Avaya Solution & Interoperability Test Lab

Configuring the AudioCodes Mediant 1000 Media Gateway and Avaya Meeting Exchange over TCP and UDP – Issue 1.0

Abstract

These Application Notes describe a compliance tested solution comprising the Avaya Meeting Exchange Enterprise S6200 Conferencing Server communicating directly with the AudioCodes Mediant 1000 Media Gateway over TCP and UDP. The AudioCodes Mediant 1000 is utilized to enable connectivity between the Avaya Meeting Exchange Enterprise S6200 Conferencing Server 5.2 SP1 and the PSTN. This configuration provides a rich set of conferencing options available on the Avaya Meeting Exchange Enterprise S6200 Conferencing Server to participants associated with the PSTN.

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

1 Introduction

These Application Notes describe a compliance tested solution comprised of the Avaya Meeting Exchange Enterprise S6200 Conferencing Server and the AudioCodes Mediant 1000 over TCP and UDP. The AudioCodes Mediant 1000 is utilized to enable connectivity between Avaya Meeting Exchange Enterprise S6200 Conferencing Server and the PSTN.

1.1 Interoperability Compliance Testing

The Avaya Meeting Exchange is SIP-based with call signalling and Media Server capability for voice conferencing. Avaya's Conferencing Applications include reservation-less, attended, event and mobile to support various IP network implementations. The following capabilities are supported by Avaya Meeting Exchange

- RFC 2833 DTMF
- In-band DTMF
- Up to 3200-user and 140-operator conferences
- Up to four digitally recorded music sources
- Codecs G.711 PCMU, G.711 PCMA

The AudioCodes Mediant 1000 system is a SIP based Voice-over-IP (VoIP) media gateway, offering integrated voice gateway functionality over IP networks. This solution addresses mid-density applications deployed in IP networks by simultaneous Voice over IP calls. The AudioCodes Mediant 1000 routes calls over the IP network using SIP signalling protocol, enabling the deployment of Voice over Packet solutions to PSTN.

1.2 Support

Audio Codes support website and contact details

Tel: +972-3-976-4000
http://www.audiocodes.com/support

2 Reference Configuration

The end to end signalling connectivity between Avaya Meeting Exchange and the PSTN is shown in **Figure 1**.

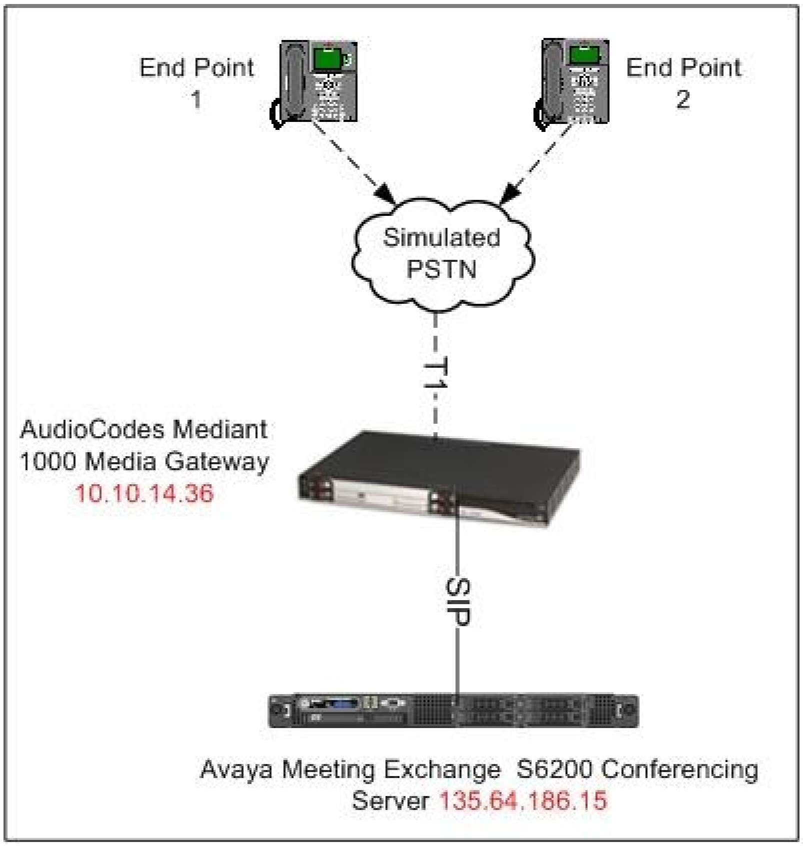


Figure 1: Network Configuration

3 Equipment and Software Validated

The following equipment and software versions were used for the sample configuration provided in these Application Notes.

Equipment	Software
Avaya Meeting Exchange Enterprise S6200 Conferencing 8510 Server	MX-5.2.1.0.4 mx-bridge-patch-5.2.1.3.1-1
Avaya Bridge Talk (BT)	5.2.0.0.4
Audio Codes Mediant 1000 Media Gateway	5.80A.033

Table 1: Hardware and Software Versions

4 Configure Avaya Meeting Exchange

The following sections describe the administrative procedures for configuring the Meeting Exchange

- **Basic configuration**
- **Enable dial out**
- **Provision DDI/DNIS for dialing into conferences**
- **Bridge Talk**

4.1 Basic Configuration

Log in to the Avaya Meeting Exchange console (PuTTY) using ssh to access the Command Line Interface (CLI) with the appropriate credentials. Configure settings that enable SIP connectivity between the Avaya Meeting Exchange and other devices by editing the **system.cfg** file as follows:

- Edit **/usr/ipcb/config/system.cfg**
- Add Meeting Exchange S6200 IP address
    - **IPAddress=("135.64.186.15")**
- Add a line to populate the From Header Field in SIP INVITE messages for Avaya Meeting Exchange in this sample configuration the values used.
    - **MyListener=<sip:6000@135.64.186.15;5060;transport=tcp>**
    - **MyListener=<sip:6000@135.64.186.15;5060;transport=udp>**

***Note**: The user field 6000 was set as default digit string ,used as the CLI for dials out originating from Avaya Meeting Exchange , this number should be aligned with dial plan. It is selected to match the user field provisioned for the **respContact** entry (see below).*

- Add a line to provide SIP Device Contact address to use for acknowledging SIP messages from the Avaya Meeting Exchange:
    - **respContact=<sip:6000@134.64.186.15:5060;transport=tcp>**
    - **respContact=<sip:6000@134.64.186.15:5060;transport=udp>**

- Add the following lines to set the Min-SE timer to **900** seconds in SIP INVITE messages from the Avaya Meeting Exchange :
    - **sessionRefreshTimerValue= 900**
    - **minSETimerValue= 900**

4.2 Enable Dial Out

To enable Dial-Out from the Avaya Meeting Exchange to the **Mediant 1000**, edit the **telnumToUri.tab** file as follows:

- Edit /usr/ipcb/config/telnumToUri.tab file with a text editor. Add a line to the file to route outbound calls from the Avaya Meeting Exchange to the **Mediant 1000**.
  "$1" substitutes the first wild card.

  ***   sip:$1@10.10.14.36:5060;transport=tcp   Mediant 1000**
  ***   sip:$1@10.10.14.36:5060;transport=udp   Mediant 1000**

4.3 Provision DDI/DNIS for Dialing into Conferences

The following steps provide examples of how to provision Scan Flow (Scheduled and Demand) conference call functions by utilizing the cbutil utility on the Avaya Meeting Exchange. To map DNIS entries, run the **cbutil** utility on Avaya Meeting Exchange as follows:

- If not already logged on, log in to the Avaya Meeting Exchange with a ssh connection using PuTTY with the appropriate credentials.

Enable Dial-In access (via passcode) to conferences provisioned on the Avaya Meeting Exchange as follows:

- Add a DNIS entry for a **scan call function** corresponding to DID **1001** by entering the following command at the command prompt:

```
cbutil add <dnis> <rg> <msg> <ps> <ucps> <func> [-l <ln> -c <cn>],
where the    variables for add command is defined as follows:
      o <dnis>    DNIS
      o <rg>      Reservation Group
      o <msg>     Annunciator message number
      o <ps>      Prompt Set number (0-20)
      o <ucps>    Use Conference Prompt Set (y/n)
      o <func>    One of: DIRECT/SCAN/ENTER/HANGUP/AUTOVL/FLEX
      o -l <"ln"> Optional line name to associate with caller
      o -c <"cn"> Optional company name to associate with caller
```

In this sample configuration:

```
S6200App->cbutil add 1001 0 247 1 n SCAN –l  Mediant 1000
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
```

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

```
[sroot@MX config]# 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
---------------- --- --- --- -- -------- ---------- -------------------
- ------------------- ----------- ----------
1001             0   247 1   N  SCAN     ENTER   SCAN

3331           0   301 1   N  DIRECT     ENTER   DIRECT
```

4.4 Bridge Talk

The following steps utilize the Avaya Bridge Talk application to provision a sample conference on the Avaya Meeting Exchange. This sample conference enables both Dial-In and Dial-Out access to audio conferencing for endpoints on the PSTN.

- **Logging into Bridge Talk**
- **Provision a dial out**
- **Provision a Conference with Auto Blast Enabled**

Note: If any of the features displayed in the Avaya Bridge Talk screen captures are not present, contact an authorized Avaya sales representative to make the appropriate changes.

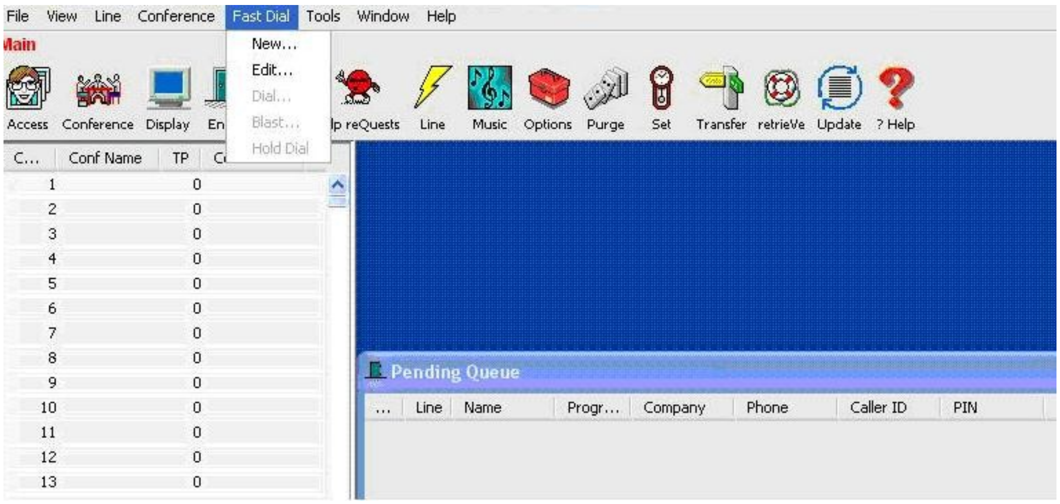
4.4.1 Logging into Bridge Talk

Invoke the Avaya Bridge Talk application as follows:

- Double-click on the desktop icon from a Personal Computer loaded with the Avaya Bridge Talk application and with network connectivity to the Avaya Meeting Exchange [Not shown].
- Enter the appropriate credentials in the **Sign-In** and **Password** fields.
- Enter the IP address of the Avaya Meeting Exchange (**135.64.186.15)** for this sample configuration) in the **Bridge** field.

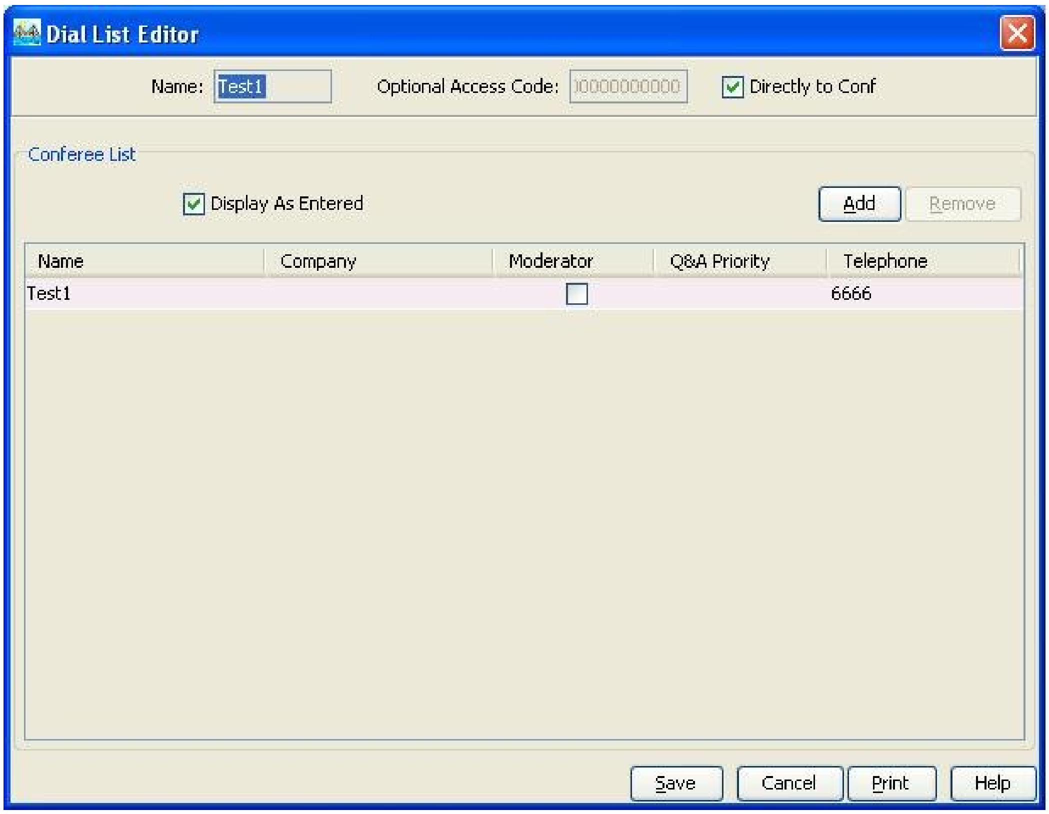
4.4.2 Provision a Dial-Out

Provision a dial list that is utilized for Dial-Out (e.g., Blast dial and Fast dial) from the Avaya Meeting Exchange.From the Avaya Bridge Talk Menu Bar, click **Fast Dial→New**.



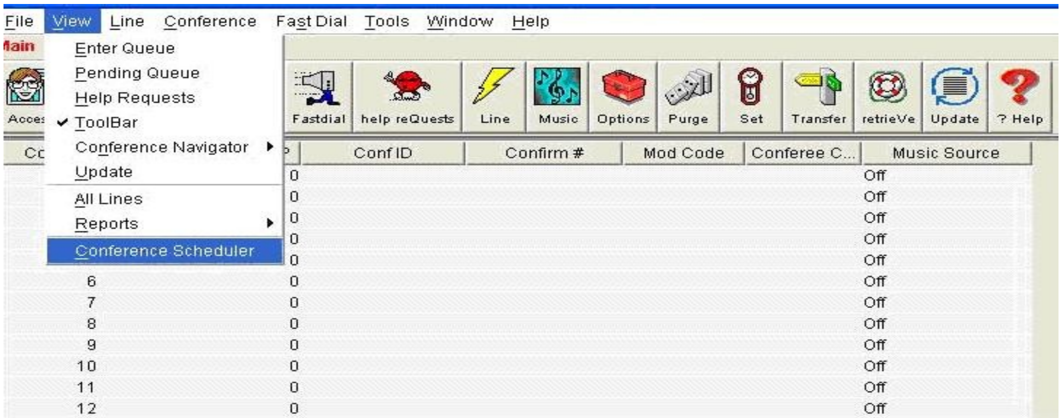
From the **Dial List Editor** window that is displayed:

- Enter a descriptive label in the **Name** field.
- Enable conference participants on the dial list to enter the conference without a passcode by checking the **Directly to Conf** box as displayed.
- Add entries to the dial list by clicking on the **Add** button and enter **Name**, **Company** and **Telephone** number for dial out for each participant. [*Optional*] *Moderator privileges may be granted to a conference participant by checking the **Moderator** box.*
- When finished, click on the **Save** button on the bottom of the screen.

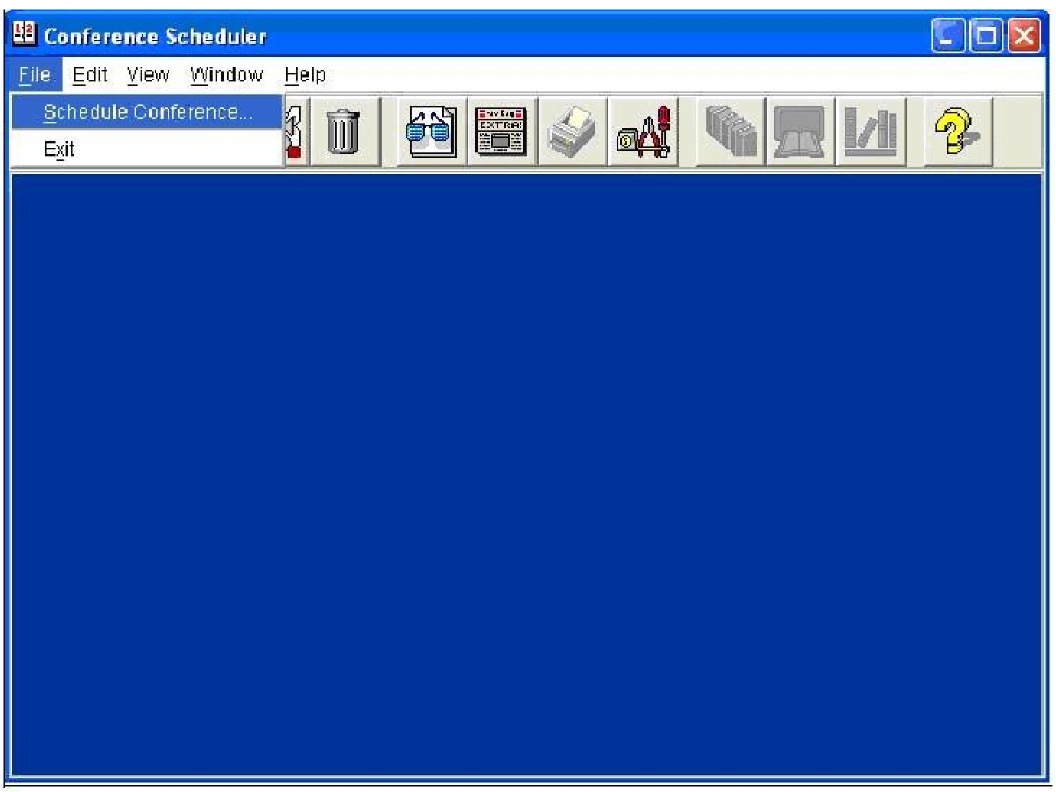


4.4.3 Provision a Conference with Auto Blast Enabled

To provision a conference with Auto Blast enabled go to the Avaya Bridge Talk Menu Bar, click **View →Conference Scheduler**.

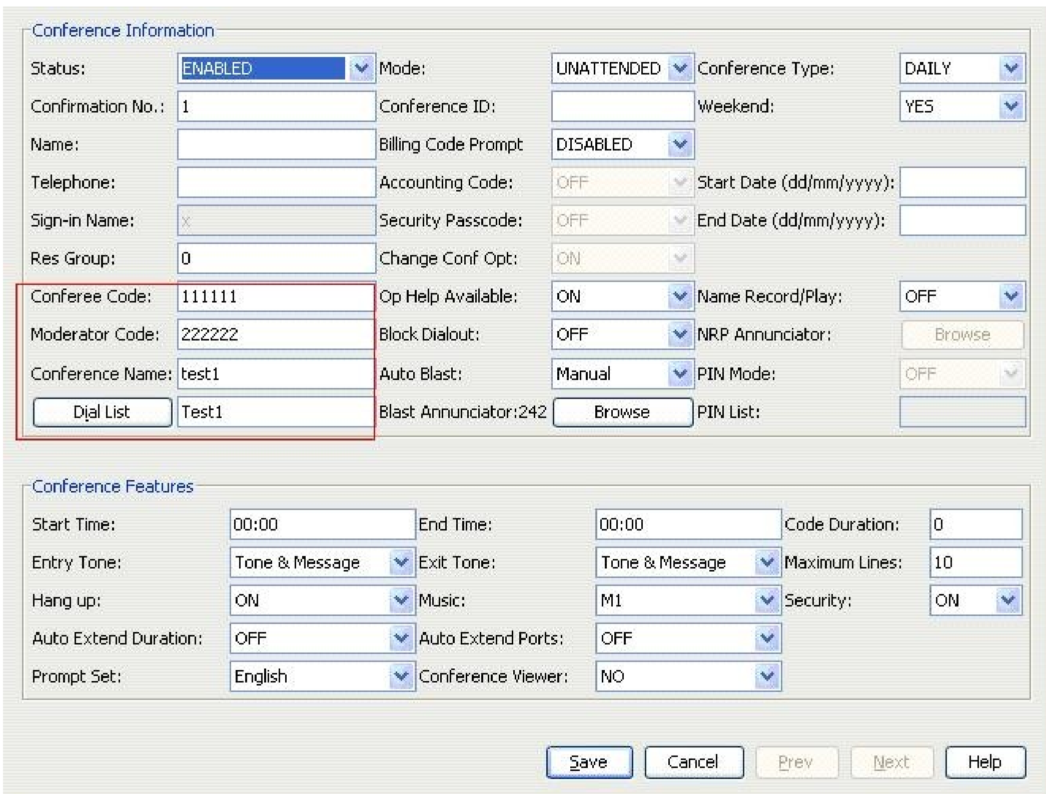


From the **Conference Scheduler** window, click **File →Schedule Conference**.



Provision a Scan Flow conference on Avaya Meeting Exchange, From the **Schedule Conference** window that is displayed, provision a conference as follows:

- Enter a unique **Conferee Code** to allow participants access to this conference.
- Enter a unique **Moderator Code** to allow participants access to this conference with moderator privileges. Enable moderator access with a passcode for this conference call by configuring the following:
  Note: *This conference remains open for participants to enter as either moderator or participant by entering the appropriate code when prompted*.
- Enter a descriptive label in the **Conference Name** field.
- Administer settings to enable an Auto Blast dial by setting **Auto Blast** to **Auto/Manual** depending on this test.
- Select a dial list by clicking on the **Dial List** button, select a dial list from the **Create, Select or Edit Dial List** window that is displayed, and click on the **Select** button (To verify Dial out and Blast Dial out) [Not Shown].
- When finished, click on the **Save** button on the bottom of the screen.



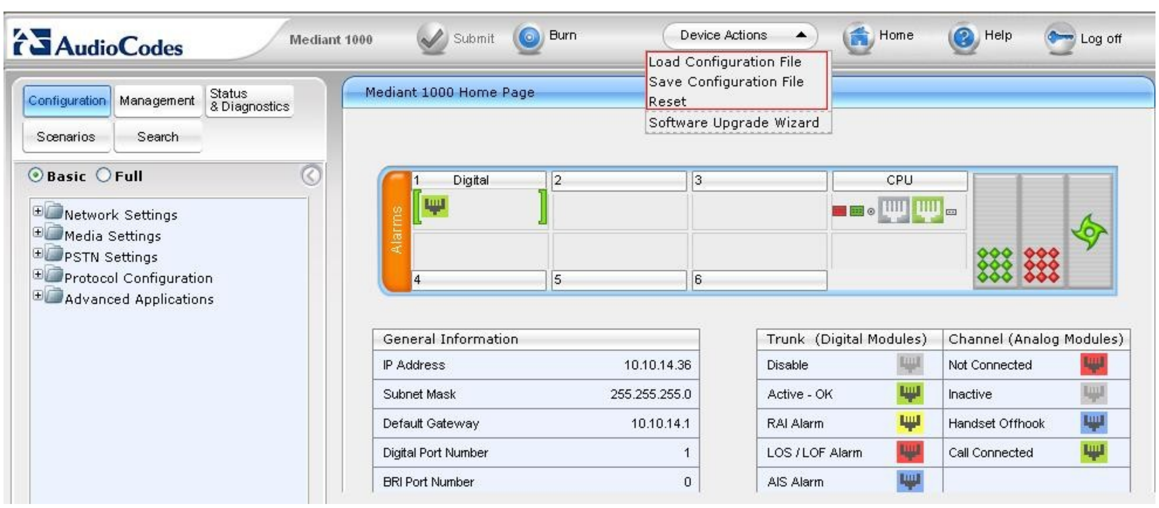
5 Configure AudioCodes Mediant 1000 Media Gateway

The following sections describe the steps for configuring the SIP trunks and call routing for the AudioCodes Mediant 1000. This configuration will enable the AudioCodes Mediant 1000 to interoperate with both the Avaya Meeting Exchange and the PSTN. Configuration is performed using the embedded web server, which supports gateway configuration, including loading of configuration files. The Embedded Web Server can be accessed from a standard web browser. Specifically, users can employ this facility to set up the gateway configuration parameters. Users also have the option to remotely reset the gateway and to permanently apply the new set of parameters.

5.1 Access to Home Page Page

From an Internet browser, navigate to **http://<ip-addr>**, where the <ip-addr> is the IP address of the AudioCodes Mediant 1000. The page below shows the **AudioCodes Mediant 1000 Home Page**. Log in to the AudioCodes Mediant 1000 with the appropriate credentials.

Note: When configuring any parameters using the Web Browser, click the **Submit** icon and the **Burn** icon to save the configuration to flash memory. These icons are found on the top of the browser page. To reset the AudioCodes Mediant 1000, click on the **Device Actions** drop down window and click **Reset** as shown below.



5.2 Configuring PSTN Settings

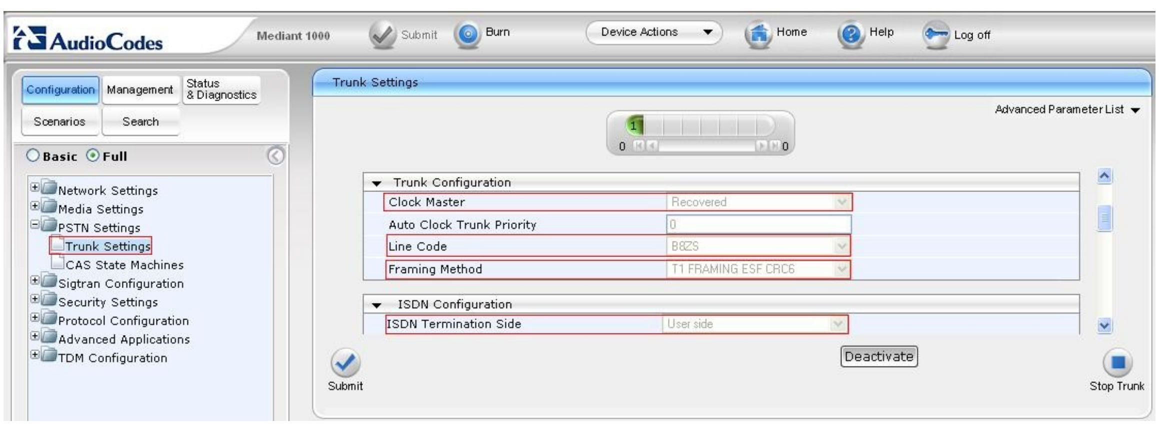
On the left pane click **Configuration a**nd navigate to **PSTN Settings → Trunk Settings**. **Note:** Click the **Stop Trunk** button to modify the selected trunk number 1(not shown). In the **General Settings** section of **Trunk Settings** configure as follows:

- Select **T1 5ESS 10 ISDN** in the **Protocol Type** field as shown below to enable connectivity to the PSTN according to requirements defined by the PSTN service provider.

5.3 Configuring Trunk Configuration

Scroll down to the **Trunk Configuration** section and configure as follows:

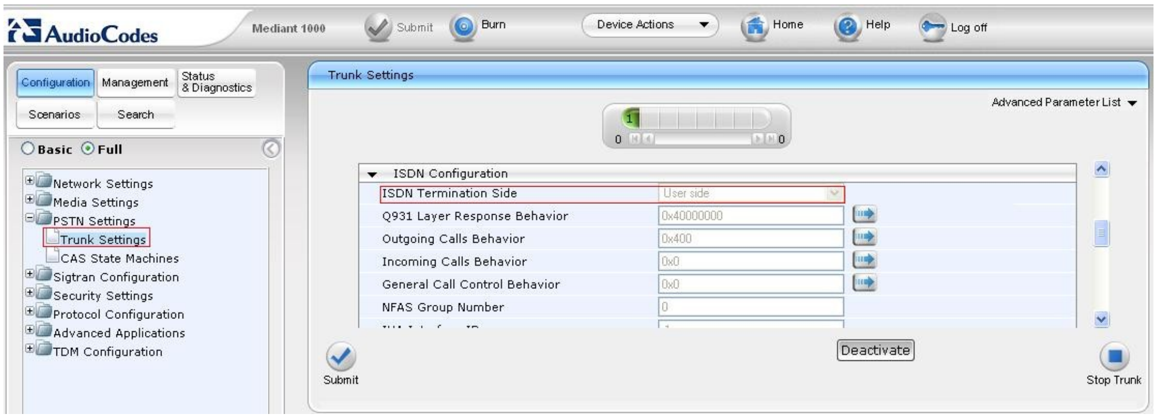
- Select **Recovered** in the **Clock Master** field as the clock is recovered from the PSTN line.
- Select **B8ZS** for **Line Code** as this is used for T1 trunks
- The **Framing Method** for T1 trunks is always Framing **T1 FRAMING ESF CRC6**.



5.4 Configuring ISDN Configuration

Scroll down to the **ISDN Configuration** section and configure as follows:

- Configure **ISDN Termination Side** to **User side** as the PSTN is configured as Network and the Mediant 1000 is the slave device.

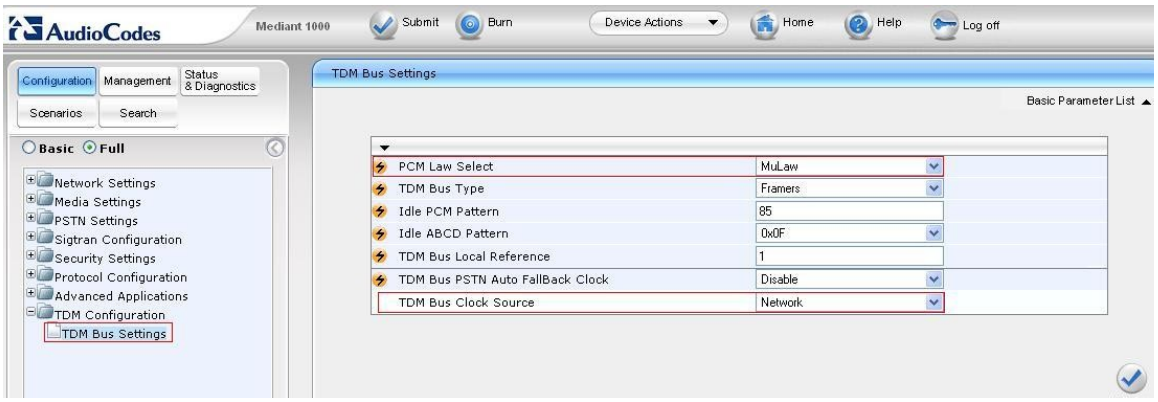


Scroll down to the remainder of the **ISDN Configuration** section and the remaining fields are default settings.

5.5 Configuring TDM Bus Settings

On the left pane click **Configuration** and navigate to **TDM Configuration → TDM Bus Settings**.

- Configure **PCM Law Select** to **A-Law** as T1 trunks are configured to this value.
- Configure the **TDM Bus PSTN Auto Fallback Clock** to **Disable**.
- Configure **TDM Bus Clock Source** to **Network** as the clocking is generated from the PSTN line.



5.6 Proxy & Registration

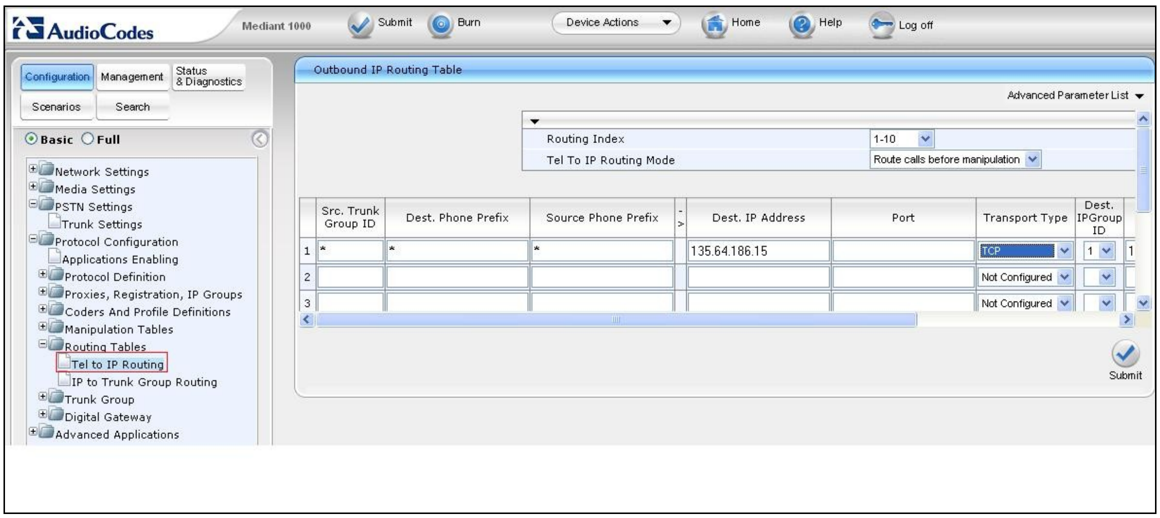
On the left pane click **Configuration** and navigate to **Protocol Configuration → Protocol Definition → Proxy & Registration**.

- Configure **Use Default Proxy** to **No** as SIP connectivity between the AudioCodes Mediant 1000 and the Meeting Exchange is direct.

5.7 Routing Calls

To administer call routing to calls originating from the PSTN to the Meeting Exchange using TCP or UDP, on the left pane click **Configurations** and navigate to **Protocol Configuration → Routing Tables → Tel to IP Routing**.

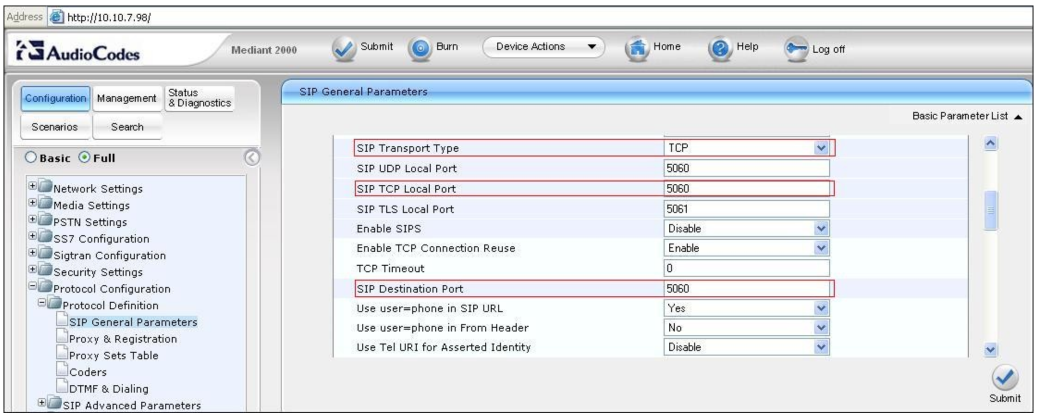
- Enter a rule in the **Dest Phone Prefix** field that matches the pattern of incoming calls to the AudioCodes Mediant 1000 from the PSTN. For these Application Notes, the rule ***** is utilized, where ***** is a wildcard and will match any digit(s), thus routing all calls to the Meeting Exchange from the PSTN.
- Enter an ***** in the **Source Phone Prefix** field to allow routine for any source telephone number Dialing-In to the Meeting Exchange from the PSTN.
- Configure table as shown below where the **Dest IP Address** is that of the Meeting Exchange.
- Configure **Transport Type** depending on which SIP transport type is under test.



5.8 Configuring SIP General Parameters for TCP

On the left pane click **Configuration** and navigate to **Protocol Configuration → Protocol Definition → SIP General Parameters**.

- Configure **SIP Transport Type** to **TCP or UDP**.
- Configure **SIP TCP Local Port** to **5060**, to enable SIP/TCP connectivity with the Meeting Exchange (see **Section 4.1**) on configuring Meeting Exchange)
- Configure the **SIP Destination Port** to port **5060**.

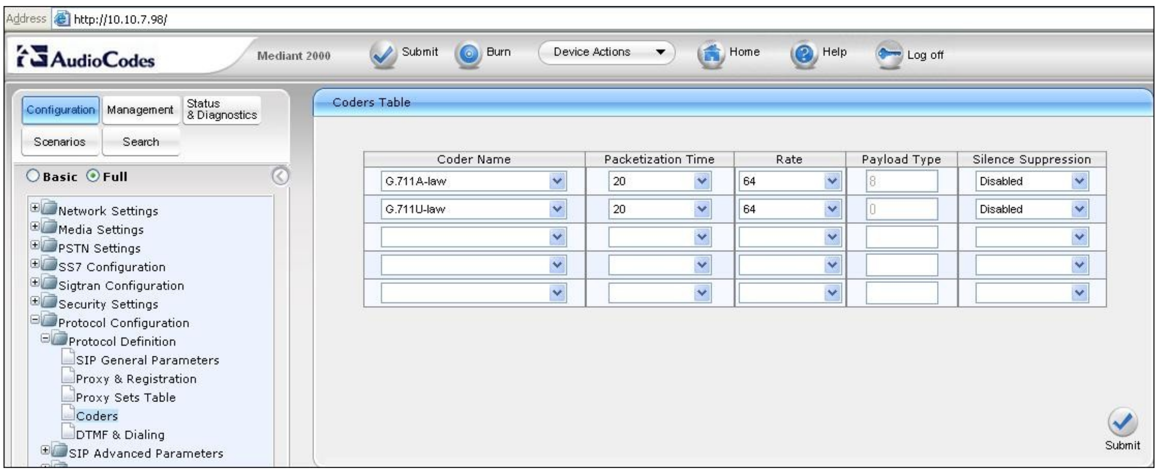


5.9 Administer Codec Preferences

To administer codec preferences, on the left pane click **Configuration** and navigate to **Protocol Configuration → Protocol Definition → Coders**.

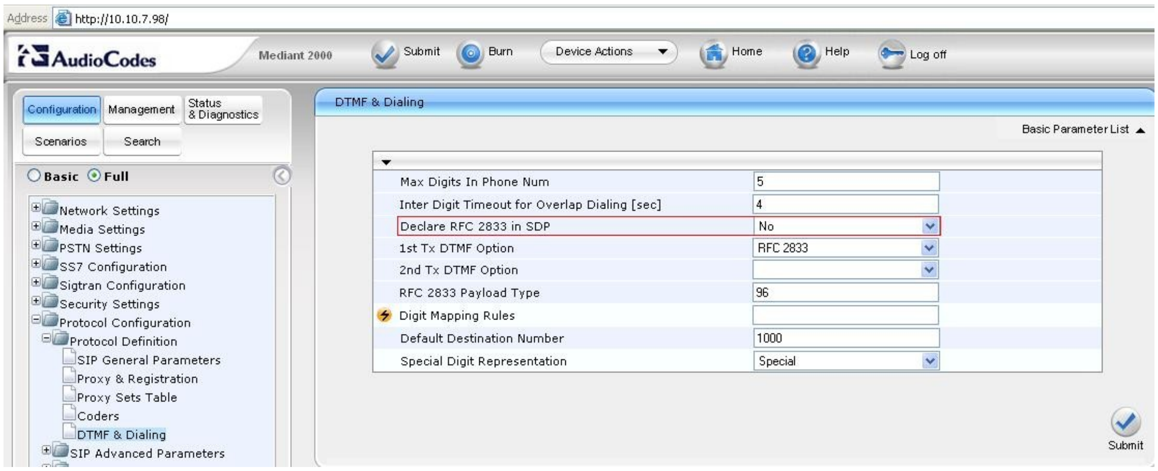
- Configure a **Coder Name** that is compatible with Avaya Meeting Exchange.
- Remaining fields are default settings.

Note: The first coder is the highest priority coder and is used by the AudioCodes Mediant 1000 whenever possible. If the Avaya Meeting Exchange cannot use the coder assigned as the first coder, the gateway attempts to use the next coder and so forth.



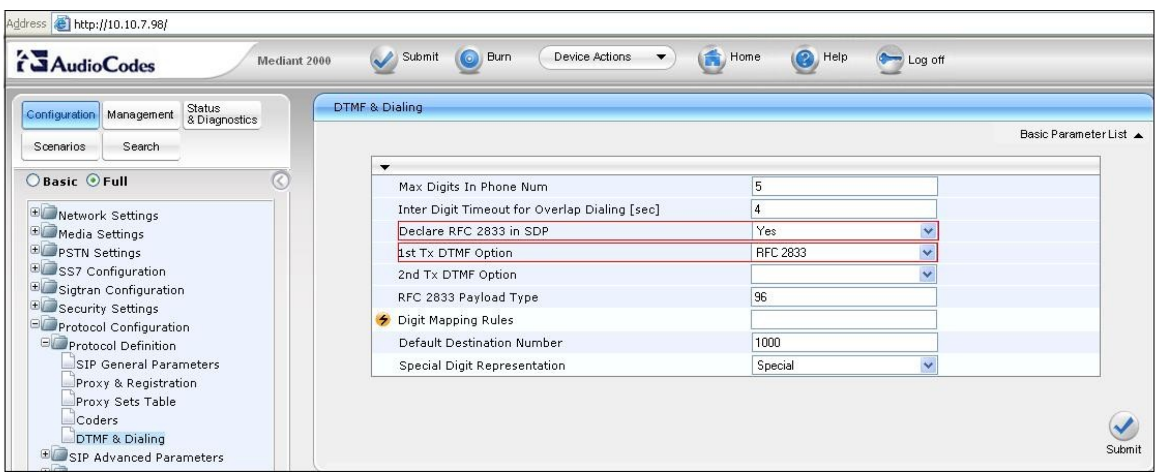
5.10 Configure In Band DTMF Settings

On the left pane click **Configuration** and navigate to **Protocol Configuration → Protocol Definition → DTMF & Dialing**. Configure **Declare RFC 2833 in SDP** to **No.**



5.11 Configure Out of Band Settings

On the left pane click **Configuration** and navigate to **Protocol Configuration → Protocol Definition → DTMF & Dialing**. Configure **Declare RFC 2833 in SDP** to **Yes** and **1st Tx DTMF Option** to **RFC 2833**.



6 General Test Approach and Test Results

The general test approach was to place calls between the Meeting Exchange and the PSTN directly via the AudioCodes Mediant 1000. The main objectives were to verify the following using the following protocols TCP and UDP.

Dial-In Conferencing

- DNIS direct call function, where conference participants enter a conference as moderator, without entering a participant-access-code (passcode).
- Scan all function, where conference participants enter a conference with a valid passcode.

Dial-Out Conferencing

- Blast Dial, auto and manual.
- Originator Dial-Out, where a conference participant is already in a conference as moderator and invokes a Dial-Out to a single participant by entering *1.
- Operator Fast Dial, where an operator can Dial-Out to a pre-provisioned dial list of one or more participant.

Audio Path

- Operator Dial-Out to establish an Audio Path.
- Operator Dial-In to establish an Audio Path.

DTMF/Touchtone commands

- *0 Request Help
- *2 (as moderator) to start/stop conference recording
- *3 to start/stop playback of conference recording
- *5 (as moderator) toggle lecture on/off
- *6 toggle mute on/off
- *7 (as moderator) toggle conference security on/off
- *8 play the roster of participant name during conference
- *93X (where X is defined from 1 to 9) to invoke a subconference
- *930 entered from a subconference to go back to the main conference
- *93# entered from a subconference (as moderator) to bring all conference participants back to the main conference
- ## (as moderator) to end the conference

7 Verification Steps

The following steps were used to verify the administrative steps presented in these Application Notes and are applicable for similar configurations in the field. The verification steps in this section validated the following:

- The Meeting Exchange configuration.
- Verify that the trunks are up on the AudioCodes Mediant 1000 by verifying the icons for those entries on the Trunk & Channel Status screen are green.
- Verify successful inbound and outbound calls between the Meeting Exchange and the PSTN

7.1 Verify Avaya Meeting Exchange Processes:

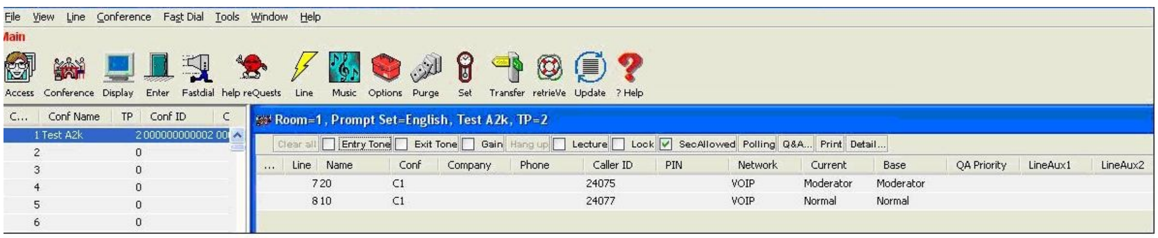
- Log in to the Meeting Exchange console to access the CLI with the appropriate credentials.
- cd to **/usr/dcb/bin**.
- At the command prompt, run the script **dcbps** and confirm all processes are running by verifying an associated Process ID (PID) for each process.

```
5946 ?    00:00:01 initdcb
6063 ?    00:00:00 log
6066 ?    00:00:00 bridgeTranslato
6067 ?    00:00:00 netservices
6074 ?    00:00:00 timer
6075 ?    00:00:00 traffic
6076 ?    00:00:00 chdbased
6077 ?    00:00:00 startd
6078 ?    00:00:00 cdr
6079 ?    00:00:00 modapid
6080 ?    00:00:00 schapid
6081 ?    00:00:12 callhand
6082 ?    00:00:00 initipcb
6087 ?    00:00:00 sipagent
6088 ?    00:00:00 msdispatcher
6089 ?    00:00:00 serverComms
6090 ?    00:00:00 softms
6091 ?    00:00:02 softms
6092 ?    00:00:00 softms
6093 ?    00:00:03 softms
6095 ?    00:00:00 softms
6096 ?    00:00:02 softms
3603 ?    00:00:04 postmaster with 23 children
```

7.2 Verify Call Routing

Verify end to end signalling/media connectivity between the Meeting Exchange and the PSTN directly via the AudioCodes Mediant 1000. This is accomplished by placing calls to and from the Meeting Exchange. This step utilizes the Avaya Bridge Talk application to verify calls to and from the Meeting Exchange are managed correctly, e.g., callers are added/removed from conferences. This step will also verify the conferencing applications provisioned

- From an endpoint on the Public Switch Telephone Network, Dial **1001** to enter a conference as **Moderator** (with passcode) while simultaneously invoking the associated Auto Blast dial feature for this conference
- If not already logged on, log in to the Avaya Bridge Talk application with the appropriate credentials
- **Double-Click on the** highlighted **Conf #** to open a **Conference Room** window.Verify conference participants are added/removed from conferences by observing the Conference Navigator and/or Conference Room windows.



8 Conclusion

These Application Notes presented a compliance-tested solution comprised of the Avaya Meeting Exchange Enterprise S6200 Conferencing Server and the AudioCodes Mediant 1000. This solution enables connectivity between the Avaya Meeting Exchange Enterprise S6200 Conferencing Server directly with AudioCodes Mediant 1000 over TCP and UDP.

9 Additional References

Avaya references, available at http://support.avaya.com

- *Meeting Exchange S6200 5.2 Administration and Maintenance S6200/S6800*
- *Avaya Meeting Exchange Enterprise Groupware Edition Version 5.2 User's Guide for Bridge Talk*

Audio Code Mediant 1000 Media Gateway for technical support and System Deployment Guide are available at http://www.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 DevConnect Program at devconnect@avaya.com.