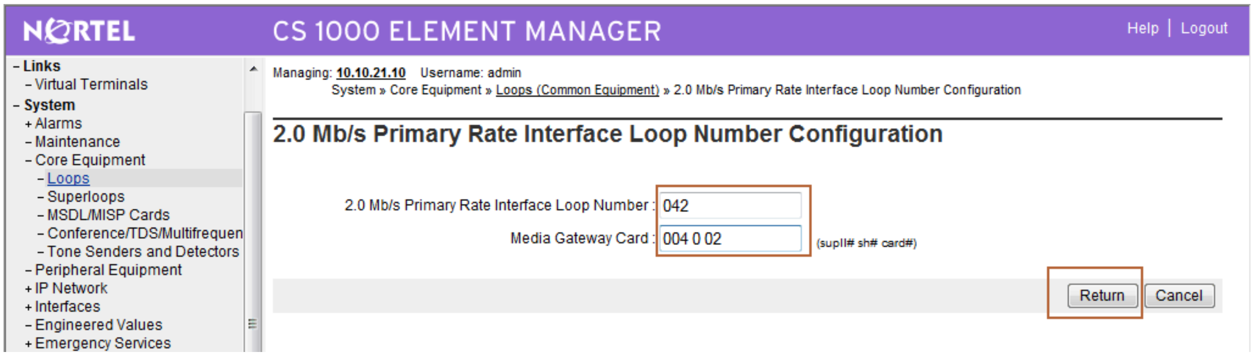
Select **System → Core Equipment → Loops** from the left pane to display the **Loops (Common Equipment)** screen.



Expand **Feature Packages** then expand **2.0 Mb/s Primary Rate Interface**. Click **Add…** on the right-hand side of **2.0 Mb/s Primary Rate Interface Loop Number.**



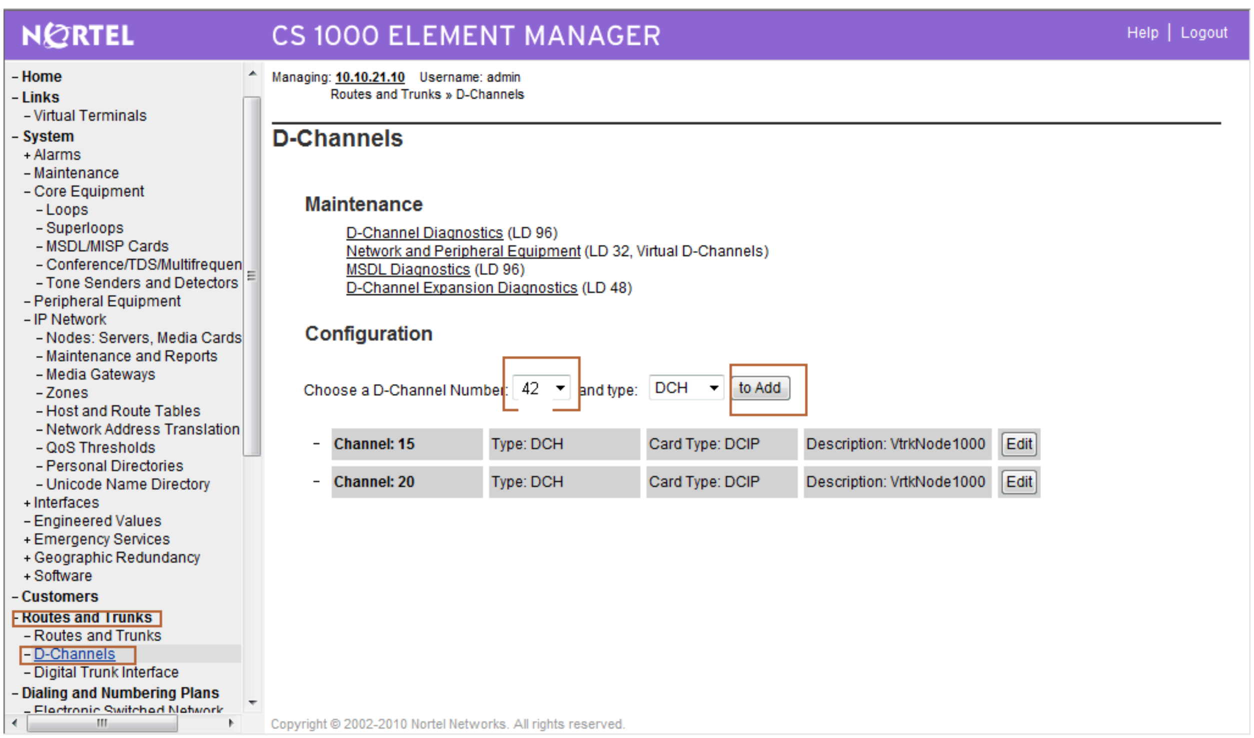
The **2.0 Mb/s Primary Rate Interface Loop Number Configuration** screen is displayed.



Select the Loop, Shelf, Cabinet, Card number corresponding to the physical slot location of the E1 PRI card. In this case, “Loop 042”, “Shelf 004”, “Cabinet 0” and “Card 02” is selected as the NTBK50AAE5 E1 2Mbps ISDN trunk card is installed in slot 2 of the Avaya MG 1000E Media Gateway. Click **Return**.

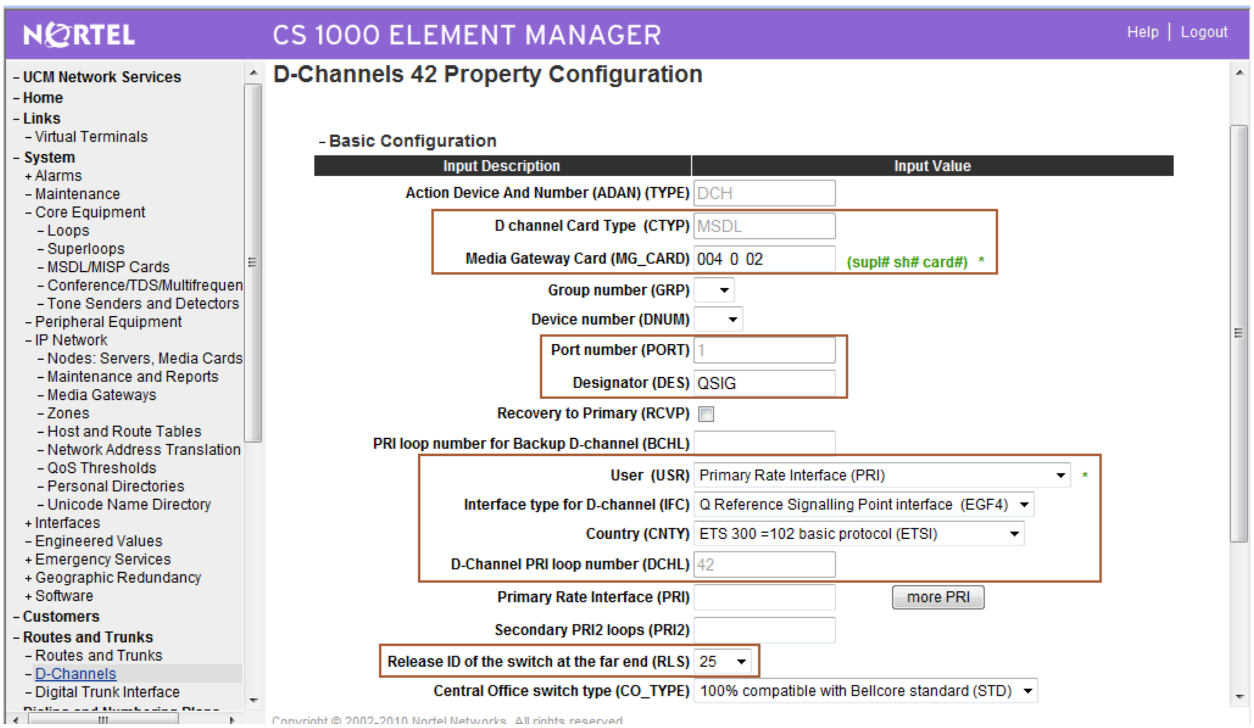
3.5 Administer D-Channel

Select **Routes and Trunks → D-Channels** from the left pane to display the **D-Channels** screen. In the **Choose a D-Channel Number** field, select an available D-channel from the drop-down list (in the sample configuration **42**). Click **to Add**.



The **D-Channels 42 Property Configuration** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields. The E1 card is installed in Shelf 004, Cabinet 0 and Slot 02.

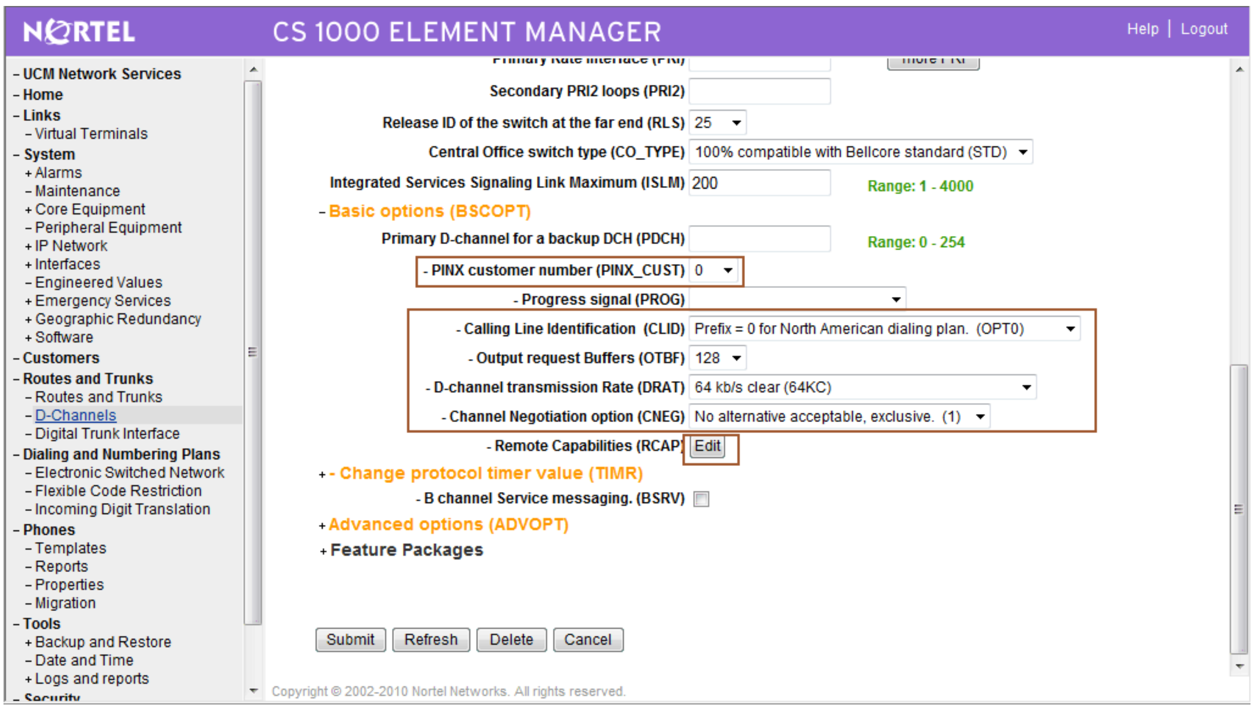
- **D channel Card Type (CTYP):** **MSDL**
- **Media Gateway Card (MG_CARD)** Type **004 0 02**
- **Port number (PORT):** **1**
- **Designator (DES):** A descriptive text
- **User (USR):** **Primary Rate Interface (PRI)**
- **Interface type for D-channel (IFC):** **Q Reference Signaling Point interface (EGF4)**
- **Country (CNTY):** **ETS 300 = 102 basic protocol ETSI**
- **D-Channel PRI loop number (DCHL)**: The digital trunk interface loop number from **Section 3.4**
- **Release ID of the switch at the far end (RLS)** **25**



Scroll down and expand **Basic options (BSCOPT).** Enter the following values for the specified fields, and retain the default values for the remaining fields.

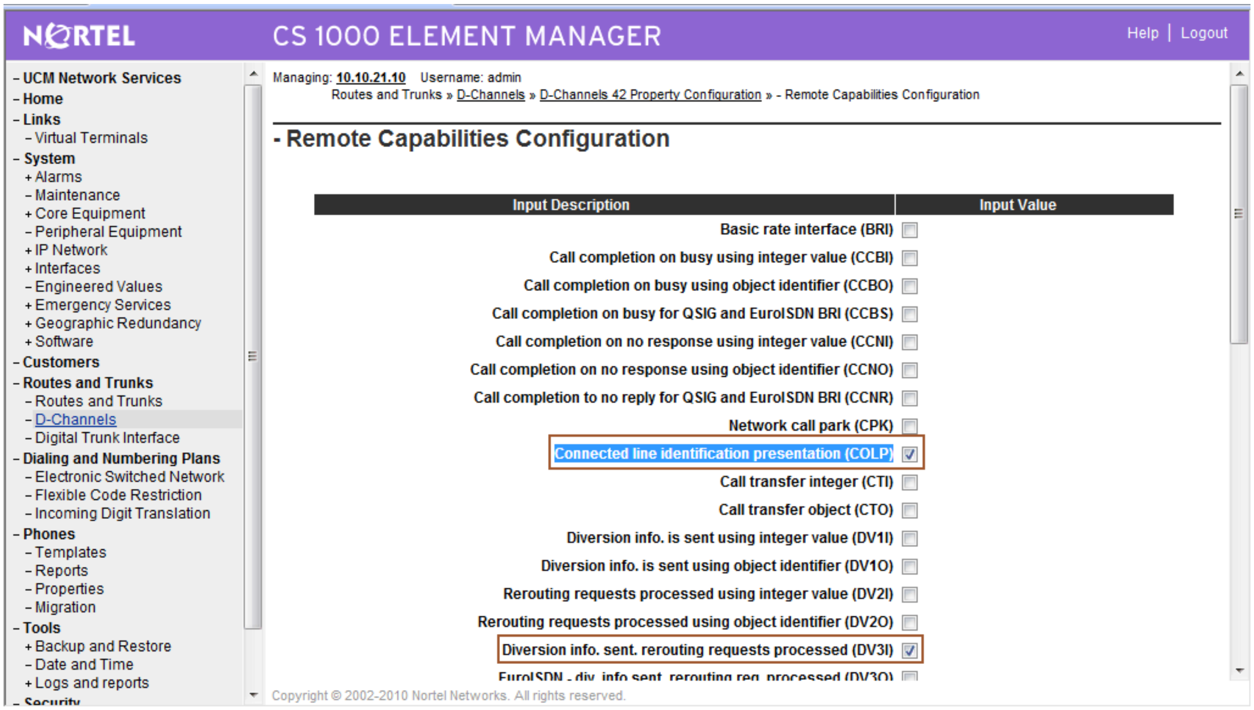
- **PINX customer number (PINX_CUST)** Select the appropriate customer number. In the sample configuration "0"
- **Calling Line Identification (CLID)** Select **Prefix = 0 for North American dialing plan (OPT0)**
- **Output request Buffers (OTBF)** Select **128**
- **D-channel transmission Rate (DRAT)** Select "64 kb/s clear (64KC)"
- **Channel Negotiation option (CNEG)** Select **No alternative acceptable, exclusive (1)**

Retain the default values in the remaining fields, and click **Edit** next to the **Remote Capabilities (RCAP)** field.

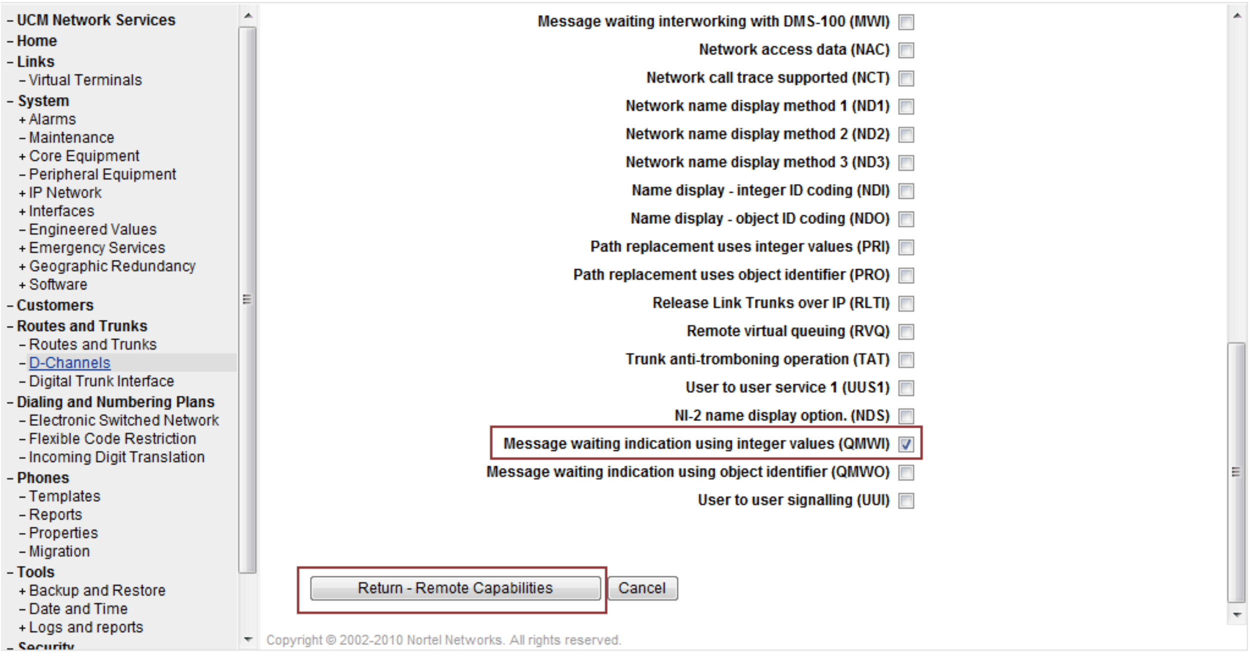


The **Remote Capabilities Configuration** screen is displayed next. Scroll down the screen as necessary to check the following capabilities:

- **Connected line identification presentation (COLP)**
- **Diversion info. sent. rerouting requests processed (DV3I)**
- **Message waiting indication using integer values (QMWI)**

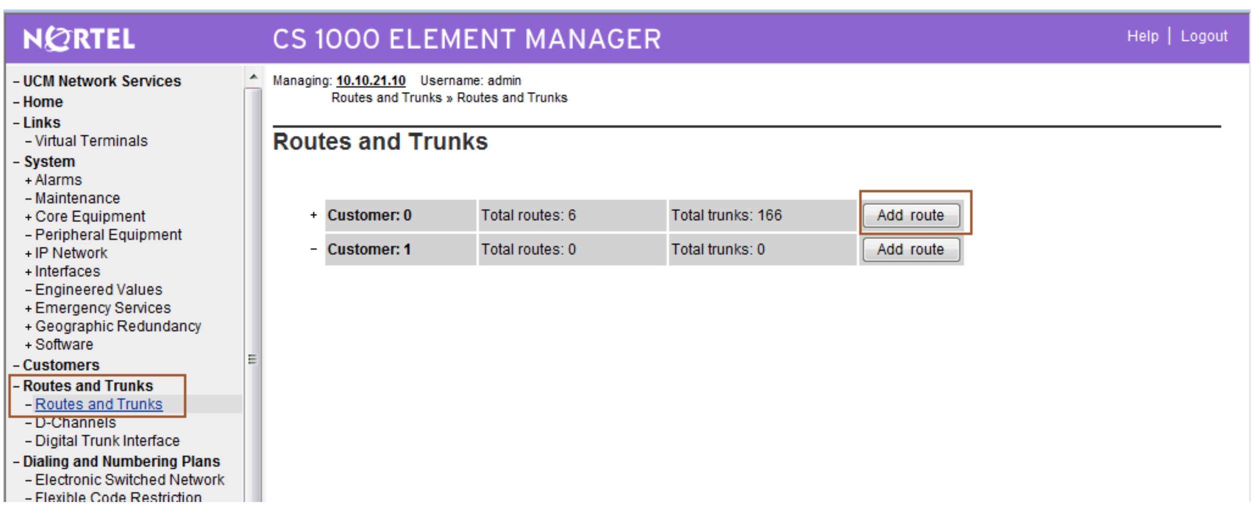


Click **Return – Remote Capabilities** at the bottom of the screen. The **D-Channels 42 Property Configuration** screen is displayed again (not shown below). Click **Submit**.



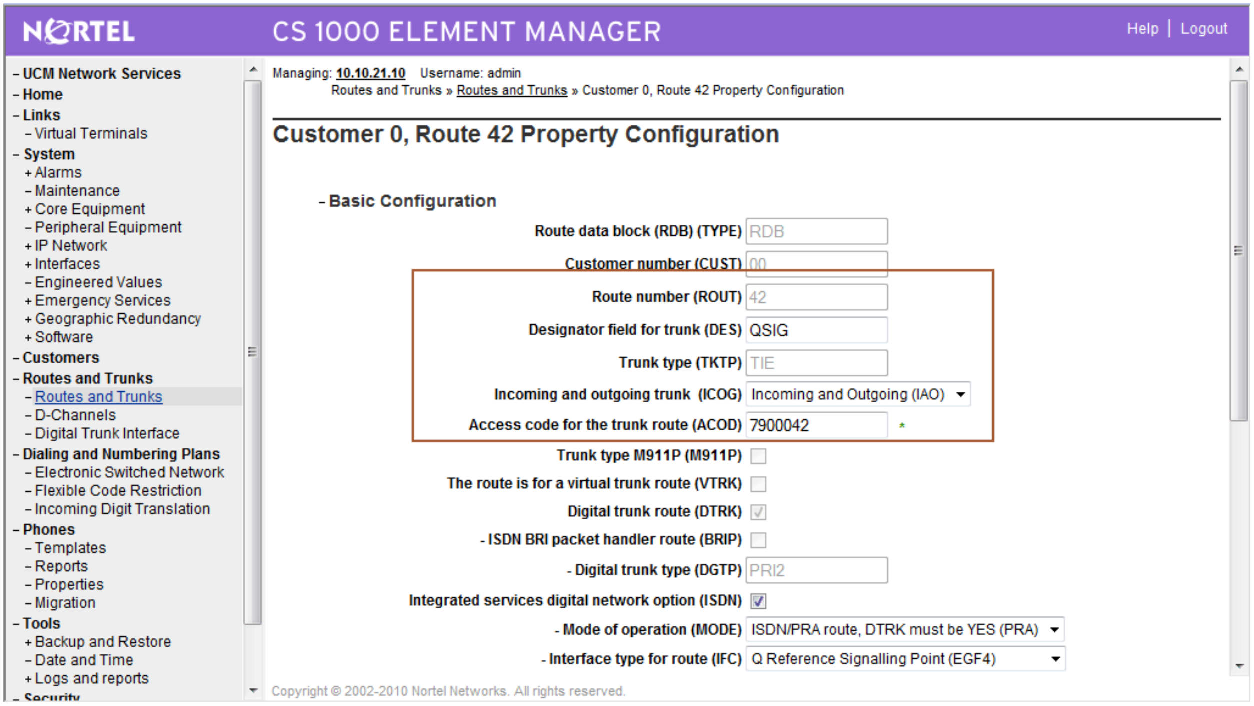
3.6 Administer Routes and Trunks

Select **Routes and Trunks → Routes and Trunks** from the left pane to display the **Routes and Trunks** screen. Next to the applicable **Customer** row, click **Add route**.



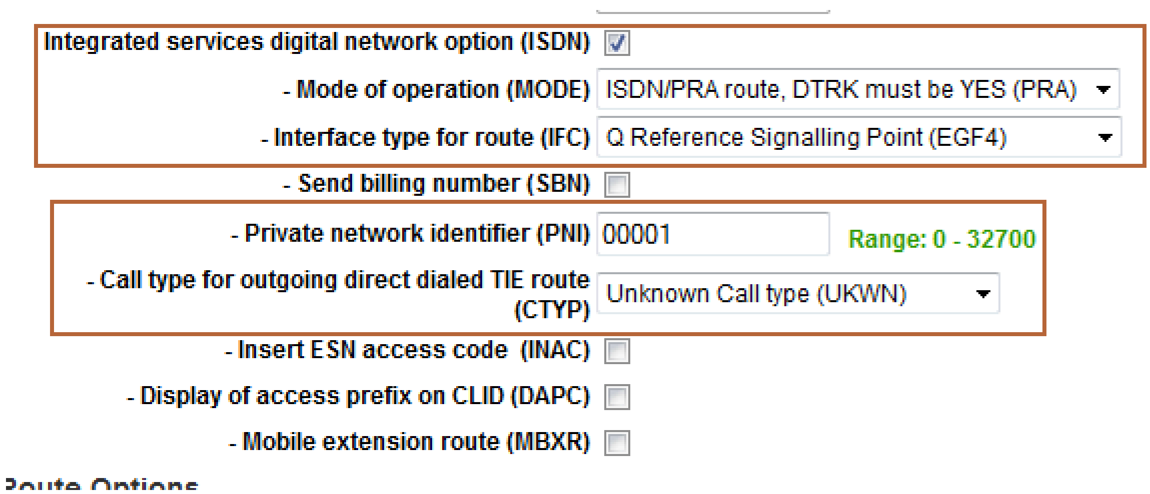
The **Customer 0, New Route Configuration** screen is displayed next. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Route Number (ROUT):** Select an available route number
- **Designator field for trunk (DES):** A descriptive text
- **Trunk Type (TKTP):** **TIE trunk data block (TIE)**
- **Incoming and Outgoing trunk (ICOG):** **Incoming and Outgoing (IAO)**
- **Access Code for the trunk route (ACOD):** An available access code

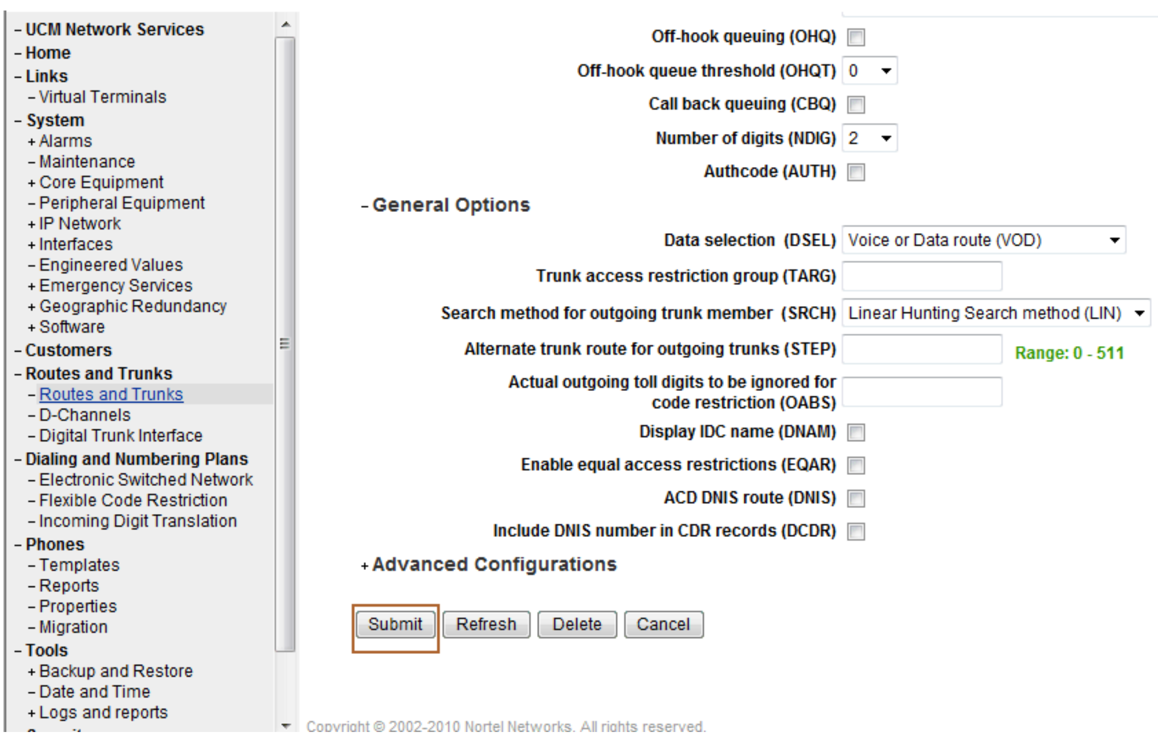


Scroll down the screen and check the **Integrated services digital network option (ISDN)** checkbox. Enter the following values for the specified fields, and retain the default values for the remaining fields.

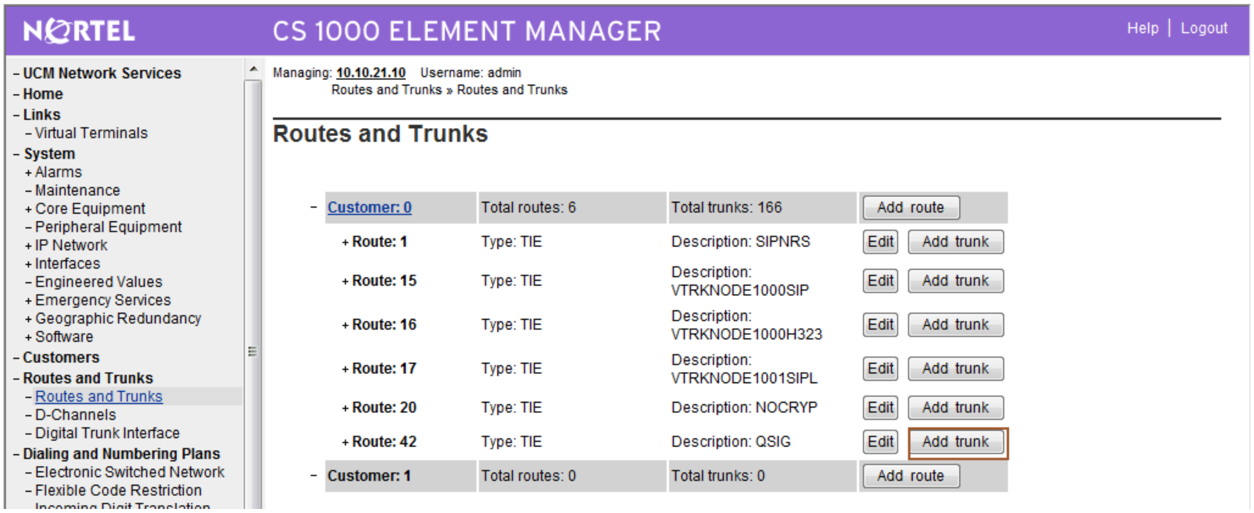
- **Mode of operation (MODE)** **ISDN/PRA route, DRTK must be YES (PRA)**
- **Interface type for route (IFC)** **Q Reference Signalling Point (EGF4)**
- **Private network identifier (PNI)** Type **00001**
- **Call type for outgoing direct dialed TIE route (CTYP)** **Unknown Call type (UKWN)**



Scroll down to the bottom of the screen, and click **Submit**.

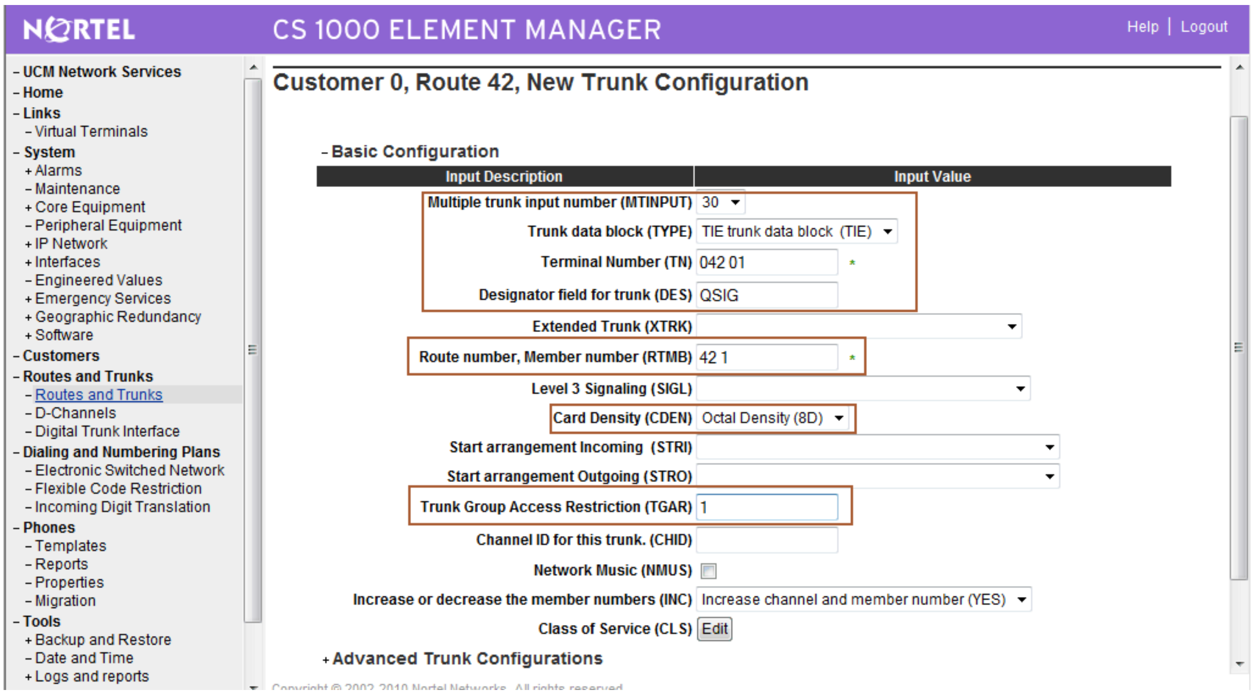


The **Routes and Trunks** screen is displayed again, and updated with the newly added route. Click the **Add trunk** button next to the newly added route.



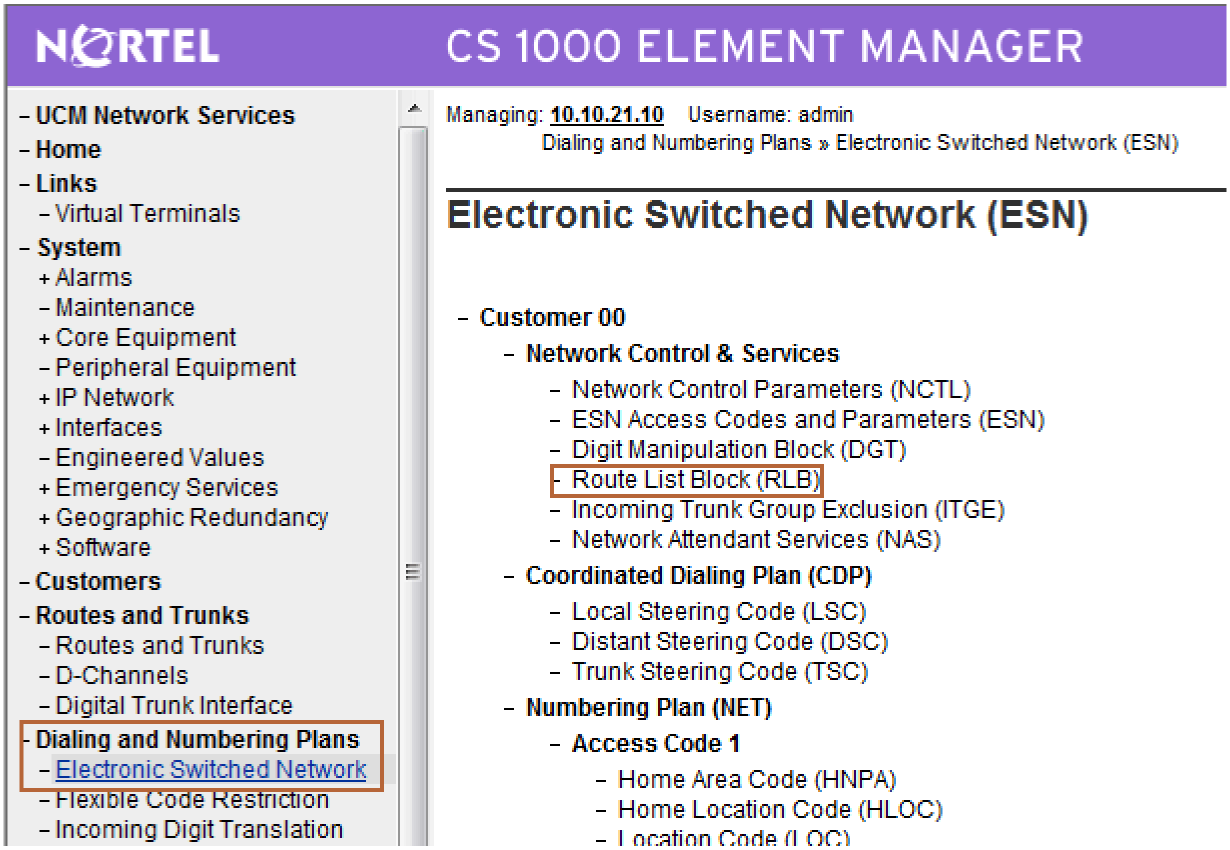
The **Customer 0, Route 42, New Trunk Configuration** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down to the bottom of the screen, and click **Submit**. The **Multiple trunk input number (MTINPUT)** field may be used to add multiple trunks in a single operation, or repeat the operation for each trunk.

- **Multiple trunk input number (MTINPUT):** Select **30**
- **Trunk data block (TYPE)** Select **TIE trunk data block (TIE)**
- **Terminal Number (TN):** The E1 loop and port number.
- **Designator field for trunk (DES):** A descriptive text.
- **Route number, Member number (RTMB):** Current route number and starting member.
- **Trunk Group Access Restriction (TGAR):** Desired trunk group access restriction level.

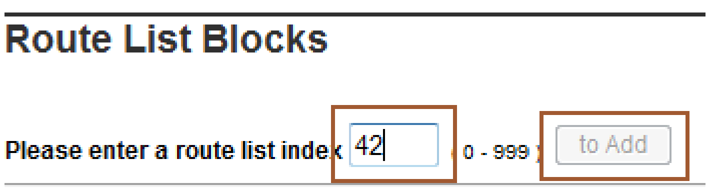


3.7 Administer Route List Block

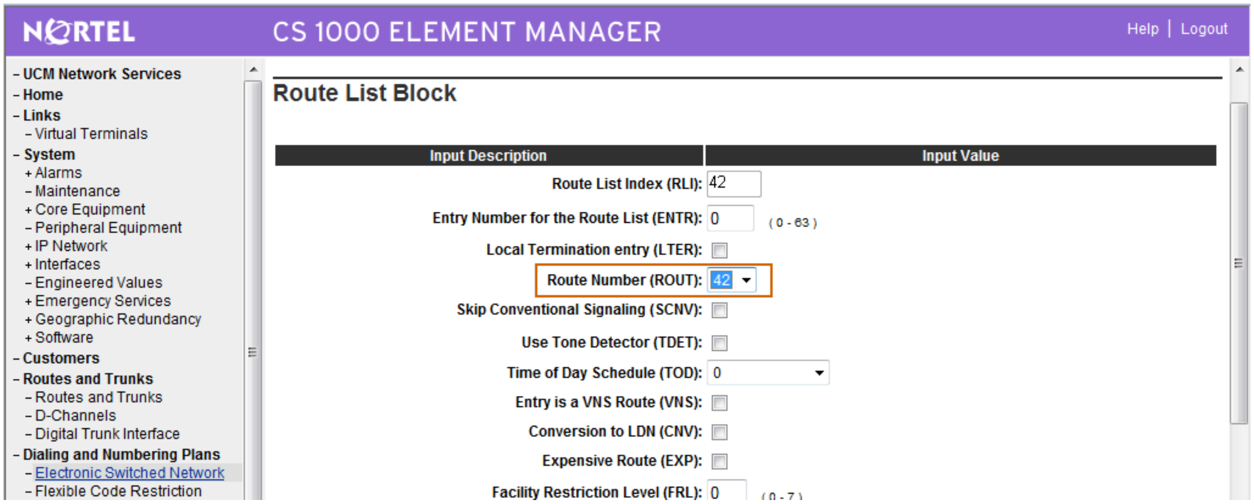
Select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Route List Block (RLB)**.



The **Route List Blocks** screen is displayed. In the **Please enter a route list index** field enter an available route list block number (in this case "42"). Click **to Add**.

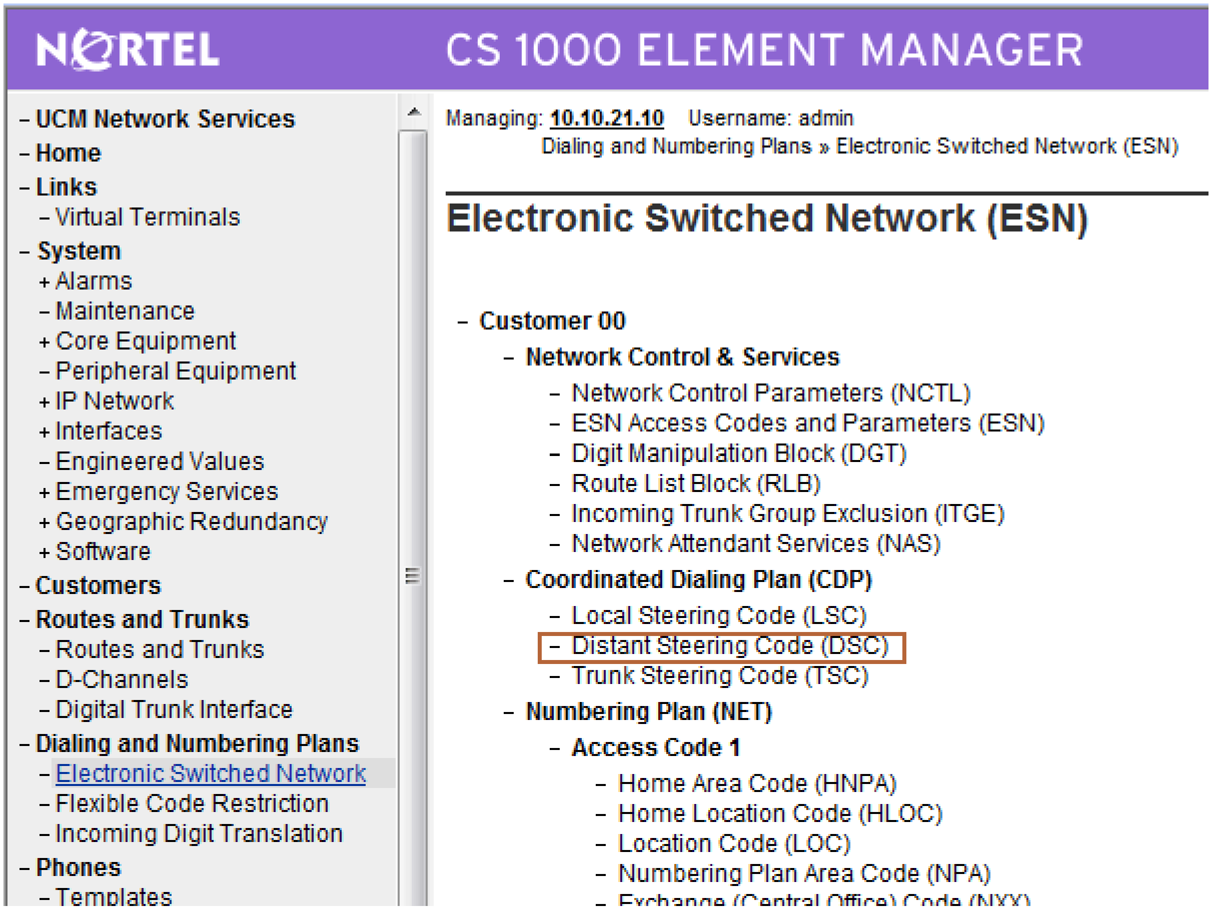


The **Route List Block** screen is updated with a listing of parameters. For the **Route Number (ROUT)** field, select the route number from **Section 3.6**. Retain the default values for the remaining fields, and scroll down to the bottom of the screen and click **Submit** (not shown).



3.8 Administer Distant Steering Code

The **Electronic Switched Network (ESN)** screen is displayed again. Select **Distant Steering Code (DSC)** to add an entry to route calls to the Avaya MX Application Server when the user dials 66666 (conference access number).

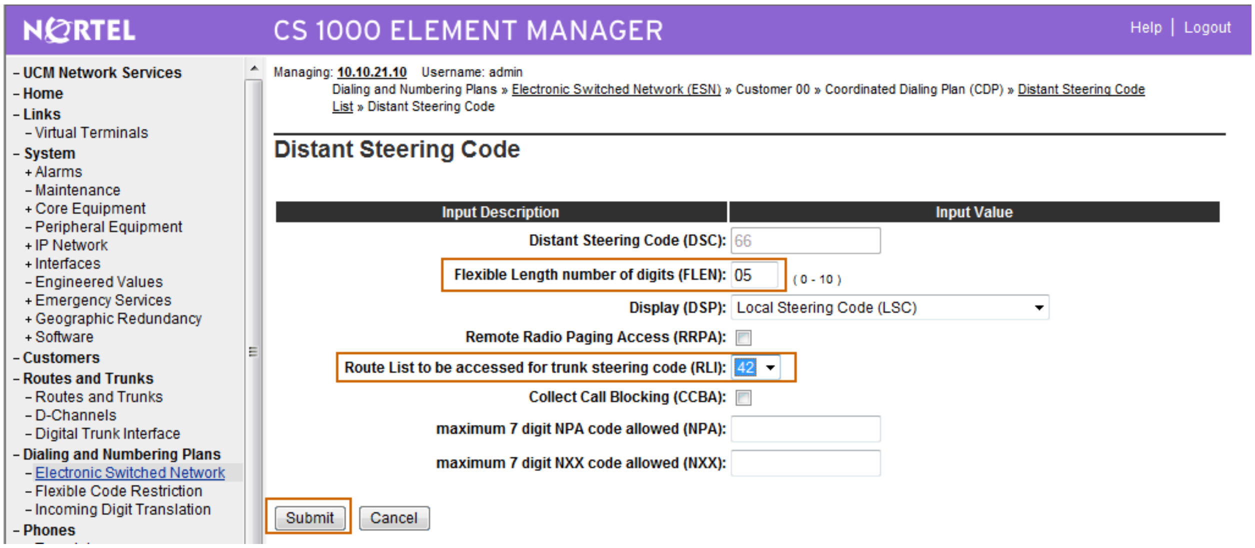


The **Distant Steering Code List** screen is displayed next. In the **Please enter a distant steering code** field, enter the dialed prefix digits to match on (in this case "66"). Click **to Add**.

The **Distant Steering Code** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Flexible Length number of digits (FLEN)** Type **05**
- **Route List to be accessed for trunk steering code (RLI)** Select the route list index in **Section 3.7** from the drop-down list.

Click **Submit**.



3.9 Enable E1 Card

Even though the E1 card can be enabled via the web based interface Element Manager, the D-channel cannot come into service unless the E1 card is enabled via the command line interface. Access the Avaya Communication Server 1000 command line interface via SSH.

The Avaya Communication Server 1000 command line interface is a character-based serial interface to the operating system and overlay programs on each system component. The program issues a prompt for input, and the system administrator enters the appropriate response through the keyboard followed by the **Return** key. The output from the Avaya Communication Server 1000 command line interface has been trimmed down in the subsequent sections in order to focus on the key settings for the configuration. Values highlighted in bold represent values entered by the system administrator.

Command	Comment
> **login** USERID? **xxxxx** PASS? **yyyyy**	Issue the login command. Enter a valid user ID. Enter a valid user password.
TTY #00 LOGGED IN xxxxx 16:50 06/4/2010	A sample response indicating successful log in.
> **ld 96** . **enl MSDL 4 0 2 FDL**	Use load 96 to enable the E1 card. Enable the E1 card with the physical slot number of the card and the option “FDL” to force parametric downloads. Please note the dot before “enl”.

3.10 Enable D-channel Automatic Establishment

Use the command line interface to enable automatic establishment for the administered D-channel.

Command	Comment
> **ld 96** . **enl auto 42**	Use load 96 to enable automatic establishment for the D-channel. Enable the D-channel automatic establishment with the D-channel number, in this case “42”.

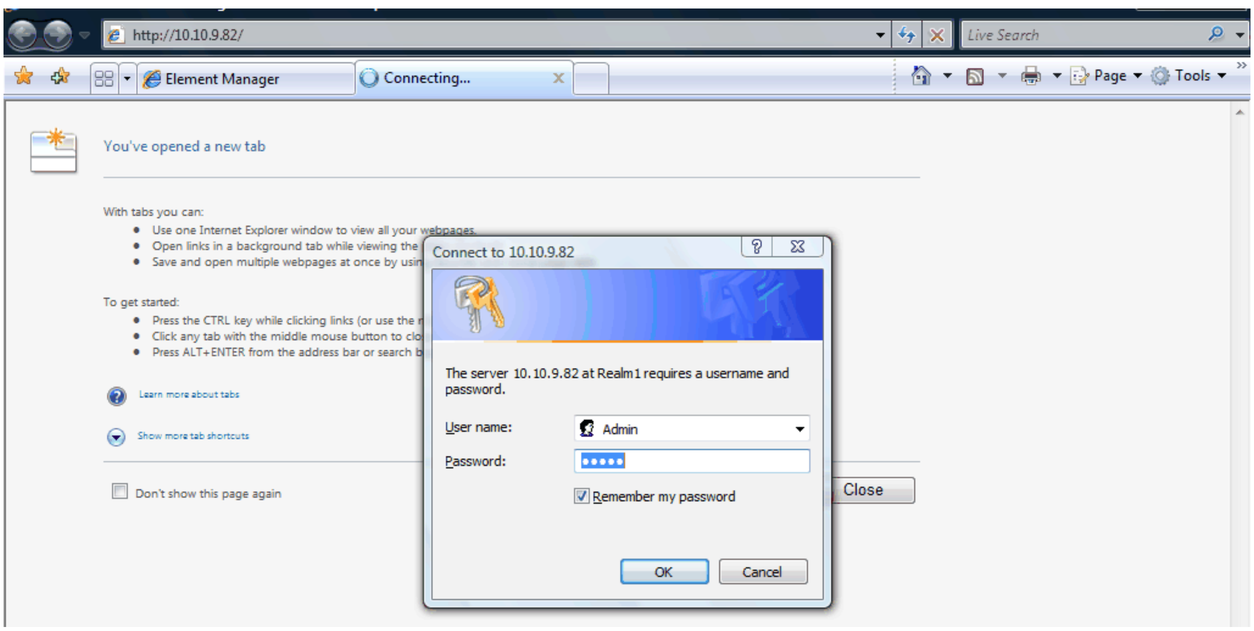
4 Configure AudioCodes Mediant 1000

The following sections describe the configuration steps required to implement E1/PRI QSIG and SIP trunks and inter-trunk routing on the AudioCodes Mediant 1000, using the web interface. It is assumed that basic hardware and software installation has been performed as described in **[5]**. This section focuses on the following configuration areas:

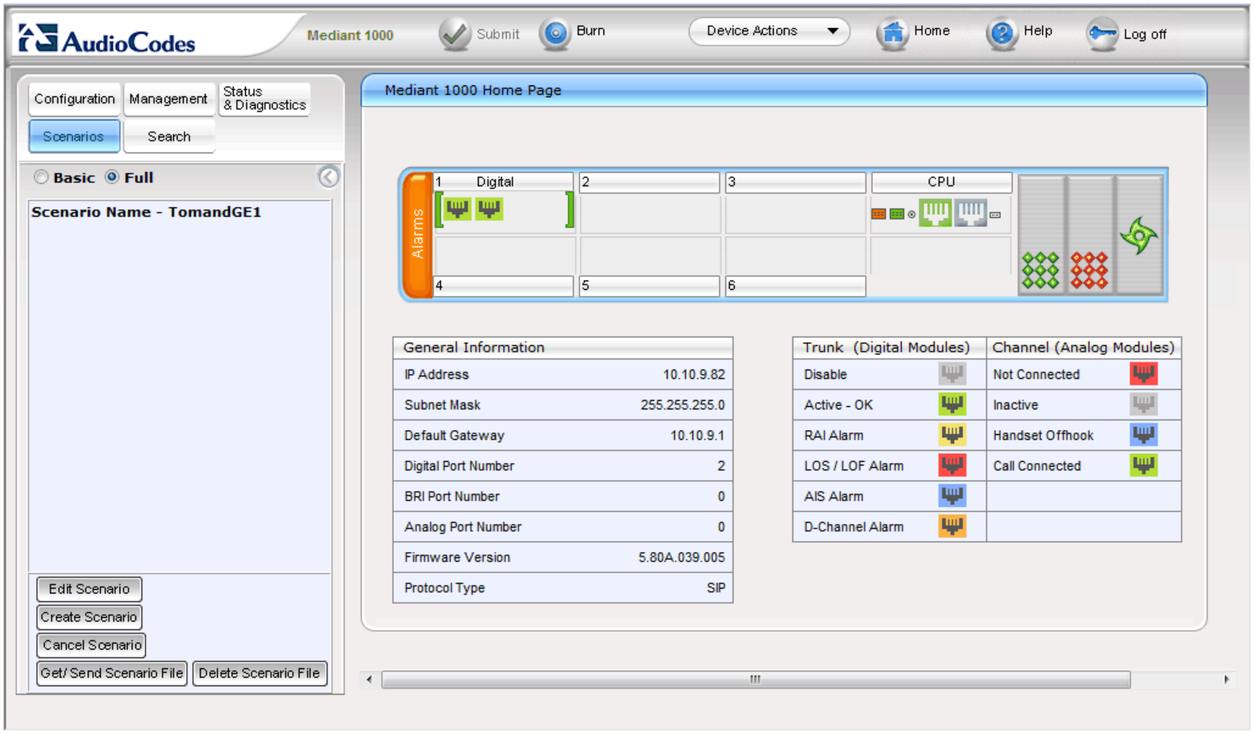
- Access Web Configuration Interface
- Configure TDM Bus Settings
- PSTN Trunk Settings
- SIP Protocol Parameters
- Routing Tables
- Configure PSTN Trunk Group
- Customize the INI File

4.1 Access Web Configuration Interface

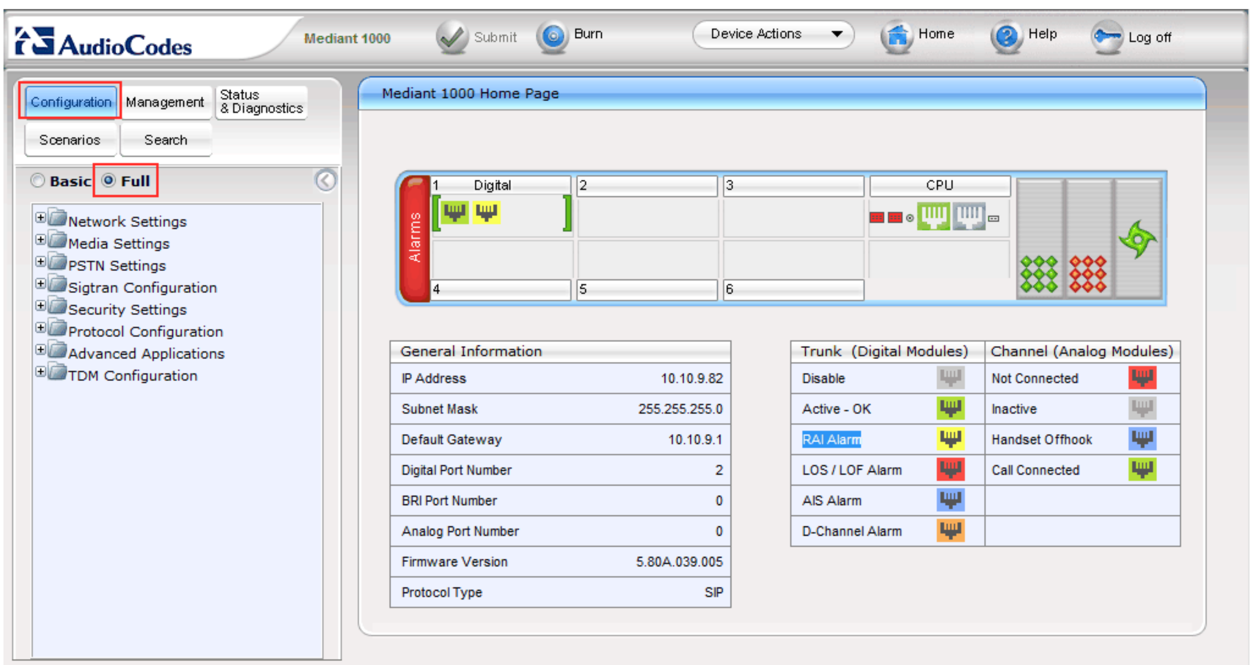
Open an instance of a web browser and connect to the AudioCodes GUI at the following address: **http://<Management IP address of the M1000>.** Log in using an appropriate Username and Password.



The **Mediant 1000 Home Page** screen is displayed.



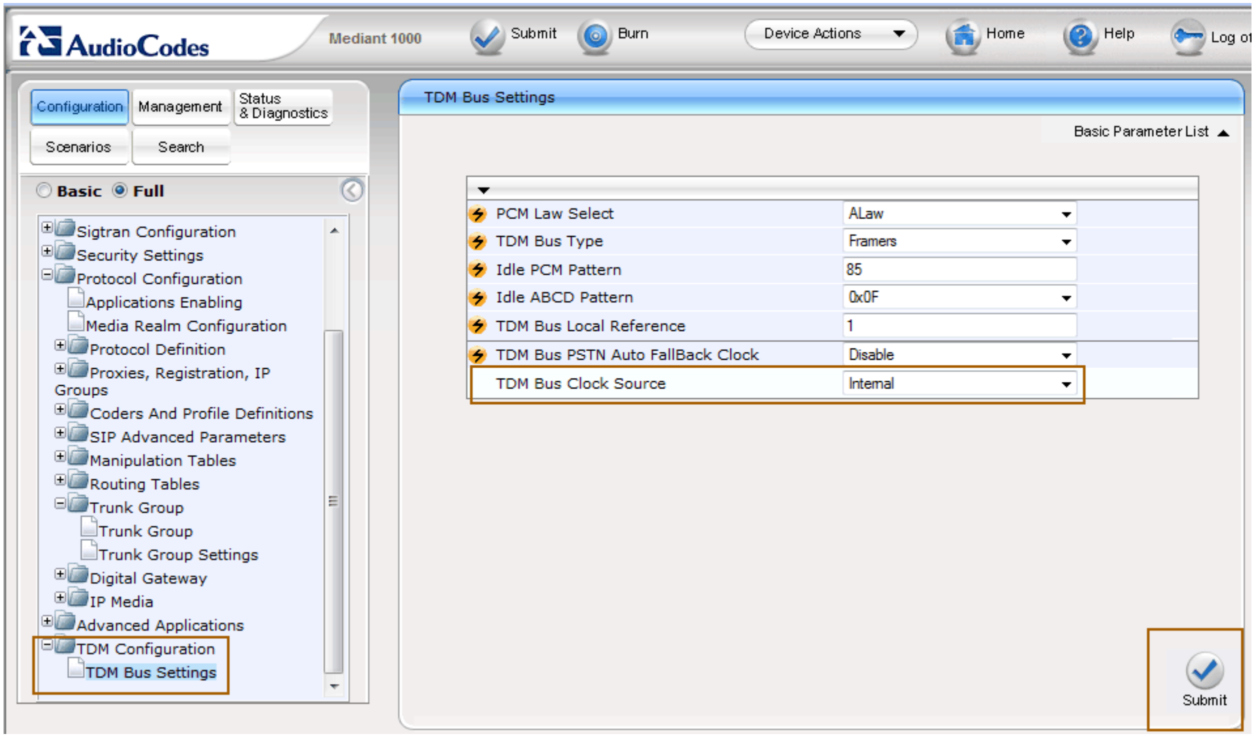
Click on **Configuration** and set the mode to **Full**. The menus on the left can be expanded as necessary to configure the appropriate features, as described in the following sections.



4.2 Configure TDM Bus Settings

Expand the **TDM Configuration** menu and click on **TDM Bus Settings**. In the sample configuration the internal clock of the M1000 Gateway provides the clocking for the E1 PRI trunk. Select the following parameters, leaving the remaining parameters at their default values.

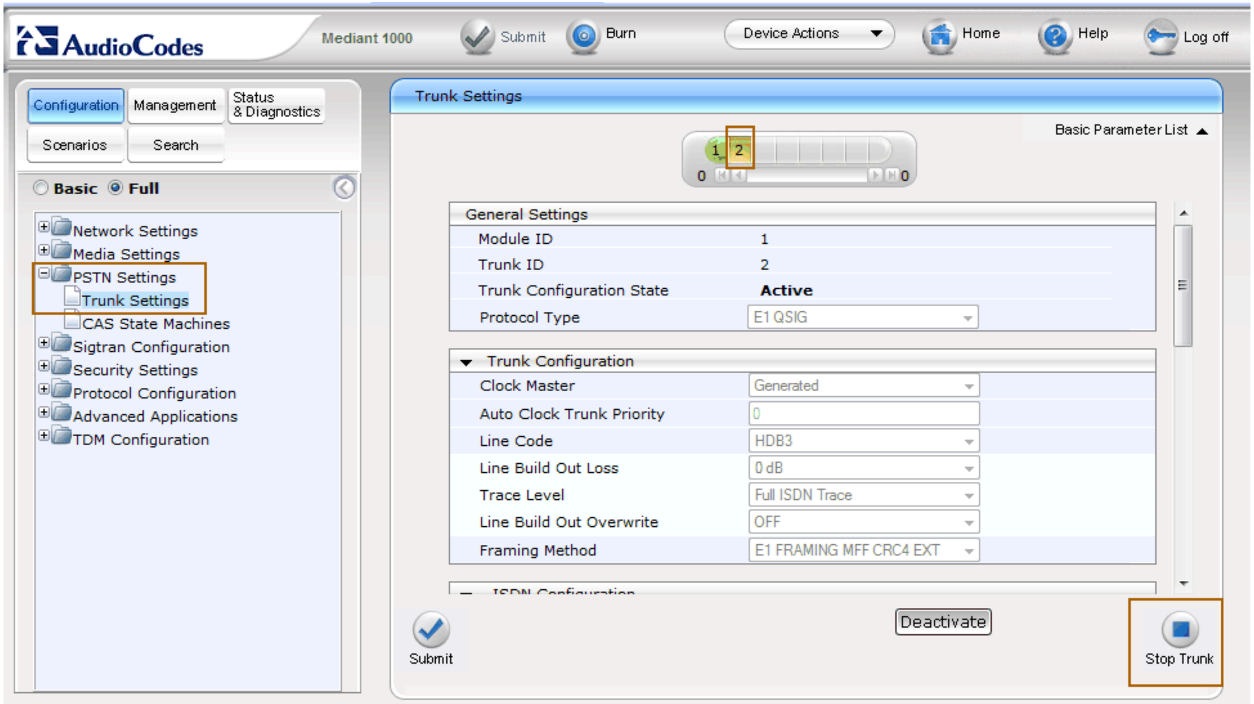
- **TDM Bus Clock Source:** Select **Internal**



4.3 Configure PSTN Trunk Settings

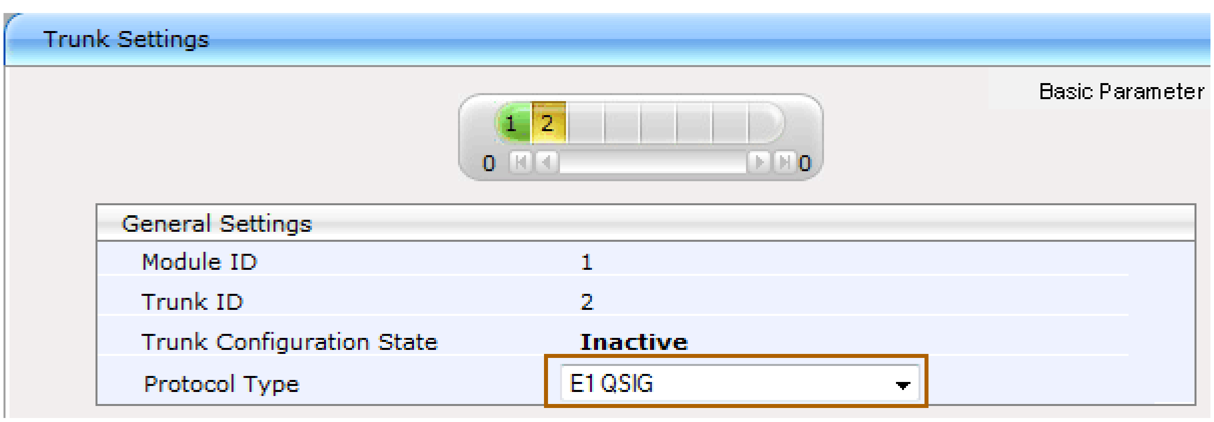
Expand the **PSTN Settings** menu and click on **Trunk Settings**. The following web page is displayed. Click on the **E1 port number**. In these application notes the second E1 port of the M1000 is connected to the E1 card of the CS1000.

Click **Stop Trunk**, which will enable editing of the parameters.



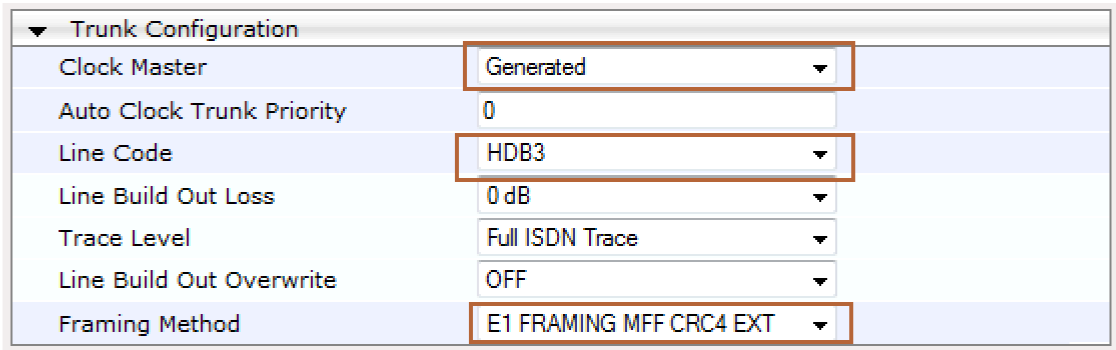
Select the following parameters, leaving the remaining parameters at their default values.
Under **General Settings**:

- **Protocol Type: E1 QSIG**



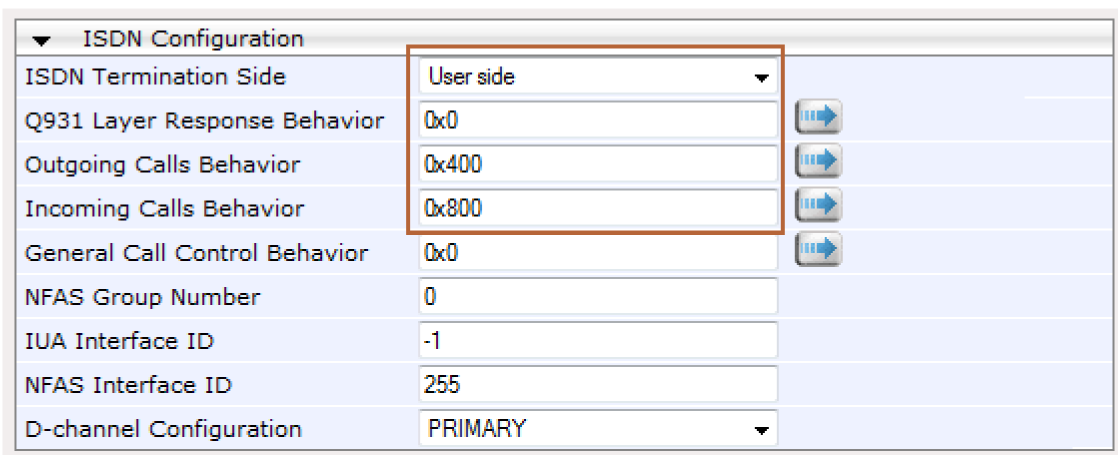
Under **Trunk Configuration**:

- **Clock Master:** **Generated**
- **Line Code:** **HDB3**
- **Framing Method:** **E1 Framing MFF CRC4 EXT**



Under **ISDN Configuration**:

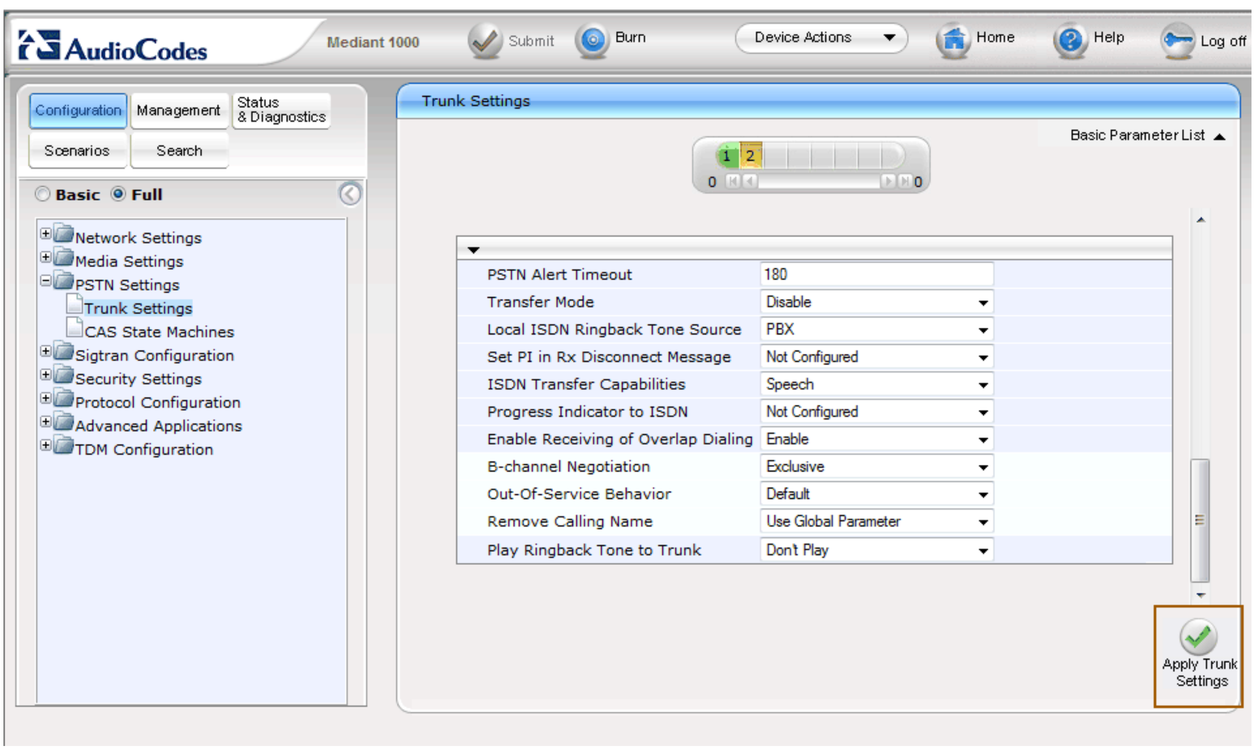
- **ISDN Termination Side:** **User side**
- **Q931 Layer Response Behavior**: **0x0**
- **Outgoing Calls Behavior:** **0x400**
- **Incoming Calls Behavior:** **0x800**



Under **Miscellaneous**:

- **ISDN Transfer Capabilities:** **Speech**
- **B-channel Negotiation:** **Exclusive**
- **Play Ringback Tone to Trunk:** **Don't Play**

Click on **Apply Trunk Settings** to save all of the above changes and put the trunk into service. Successful trunk configuration will be indicated by the green status indications for the trunk board, as shown in the first figure in **Sections 4.1**.



4.4 Configure SIP Protocol Parameters

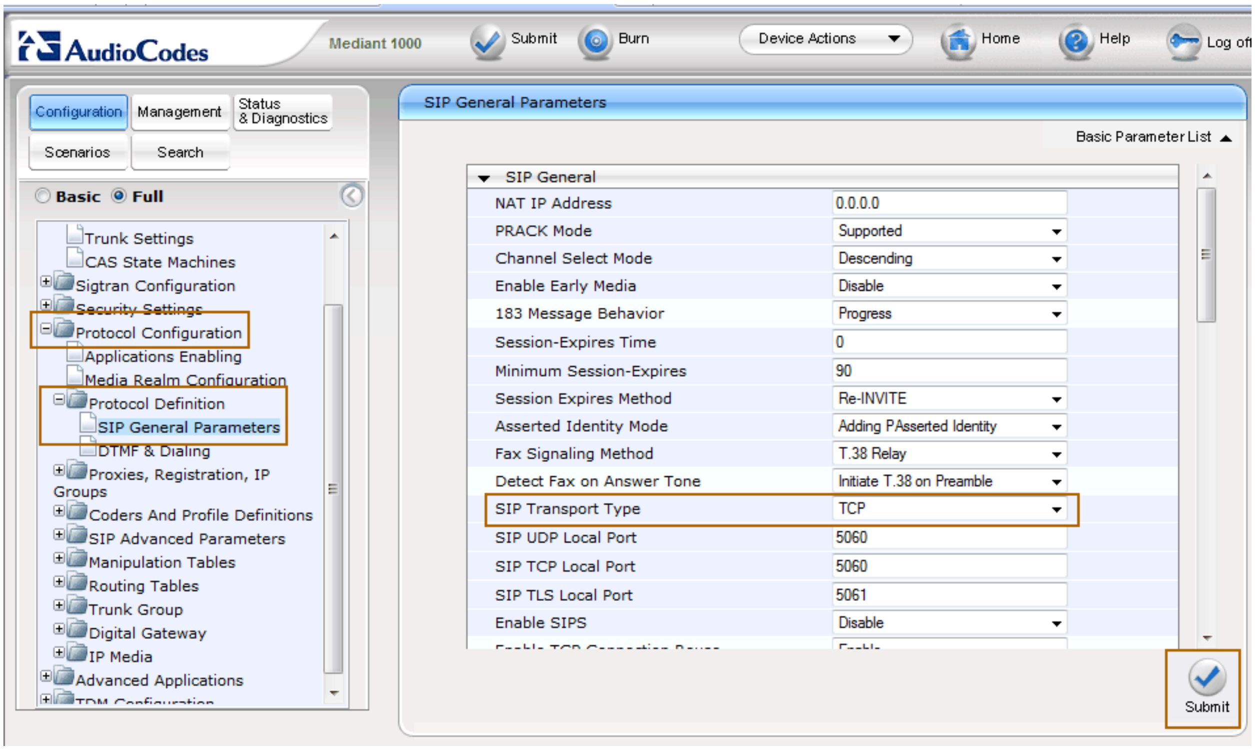
To configure the SIP parameters used when signaling with Meeting Exchange Enterprise Application Server, expand the **Protocol Configuration** menu followed by the **Protocol Definition** menu.

4.4.1 General Parameters

Click on **SIP General Parameters**. Set the following parameters, leaving the remaining parameters at their default values.

Under **SIP General**:

- **SIP Transport Type:** TCP

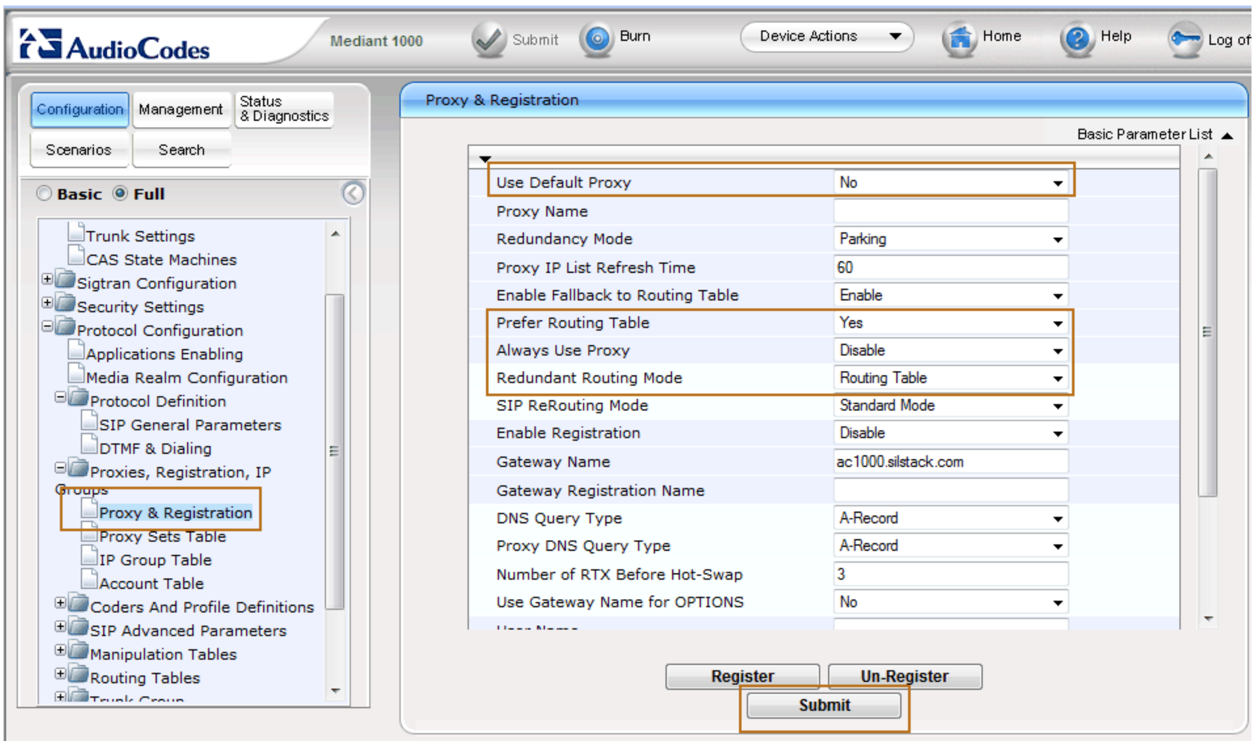


Click on **Submit** to save these changes.

4.4.2 Proxy & Registration Parameters

These application notes use the call routing table of the M1000 Gateway for outbound calls rather than using a Default SIP Proxy. Manually configured call routing is useful when the dialed number rarely changes or when a single number is dialed by a large number of users (conference bridge access number or voice mail pilot number). Click on **Proxy, Registration**, **IP Groups** on the left. Set the following parameters, leaving the remaining parameters at their default values.

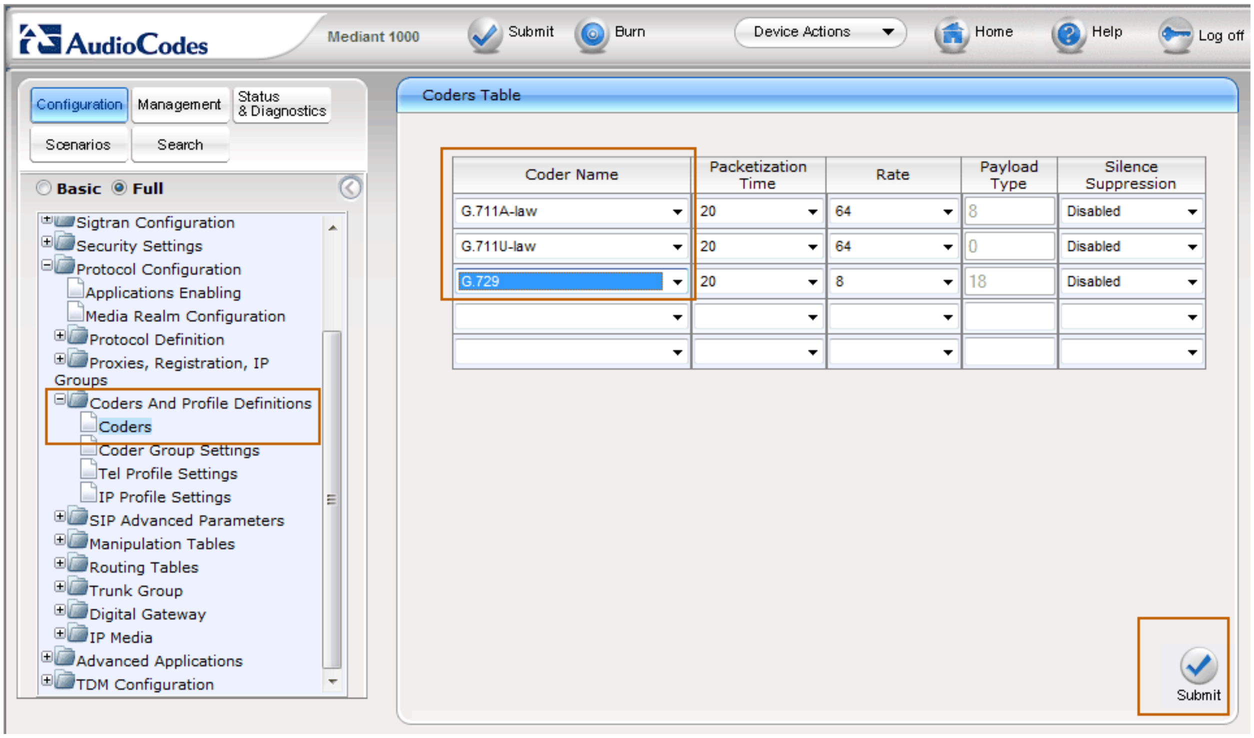
- **Use Default Proxy:** **No**
- **Proxy Name:** Leave blank
- **Prefer Routing Table:** **Yes**
- **Always Use Proxy:** **Disable**
- **Redundant Routing Mode:** **Routing Table**



Click on **Submit** to save these changes.

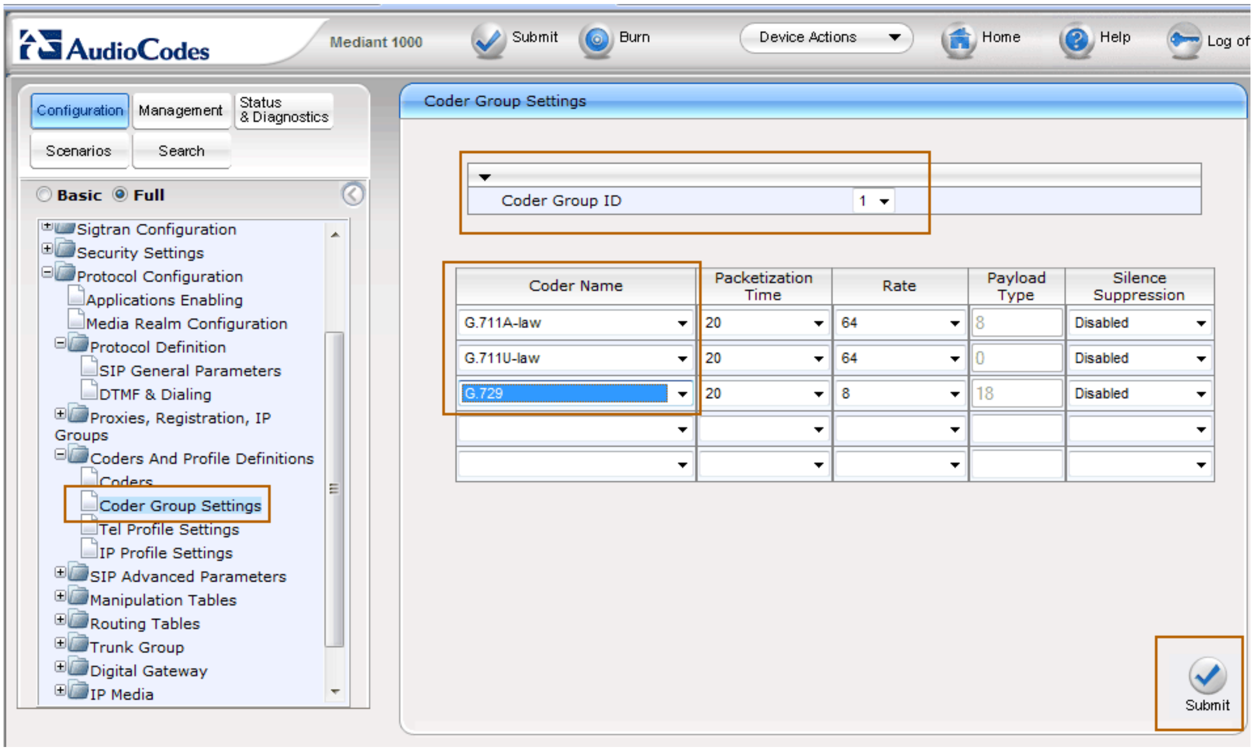
4.4.3 Audio Codecs

Select **Coders and Profile Definitions → Coders** on the left pane. In the rows of the table that are displayed, enter the desired codecs in order of preference. In the sample configuration, G.711 A-law, G.711 U-law and G.729 audio codecs were tested. Click on **Submit** to save these changes.



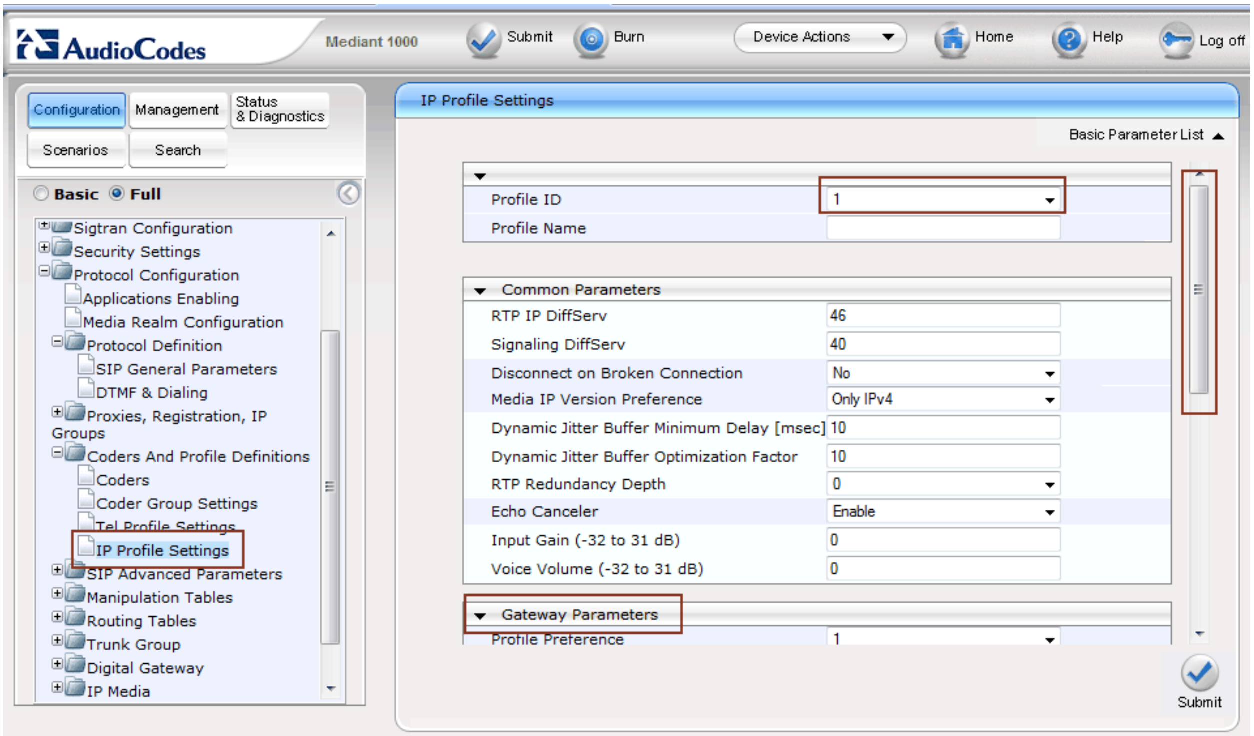
4.4.4 Coder Group Settings

Select **Coder Group Settings** on the left pane. Set **Coder Group ID** to **1**. Repeat the steps from **Section 4.3.3** to configure the supported codecs.



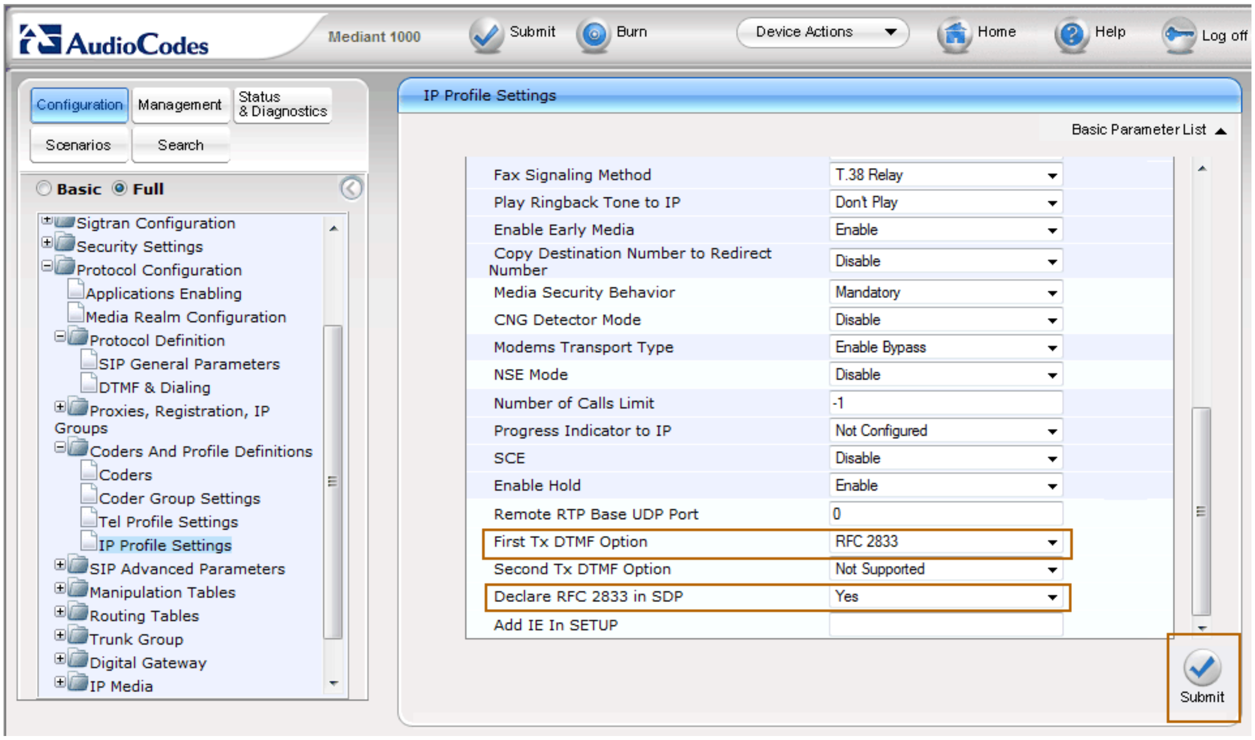
4.4.5 IP Profile Settings

Select **IP Profile Settings**. Set **Profile ID** to **1**. Scroll down to the **Gateway Parameters** section.



Set the following parameters, leaving the remaining parameters at their default values. Click on **Submit** to save these changes.

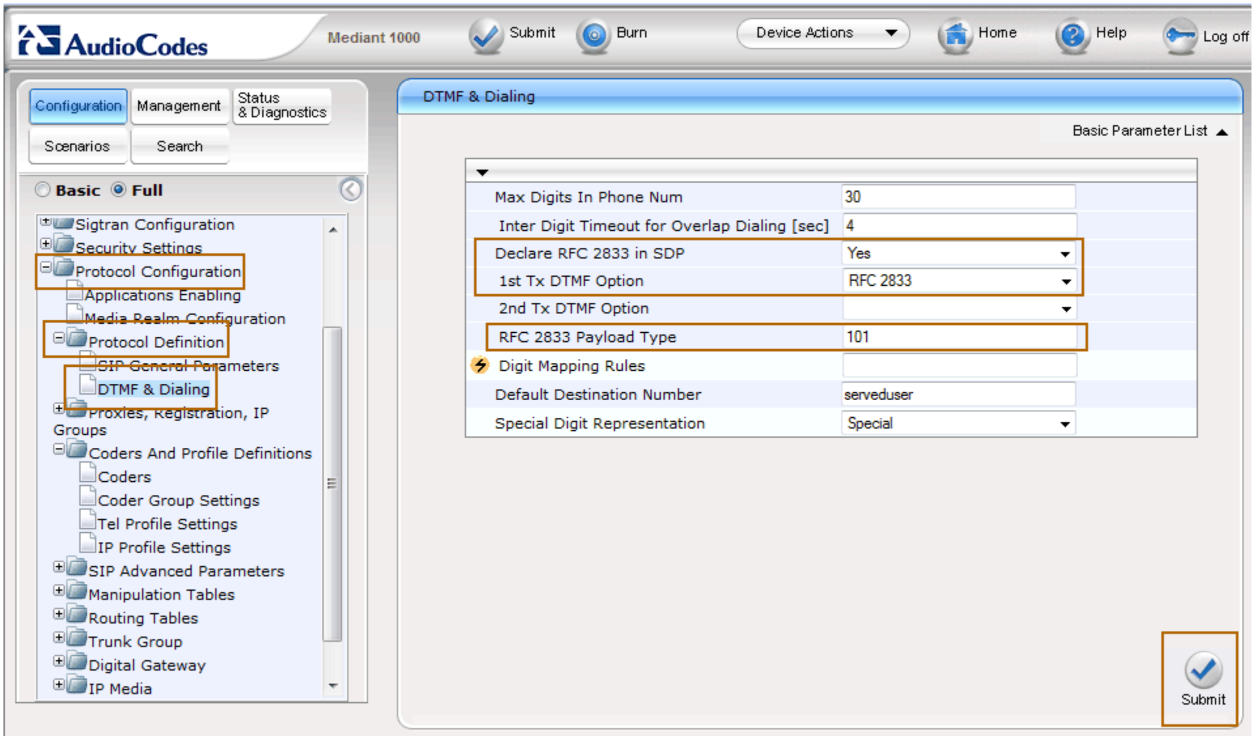
- **First Tx DTMF Option: RFC 2833**
- **Declare RFC 2833 in SDP: Yes**



4.4.6 DTMF Signaling

Select **Protocol Configuration → Protocol Definition →DTMF & Dialing** on the left. Set the following parameters, leaving the remaining parameters at their default values. Click on **Submit** to save these changes.

- **Declare RFC 2833 in SDP: Yes**
- **1st Tx DTMF Option: RFC 2833**
- **RFC 2833 Payload Type: 101**

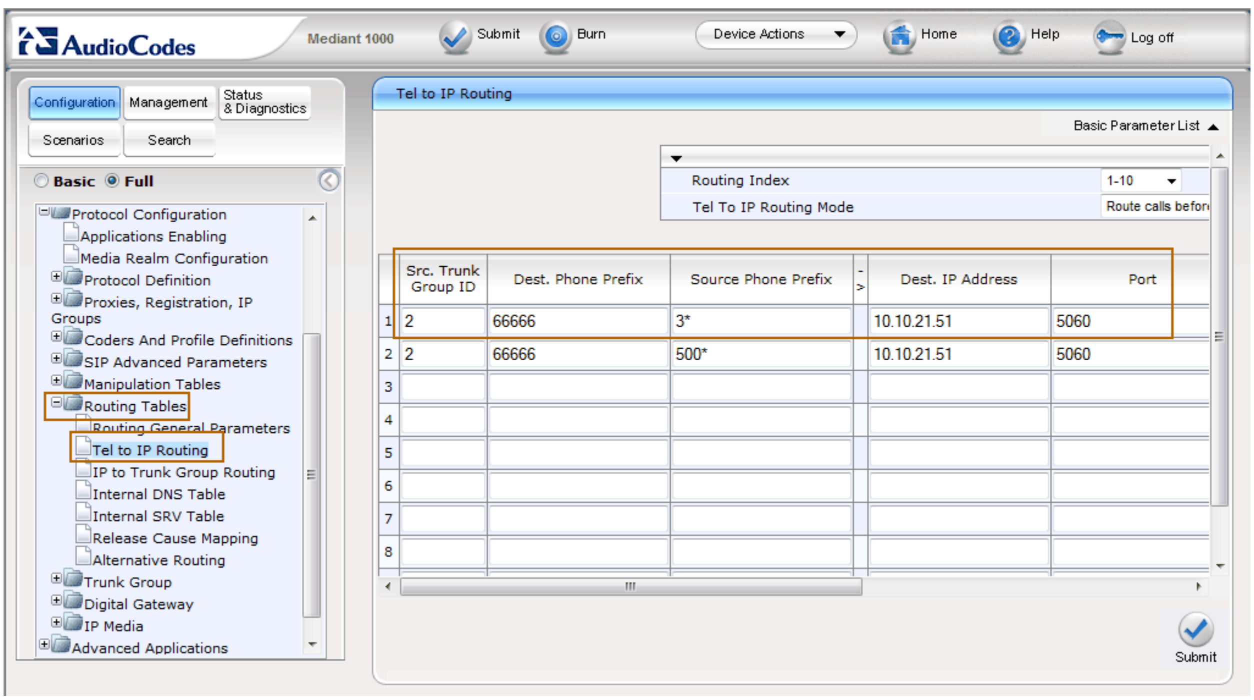


4.5 Configure Routing Tables

To configure the tables used for routing calls between the E1 and SIP interfaces, expand the **Routing Tables** menu under **Protocol Configuration** on the left. Since use of a SIP proxy was disabled in **Section 4.3.2**, the **Tel to IP Routing** needs to be configured. All calls from the PSTN and the enterprise users (Avaya 11xx series IP phones) are routed to the MX Enterprise Application Server based on the dialed number.

The DNIS in the sample configuration is the conference access number 66666. To configure E1 to SIP call routing, click on **IP to Trunk Group Routing** on the left. Set the following parameters in Row 1, leaving the remaining parameters at their default values. These values specify that all TDM calls are to be routed to the MX Enterprise Application Server via the SIP interface of the M1000 Gateway.

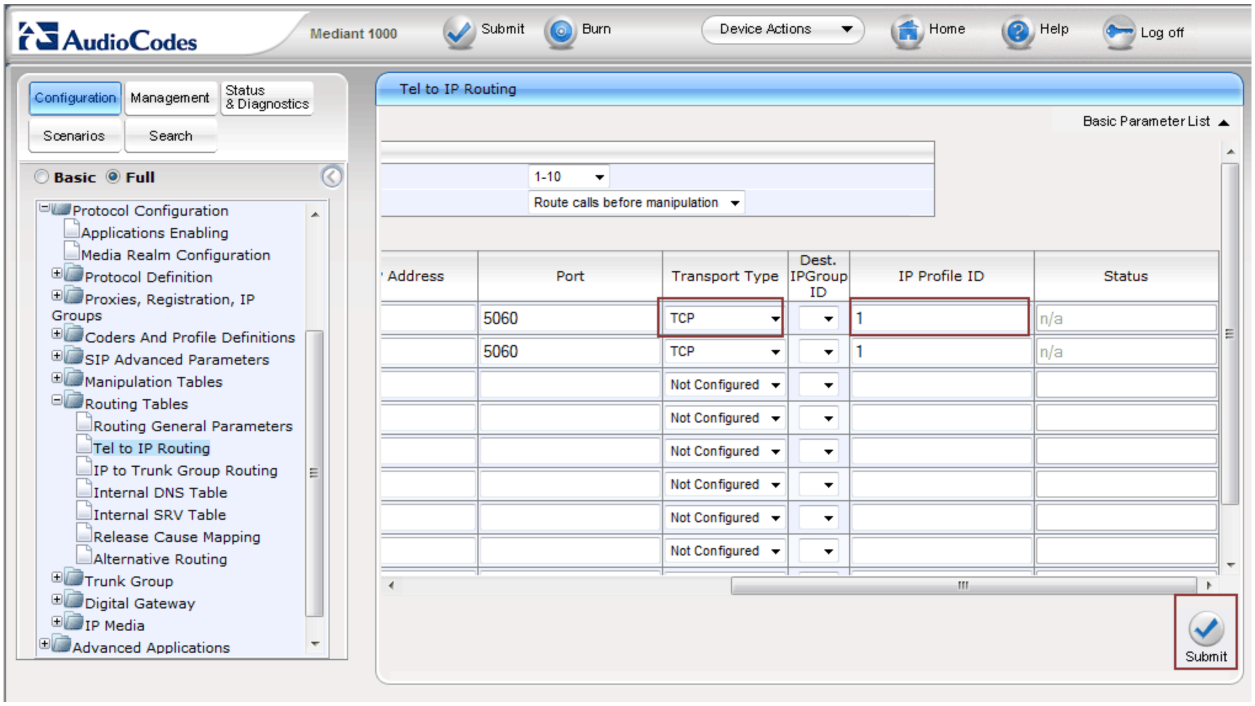
- **Source Trunk Group ID:** Type **2**, as the second E1 port of the M1000 Gateway is connected to the CS1000
- **Dest. Phone Prefix:** Type **66666.** The conference access number is set to 66666 in **Section 5.3**
- **Source Phone Prefix:** Type **3*.** The enterprise users are in the 3xxx extension range
- **Dest. IP Address:** Type **10.10.21.51,** which is the IP address of the MX Application Server
- **Port:** Type **5060**



Scroll right. Set the following parameters in Row 1, leaving the remaining parameters at their default values.

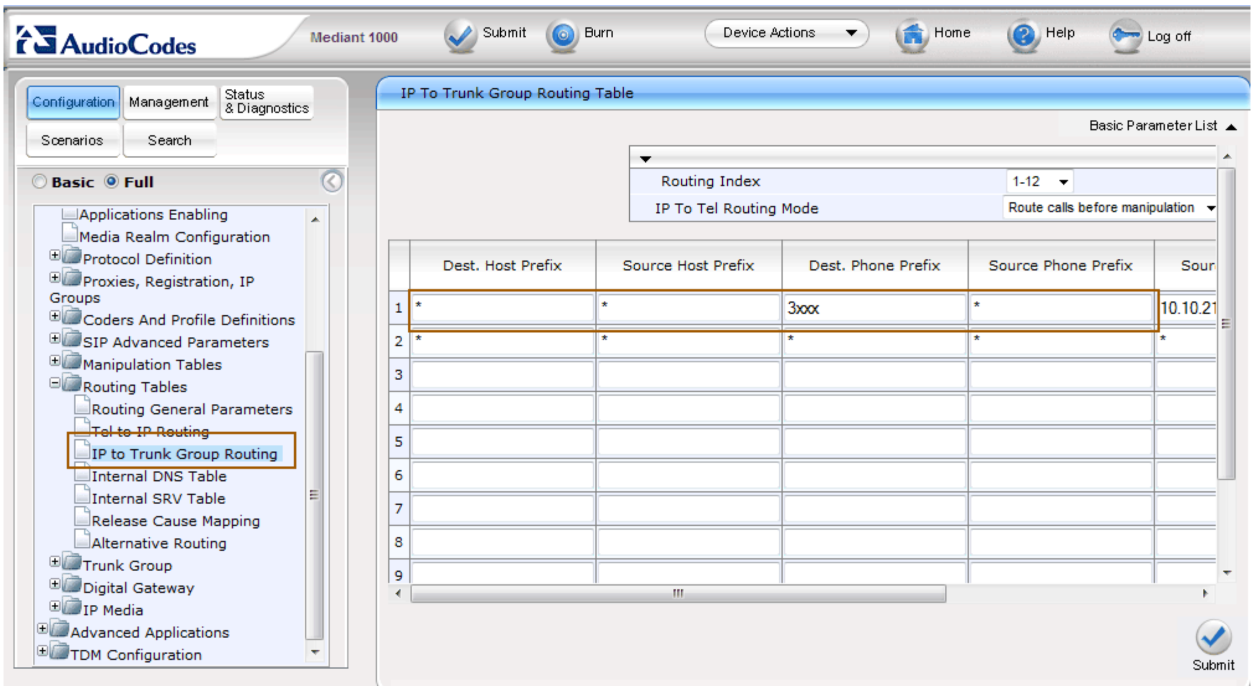
- **Transport Type:** Select **TCP**
- **IP Profile ID:** Type **1**, to associate the audio codecs we configured in **Section 4.4.4** and bound to a profile in **Section 4.4.5** with this particular route

Click on **Submit** to save these changes.



To configure routing from SIP to E1, click on **IP to Trunk Group Routing** on the left. Set the following parameters in Row 1, leaving the remaining parameters at their default values. These values specify that all SIP calls are to be routed to the E1 PRI interface (Trunk Group 2).

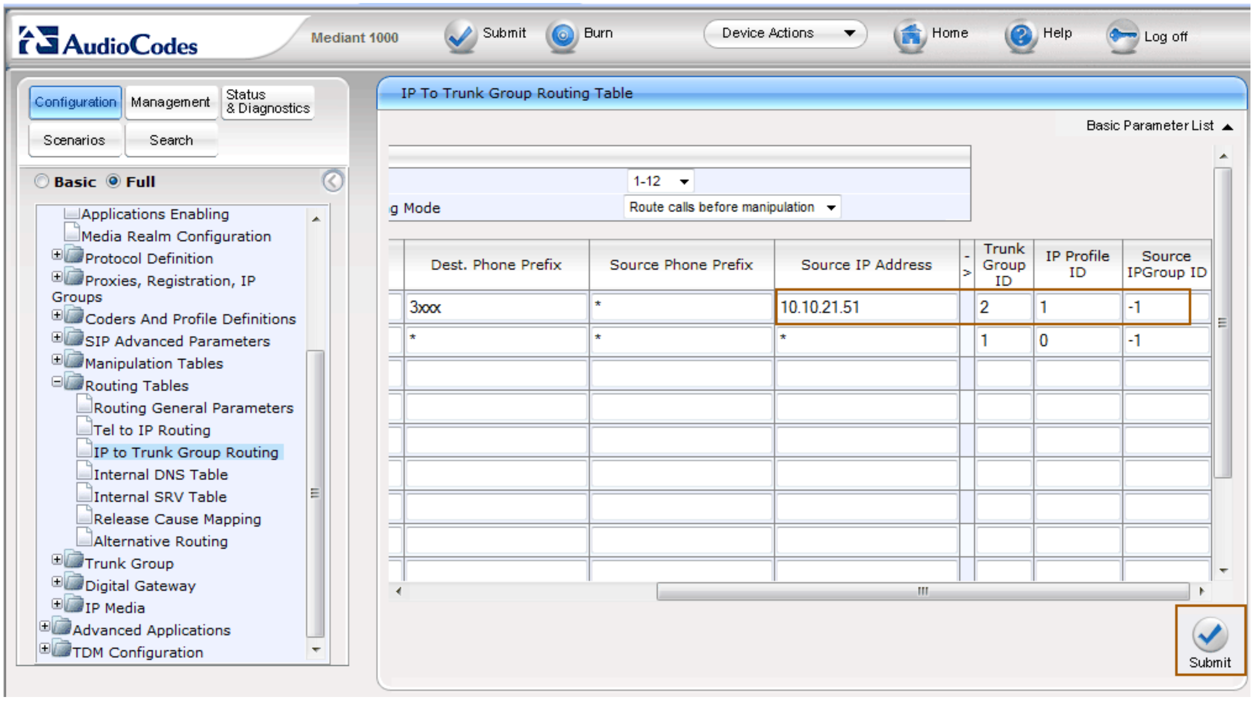
- **Dest. Host Prefix:** *
- **Source Host Prefix:** *
- **Dest. Phone Prefix** 3xxx
- **Source Phone Prefix:** *



Scroll right. Set the following parameters in Row 1, leaving the remaining parameters at their default values.

- **Source IP Address:** **10.10.21.51**
- **Trunk Group ID:** **2**
- **IP Profile ID:** **1**
- **Source IPGroup ID:** **-1**

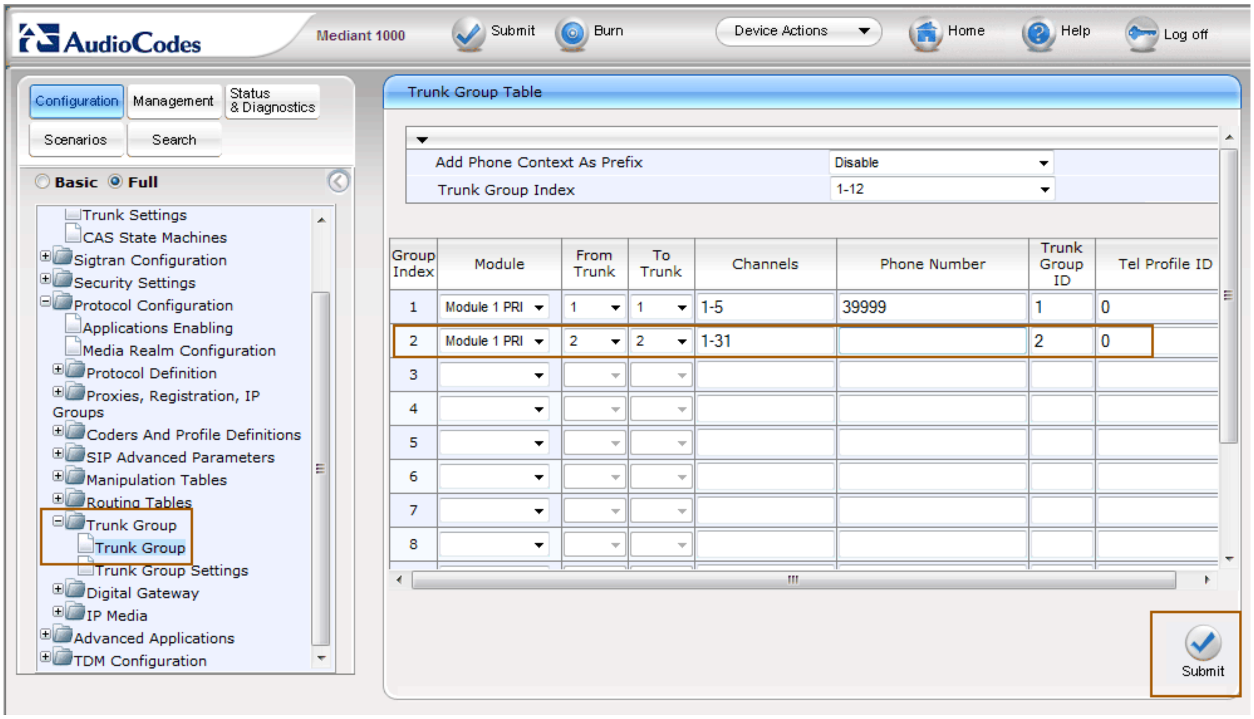
Click on **Submit** to save these changes.



4.6 Configure PSTN Trunk Group

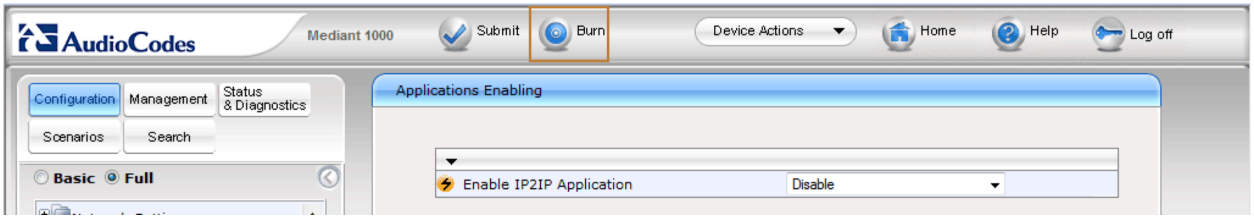
To configure the trunk group associated with the E1 PR1 port configured in **Section 4.3** expand the **Trunk Group** menu under **Protocol Configuration** on the left. Click on **Trunk Group** and set the following parameters for **Group Index** 1, leaving the remaining parameters at their default values. Click on **Submit** to save these changes.

- **Module:** Select Module 1 PRI
- **From Trunk:** Select **2**
- **To Trunk:** Select **2**
- **Channels:** Type **1-31**
- **Phone Number:** Enter a logical phone number that will be used if a call from the PSTN does not contain a calling number (optional)
- **Trunk Group ID:** Type **2**

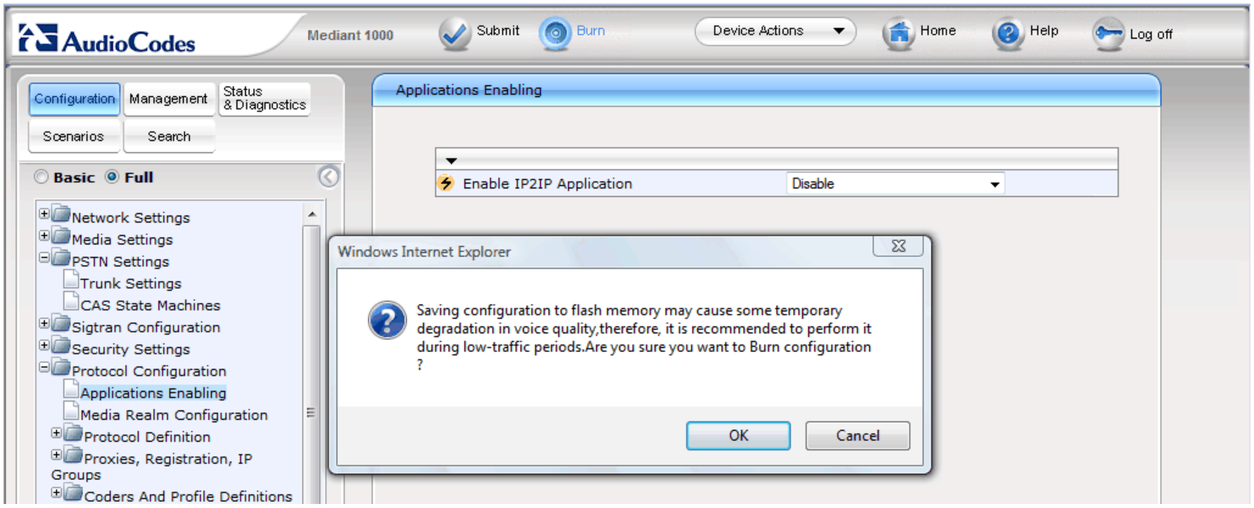


4.7 Save the Configuration

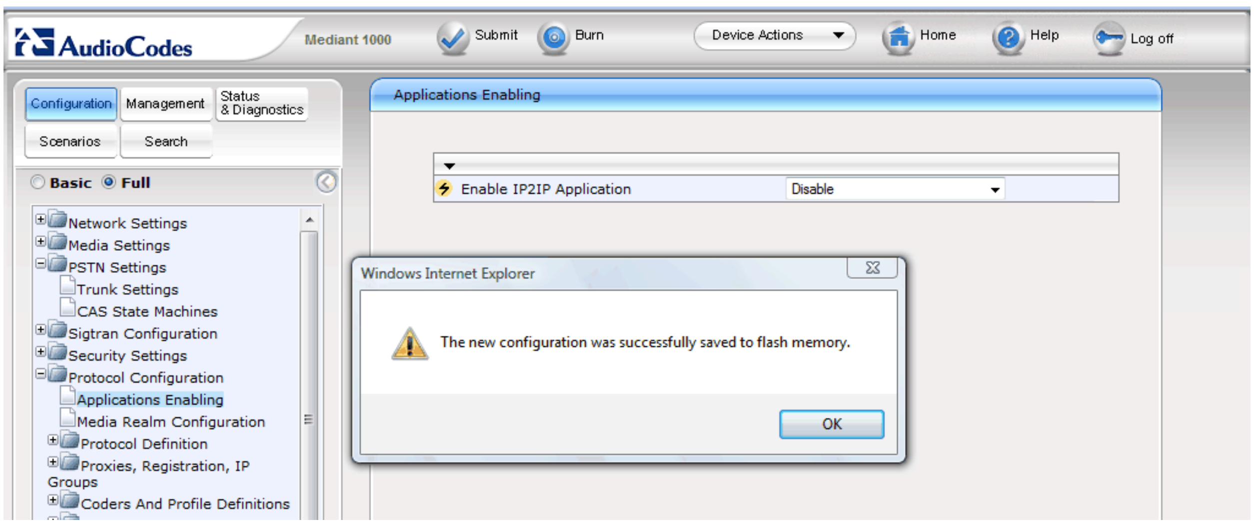
Click on **Burn** on the AudioCodes Toolbar.



Click **OK** to confirm the below message.



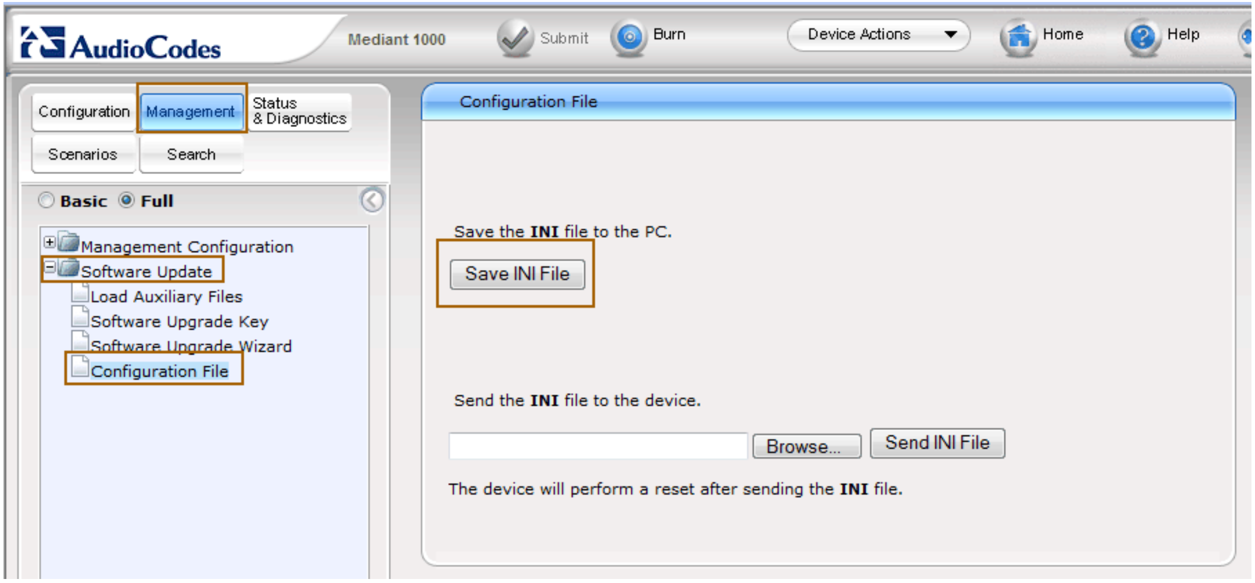
Click **OK** to acknowledge the below prompt.



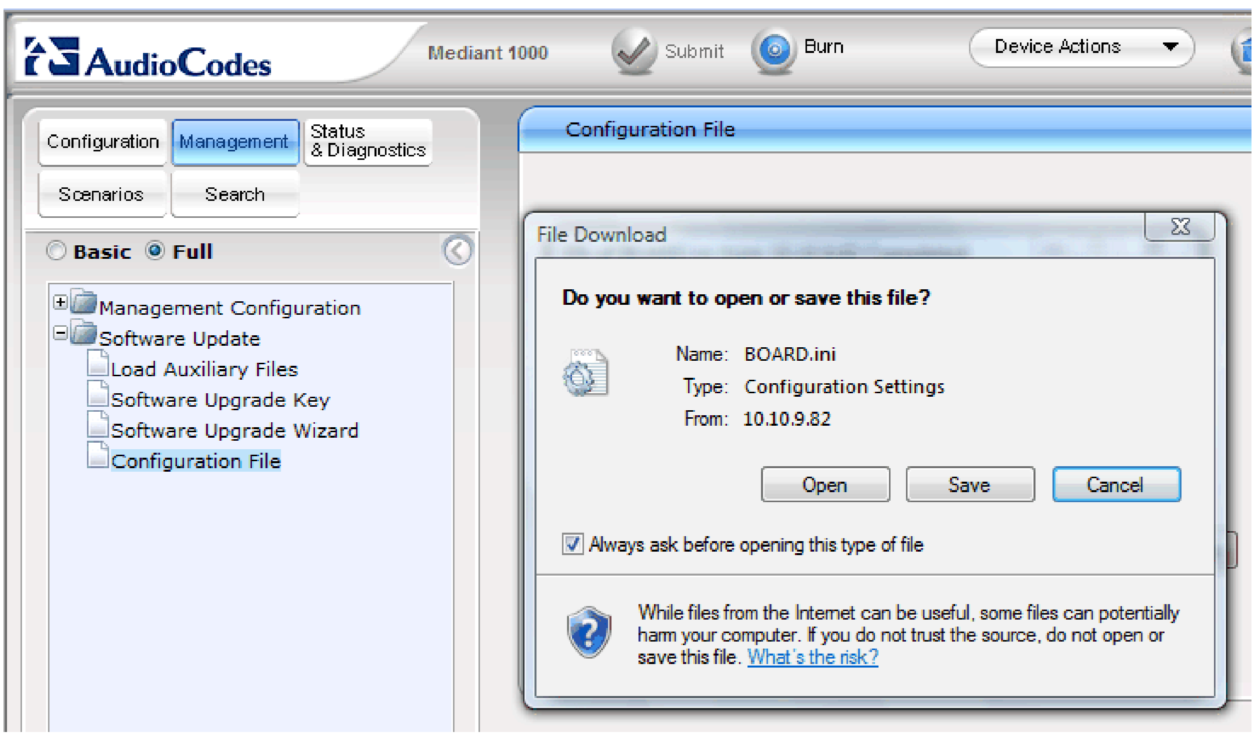
4.8 Customize the INI File

Some of the more advanced parameters are not configurable via the Web GUI of the M1000 Gateway. In this case the configuration file of the M1000 Gateway (aka. INI file) can be downloaded from the gateway and edited by a text editor. The INI file needs to be customized in order to comply with the B-channel selection method used by the CS1000 in QSIG trunks as per **[4]**.

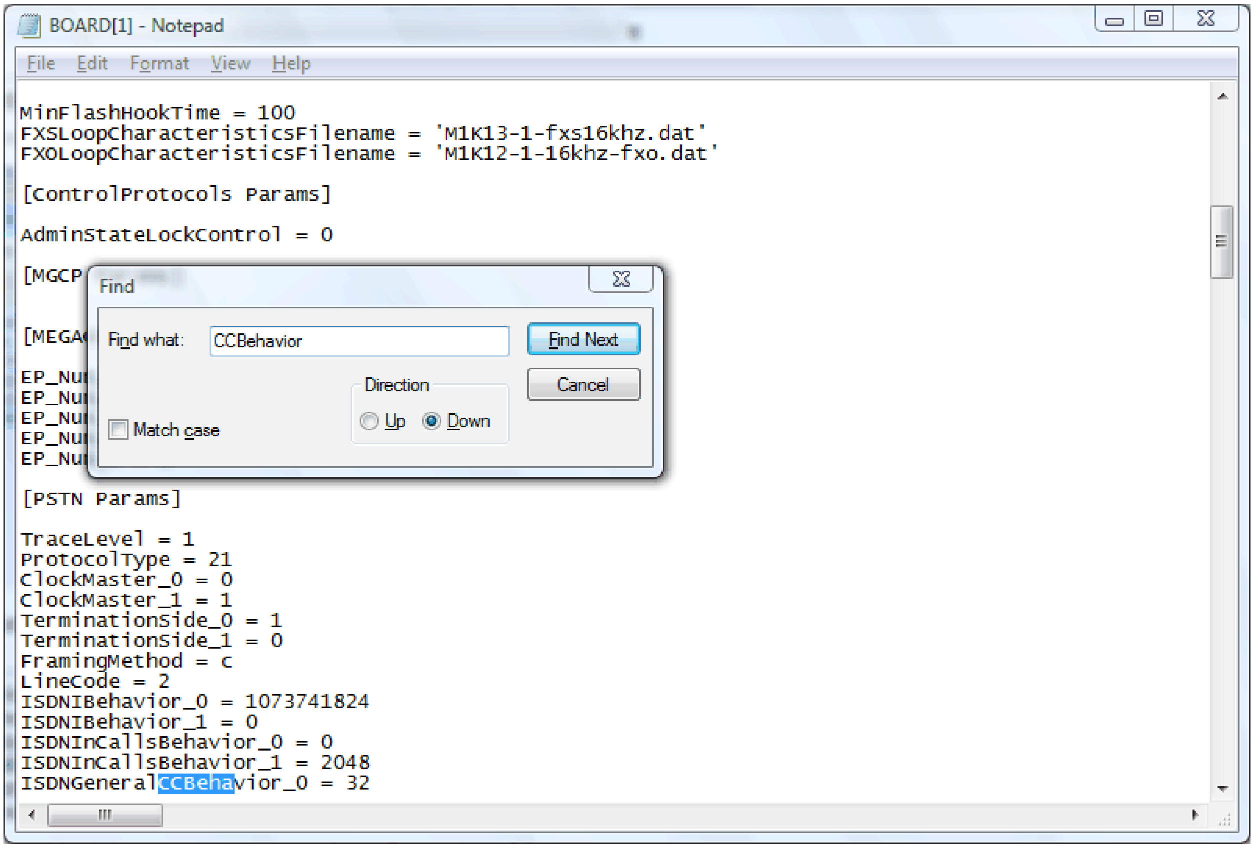
Click on **Management**. Expand the **Software Update** menu and select **Configuration File**. Click on **Save INI File** to download the configuration file of the M1000 Gateway.



Save and open the file for editing using a text editor on the local PC.



Search for the **CCBehavor** string the INI file.



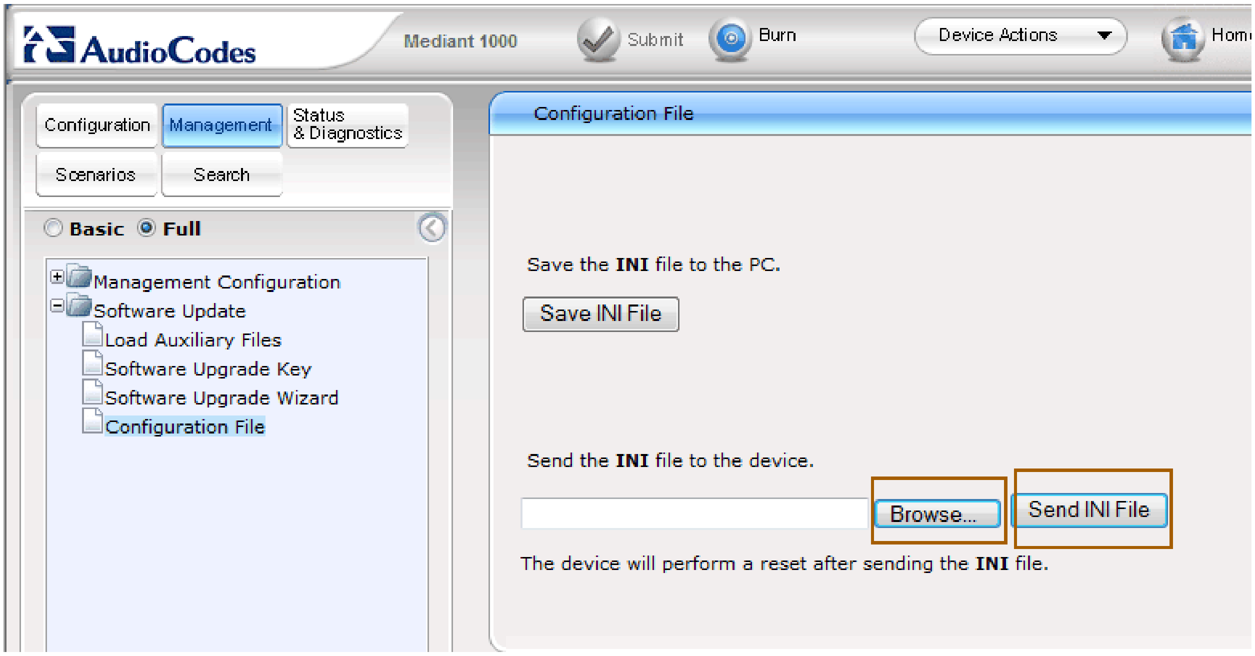
In the sample configuration a dual port E1 module was used in slot 1 of the M1000 Gateway. The first E1 port is named **ISDNGeneralCCBehavior_0**, while the second E1 port is identified as **ISDNGeneralCCBehavior_1** in the INI file. Change the **ISDNGeneralCCBehavior_1** parameter from the default **32** to **0** and save the file on the local PC.

BOARD[1] - Notepad

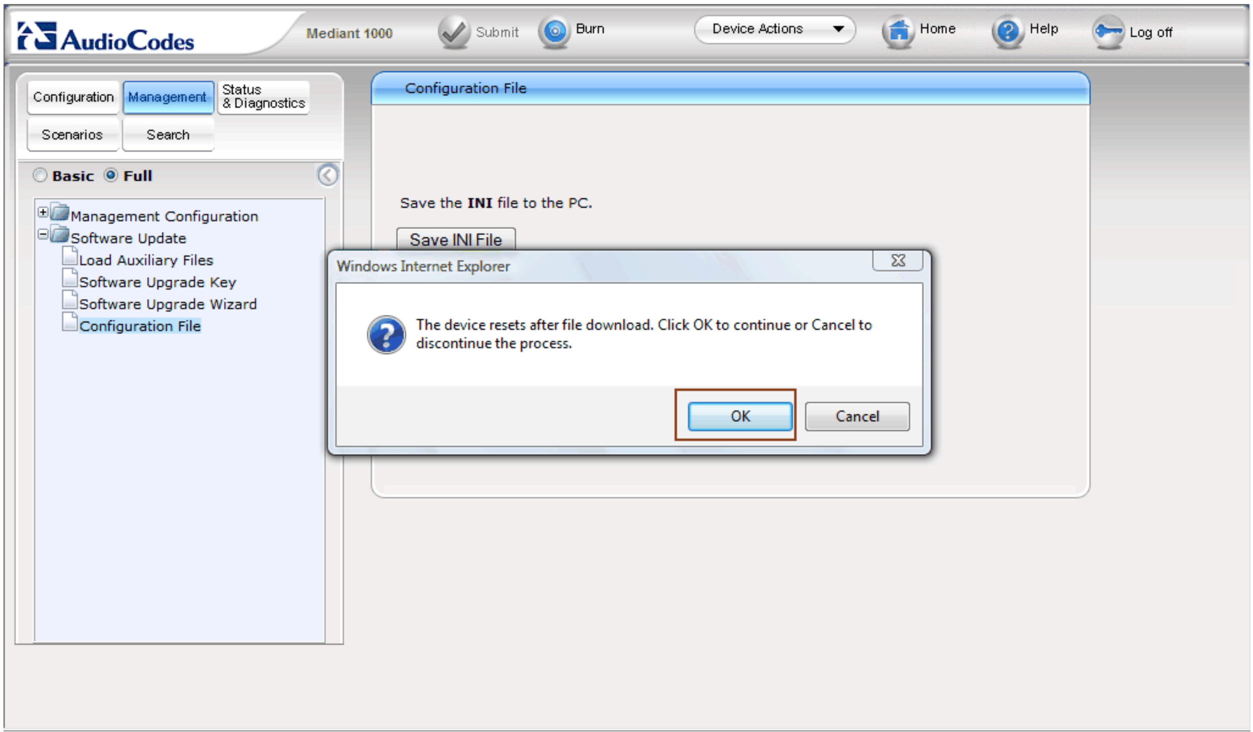
File Edit Format View Help

```
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]

TraceLevel = 1
ProtocolType = 21
ClockMaster_0 = 0
ClockMaster_1 = 1
TerminationSide_0 = 1
TerminationSide_1 = 0
FramingMethod = c
LineCode = 2
ISDNIBehavior_0 = 1073741824
ISDNIBehavior_1 = 0
ISDNInCallsBehavior_0 = 0
ISDNInCallsBehavior_1 = 2048
ISDNGeneralCCBehavior_0 = 32
ISDNGeneralCCBehavior_1 = 0
PSTNReserved3 = 8
TrunkAdministrativeState_0 = 2
TrunkAdministrativeState_1 = 0
DPNSSBehavior = 12
CasTrunkDialPlanName = ''
```

Under **Software Update →Configuration File**, click on **Browse…** and select the INI file on the local PC. Click **Send INI File** to upload the file to the gateway.



Click **OK** when the below pop-up appears to confirm a reboot. The M1000 Gateway configuration is complete.



5 Configure Avaya Meeting Exchange Enterprise Application Server

This section describes the steps for configuring SIP trunks between Avaya Meeting Exchange Enterprise Application Server and AudioCodes Mediant 1000 Modular Media Gateway. This configuration will allow both moderators and participants to access a conference and also for operators to dial participants who have difficulties accessing a conference. It's assumed, that Meeting Exchange is installed, configured and licensed as per **[1].**

The following steps describe the administrative procedures for configuring Avaya Meeting Exchange Enterprise Application Server:

- Configure SIP Listener
- Configure Dialout
- Configure DNIS Mappings
- Configure Incoming SIP URI Conversion Rules
- Restart the Conference Bridge

The following instructions assume the user is logged in to the Avaya Meeting Exchange Enterprise Application Server Linux console using SSH. It is highly recommended to backup the Meeting Exchange configuration files before editing.

5.1 Configure SIP Listener

The SIP signaling messages in the sample configuration are transmitted between the Avaya Meeting Exchange Enterprise Application Server and the SIP interface of the M1000 Gateway over a TCP channel. Configure the following settings to enable SIP connectivity on the Meeting Exchange Enterprise Application Server:

- Edit **/usr/ipcb/config/system.cfg** using the Linux vi tool or download the file to your local machine using a Secure Copy Protocol (SCP) client (i.e.: WinSCP) for editing.
- Add the IP address of the Meeting Exchange Enterprise Application Server: **IPAddress=10.10.21.51** as shown below
- Add a line to populate the From Header Field in SIP INVITE messages. The following SIP URI will be displayed when the conference operator calls a participant: **MyListener=sip:6000@10.10.21.51**
- Add a line to provide a SIP Device Contact address to use for acknowledging SIP messages: **respContact=sip:6000@10.10.21.51**

```
# ip address of the server
IPAddress=10.10.21.50

# request we will be listening to
MyListener=sip:6000@10.10.21.51

# if this setting is populated will Overwrite the contact field in responses
respContact=sip:6000@10.10.21.51
MaxChannelCount=3200
```

5.2 Configure Dialout

The FQDN of the M1000 gateway must be configured on Meeting Exchange Enterprise Application Server for dialout to work. The Application Server must be able to resolve the FQDN to an IP address, by either using its own hosts file or an external DNS server. For the sample configuration an external DNS server was authoritative for the **m1k.avaya.com** zone and had the following entry in its zone file:

```
m1k.avaya.com IN     A      10.10.9.82
```

Edit **/usr/ipcb/config/telnumToUri.tab** file with a text editor. Add the following line to the file to route outbound calls from the Avaya Meeting Exchange Enterprise Application Server to the extension range administered in **Section 4.5** of the IP to Tel Routing page of the M1000 Gateway.

```
        3*                sip:$0@m1k.avaya.com:5060;transport=tcp   default
```

5.3 Configure DNIS Mappings

The DNIS is the number that the phone users dial to access a conference. To map DNIS entries, run the **cbutil** utility on Avaya Meeting Exchange Enterprise Application Server as follows:

- Add 66666 as a new DNIS entry using the following command:

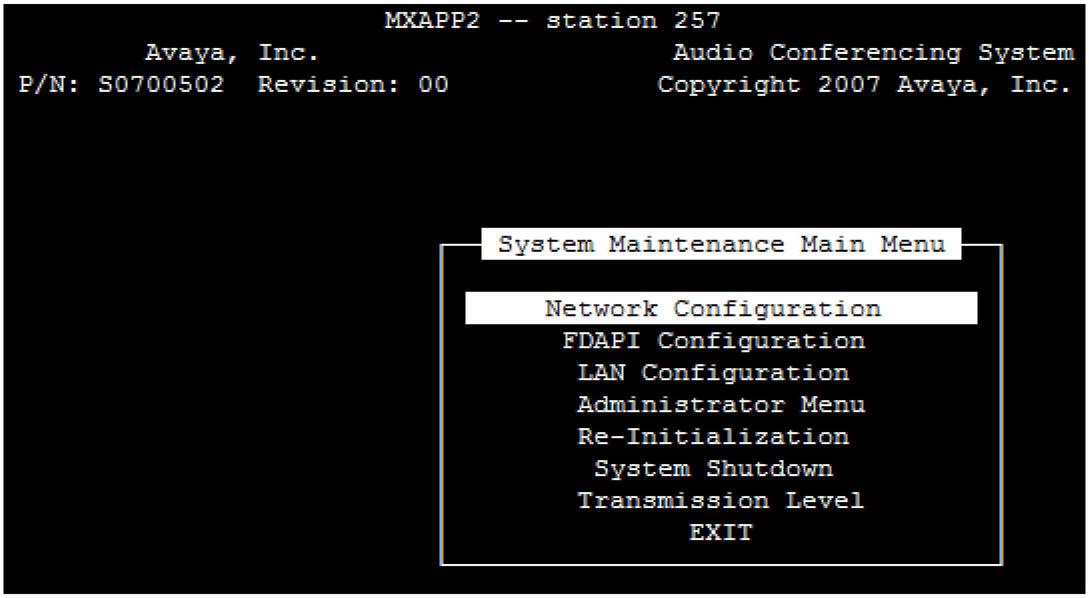
```
[mx6200-a ~]# cbutil add 66666 0 247 1 N SCAN
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
```

At the command prompt, enter **cbutil list** to verify the DNIS entries provisioned.

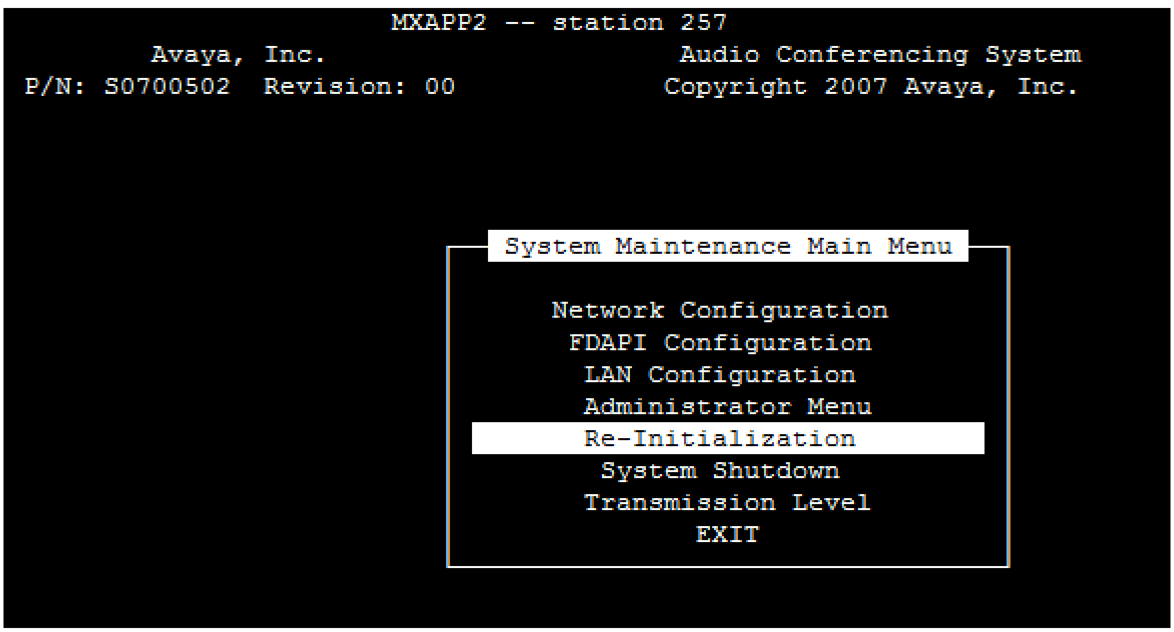
```
[mx6200-a ~]# cbutil list
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

DNIS   Grp Msg PS  CP Function On Failure Line Name Company Name Room Start  Room End
------ --- --- --- -- -------- ---------- --------- ------------ ----------- --------
66666   0   247 1   N  SCAN     ENTER                             0           0
```

5.4 Restart the Conference Bridge

After the configuration changes are made, restart the Meeting Exchange Enterprise Application Server. Log in to the MX Application Server using the **dcbmaint** account. Issue the **dcbmaint** command. The **System Maintenance Main Menu** screen is displayed.



Navigate to **Re-Initialization**.



Press **Enter** and at the prompt type **yes**.

```
                        MXAPP2 -- station 257
         Avaya, Inc.                        Audio Conferencing System
P/N: S0700502  Revision: 00                Copyright 2007 Avaya, Inc.



                                   System Re-Initialization

                                ARE YOU SURE? (yes/NO):
```

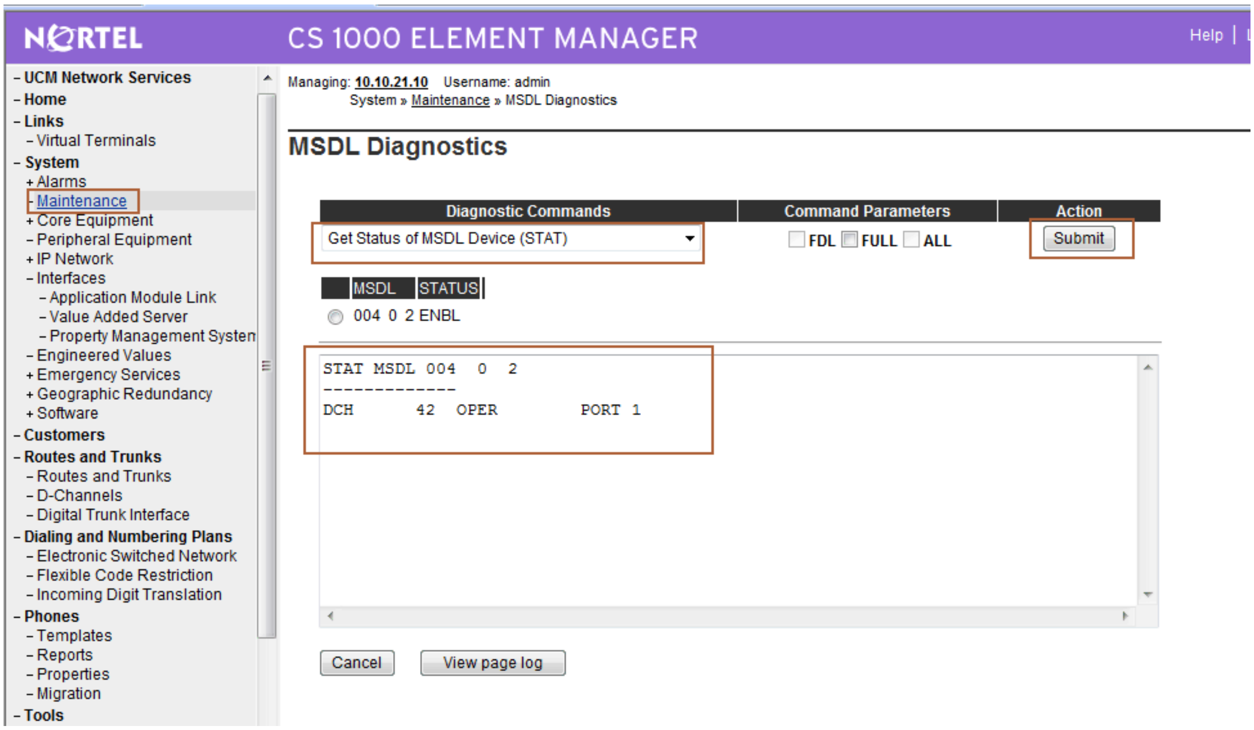
The **dcbmaint** utility terminates and the following message is displayed:

```
[dcbmaint@MXAPP2 sroot]$ dcbmaint
System Shutdown in Progress...
Please Stand By.

[dcbmaint@MXAPP2 sroot]$
```

6 Verification Steps

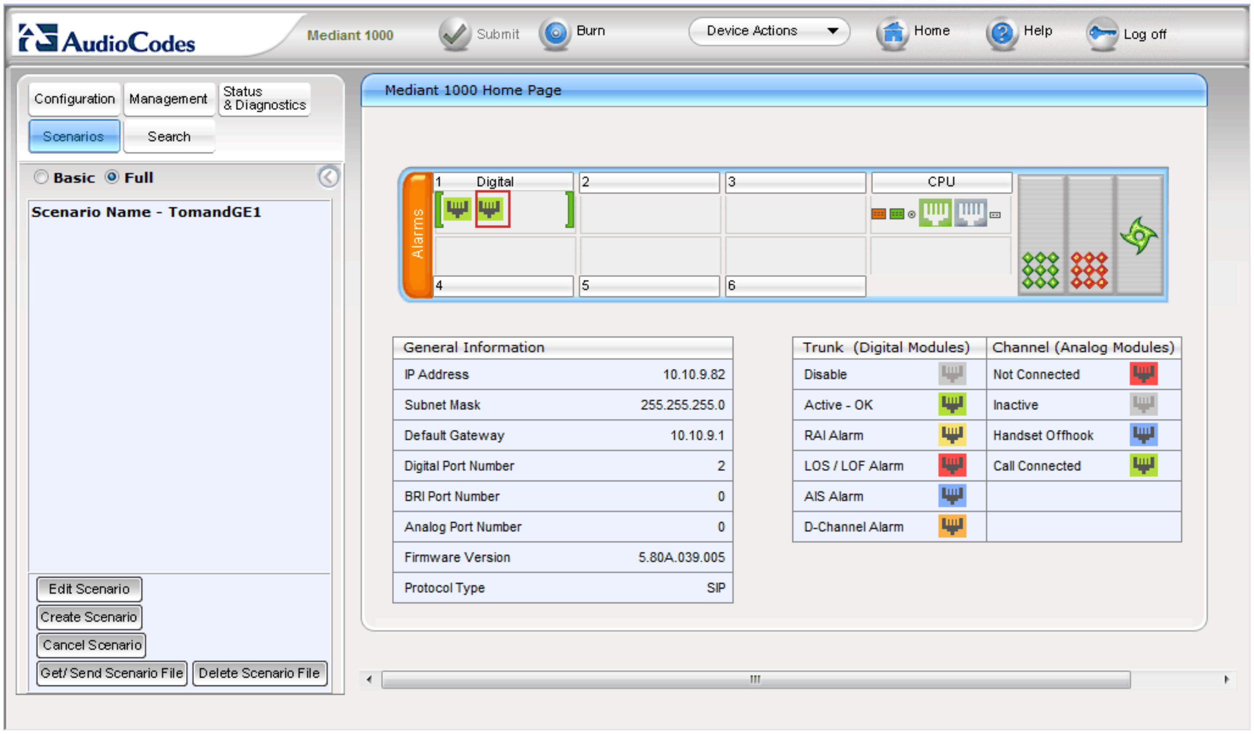
6.1 Verify Avaya Communication Server 1000

Select **System →Maintenance** on the left. Click **Get Status of MSDL Device (STAT)**. Click **Submit**. Verify that the D-channel of the QSIG trunk is operational. In the sample configuration **DCH 42** was used to route calls to the M1000 Gateway.



6.2 Verify AudioCodes Mediant 1000

Log in through the Web GUI and verify that the connector icon of the E1 port, shown below in Slot 1, is green in color. If not, use the colored legend on the page to determine what the error condition is and check the cabling and signaling parameters (e.g., framing, line code, clock master, network/user, etc.) of the AudioCodes M1000 and Avaya Communication Server 1000.



7 Verification Scenarios

Verification scenarios for the configuration described in these Application Notes included:

- Conference calls between various telephones on the Avaya Communication Server 1000E can be made using G.711MU/A and G.729. Proper display of the calling and called party name and number information was verified for all telephones.
- Dialout from the Operator phone to conference participants.

8 Conclusion

As illustrated in these Application Notes, Avaya Communication Server 1000 front-ended by the AudioCodes Mediant 1000 can be integrated with Avaya SIP products, including Avaya Meeting Exchange Enterprise 5.2. The following is a list of interoperability items to note:

- The ISDNGeneralCCBehavior value in the Audiocodes M1000 INI file needs to be set to "0" as the default value of 32 applies only to ETSI E1 lines (30B+D). This parameter enables handling the differences between the newer QSIG standard (ETS 300-172) and other ETSI-based standards (ETS 300-102 and ETS 300-403) in the conversion of B-channel ID values into timeslot values.
- The M1000 rejects incoming SIP calls when video codecs are offered in the SDP of an incoming INVITE from the MX. Video codecs are not supported by the M1000 Gateway in the current release.

9 Additional References

This section references the product documentation relevant to these Application Notes.

Avaya Meeting Exchange Support Documents:

[1] *Administering Meeting Exchange™ 5.2 Servers*, Doc ID 04-603419, 16-Nov-2009, available at http://support.avaya.com/css/P8/documents/100068644

Avaya Application Notes:

[2] *Configure an Avaya Centralized Messaging Solution with Avaya Communication Manager and Nortel Communication Server 1000 – Issue 1.0*, available at http://www.avaya.com.

[3] *Front-Ending Nortel Communication Server 1000 with an AudioCodes Mediant 1000 Modular Media Gateway to Support SIP Trunks to Avaya Aura™ Session Manager with Avaya Aura™ Communication Manager 5.2.1 as an Access Element – Issue 1.1*, available at http://www.avaya.com.

Avaya CS1000 Documentation

[4] *Nortel Communication Server 1000 - ISDN Primary Rate Interface Features Fundamentals* - NN43001-569 03.01 - 11 May 2009, available at http://www.nortel.com

AudioCodes Mediant 1000 Documents:

[5] *Manual Version 5.0 - Document #: LTRT-83301* available at http://AudioCodes.com

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 Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com