



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**).

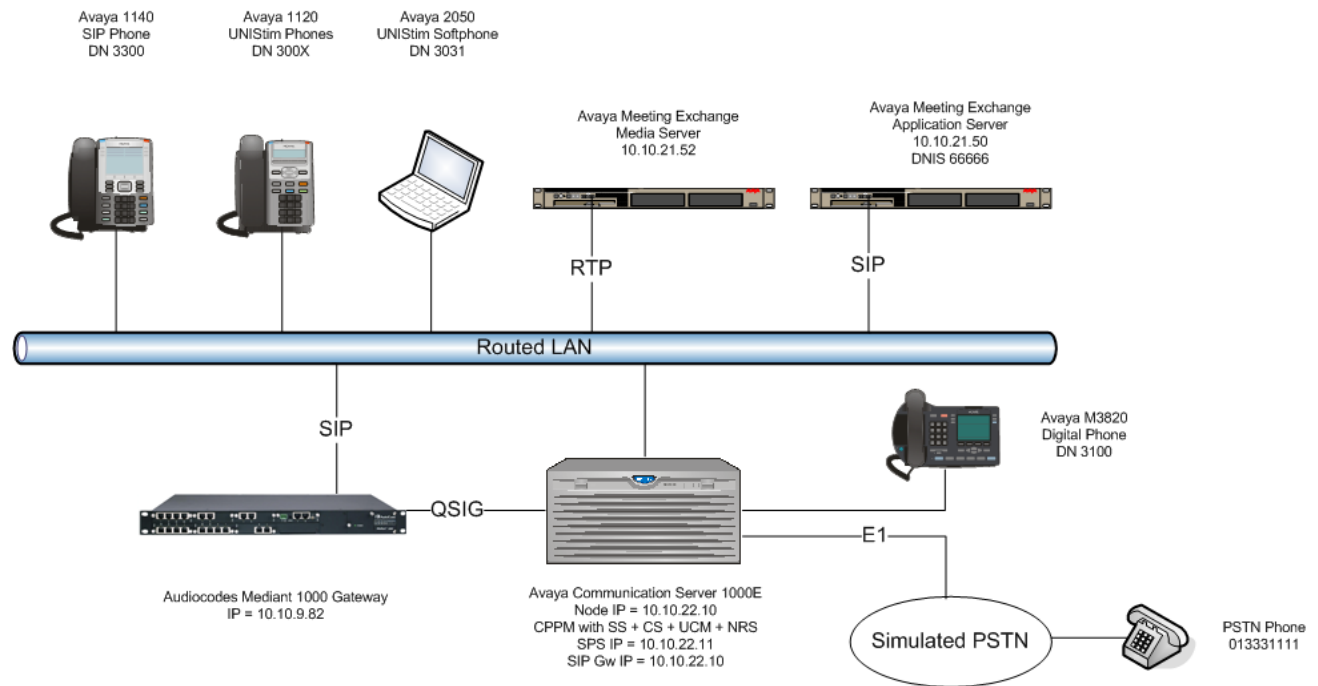


Figure 1 – CS1000 with AudioCodes and Meeting Exchange Enterprise

2 Equipment and Software Validated

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

Equipment	Software Version
Avaya Communication Server 1000E <ul style="list-style-type: none">• 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.

Use this page to access the server by IP address. You will need to log in again when switching to another server, even if it is in the same security domain.

Important: Only accounts which have been previously created in the primary security server are allowed. Expired or reset passwords that normally must be changed during login will fail authentication in this mode (use the link to manual password change instead). Local OS-authenticated User IDs cannot be used.

User ID:

Password:

[Change Password](#)

[Go to central login for Single Sign-On](#)

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**”.

NORTEL UNIFIED COMMUNICATIONS MANAGEMENT [Help](#) | [Logout](#)

Host Name: 10.10.22.19 Software Version: 02.00.0055.00(3266) User Name admin

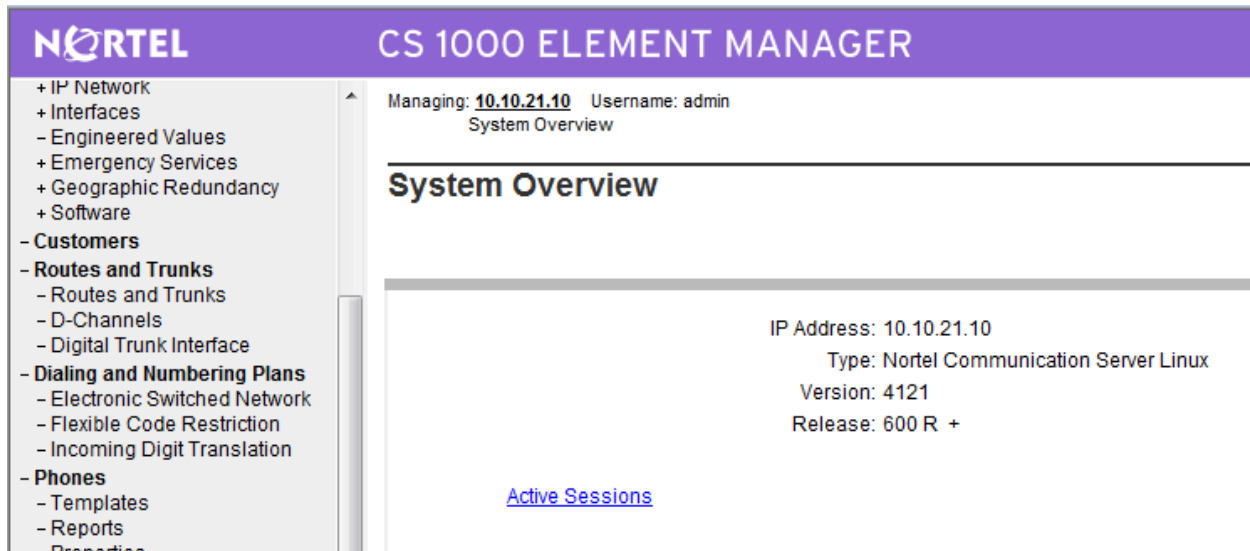
Elements

New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service.

	Element Name	Element Type	Release	Address	Description
1	EM on ssg	CS1000	6.0	10.10.21.10	New element.
2	EM on bodag-r022011	CS1000	6.0	10.10.21.10	New element.
3	ssq.avaya.com (primary)	Linux Base	6.0	10.10.22.19	Base OS element.
4	bodag-r022011.cs1k.avaya.com (member)	Linux Base	6.0	10.10.22.11	Base OS element.
5	10.10.21.12	Media Gateway Controller	6.0	10.10.21.12	New element.
6	NRSM on bodag-r022011	Network Routing Service	6.0	10.10.21.10	New element.
7	NRSM on ssg	Network Routing Service	6.0	10.10.21.19	New element.

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)**

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 10.10.21.10 Username: admin
Tools » Logs and reports » Equipped Feature Packages

Equipped Feature Packages

	Package Description	Package Name	Package Number
16	Digit Display	DDSP	19
17	Office Data Administration System	ODAS	20
18	Dial Intercom	DI	21
19	Direct Inward System Access	DISA	22
20	Charge Account for CDR	CHG	23
21	Charge Account/Authorization code	CAB	24
22	Basic Authorization code	BAUT	25
23	Centralized Attendant Service (Main)	CASM	26
24	Centralized Attendant Service (Remote)	CASR	27
25	Basic Queuing	BQUE	28
26	Network Traffic must have NWK packages.	NTRF	29
27	Network Class of Service	NCOS	32
28	Call Park	CPRK	33
29	System Speed Call	SSC	34
30	UST	UST	35

Items per page: 100 First | Prev | Next | Last

Copyright © 2002-2010 Nortel Networks. All rights reserved.

3.4 Administer E1 Card

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

The screenshot displays the Nortel CS 1000 Element Manager web interface. The top header bar is purple with the Nortel logo on the left, 'CS 1000 ELEMENT MANAGER' in the center, and 'Help | Logout' on the right. Below the header, a status bar indicates 'Managing: 10.10.21.10 Username: admin' and the breadcrumb 'System » Core Equipment » Loops (Common Equipment)'. The left sidebar contains a tree view with categories: Home, Links, System, Core Equipment, and IP Network. Under 'Core Equipment', 'Loops' is selected and highlighted. The main content area is titled 'Loops (Common Equipment)' and contains a section for '- Basic IP Configuration'. This section includes several input fields and buttons: 'Change to Common Equipment parameters' with a dropdown set to 'CEQU'; 'Extended Conference/TDS/MFS' with an 'Edit...' button; 'TDS Loop Number' with a text field containing '104 004 0 MGC' and 'Edit...' and 'Add...' buttons; 'Conference Loop Numbers' with a text field containing '105 004 0 30 MGC' and 'Edit...' and 'Add...' buttons; and 'Digital Trunk Interface Loop Number' with an 'Add...' button. Below this is a section for '+ Feature Packages' which is currently empty. At the bottom right of the configuration area are 'Save' and 'Cancel' buttons.

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**.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: **10.10.21.10** Username: admin
System » Core Equipment » Loops (Common Equipment)

Loops (Common Equipment)

- Basic IP Configuration

Change to Common Equipment parameters :

Extended Conference/TDS/MFS :

TDS Loop Number :

Conference Loop Numbers :

Digital Trunk Interface Loop Number :

- Feature Packages

- Integrated Digital Access

Package: 122 -- Unequipped	<input type="button" value="To Order"/>
Package: 129	
- Dial Tone Detection

Package: 138 -- Unequipped	<input type="button" value="To Order"/>
Package: 154	
- 2.0 Mb/s Primary Rate Interface

2.0 Mb/s Primary Rate Interface Loop Number	<input type="text" value="042 004 0 02"/>	<input type="button" value="Edit..."/>	<input type="button" value="Add..."/>
---	---	--	---------------------------------------

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

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: **10.10.21.10** Username: admin
System » Core Equipment » **Loops (Common Equipment)** » 2.0 Mb/s Primary Rate Interface Loop Number Configuration

2.0 Mb/s Primary Rate Interface Loop Number Configuration

2.0 Mb/s Primary Rate Interface Loop Number:

Media Gateway Card: (supl# sh# card#)

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**.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: **10.10.21.10** Username: admin
Routes and Trunks » **D-Channels**

D-Channels

Maintenance

- [D-Channel Diagnostics \(LD 96\)](#)
- [Network and Peripheral Equipment \(LD 32, Virtual D-Channels\)](#)
- [MSDL Diagnostics \(LD 96\)](#)
- [D-Channel Expansion Diagnostics \(LD 48\)](#)

Configuration

Choose a D-Channel Number: and type:

Channel	Type	Card Type	Description	Edit
Channel: 15	Type: DCH	Card Type: DCIP	Description: VtrkNode1000	<input type="button" value="Edit"/>
Channel: 20	Type: DCH	Card Type: DCIP	Description: VtrkNode1000	<input type="button" value="Edit"/>

Copyright © 2002-2010 Nortel Networks. All rights reserved.

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**

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

D-Channels 42 Property Configuration

- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D channel Card Type (CTYP)	MSDL
Media Gateway Card (MG_CARD)	004 0 02 (supl# sh# card#) *
Group number (GRP)	
Device number (DNUM)	
Port number (PORT)	1
Designator (DES)	QSIG
Recovery to Primary (RCVP)	<input type="checkbox"/>
PRI loop number for Backup D-channel (BCHL)	
User (USR)	Primary Rate Interface (PRI) *
Interface type for D-channel (IFC)	Q Reference Signalling Point interface (EGF4) ▼
Country (CNTY)	ETS 300 =102 basic protocol (ETSI) ▼
D-Channel PRI loop number (DCHL)	42
Primary Rate Interface (PRI)	<input type="button" value="more PRI"/>
Secondary PRI2 loops (PRI2)	
Release ID of the switch at the far end (RLS)	25 ▼
Central Office switch type (CO_TYPE)	100% compatible with Bellcore standard (STD) ▼

Copyright © 2002-2010 Nortel Networks. All rights reserved.

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.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Primary Rate Interface (PRI) [Field]

Secondary PRI2 loops (PRI2) [Field]

Release ID of the switch at the far end (RLS) 25

Central Office switch type (CO_TYPE) 100% compatible with Bellcore standard (STD)

Integrated Services Signaling Link Maximum (ISLM) 200 Range: 1 - 4000

- Basic options (BSCOPT)

Primary D-channel for a backup DCH (PDCH) Range: 0 - 254

- PINX customer number (PINX_CUST) 0

- Progress signal (PROG) [Field]

- Calling Line Identification (CLID) Prefix = 0 for North American dialing plan. (OPT0)

- Output request Buffers (OTBF) 128

- D-channel transmission Rate (DRAT) 64 kb/s clear (64KC)

- Channel Negotiation option (CNEG) No alternative acceptable, exclusive. (1)

- Remote Capabilities (RCAP) [Edit]

+ - Change protocol timer value (TIMR)

- B channel Service messaging. (BSRV) [Field]

+ Advanced options (ADVOPT)

+ Feature Packages

[Submit] [Refresh] [Delete] [Cancel]

Copyright © 2002-2010 Nortel Networks. All rights reserved.

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)**

Managing: 10.10.21.10 Username: admin
Routes and Trunks » D-Channels » D-Channels 42 Property Configuration » - Remote Capabilities Configuration

- Remote Capabilities Configuration

Input Description	Input Value
Basic rate interface (BRI)	<input type="checkbox"/>
Call completion on busy using integer value (CCBI)	<input type="checkbox"/>
Call completion on busy using object identifier (CCBO)	<input type="checkbox"/>
Call completion on busy for QSIG and EuroISDN BRI (CCBS)	<input type="checkbox"/>
Call completion on no response using integer value (CCNI)	<input type="checkbox"/>
Call completion on no response using object identifier (CCNO)	<input type="checkbox"/>
Call completion to no reply for QSIG and EuroISDN BRI (CCNR)	<input type="checkbox"/>
Network call park (CPK)	<input type="checkbox"/>
Connected line identification presentation (COLP)	<input checked="" type="checkbox"/>
Call transfer integer (CTI)	<input type="checkbox"/>
Call transfer object (CTO)	<input type="checkbox"/>
Diversion info. is sent using integer value (DV1I)	<input type="checkbox"/>
Diversion info. is sent using object identifier (DV1O)	<input type="checkbox"/>
Rerouting requests processed using integer value (DV2I)	<input type="checkbox"/>
Rerouting requests processed using object identifier (DV2O)	<input type="checkbox"/>
Diversion info. sent. rerouting requests processed (DV3I)	<input checked="" type="checkbox"/>
EuroISDN - div info sent rerouting req processed (DV3O)	<input type="checkbox"/>

Copyright © 2002-2010 Nortel Networks. All rights reserved.

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**.

UCM Network Services

- Home
- Links
 - Virtual Terminals
- System
 - Alarms
 - Maintenance
 - Core Equipment
 - Peripheral Equipment
 - IP Network
 - Interfaces
 - Engineered Values
 - Emergency Services
 - Geographic Redundancy
 - Software
- Customers
- Routes and Trunks
 - Routes and Trunks
 - D-Channels**
 - Digital Trunk Interface
- Dialing and Numbering Plans
 - Electronic Switched Network
 - Flexible Code Restriction
 - Incoming Digit Translation
- Phones
 - Templates
 - Reports
 - Properties
 - Migration
- Tools
 - Backup and Restore
 - Date and Time
 - Logs and reports
- Security

Message waiting interworking with DMS-100 (MWI) ☐

Network access data (NAC) ☐

Network call trace supported (NCT) ☐

Network name display method 1 (ND1) ☐

Network name display method 2 (ND2) ☐

Network name display method 3 (ND3) ☐

Name display - integer ID coding (NDI) ☐

Name display - object ID coding (NDO) ☐

Path replacement uses integer values (PRI) ☐

Path replacement uses object identifier (PRO) ☐

Release Link Trunks over IP (RLTI) ☐

Remote virtual queuing (RVQ) ☐

Trunk anti-tromboning operation (TAT) ☐

User to user service 1 (UUS1) ☐

NI-2 name display option. (NDS) ☐

Message waiting indication using integer values (QMVI) ☒

Message waiting indication using object identifier (QMWI) ☐

User to user signalling (UUI) ☐

Return - Remote Capabilities Cancel

Copyright © 2002-2010 Nortel Networks. All rights reserved.

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**.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 10.10.21.10 Username: admin
Routes and Trunks » Routes and Trunks

Routes and Trunks

+ Customer: 0	Total routes: 6	Total trunks: 166	Add route
- Customer: 1	Total routes: 0	Total trunks: 0	Add route

UCM Network Services

- Home
- Links
 - Virtual Terminals
- System
 - Alarms
 - Maintenance
 - Core Equipment
 - Peripheral Equipment
 - IP Network
 - Interfaces
 - Engineered Values
 - Emergency Services
 - Geographic Redundancy
 - Software
- Customers
- Routes and Trunks**
 - Routes and Trunks**
 - D-Channels
 - Digital Trunk Interface
- Dialing and Numbering Plans
 - Electronic Switched Network
 - Flexible Code Restriction

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

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 10.10.21.10 Username: admin
Routes and Trunks » Routes and Trunks » Customer 0, Route 42 Property Configuration

Customer 0, Route 42 Property Configuration

- Basic Configuration

Route data block (RDB) (TYPE)

Customer number (CUST)

Route number (ROUT)

Designator field for trunk (DES)

Trunk type (TKTP)

Incoming and outgoing trunk (ICOG)

Access code for the trunk route (ACOD)

Trunk type M911P (M911P) ☐

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

Digital trunk route (DTRK) ☒

ISDN BRI packet handler route (BRIP) ☐

Digital trunk type (DGTP)

Integrated services digital network option (ISDN) ☒

Mode of operation (MODE)

Interface type for route (IFC)

Copyright © 2002-2010 Nortel Networks. All rights reserved.

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)

Integrated services digital network option (ISDN) ☒

- Mode of operation (MODE) ISDN/PRA route, DTRK must be YES (PRA) ▼

- Interface type for route (IFC) Q Reference Signalling Point (EGF4) ▼

- Send billing number (SBN) ☐

- Private network identifier (PNI) 00001 Range: 0 - 32700

- Call type for outgoing direct dialed TIE route (CTYP) Unknown Call type (UKWN) ▼

- Insert ESN access code (INAC) ☐

- Display of access prefix on CLID (DAPC) ☐

- Mobile extension route (MBXR) ☐

Route Options

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

Off-hook queuing (OHQ) ☐

Off-hook queue threshold (OHQT) 0 ▼

Call back queuing (CBQ) ☐

Number of digits (NDIG) 2 ▼

Authcode (AUTH) ☐

- General Options

Data selection (DSEL) Voice or Data route (VOD) ▼

Trunk access restriction group (TARG)

Search method for outgoing trunk member (SRCH) Linear Hunting Search method (LIN) ▼

Alternate trunk route for outgoing trunks (STEP) Range: 0 - 511

Actual outgoing toll digits to be ignored for code restriction (OABS)

Display IDC name (DNAM) ☐

Enable equal access restrictions (EQAR) ☐

ACD DNIS route (DNIS) ☐

Include DNIS number in CDR records (DCDR) ☐

+ Advanced Configurations

Submit Refresh Delete Cancel

Copyright © 2002-2010 Nortel Networks. All rights reserved.

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.

CS 1000 ELEMENT MANAGER
Help | Logout

- UCM Network Services
- Home
- Links
 - Virtual Terminals
- System
 - + Alarms
 - Maintenance
 - + Core Equipment
 - Peripheral Equipment
 - + IP Network
 - + Interfaces
 - Engineered Values
 - + Emergency Services
 - + Geographic Redundancy
 - + Software
- Customers
- Routes and Trunks
 - [Routes and Trunks](#)
 - D-Channels
 - Digital Trunk Interface
- Dialing and Numbering Plans
 - Electronic Switched Network
 - Flexible Code Restriction
 - Incoming Digit Translation

Managing: **10.10.21.10** Username: admin
Routes and Trunks » Routes and Trunks

Routes and Trunks

- Customer: 0	Total routes: 6	Total trunks: 166	Add route	
+ Route: 1	Type: TIE	Description: SIPNRS	Edit	Add trunk
+ Route: 15	Type: TIE	Description: VTRKNODE1000SIP	Edit	Add trunk
+ Route: 16	Type: TIE	Description: VTRKNODE1000H323	Edit	Add trunk
+ Route: 17	Type: TIE	Description: VTRKNODE1001SIPL	Edit	Add trunk
+ Route: 20	Type: TIE	Description: NOCRYP	Edit	Add trunk
+ Route: 42	Type: TIE	Description: QSIG	Edit	Add trunk
- Customer: 1	Total routes: 0	Total trunks: 0	Add 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.

Nortel CS 1000 ELEMENT MANAGER Help | Logout

Customer 0, Route 42, New Trunk Configuration

- Basic Configuration

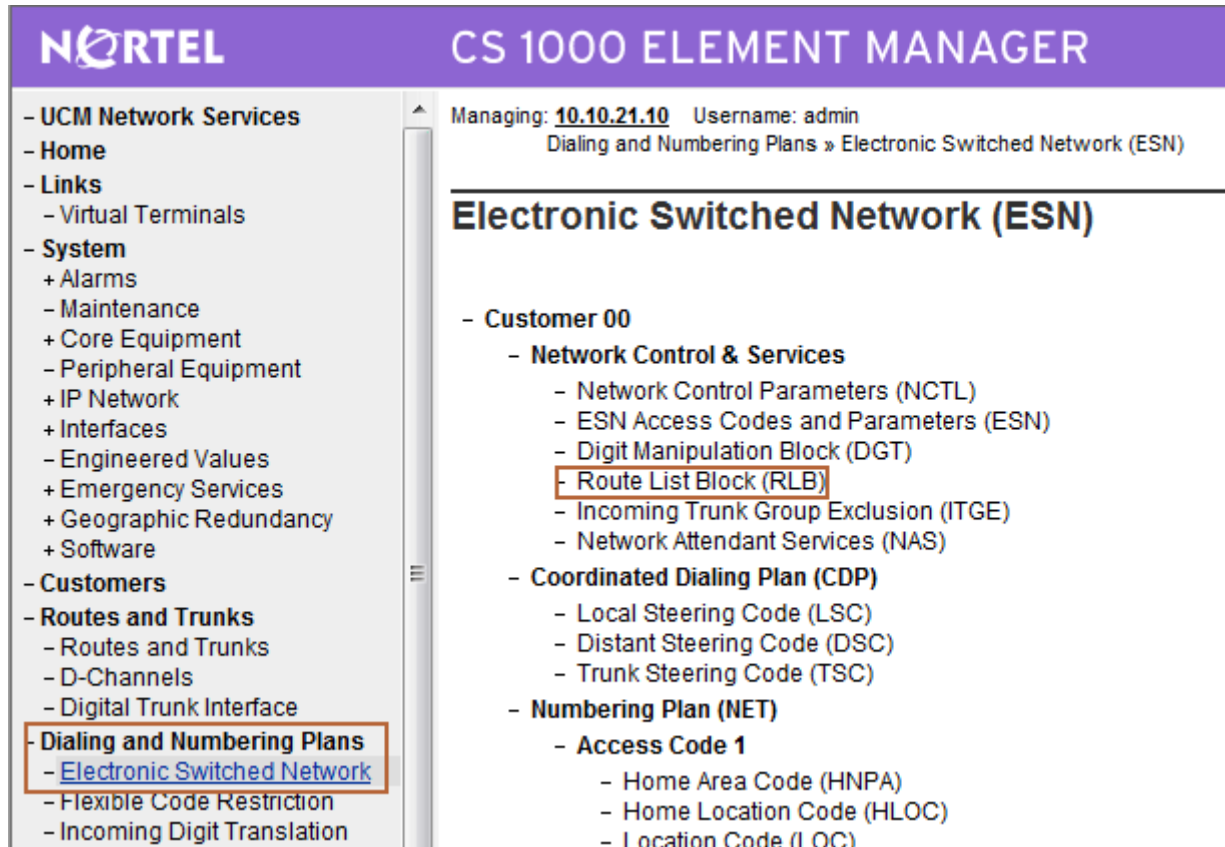
Input Description	Input Value
Multiple trunk input number (MTINPUT)	30
Trunk data block (TYPE)	TIE trunk data block (TIE)
Terminal Number (TN)	042 01
Designator field for trunk (DES)	QSIG
Extended Trunk (XTRK)	
Route number, Member number (RTMB)	42 1
Level 3 Signaling (SIGL)	
Card Density (CDEN)	Octal Density (8D)
Start arrangement Incoming (STRI)	
Start arrangement Outgoing (STRO)	
Trunk Group Access Restriction (TGAR)	1
Channel ID for this trunk. (CHID)	
Network Music (NMUS)	<input type="checkbox"/>
Increase or decrease the member numbers (INC)	Increase channel and member number (YES)
Class of Service (CLS)	Edit

+ Advanced Trunk Configurations

Copyright © 2007-2010 Nortel Networks. All rights reserved.

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**.

Route List Blocks

Please enter a route list index 0 - 999

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).

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

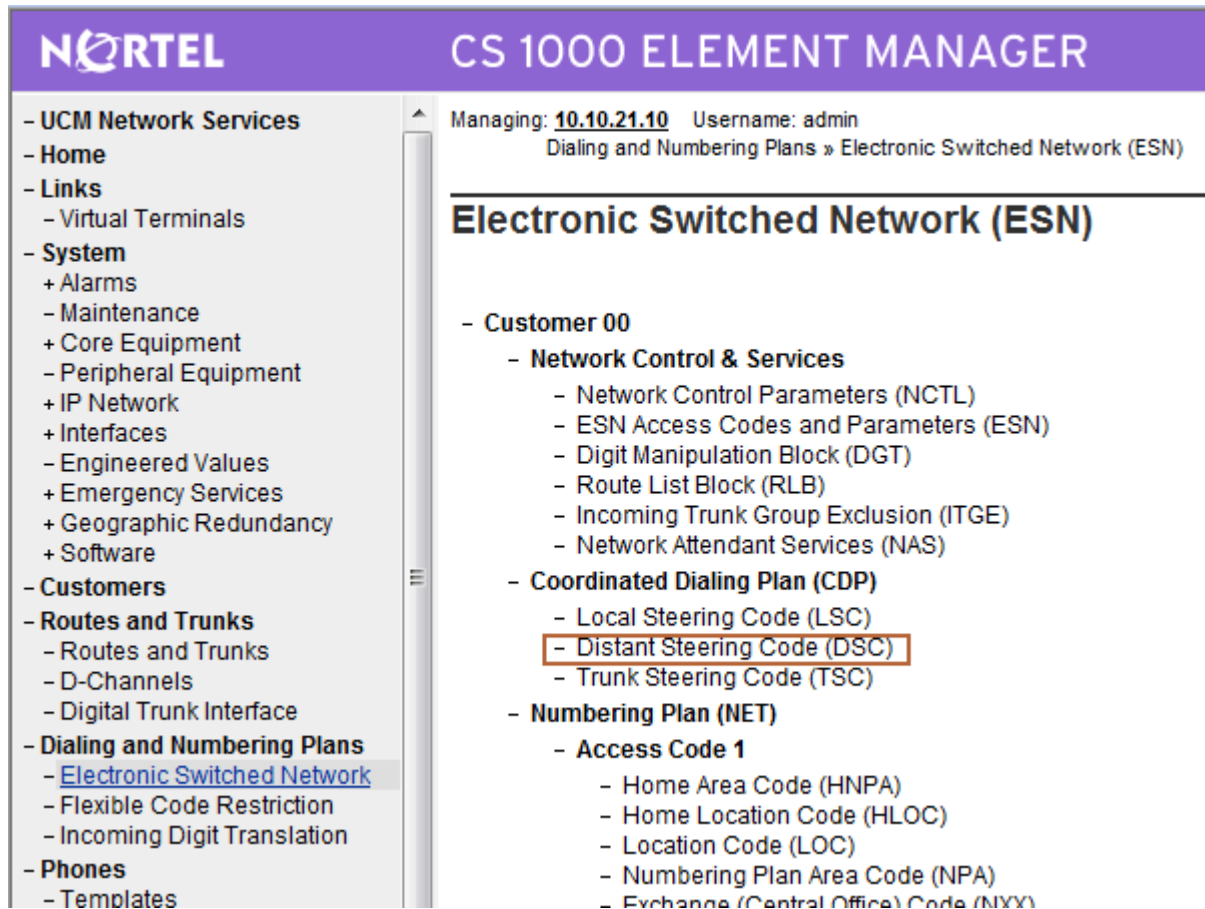
- UCM Network Services
- Home
- Links
 - Virtual Terminals
- System
 - + Alarms
 - Maintenance
 - + Core Equipment
 - Peripheral Equipment
 - + IP Network
 - + Interfaces
 - Engineered Values
 - + Emergency Services
 - + Geographic Redundancy
 - + Software
- Customers
- Routes and Trunks
 - Routes and Trunks
 - D-Channels
 - Digital Trunk Interface
- Dialing and Numbering Plans
 - Electronic Switched Network
 - Flexible Code Restriction

Route List Block

Input Description	Input Value
Route List Index (RLI):	42
Entry Number for the Route List (ENTR):	0 (0 - 63)
Local Termination entry (LTER):	<input type="checkbox"/>
Route Number (ROUT):	42
Skip Conventional Signaling (SCNV):	<input type="checkbox"/>
Use Tone Detector (TDET):	<input type="checkbox"/>
Time of Day Schedule (TOD):	0
Entry is a VNS Route (VNS):	<input type="checkbox"/>
Conversion to LDN (CNV):	<input type="checkbox"/>
Expensive Route (EXP):	<input type="checkbox"/>
Facility Restriction Level (FRL):	0 (0 - 7)

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**.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: [10.10.21.10](#) Username: admin
Dialing and Numbering Plans » [Electronic Switched Network \(ESN\)](#) » Customer 00 » Coordinated Dialing Plan (CDP) » [Distant Steering Code List](#) » Distant Steering Code

Distant Steering Code

Input Description	Input Value
Distant Steering Code (DSC):	66
Flexible Length number of digits (FLEN):	05 (0 - 10)
Display (DSP):	Local Steering Code (LSC)
Remote Radio Paging Access (RRPA):	<input type="checkbox"/>
Route List to be accessed for trunk steering code (RLI):	42
Collect Call Blocking (CCBA):	<input type="checkbox"/>
maximum 7 digit NPA code allowed (NPA):	
maximum 7 digit NXX code allowed (NXX):	

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? xxxxxx PASS? yyyyy TTY #00 LOGGED IN xxxxxx 16:50 06/4/2010 > ld 96 . enl MSDL 4 0 2 FDL	Issue the login command. Enter a valid user ID. Enter a valid user password. A sample response indicating successful log in. 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

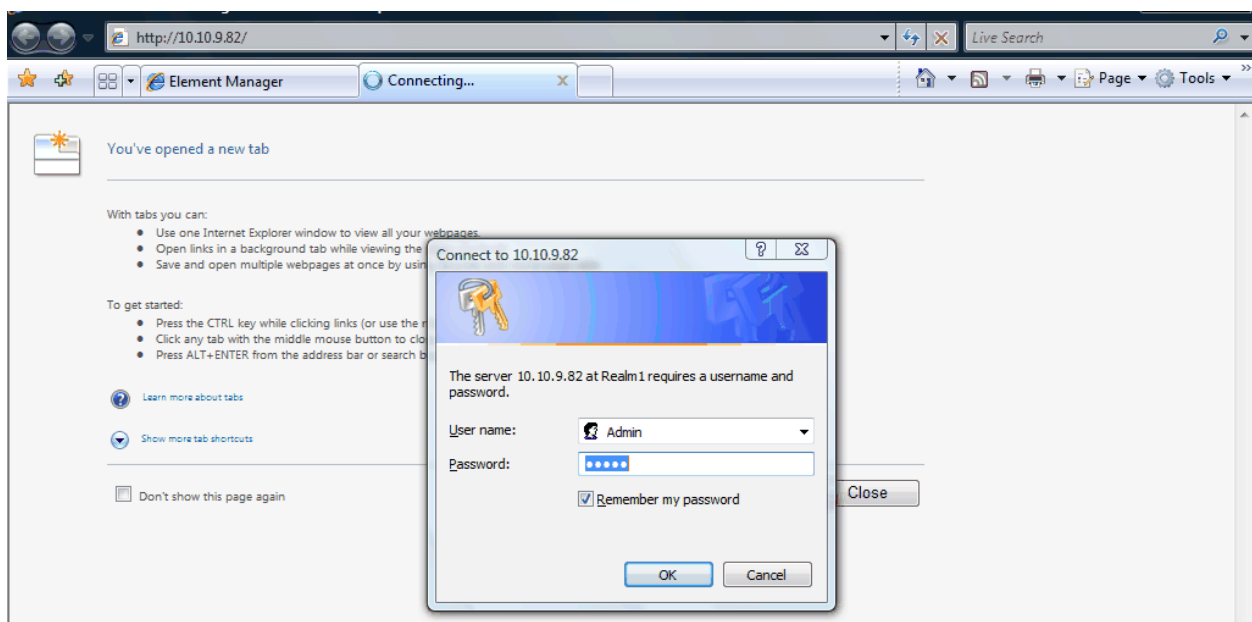
The following sections describe the configuration steps required to implement E1/PRI QSIG and SIP trunks and inter-trunk routing on the AudioCodes Mediant 1000, using the web interface. It is assumed that basic hardware and software installation has been performed as described in [5].

This section focuses on the following configuration areas:

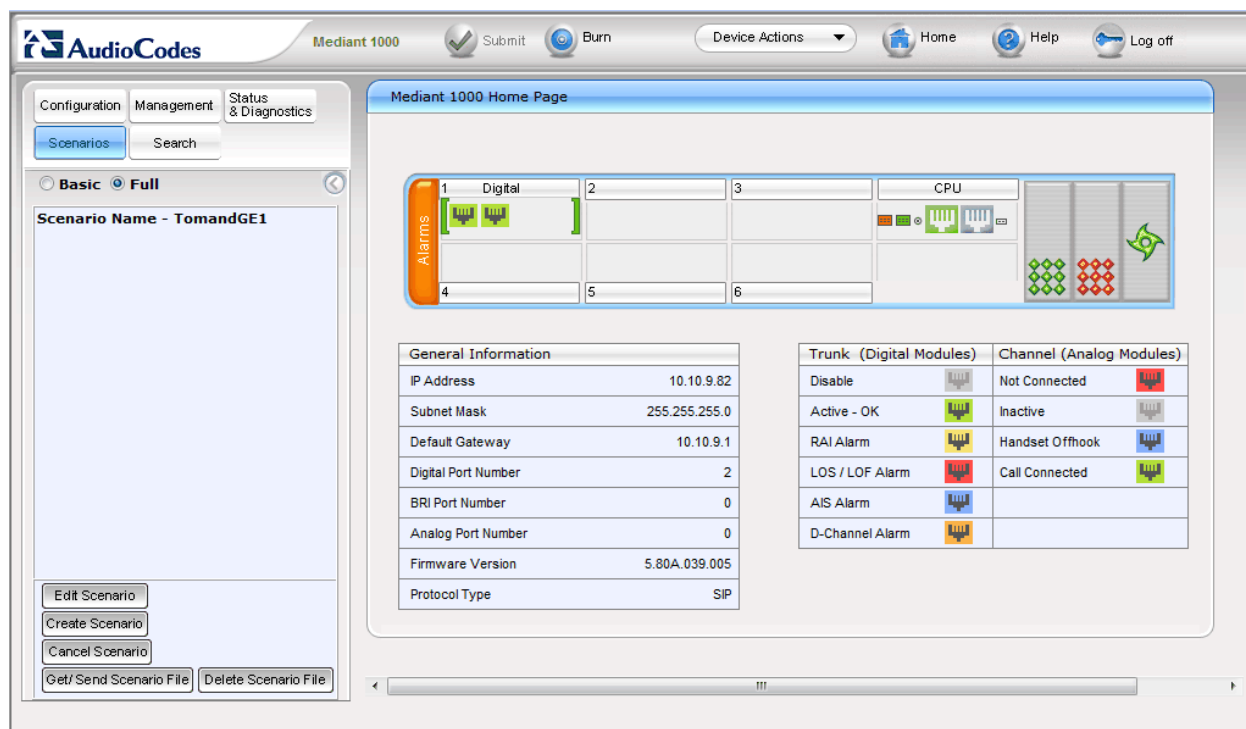
- Access Web Configuration Interface
- Configure TDM Bus Settings
- PSTN Trunk Settings
- SIP Protocol Parameters
- Routing Tables
- Configure PSTN Trunk Group
- Customize the INI File

4.1 Access Web Configuration Interface

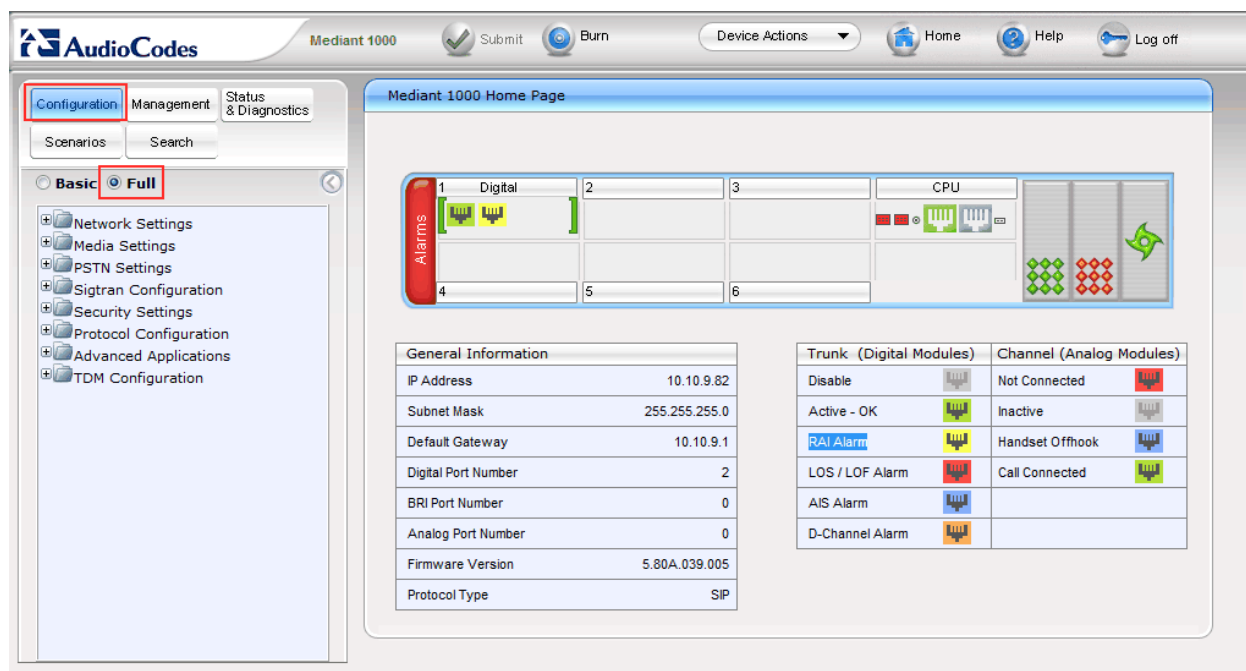
Open an instance of a web browser and connect to the AudioCodes GUI at the following address: **http://<Management IP address of the M1000>**. Log in using an appropriate Username and Password.



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**

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a tree view with the following structure:

- Configuration
 - Scenarios
 - Search
 - Basic
 - Sigtran Configuration
 - Security Settings
 - Protocol Configuration
 - Applications Enabling
 - Media Realm Configuration
 - Protocol Definition
 - Proxies, Registration, IP Groups
 - Coders And Profile Definitions
 - SIP Advanced Parameters
 - Manipulation Tables
 - Routing Tables
 - Trunk Group
 - Trunk Group
 - Trunk Group Settings
 - Digital Gateway
 - IP Media
 - Advanced Applications
 - TDM Configuration
 - TDM Bus Settings
 - Full
- Management
- Status & Diagnostics

The main content area is titled "TDM Bus Settings" and displays a "Basic Parameter List" table:

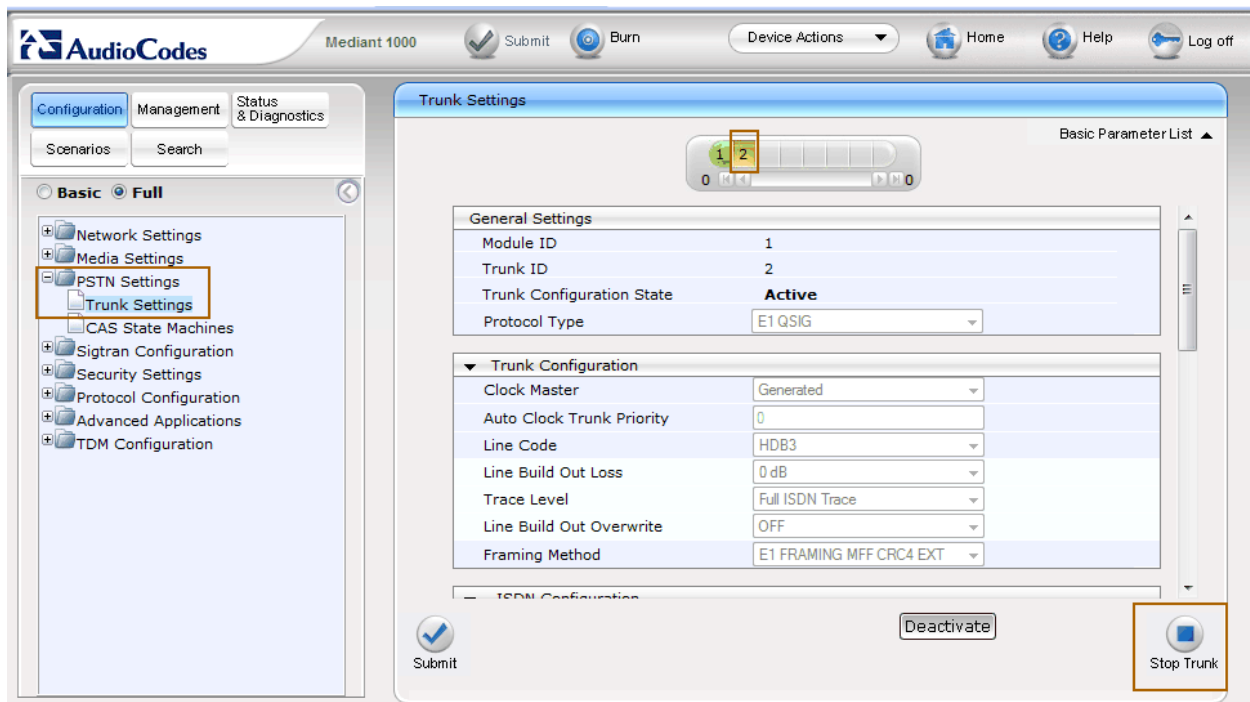
Parameter	Value
PCM Law Select	ALaw
TDM Bus Type	Framers
Idle PCM Pattern	85
Idle ABCD Pattern	0x0F
TDM Bus Local Reference	1
TDM Bus PSTN Auto FallBack Clock	Disable
TDM Bus Clock Source	Internal

A "Submit" button is located at the bottom right of the configuration area.

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.

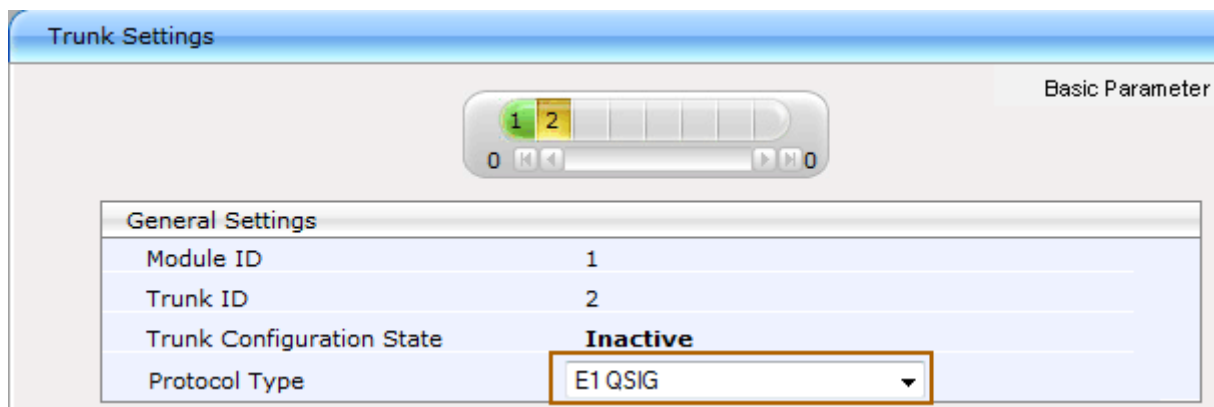
Click **Stop Trunk**, which will enable editing of the parameters.



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**

▼ Trunk Configuration	
Clock Master	Generated
Auto Clock Trunk Priority	0
Line Code	HDB3
Line Build Out Loss	0 dB
Trace Level	Full ISDN Trace
Line Build Out Overwrite	OFF
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**

▼ ISDN Configuration	
ISDN Termination Side	User side
Q931 Layer Response Behavior	0x0
Outgoing Calls Behavior	0x400
Incoming Calls Behavior	0x800
General Call Control Behavior	0x0
NFAS Group Number	0
IUA Interface ID	-1
NFAS Interface ID	255
D-channel Configuration	PRIMARY

Under **Miscellaneous**:

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

PSTN Alert Timeout	180
Transfer Mode	Disable
Local ISDN Ringback Tone Source	PBX
Set PI in Rx Disconnect Message	Not Configured
ISDN Transfer Capabilities	Speech
Progress Indicator to ISDN	Not Configured
Enable Receiving of Overlap Dialing	Enable
B-channel Negotiation	Exclusive
Out-Of-Service Behavior	Default
Remove Calling Name	Use Global Parameter
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**.

AudioCodes Mediant 1000

Configuration Management Status & Diagnostics

Scenarios Search

Basic Full

Network Settings

Media Settings

PSTN Settings

Trunk Settings

CAS State Machines

Sigtran Configuration

Security Settings

Protocol Configuration

Advanced Applications

TDM Configuration

Trunk Settings

Basic Parameter List

PSTN Alert Timeout	180
Transfer Mode	Disable
Local ISDN Ringback Tone Source	PBX
Set PI in Rx Disconnect Message	Not Configured
ISDN Transfer Capabilities	Speech
Progress Indicator to ISDN	Not Configured
Enable Receiving of Overlap Dialing	Enable
B-channel Negotiation	Exclusive
Out-Of-Service Behavior	Default
Remove Calling Name	Use Global Parameter
Play Ringback Tone to Trunk	Don't Play

Apply Trunk Settings

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

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a navigation tree with the following items: Configuration, Management, Status & Diagnostics, Scenarios, Search, Basic, Full, Trunk Settings, CAS State Machines, Sigtran Configuration, Security Settings, Protocol Configuration, Applications Enabling, Media Realm Configuration, Protocol Definition, SIP General Parameters, DTMF & Dialing, Proxies, Registration, IP Groups, Coders And Profile Definitions, SIP Advanced Parameters, Manipulation Tables, Routing Tables, Trunk Group, Digital Gateway, IP Media, Advanced Applications, and TDM Configuration. The 'SIP General Parameters' item is highlighted. The main area displays the 'SIP General Parameters' configuration page. The parameters are listed in a table with the following values:

SIP General	
NAT IP Address	0.0.0.0
PRACK Mode	Supported
Channel Select Mode	Descending
Enable Early Media	Disable
183 Message Behavior	Progress
Session-Expires Time	0
Minimum Session-Expires	90
Session Expires Method	Re-INVITE
Asserted Identity Mode	Adding PAsserted Identity
Fax Signaling Method	T.38 Relay
Detect Fax on Answer Tone	Initiate T.38 on Preamble
SIP Transport Type	TCP
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPs	Disable
Enable TCP Connection Reuse	Enable

A 'Submit' button is located at the bottom right of the configuration page.

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

The screenshot shows the AudioCodes Mediant 1000 configuration web interface. On the left is a navigation tree with categories like Configuration, Management, and Status & Diagnostics. Under Configuration, the 'Proxy & Registration' option is selected and highlighted with an orange box. The main area displays a 'Basic Parameter List' for Proxy & Registration. Several parameters are highlighted with orange boxes: 'Use Default Proxy' (set to No), 'Prefer Routing Table' (set to Yes), 'Always Use Proxy' (set to Disable), and 'Redundant Routing Mode' (set to Routing Table). Other parameters include Proxy Name, Redundancy Mode (Parking), Proxy IP List Refresh Time (60), Enable Fallback to Routing Table (Enable), SIP ReRouting Mode (Standard Mode), Enable Registration (Disable), Gateway Name (ac1000.silstack.com), Gateway Registration Name, DNS Query Type (A-Record), Proxy DNS Query Type (A-Record), Number of RTX Before Hot-Swap (3), and Use Gateway Name for OPTIONS (No). At the bottom of the main area are buttons for 'Register', 'Un-Register', and 'Submit' (which is highlighted with an orange box).

Basic Parameter List	
Use Default Proxy	No
Proxy Name	
Redundancy Mode	Parking
Proxy IP List Refresh Time	60
Enable Fallback to Routing Table	Enable
Prefer Routing Table	Yes
Always Use Proxy	Disable
Redundant Routing Mode	Routing Table
SIP ReRouting Mode	Standard Mode
Enable Registration	Disable
Gateway Name	ac1000.silstack.com
Gateway Registration Name	
DNS Query Type	A-Record
Proxy DNS Query Type	A-Record
Number of RTX Before Hot-Swap	3
Use Gateway Name for OPTIONS	No

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.

The screenshot shows the AudioCodes Mediant 1000 configuration interface. On the left, a tree view under 'Configuration' shows 'Coders And Profile Definitions' expanded, with 'Coders' selected. The main area displays a 'Coders Table' with the following data:

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711A-law	20	64	8	Disabled
G.711U-law	20	64	0	Disabled
G.729	20	8	18	Disabled

A 'Submit' button is located in the bottom right corner of the interface.

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.

The screenshot displays the AudioCodes Mediant 1000 web interface. The left navigation pane is expanded to show the 'Coders' section, with 'Coder Group Settings' selected. The main content area is titled 'Coder Group Settings'. At the top, there is a dropdown menu for 'Coder Group ID' with the value '1' selected. Below this is a table with the following columns: 'Coder Name', 'Packetization Time', 'Rate', 'Payload Type', and 'Silence Suppression'. The table contains three rows of data, with the third row, 'G.729', highlighted. A 'Submit' button is located at the bottom right of the interface.

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711A-law	20	64	8	Disabled
G.711U-law	20	64	0	Disabled
G.729	20	8	18	Disabled

4.4.5 IP Profile Settings

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

The screenshot displays the AudioCodes Mediant 1000 web interface for configuring IP Profile Settings. The left-hand navigation pane shows a tree structure under the 'Full' configuration tab, with 'IP Profile Settings' selected and highlighted with a red box. The main content area is titled 'IP Profile Settings' and includes a 'Basic Parameter List' header. The 'Profile ID' is set to 1, and the 'Profile Name' field is empty. Below this, the 'Common Parameters' section is expanded, showing a list of parameters and their values:

Parameter	Value
RTP IP DiffServ	46
Signaling DiffServ	40
Disconnect on Broken Connection	No
Media IP Version Preference	Only IPv4
Dynamic Jitter Buffer Minimum Delay [msec]	10
Dynamic Jitter Buffer Optimization Factor	10
RTP Redundancy Depth	0
Echo Canceled	Enable
Input Gain (-32 to 31 dB)	0
Voice Volume (-32 to 31 dB)	0

Below the 'Common Parameters' section, the 'Gateway Parameters' section is also expanded, showing the 'Profile Preference' set to 1. A 'Submit' button is located at the bottom right of the form.

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**

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a tree view with categories like Configuration, Management, and Status & Diagnostics. Under Configuration, the 'Full' tab is selected, and the 'IP Profile Settings' option is highlighted. The main panel displays a list of parameters for the IP Profile. The following table represents the data visible in the interface:

Basic Parameter List	
Fax Signaling Method	T.38 Relay
Play Ringback Tone to IP	Don't Play
Enable Early Media	Enable
Copy Destination Number to Redirect Number	Disable
Media Security Behavior	Mandatory
CNG Detector Mode	Disable
Modems Transport Type	Enable Bypass
NSE Mode	Disable
Number of Calls Limit	-1
Progress Indicator to IP	Not Configured
SCE	Disable
Enable Hold	Enable
Remote RTP Base UDP Port	0
First Tx DTMF Option	RFC 2833
Second Tx DTMF Option	Not Supported
Declare RFC 2833 in SDP	Yes
Add IE In SETUP	

The 'Submit' button is located at the bottom right of the configuration panel.

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**

The screenshot shows the AudioCodes Mediant 1000 configuration interface. On the left, the 'Configuration' tab is active, and the 'Full' view is selected. The navigation tree on the left shows the following structure:

- Configuration
- Management
- Status & Diagnostics
- Scenarios
- Search
- Basic
- Full
- Sigtran Configuration
- Security Settings
- Protocol Configuration
- Applications Enabling
- Media Realm Configuration
- Protocol Definition
- SIP General Parameters
- DTMF & Dialing
- Proxies, Registration, IP Groups
- Coders And Profile Definitions
- Coders
- Coder Group Settings
- Tel Profile Settings
- IP Profile Settings
- SIP Advanced Parameters
- Manipulation Tables
- Routing Tables
- Trunk Group
- Digital Gateway
- IP Media

The 'DTMF & Dialing' configuration page is displayed on the right. It contains a table of parameters with the following values:

Basic Parameter List ▲	
Max Digits In Phone Num	30
Inter Digit Timeout for Overlap Dialing [sec]	4
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option	RFC 2833
2nd Tx DTMF Option	
RFC 2833 Payload Type	101
Digit Mapping Rules	
Default Destination Number	serveduser
Special Digit Representation	Special

A 'Submit' button is located at the bottom right of the configuration page.

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**

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar displays the 'Configuration' menu with 'Routing Tables' expanded. The main area shows the 'Tel to IP Routing' configuration page. A table with 6 columns (Index, Src. Trunk Group ID, Dest. Phone Prefix, Source Phone Prefix, Dest. IP Address, Port) is displayed. Row 1 is highlighted with an orange border and contains the values: 1, 2, 66666, 3*, 10.10.21.51, 5060. Row 2 contains: 2, 2, 66666, 500*, 10.10.21.51, 5060. The interface includes a 'Submit' button at the bottom right.

	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Port
1	2	66666	3*	10.10.21.51	5060
2	2	66666	500*	10.10.21.51	5060
3					
4					
5					
6					
7					
8					

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.

The screenshot shows the AudioCodes Mediant 1000 web interface. The left sidebar contains a tree view with the following structure:

- Configuration
 - Applications Enabling
 - Media Realm Configuration
 - Protocol Definition
 - Proxies, Registration, IP Groups
 - Coders And Profile Definitions
 - SIP Advanced Parameters
 - Manipulation Tables
 - Routing Tables
 - Routing General Parameters
 - Tel to IP Routing**
 - IP to Trunk Group Routing
 - Internal DNS Table
 - Internal SRV Table
 - Release Cause Mapping
 - Alternative Routing
 - Trunk Group
 - Digital Gateway
 - IP Media
 - Advanced Applications

The main content area is titled 'Tel to IP Routing'. It features a 'Basic Parameter List' section with a table. The table has the following columns: Address, Port, Transport Type, Dest. IP Group ID, IP Profile ID, and Status. The first two rows are populated with Port 5060, Transport Type TCP, and IP Profile ID 1. The Status column shows 'n/a' for these rows. The remaining rows are empty or show 'Not Configured' for Transport Type. A 'Submit' button is located in the bottom right corner of the interface.

Address	Port	Transport Type	Dest. IP Group ID	IP Profile ID	Status
	5060	TCP		1	n/a
	5060	TCP		1	n/a
		Not Configured			
		Not Configured			
		Not Configured			
		Not Configured			
		Not Configured			
		Not Configured			

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:** *

The screenshot shows the AudioCodes Mediant 1000 configuration interface. On the left, the 'Configuration' tab is active, and the 'IP to Trunk Group Routing' option is selected under the 'Routing Tables' section. The main area displays the 'IP To Trunk Group Routing Table' with a 'Basic Parameter List' dropdown set to 'Routing Index' (1-12) and 'IP To Tel Routing Mode' (Route calls before manipulation). The table has five columns: 'Dest. Host Prefix', 'Source Host Prefix', 'Dest. Phone Prefix', 'Source Phone Prefix', and 'Source'. Row 1 is highlighted with an orange border and contains the following values: Dest. Host Prefix: *, Source Host Prefix: *, Dest. Phone Prefix: 3xxx, Source Phone Prefix: *, and Source: 10.10.21. The 'Submit' button is located at the bottom right of the interface.

	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source
1	*	*	3xxx	*	10.10.21
2	*	*	*	*	*
3					
4					
5					
6					
7					
8					
9					

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.

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a tree view with categories like Configuration, Management, and Status & Diagnostics. Under Configuration, the 'Full' tab is selected, and the 'Routing Tables' section is expanded, showing 'IP to Trunk Group Routing' as the active configuration.

The main area displays the 'IP To Trunk Group Routing Table'. At the top, there are dropdown menus for '1-12' and 'Route calls before manipulation'. Below these is a table with the following columns: Dest. Phone Prefix, Source Phone Prefix, Source IP Address, Trunk Group ID, IP Profile ID, and Source IPGroup ID.

Dest. Phone Prefix	Source Phone Prefix	Source IP Address	Trunk Group ID	IP Profile ID	Source IPGroup ID
3xxx	*	10.10.21.51	2	1	-1
*	*	*	1	0	-1

At the bottom right of the interface, there is a 'Submit' button with a checkmark icon.

4.6 Configure PSTN Trunk Group

To configure the trunk group associated with the E1 PRI 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

The screenshot shows the AudioCodes Mediant 1000 configuration interface. On the left, the 'Configuration' tab is selected, and the 'Trunk Group' menu item under 'Protocol Configuration' is highlighted. The main area displays the 'Trunk Group Table' with a table of 8 rows. The first row is highlighted, and the second row is selected. The table has columns for Group Index, Module, From Trunk, To Trunk, Channels, Phone Number, Trunk Group ID, and Tel Profile ID.

Group Index	Module	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	Tel Profile ID
1	Module 1 PRI	1	1	1-5	39999	1	0
2	Module 1 PRI	2	2	1-31		2	0
3							
4							
5							
6							
7							
8							

A 'Submit' button is located at the bottom right of the interface.

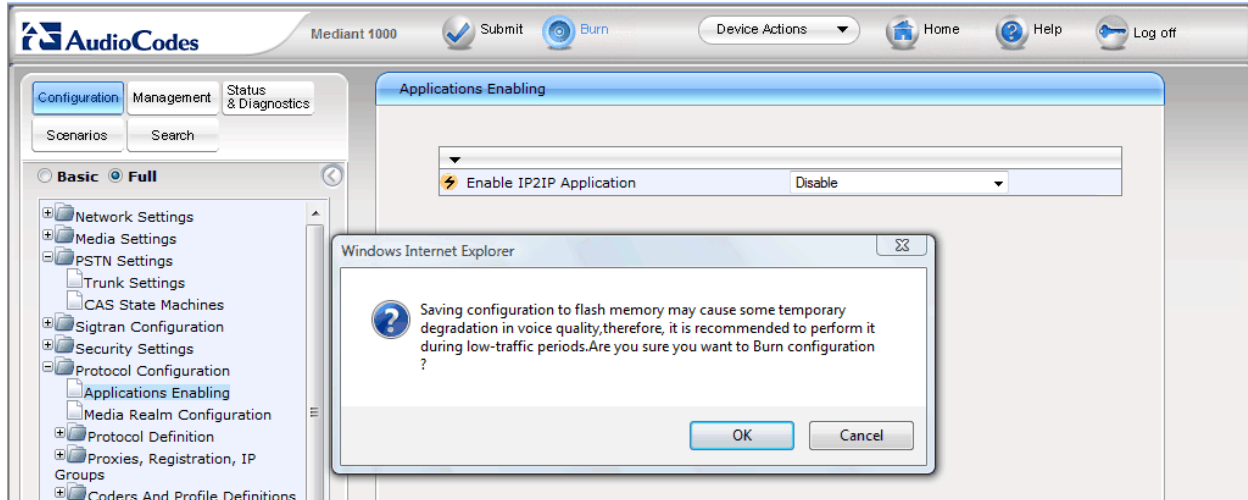
4.7 Save the Configuration

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

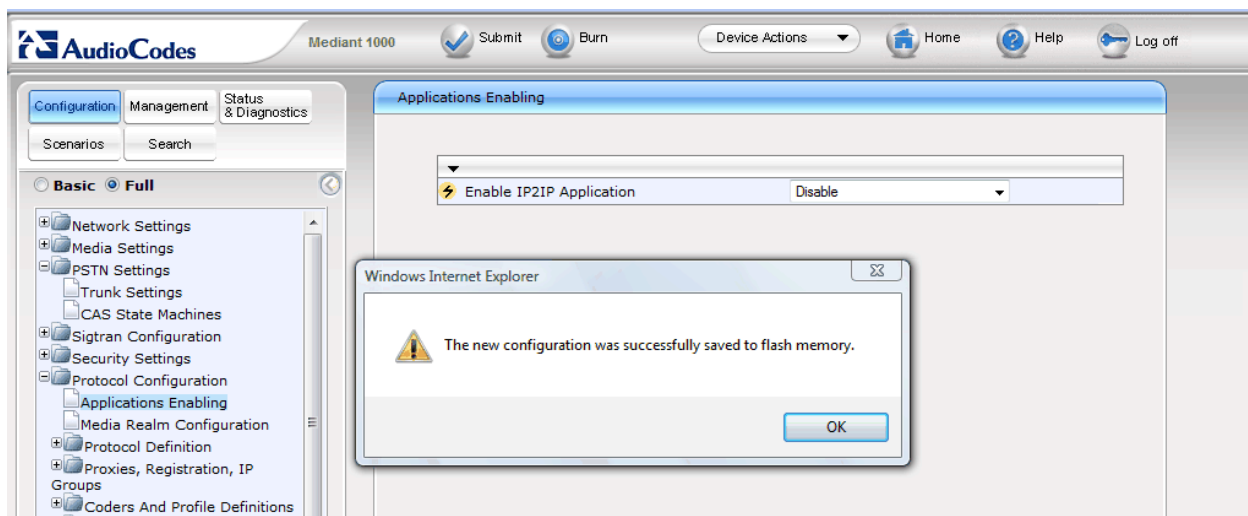
The screenshot shows the AudioCodes Mediant 1000 configuration interface. The 'Burn' button on the toolbar is highlighted. The main area displays the 'Applications Enabling' section with a table of 1 row. The table has columns for Application Name and Status.

Application Name	Status
Enable IP2IP Application	Disable

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



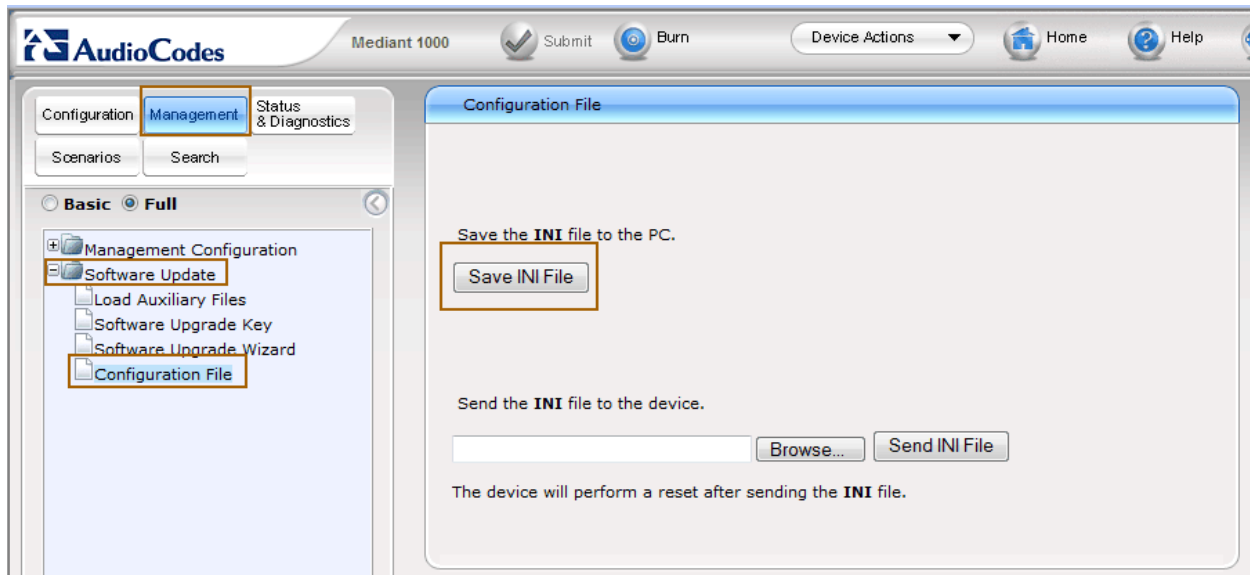
Click **OK** to acknowledge the below prompt.



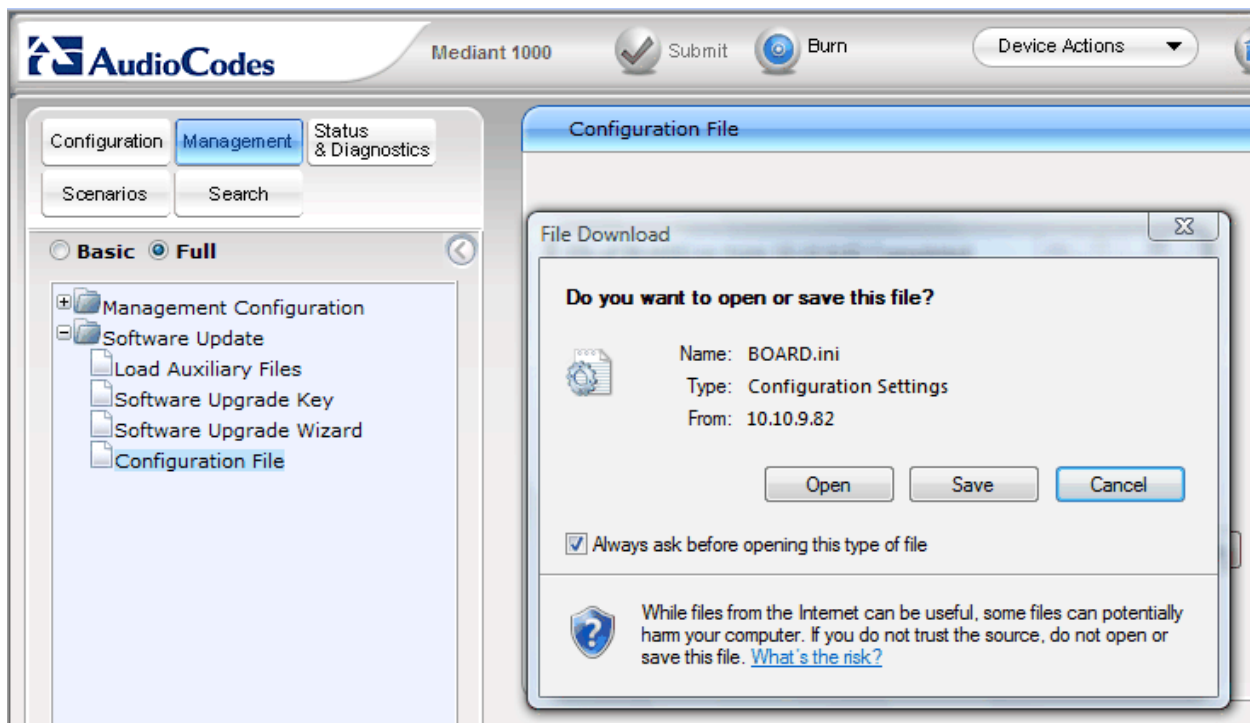
4.8 Customize the INI File

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].

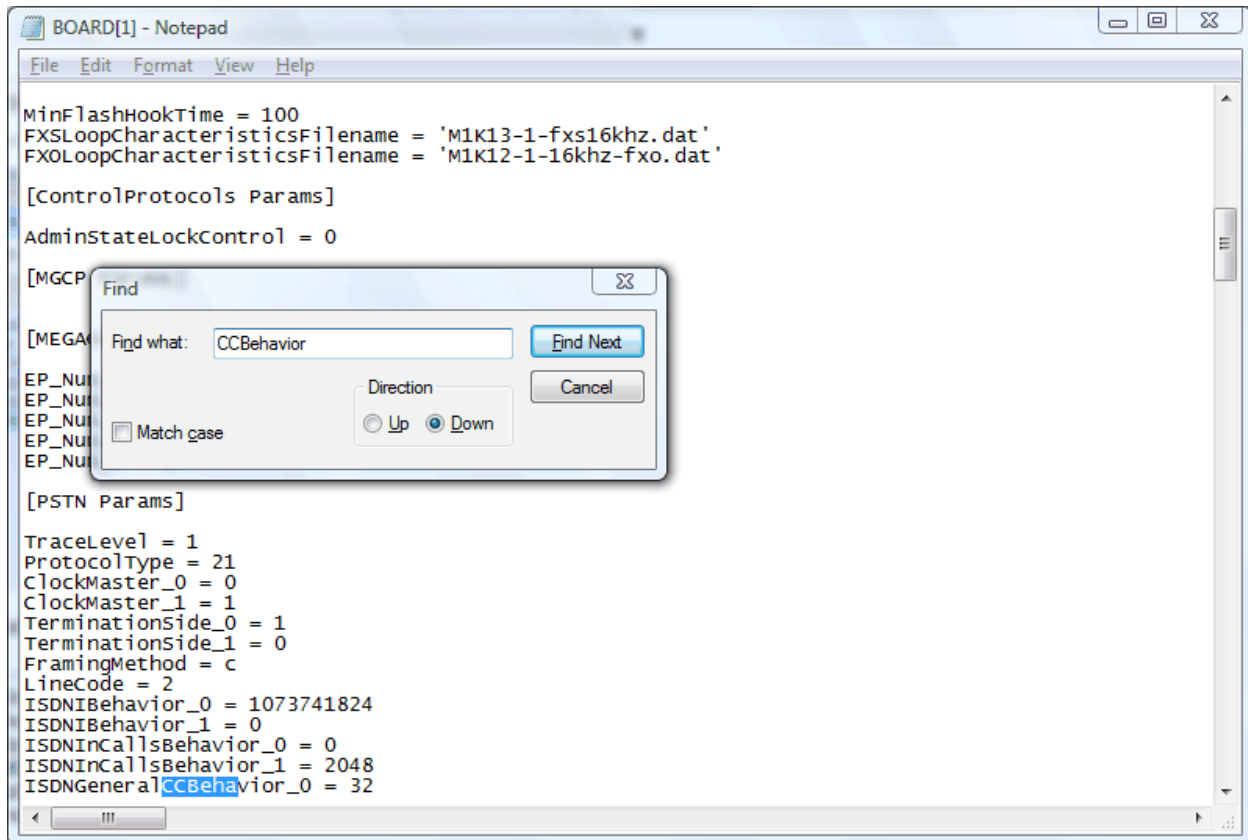
Click on **Management**. Expand the **Software Update** menu and select **Configuration File**. Click on **Save INI File** to download the configuration file of the M1000 Gateway.



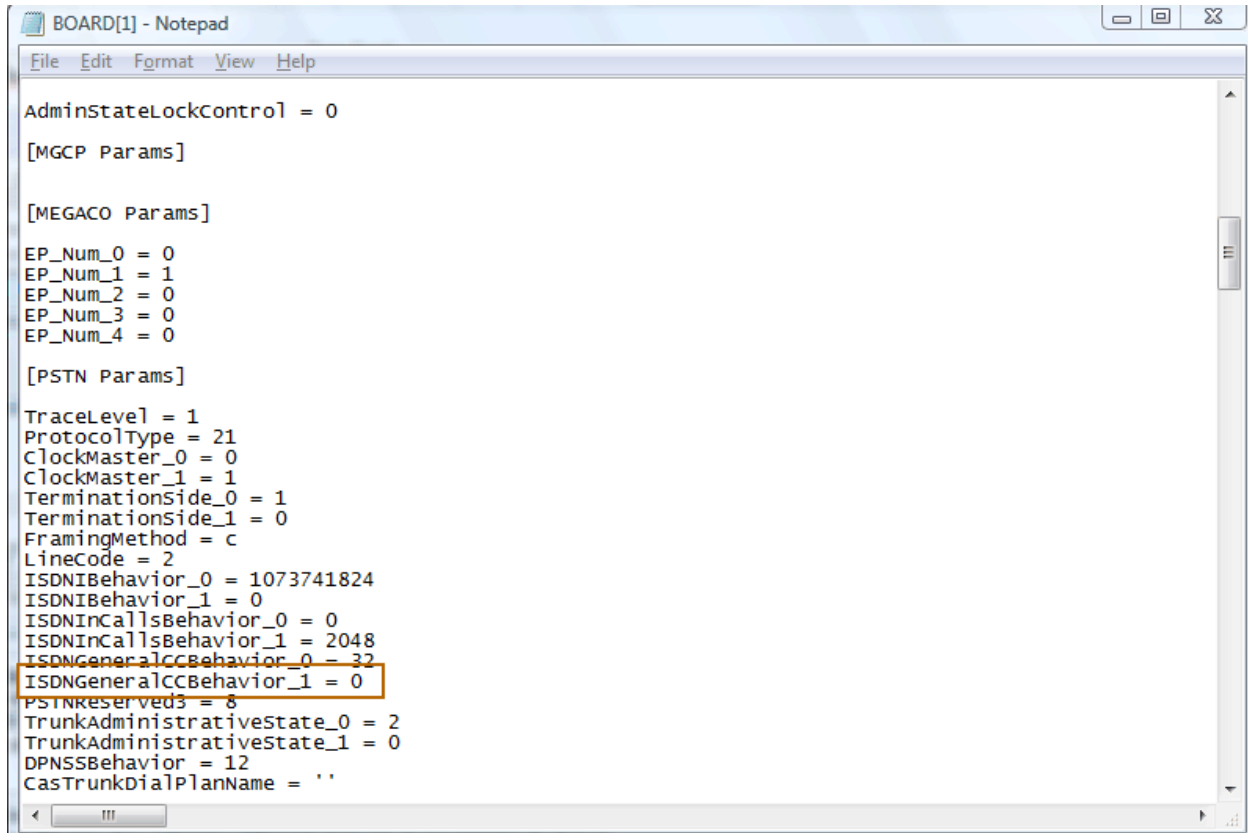
Save and open the file for editing using a text editor on the local PC.



Search for the **CCBehavior** 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

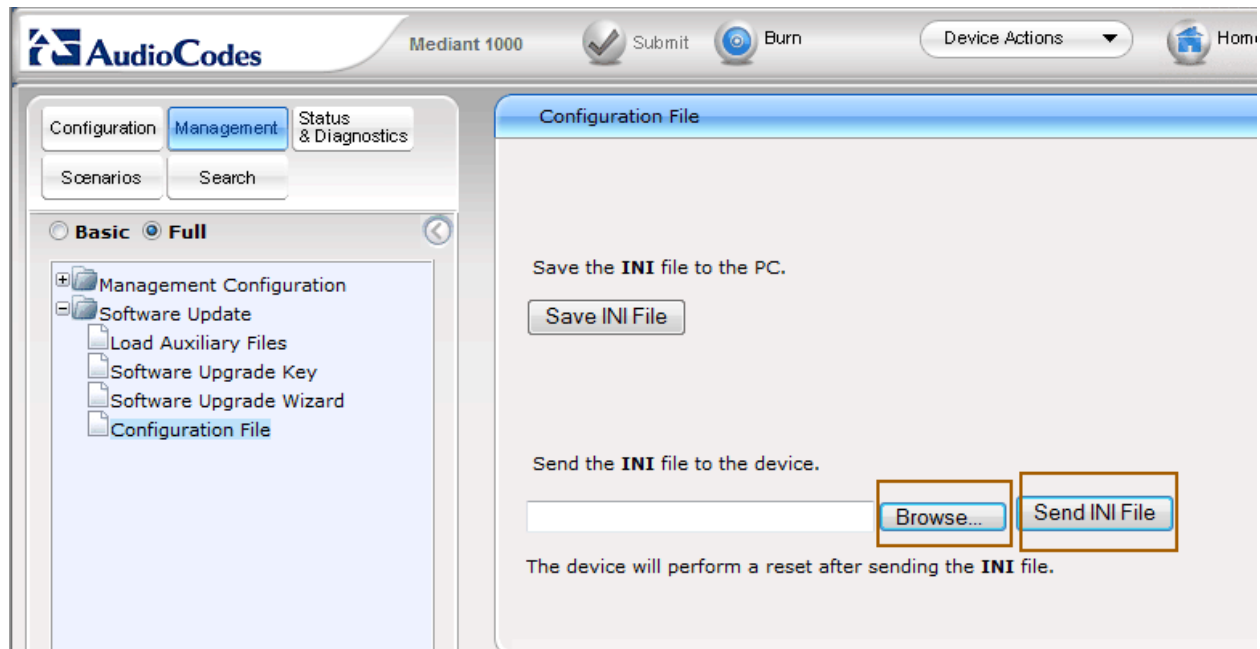
[MGCP Params]

[MEGACO Params]

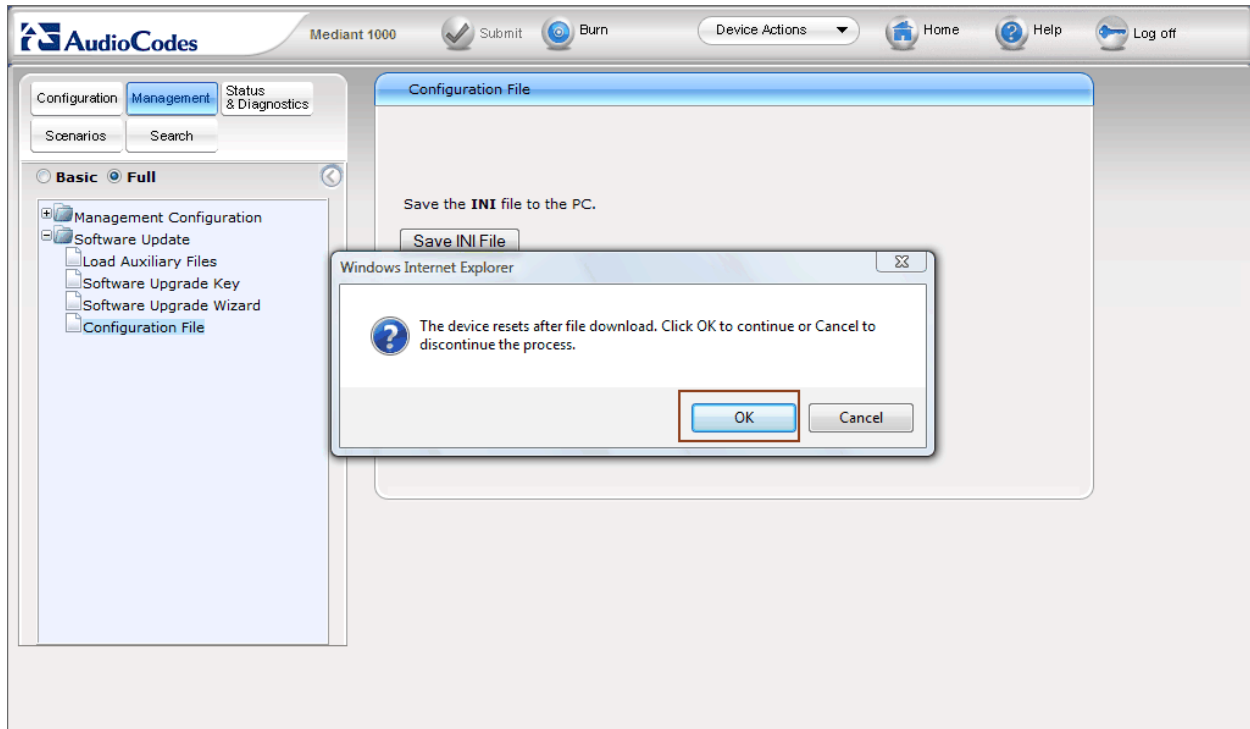
EP_Num_0 = 0
EP_Num_1 = 1
EP_Num_2 = 0
EP_Num_3 = 0
EP_Num_4 = 0

[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
DNSSBehavior = 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 **mlk.avaya.com** zone and had the following entry in its zone file:

```
mlk.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@mlk.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.

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

System Maintenance Main Menu
Network Configuration
FDAPI Configuration
LAN Configuration
Administrator Menu
Re-Initialization
System Shutdown
Transmission Level
EXIT
```

Navigate to **Re-Initialization**.

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

System Maintenance Main Menu
Network Configuration
FDAPI Configuration
LAN Configuration
Administrator Menu
Re-Initialization
System Shutdown
Transmission Level
EXIT
```

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.

NORTEL CS 1000 ELEMENT MANAGER Help | U

Managing: **10.10.21.10** Username: admin
System » **Maintenance** » MSDL Diagnostics

MSDL Diagnostics

Diagnostic Commands	Command Parameters	Action
Get Status of MSDL Device (STAT)	<input type="checkbox"/> FDL <input checked="" type="checkbox"/> FULL <input type="checkbox"/> ALL	Submit

MSDL STATUS
004 0 2 ENBL

```
STAT MSDL 004 0 2
-----
DCH      42  OPER      PORT 1
```

Cancel **View page log**

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.

Mediant 1000 Home Page

Alarms

1	Digital	2	3	CPU
4	5	6		

General Information

IP Address	10.10.9.82
Subnet Mask	255.255.255.0
Default Gateway	10.10.9.1
Digital Port Number	2
BRI Port Number	0
Analog Port Number	0
Firmware Version	5.80A.039.005
Protocol Type	SIP

Trunk (Digital Modules)

Disable	Not Connected
Active - OK	Inactive
RAI Alarm	Handset Offhook
LOS / LOF Alarm	Call Connected
AIS Alarm	
D-Channel Alarm	

Channel (Analog Modules)

Disable	Not Connected
Active - OK	Inactive
RAI Alarm	Handset Offhook
LOS / LOF Alarm	Call Connected
AIS Alarm	
D-Channel Alarm	

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>

©2010 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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