Avaya Solution & Interoperability Test Lab

Configuring ISDN-PRI connectivity between Avaya Communication Manager and Avaya Meeting Exchange (S6200) - Issue 1.0

Abstract

These Application Notes present the provisioning required for configuring ISDN-PRI connectivity between Avaya Communication Manager and Avaya Meeting Exchange (S6200). By employing this configuration, calls originating from Avaya Communication Manager may be terminated on Avaya Meeting Exchange (i.e., 'Dial-In'). Conversely, call origination from Avaya Meeting Exchange to Avaya Communication Manager (i.e., 'Dial-Out') is also supported. ISDN-PRI connectivity is enabled via utilization of an AudioCodes TrunkPack (TP)-260/SIP PCI Media Gateway Board. This configuration leverages the flexibility offered by Avaya Communication Manager to support a rich set of conferencing options provided by Avaya Meeting Exchange.

1. Introduction

These Application Notes present the provisioning required for configuring ISDN-PRI connectivity between Avaya Communication Manager and Avaya Meeting Exchange (S6200). By employing this configuration, calls originating from Avaya Communication Manager may be terminated on Avaya Meeting Exchange (i.e., 'Dial-In'). Conversely, call origination from Avaya Meeting Exchange to Avaya Communication Manager (i.e., 'Dial-Out') is also supported. ISDN-PRI connectivity is enabled via utilization of an AudioCodes TrunkPack (TP)-260/SIP PCI Media Gateway Board. This configuration leverages the flexibility offered by Avaya Communication Manager to support a rich set of conferencing options provided by Avaya Meeting Exchange.
Note the convention for Dial-In/Dial-Out assigns Avaya Meeting Exchange as the point of reference; e.g., *Dial-In to Avaya Meeting Exchange*, *Dial-Out from Avaya Meeting Exchange*.

This configuration maximizes the inherent flexibility of protocols supported on Avaya Communication Manager by enabling any station or trunk type associated with Avaya Communication Manager to interoperate with Avaya Meeting Exchange. Also, Avaya Meeting Exchange supports a rich selection of features to enable a wide selection of conferencing requirements.

The following applications have been verified for Dial-In conferencing:

- ON DEMAND
    - With a DNIS **scan** function provisioned (e.g., conferees enter a conference with a passcode).
- FLEX
    - With a DNIS **scan** function provisioned (e.g., conferees enter a conference with a passcode).
- UNATTENDED
    - With a DNIS **direct** function provisioned (e.g., conferees enter a conference as moderator without a passcode).
    - With a DNIS **direct** function provisioned and Auto Blast feature enabled. (e.g. a conferee enters a conference as moderator without a passcode and simultaneously initiates an Auto Blast Dial to a pre-provisioned FastDialList. Conferees on the FastDialList are automatically entered into the conference without a passcode. This conference remains open for others to join via passcode.
    - With a DNIS **scan** function provisioned (e.g., passcode required).
- ATTENDED

The following applications have been verified for Dial-Out conferencing:

- Auto Blast Dial with a DNIS Direct to generate new conference.
- Blast Dial to add participants to existing conference.
- Manual (one-time) dial and add participant to existing conference.

These Application Notes will focus on the administrative steps required for configuring the following Network Elements in support of the configuration depicted in **Figure 1**.

- Avaya Communication Manager.
- Avaya Meeting Exchange.
- AudioCodes TrunkPack (TP)-260/SIP PCI Media Gateway Board.

These Application Notes will also present the provisioning required to enable the aforementioned conferencing applications.

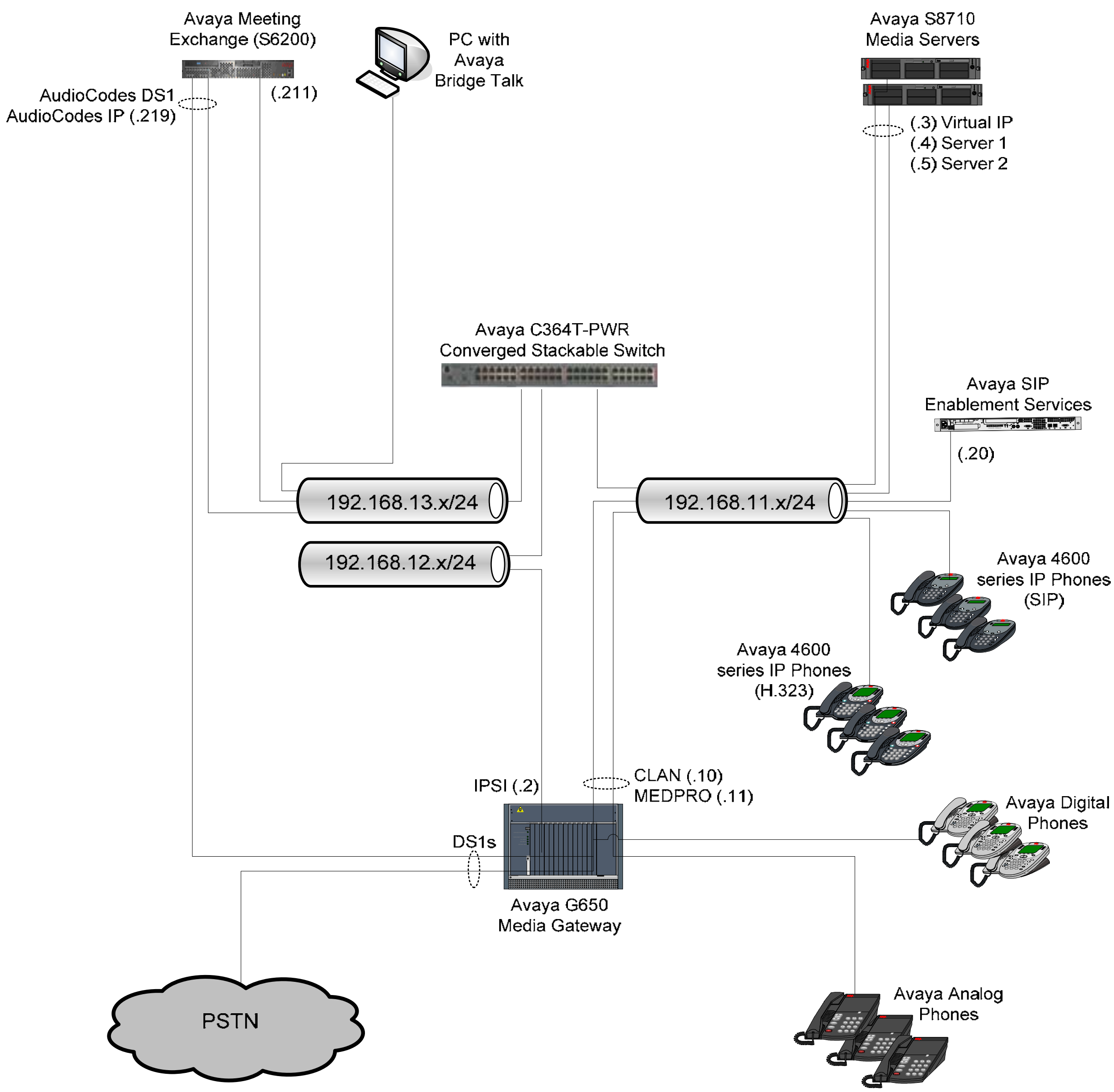


Figure 1: Network Configuration

2. Equipment and Software Validated

The following equipment and software versions were used for the configuration:

Equipment	Software
Avaya S8710 Media Servers (Duplex configuration)	R013x.01.0.628.6
Avaya G650 Media Gateway • Avaya TN2312BP (IPSI) • Avaya TN799DP (C-LAN) • Avaya TN2302AP (MEDPRO) • Avaya TN464F (DS1)	 HW12 FW031 HW01 FW017 HW20 FW112 000010
Avaya Meeting Exchange (S6200) • AudioCodes TrunkPack (TP)-260/SIP PCI Media Gateway Board	40002n V4.40.240.454
Avaya SIP Enablement Services	SES03.1-03.1.018.0
Avaya C364T-PWR Converged Stackable Switch	V4.5.14
Avaya Bridge Talk	4.1.01b
Avaya 4620 IP Telephones	2.3 (H.323)
Avaya 4602 IP Telephones	2.2 (SIP)
Avaya Analog Telephones	--
Avaya Digital Telephones	--

Table 1: Hardware and Software Versions

3. Avaya Communication Manager Configuration

This section describes the steps required for configuring Avaya Communication Manager to interoperate with Avaya Meeting Exchange (see **Figure 1**).

The following conditions are assumed as entry criteria to this section:

- IP network connectivity is configured.
- Login and password credentials are available.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.1</td><td>Log In to the S8710 Virtual IP and open a SAT session.</td></tr>
<tr><td rowspan="3">3.2</td><td>Verify Licensing for OPTIONAL FEATURES

Issue the command “display system-parameters customer-options”.
• Verify system is licensed for ISDN-PRI.</td></tr>
<tr><td><pre>
Page 4 of 10
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y   IP Stations? y
           Enable 'dadmin' Login? y   Internet Protocol (IP) PNC? n
           Enhanced Conferencing? y   ISDN Feature Plus? n
                  Enhanced EC500? y   ISDN Network Call Redirection? n
    Enterprise Survivable Server? n   ISDN-BRI Trunks? n
       Enterprise Wide Licensing? n   ISDN-PRI? y
              ESS Administration? n   Local Survivable Processor? n
          Extended Cvg/Fwd Admin? n   Malicious Call Trace? n
     External Device Alarm Admin? n   Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n   Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? n   Multifrequency Signaling? y
      Global Call Classification? n Multimedia Appl. Server Interface (MASI)? n
             Hospitality (Basic)? y   Multimedia Call Handling (Basic)? y
  Hospitality (G3V3 Enhancements)? n   Multimedia Call Handling (Enhanced)? y
                       IP Trunks? y

           IP Attendant Consoles? n
        (NOTE: You must logoff & login to effect the permission changes.)
</pre></td></tr>
<tr><td></td></tr>
</table>

Step	Description
3.3	Proceed to Page 8: • Verify system is licensed for **QSIG**.
	Page 8 of 10 QSIG OPTIONAL FEATURES **Basic Call Setup? y** **Basic Supplementary Services? y** Centralized Attendant? n Interworking with DCS? n Supplementary Services with Rerouting? n Transfer into QSIG Voice Mail? n Value-Added (VALU)? n

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.4</td><td>
Configure a DS1 CIRCUIT PACK.

Place a DS1 Board in cabinet Location: 1A09.
Issue the command “add ds1 1a09”, and administer settings as per below.
<pre>
Page 1 of 2
                                DS1 CIRCUIT PACK

            Location: 01A09   Name: DS1 to S6200
            Bit Rate: 1.544   Line Coding: b8zs
   Line Compensation: 1   Framing Mode: esf
      Signaling Mode: isdn-pri
             Connect: pbx   Interface: peer-master
   TN-C7 Long Timers? n   Peer Protocol: Q-SIG
Interworking Message: PROGress   Side: a
Interface Companding: mulaw   CRC? n
           Idle Code: 11111111
                              DCP/Analog Bearer Capability: 3.1kHz

                                           T303 Timer(sec): 4

      Slip Detection? n   Near-end CSU Type: other
</pre>
</td></tr>
<tr><td>3.5</td><td>
Configure an ISDN-PRI SIGNALING GROUP.

Issue the command “add signaling-group <number>” (for these Application Notes, number == 3), and administer settings as per below.
<pre>
Page 1 of 1
                                SIGNALING GROUP

 Group Number: 3   Group Type: isdn-pri
                     Associated Signaling? y   Max number of NCA TSC: 0
                        Primary D-Channel: 01A0924   Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC:
       Trunk Group for Channel Selection:
          Supplementary Service Protocol: b
</pre>
</td></tr>
</table>

Step	Description
3.6	Configure an ISDN-PRI **TRUNK GROUP**. Issue the command "**add trunk-group <number>**" (for these Application Notes, **number = 3**), and administer settings as per below.

```
Page 1 of 21
                                TRUNK GROUP

Group Number: 3                    Group Type: isdn          CDR Reports: y
  Group Name: S6200 ISDN PRI              COR: 1        TN: 1        TAC: 103
   Direction: two-way       Outgoing Display? y        Carrier Medium: PRI/BRI
 Dial Access? n               Busy Threshold: 255      Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n           TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

Step	Description
3.7	Proceed to Page 2 and administer 'hunting' for the **TRUNK GROUP** as per below.

```
Page 2 of 21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: b     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: ascend
                                                   Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                 Format:
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
```

Step	Description
3.8	Proceed to Page 5 and administer the members for the **TRUNK GROUP** as per below.

```
Page 5 of 21
                                  TRUNK GROUP
                                       Administered Members (min/max):   1/23
GROUP MEMBER ASSIGNMENTS                   Total Administered Members:  23

       Port    Code Sfx Name        Night           Sig Grp
  1: 01A0901  TN464  F                                3
  2: 01A0902  TN464  F                                3
  3: 01A0903  TN464  F                                3
  4: 01A0904  TN464  F                                3
  5: 01A0905  TN464  F                                3
  6: 01A0906  TN464  F                                3
  7: 01A0907  TN464  F                                3
  8: 01A0908  TN464  F                                3
  9: 01A0909  TN464  F                                3
 10: 01A0910  TN464  F                                3
 11: 01A0911  TN464  F                                3
 12: 01A0912  TN464  F                                3
 13: 01A0913  TN464  F                                3
 14: 01A0914  TN464  F                                3
 15: 01A0915  TN464  F                                3
 16: 01A0916  TN464  F                                3
 17: 01A0917  TN464  F                                3
 18: 01A0918  TN464  F                                3
 19: 01A0919  TN464  F                                3
 20: 01A0920  TN464  F                                3
 21: 01A0921  TN464  F                                3
 22: 01A0922  TN464  F                                3
 23: 01A0923  TN464  F                                3
```

Step	Description
3.9	Configure the **DIAL PLAN ANALYSIS TABLE** to send any digit string with a 'leading' **5** of **3** digits in **Total Length** to **aar**. Issue the command "**change dialplan analysis**", and administer settings as per below.

```
Page 1 of 12
                             DIAL PLAN ANALYSIS TABLE
                                                            Percent Full:    1

      Dialed  Total  Call       Dialed  Total  Call       Dialed  Total  Call
      String  Length Type       String  Length Type       String  Length Type
        0       1    attd
        1       3    dac
        2       5    ext
        3       5    ext
        4       3    aar
        5       3    aar
        6       3    ext
        7       4    ext
        7       5    ext
        8       1    fac
        9       1    fac
        *       3    fac
        #       3    fac
```

Step	Description
3.10	Configure the **AAR ANALYSIS TABLE** to send the following **Dialed Strings** to **Route Pattern 3**. Issue the command "**change aar analysis 5**", and administer settings as per below. Note: • Dialed String **503** will be used by Avaya Meeting Exchange for a **scan** function (see **Step 4.7**). • Dialed Strings **555** and **556** will be used by Avaya Meeting Exchange for a **direct** function (see **Step 4.8**).

```
Page 1 of 2
                             AAR DIGIT ANALYSIS TABLE
                                                            Percent Full:    1

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    501                    3    3     1       aar          n
    502                    3    3     2       aar          n
    503                    3    3     3       aar          n
    555                    3    3     3       aar          n
    556                    3    3     3       aar          n
```

Step	Description
3.11	Configure a **ROUTE PATTERN** to 'route' to **Grp No 3** (Trunk Group 3, see **Step 3.6**). Issue the command "**change route-pattern 3**", and administer settings as per below.

```
Page 1 of 3
                    Pattern Number: 3   Pattern Name: S6200 PRI
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 3    0                    0                                        n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 3 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                         none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

4. Avaya Meeting Exchange Configuration

This section describes the steps required for configuring Avaya Meeting Exchange to interoperate with Avaya Communication Manager (see **Figure 1**).

The following conditions are assumed as entry criteria to this section:

- IP network connectivity is configured.
- Login and password credentials are available.
- Avaya Bridge Talk is installed.

Step	Description
4.1	**Log in** to the Avaya Meeting Exchange Server.
4.2	Configure settings that relate to the 'presence' of Avaya Meeting Exchange within the SIP network by editing the **system.cfg** file as follows: • cd to **/usr/ipcb/config** • Edit the **system.cfg** file with a text editor, e.g., vi. • Add a line to identify the IP Address of Avaya Meeting Exchange (as defined in the **/etc/hosts** file), e.g., **IPAddress=192.168.13.211** • Add a line to identify the SIP request URI, e.g., **MyListener=sip:conf-bridge@192.168.13.211** Note: The name **conf-bridge** is a label.
4.3	To 'map' incoming calls to Avaya Meeting Exchange with a corresponding DDI value, edit the **UriToTelnum.tab** file as follows: • cd to **/usr/ipcb/config** • Edit the **UriToTelnum.tab** file with a text editor, e.g., vi. • Add a line to allow Dial-In from Avaya Communication Manager by matching and converting incoming SIP URIs in the SIP Invite message to DDI values; e.g., **"*<sip:*@*" $2** where **"*<sip:*@*"** will match the incoming SIP URI and **$2** will utilize the variable contained in the second ***** as the DDI value for the call. • To allow an undefined caller to enter a help queue for operator assistance, administer for the condition of an undefined SIP URI header by adding a wildcard entry as the last line in the file, e.g., *** $0** Note: Entries in this file are read sequentially, therefore, it follows that the line *** $0** must be the last line in the file. Otherwise, all calls to Avaya Meeting Exchange would match the wildcard and thus go to the help queue.

Step	Description
4.4	To configure 'routing' of outbound call from Avaya Meeting Exchange, edit the **telnumToUri.tab** file as follows: • cd to **/usr/ipcb/config** • Edit the **telnumToUri.tab** file with a text editor, e.g., vi. • Add a line to the file to allow for Dial-Out from Avaya Meeting Exchange, e.g., * **sip:$0@192.168.13.219** where ***** will allow any dialed digits to be sent to the default gateway: **192.168.13.219** (where 192.168.13.219 is the IP Address of the AudioCodes Media Gateway). Therefore, if 123 were dialed, the SIP URI would be defined as: sip:123@192.168.13.219.

The following steps will show how to provision conferences on Avaya Meeting Exchange.

Step	Description
4.5	To 'map' **DDI** values (obtained in **Step 4.3**) to **DNIS** entries run the **cbutil** utility as follows: • At the command prompt enter **tcsh** to set the environment on Avaya Meeting Exchange.
4.6	• At the command prompt enter **cbutil list** to verify DNIS entries provisioned on Avaya Meeting exchange. Note: • An optional 'wildcard' DNIS entry (**???**) is present to catch any unmatched **DDI** values.

```
S6200>cbutil list
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

DNIS             Msg PS  Function Line Name            Company Name
---------------- --- --- -------- -------------------- --------------------
???              208 1   ENTER
```

Step	Description
4.7	• At the command prompt enter **cbutil add** to add a DNIS entry for a **scan** function for DNIS **503**.

```
S6200>cbutil add 503 1 1 scan
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
```

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.8</td><td>

- At the command prompt enter **cbutil add** to add a DNIS entry for a **direct** function for DNIS **555**.

<pre>
S6200>cbutil add 555 0 1 direct
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
</pre>

- Repeat to add **direct** function for **556**.

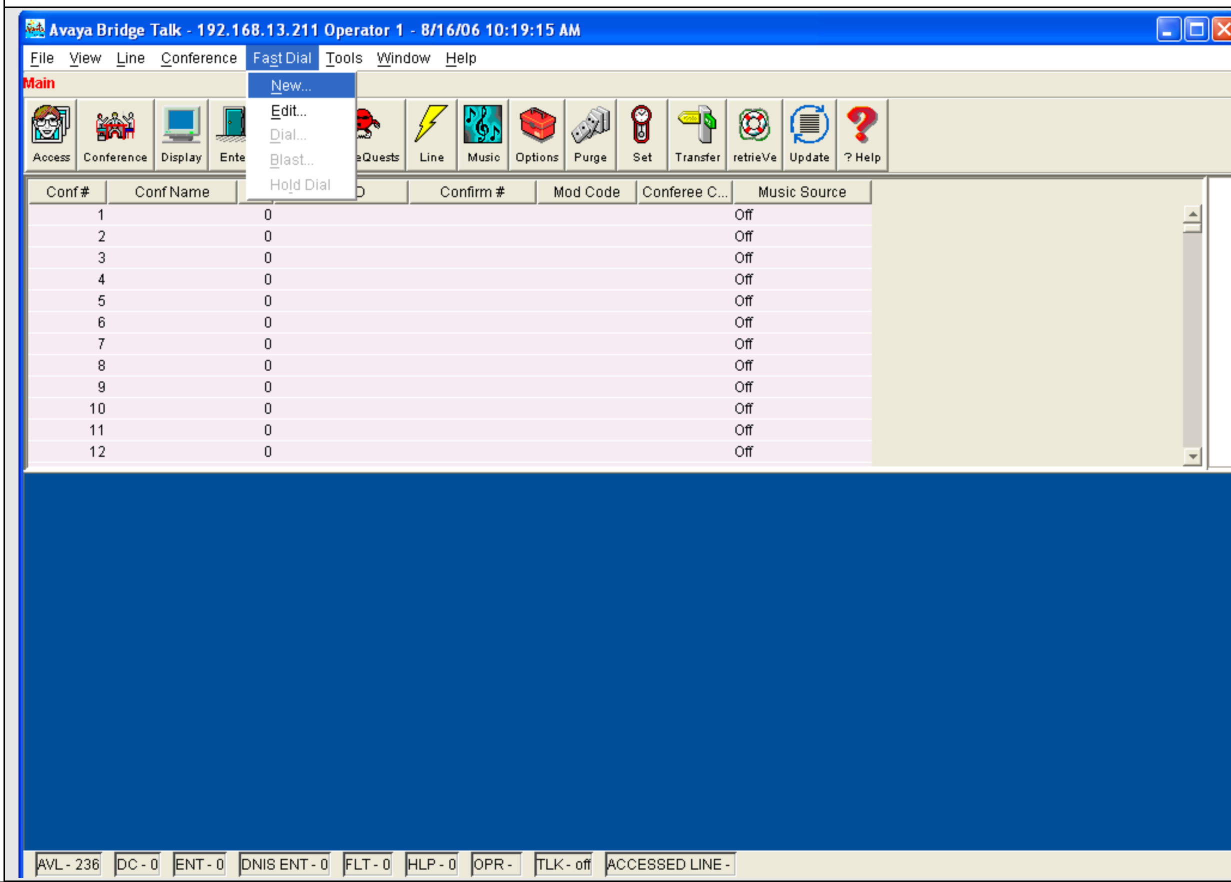
</td></tr>
<tr><td>4.9</td><td>

- At the command prompt enter **cbutil list** to verify DNIS entries provisioned on Avaya Meeting exchange.

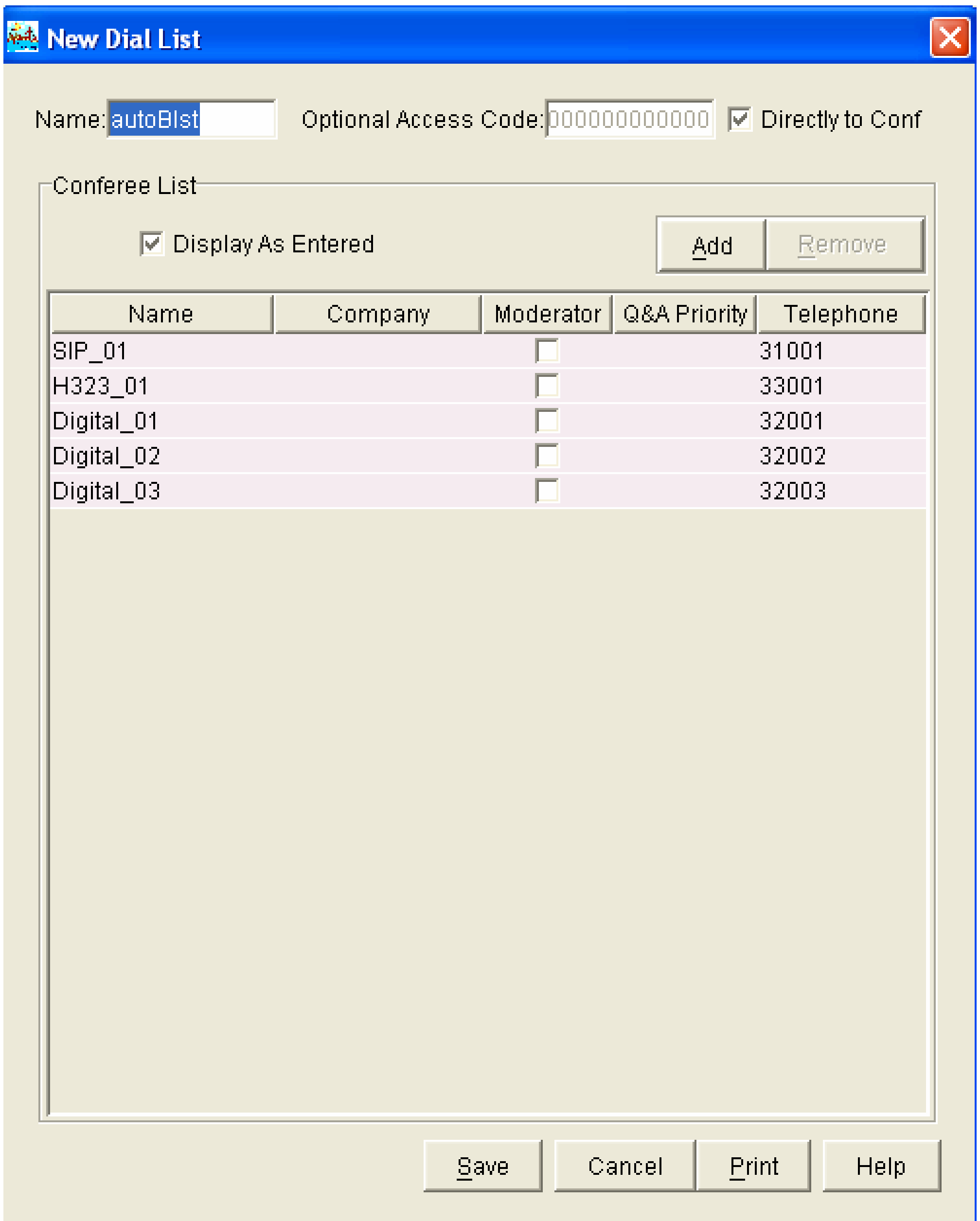
<pre>
S6200>cbutil list
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

DNIS   Msg PS Function Line Name   Company Name
---------------- --- --- -------- -------------------- --------------------
503   1   1   SCAN
555   0   1   DIRECT
556   0   1   DIRECT
???   208 1   ENTER
</pre>

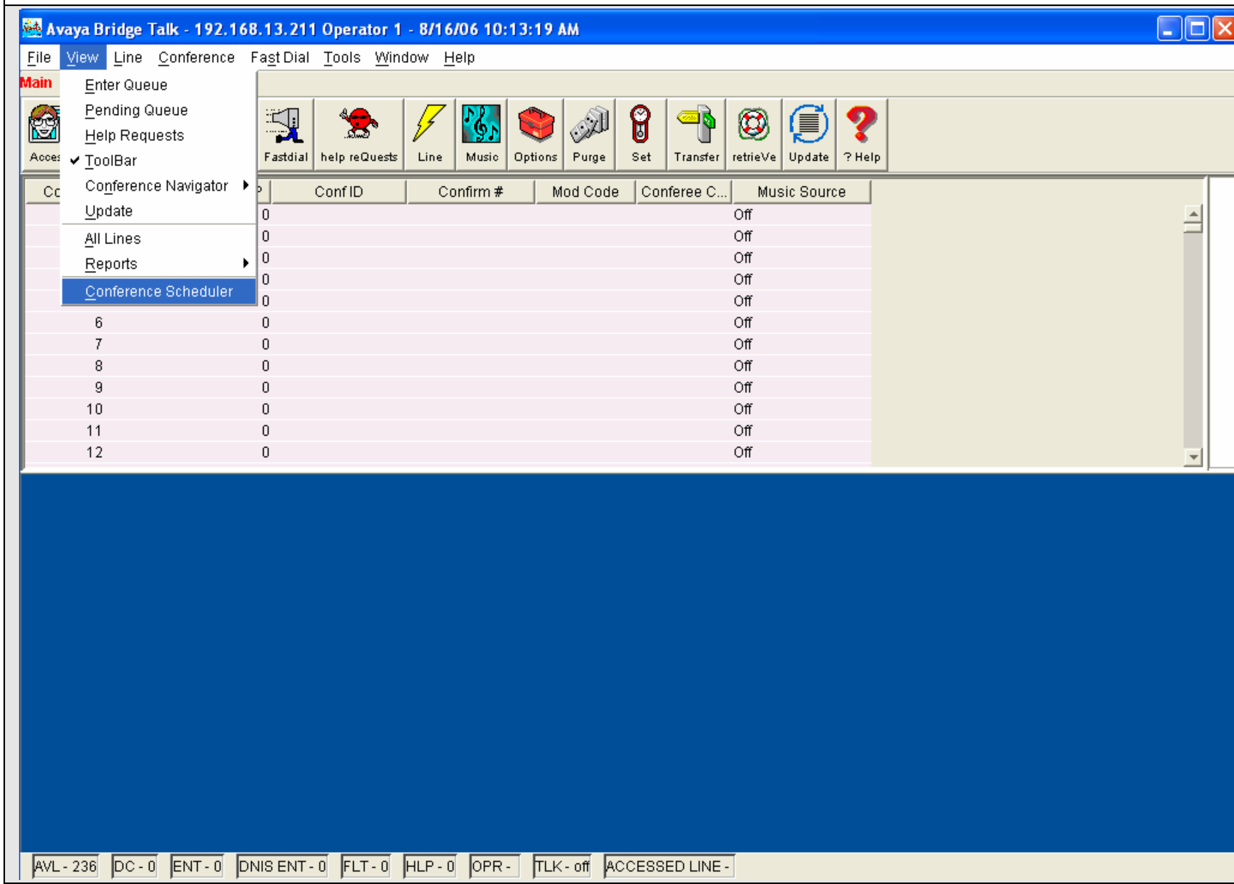
</td></tr>
<tr><td>4.10</td><td>Reboot Avaya Meeting Exchange to make change take effect.</td></tr>
</table>

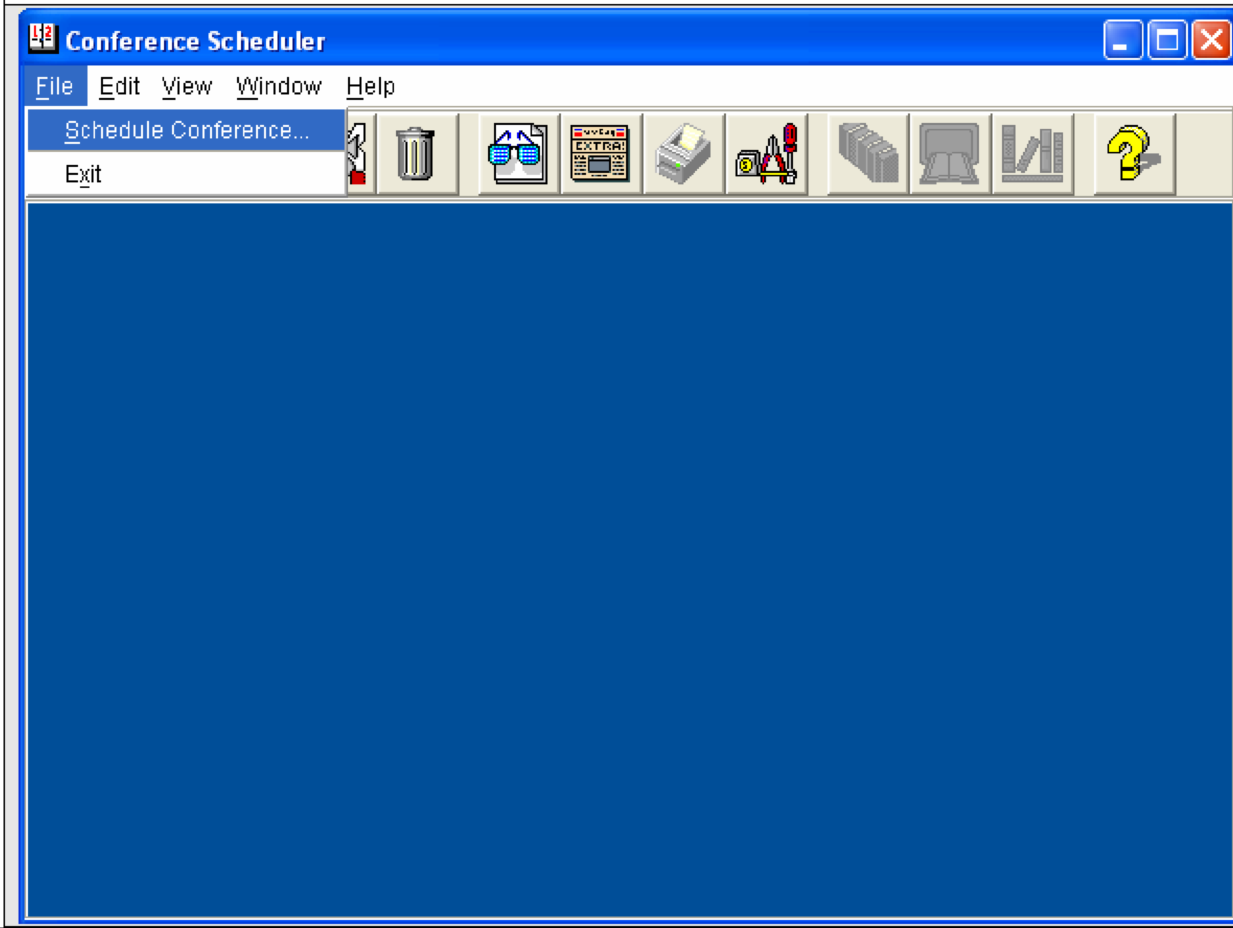
Step	Description
4.11	To provision conferences on Avaya Meeting Exchange: • Open the Avaya Bridge Talk Application and **Log in**. Note: • The following steps will detail how to provision an Auto Blast dial conference using DNIS entry **556** (see **Step 4.9**).
4.12	From the Avaya Bridge Talk **Menu Bar**: • Click **Fast Dial**, **New**.
	

Step	Description
4.13	From the **New Dial List** window: • Check the **Directly to Conf** box to allow conferees to enter a conference without a passcode. • Add conferees to ‘Blast Dial’ by clicking the **Add** button for each entry. ○ Give moderator privileges to a conferee by checking the Moderator box. • When finished, click the **Save** button on the bottom of the screen.



Step	Description
4.14	From the Avaya Bridge Talk **Menu Bar**: • Click **View**, **Conference Scheduler**.



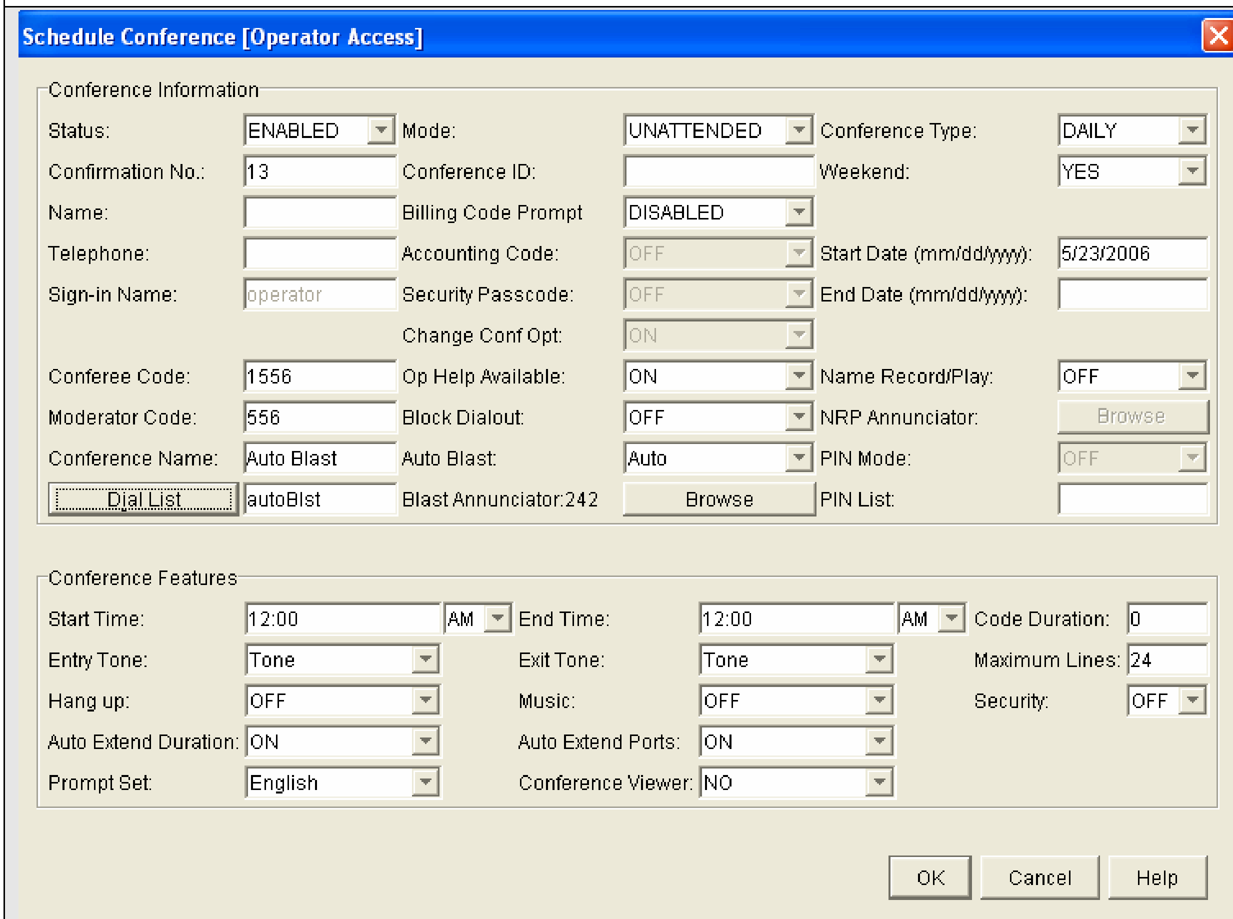
Step	Description
4.15	From the **Conference Scheduler** window: • Click **File**, **Schedule Conference**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.16</td><td>From the Schedule Conference window:

Administer settings for a conference with a DNIS direct function provisioned and Auto Blast feature enabled as per below.
When finished, click the OK button on the bottom of the screen.

Note:

If Auto Blast button is not present, contact Avaya Services.
Dial List is form Step 4.13.
To allow moderator access without a passcode, the Moderator Code (556) must have a DNIS entry for 556 with direct function provisioned (see Step 4.9).



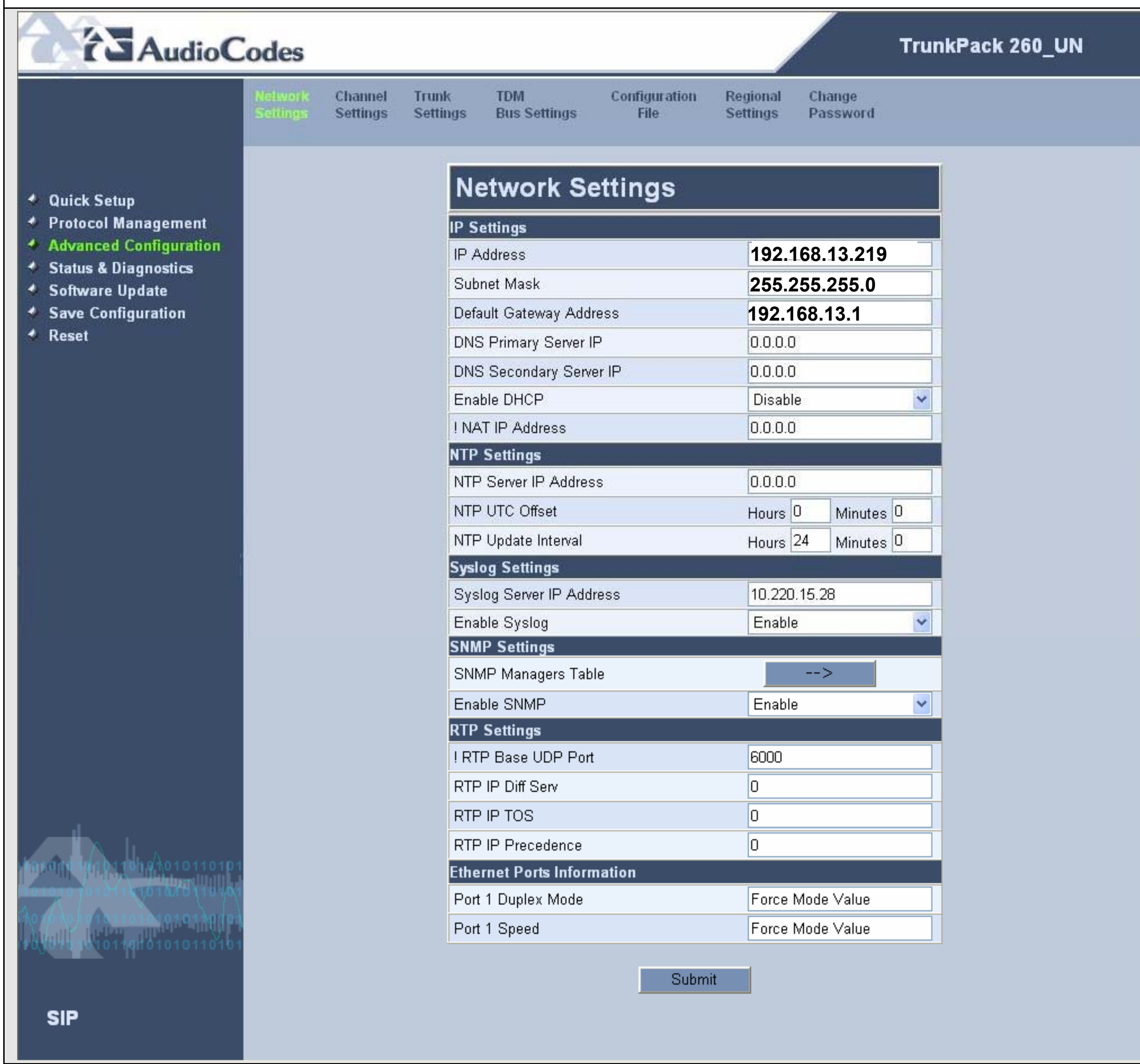
</td></tr>
</table>

5. Configure the AudioCodes Media Gateway

This section describes the steps required for configuring the AudioCodes Media Gateway to interoperate with:

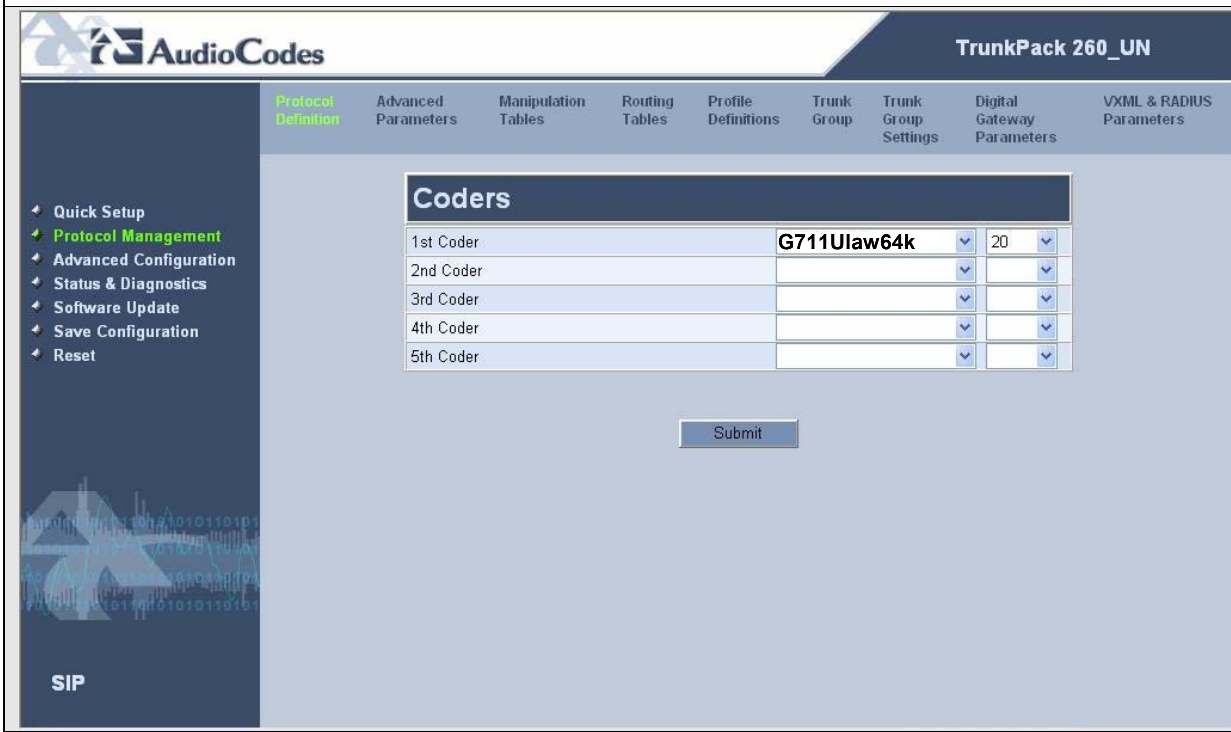
- Avaya Meeting Exchange via SIP (see **Figure 1**).
- Avaya Communication Manager via ISDN-PRI (see **Figure 1**).

Step	Description
5.1	Administer settings for Avaya Meeting Exchange as follows: • Open a web browser and enter the following URL: **http://<default IP Address>** • **Log in** to the AudioCodes Media Gateway. Note: • To obtain default IP Address, login and password information, see **Additional References**: *TP-260 UN SIP User's Manual*.

Step	Description
5.2	Configure **Network Settings** as follows: • Click **Advanced Configuration**. • Click **Network Settings**. • Administer settings as per below. • When finished, click the **Submit** button on the bottom of the screen. Note: • The **Network Settings** configured for the Audio Codes Media Gateway must have layer 3 connectivity with Avaya Meeting Exchange.
	

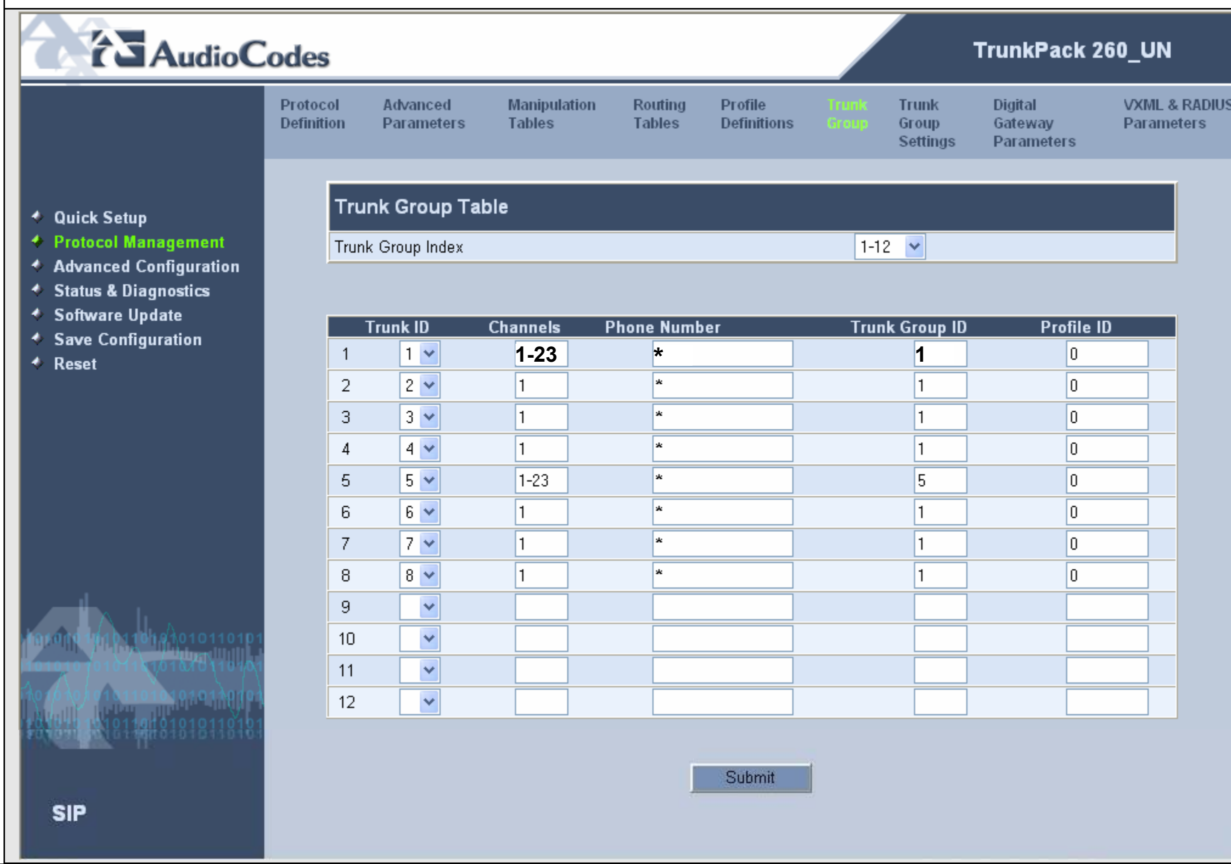
Step	Description
5.3	Configure **Proxy & Registration** parameters as follows: • Click **Protocol Management**. • Click **Protocol Definition, Proxy & Registration**. • Administer settings as per below. • When finished, click the **Submit** button on the bottom of the screen.

Step	Description
5.4	Configure **Coders** as follows: • Click **Protocol Management**. • Click **Protocol Definition, Coders**. • Administer settings as per below. • When finished, click the **Submit** button on the bottom of the screen. Note: • Configure a **Coder** that is supported on Avaya Meeting Exchange; either **g711Ulaw64k**, or **g711Alaw64k**.



Step	Description
5.5	Configure **Trunk Settings** to interoperate with Avaya Communication Manager (see **Step 3.4**) as follows: • Click **Advanced Configuration**. • Click **Trunk Settings**. • Select **Trunk Number 1**. • Administer settings as per below. • When finished, click the **Apply Trunk Settings** button on the bottom of the screen.

Step	Description
5.6	Configure **Trunk Group Table** as follows: • Click **Protocol Management**. • Click **Trunk Group**. • Administer settings as per below. • When finished, click the **Submit** button on the bottom of the screen. Note: • **1-23** channels for **Trunk ID 5** are provisioned due to Channel 24 being utilized as a signaling channel for QSIG.

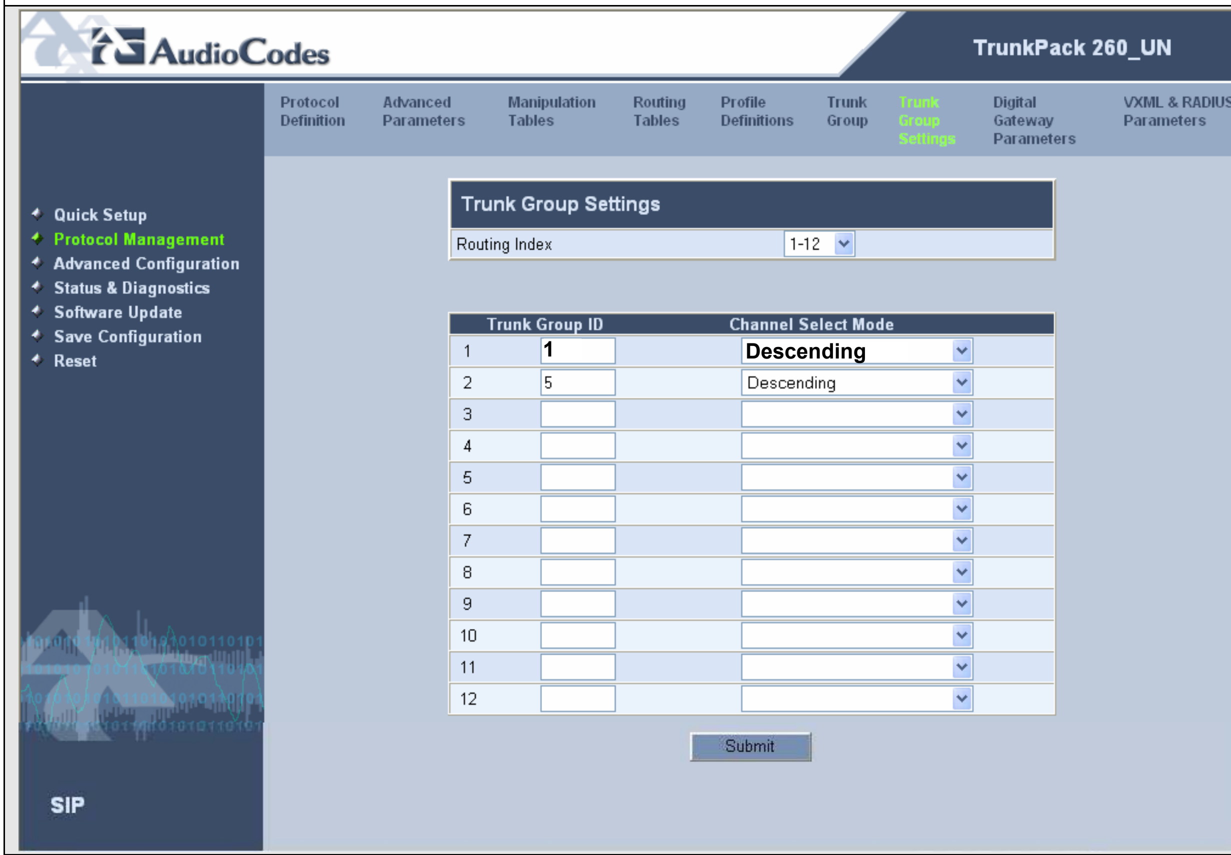


<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.7</td><td>
Configure Trunk Group Settings as follows:

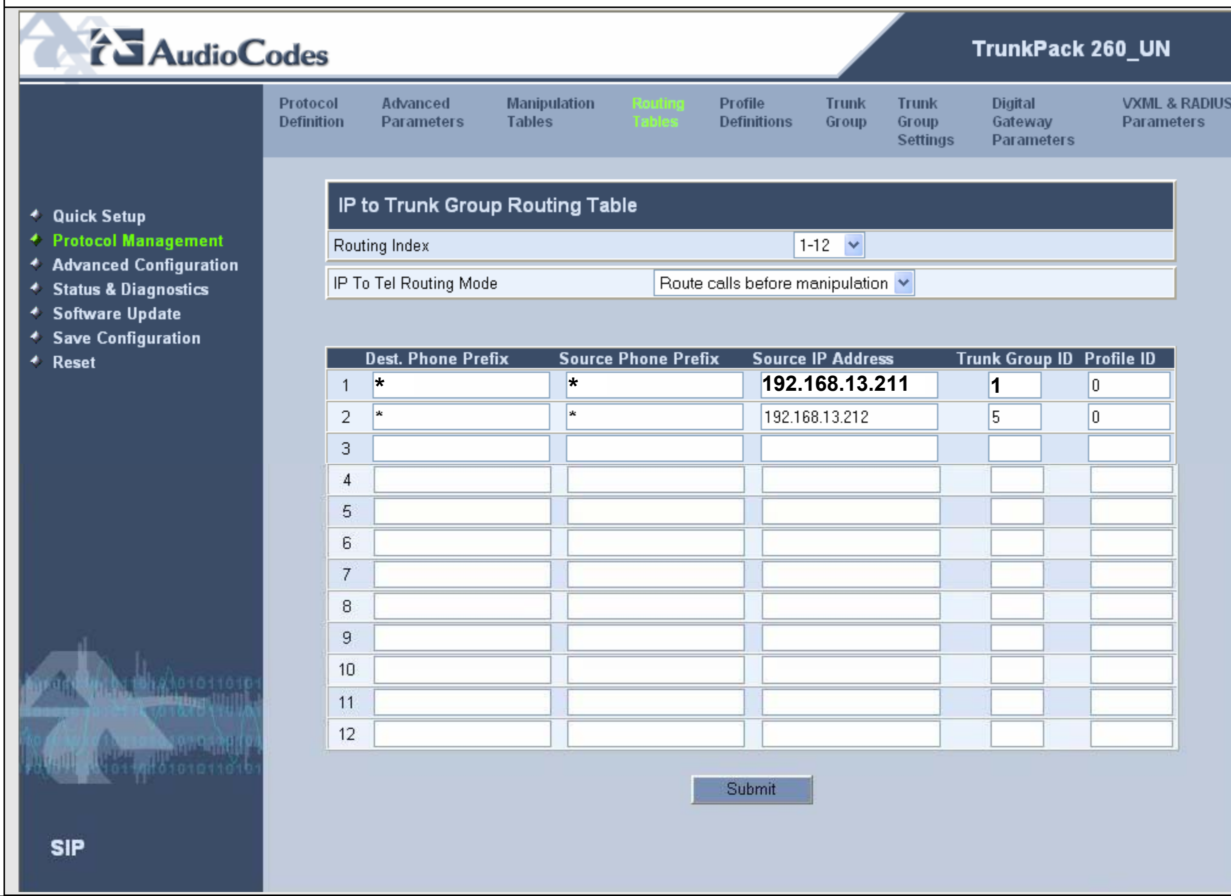
Click Protocol Management.
Click Trunk Group Settings.
Administer settings as per below.
When finished, click the Submit button on the bottom of the screen.

Note:

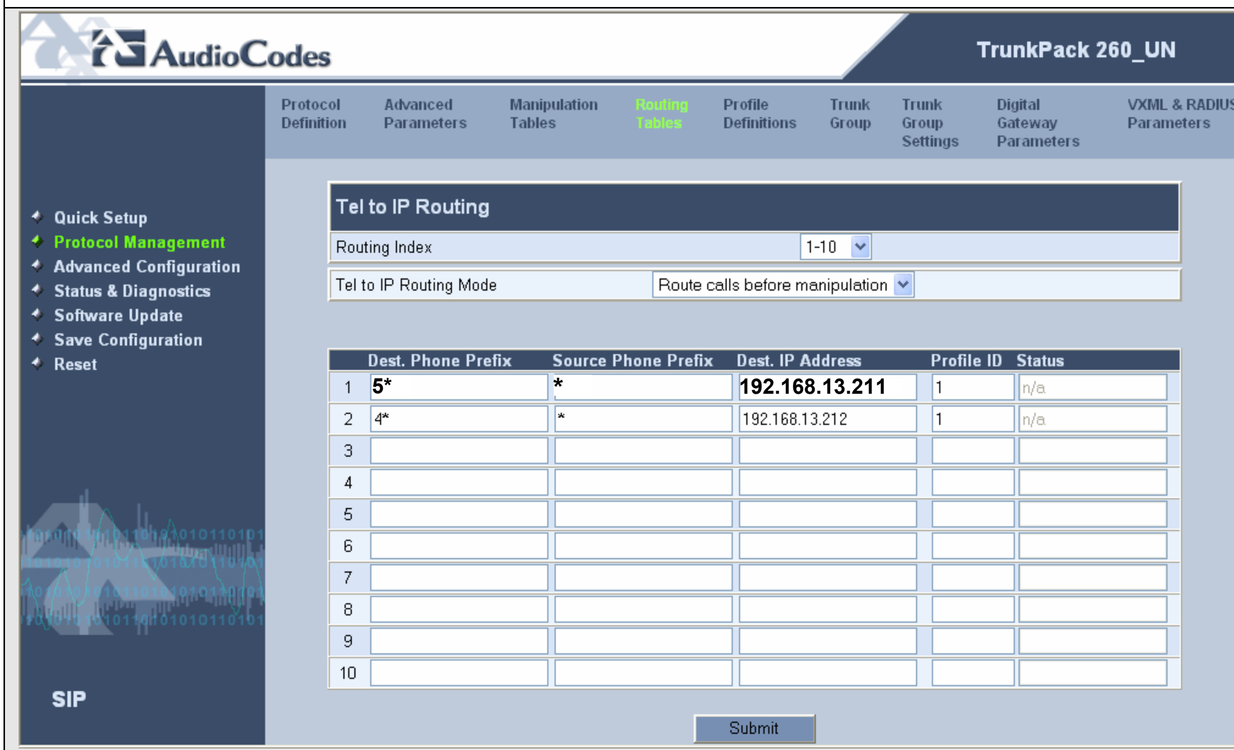
The Channel Select Mode is provisioned to ‘hunt’ in the opposite direction as Avaya Communication Manager (see Step 3.7).

</td></tr>
<tr><td></td><td>


</td></tr>
</table>

Step	Description
5.8	To enable Dial-Out from Avaya Meeting Exchange, configure the **IP to Trunk Group Routing Table** as follows: • Click **Protocol Management**. • Click **Routing Tables, IP to Trunk Group Routing**. • Administer settings as per below. • When finished, click the **Submit** button on the bottom of the screen. Note: • All calls originating from **Source IP Address** = **192.168.13.211** (Avaya Meeting Exchange) are 'routed' to **Trunk Group ID** = **1** (the trunk that connects to Avaya Communication Manager).
	

Step	Description
5.9	To allow Dial-In to Avaya Meeting Exchange, configure **Tel to IP Routing** as follows: • Click **Protocol Management**. • Click **Routing Tables, Tel to IP Routing**. • Administer settings as per below. • When finished, click the **Submit** button on the bottom of the screen. Note: • All calls with **Dest. Phone Prefix** = **5*** are 'routed' to **Dest. IP Address** = **192.168.13.211** (Avaya Meeting Exchange).



6. Verification Steps

The following steps can be used to verify the configuration described in these Application Notes.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.1</td><td>Verify all members for the ISDN trunk group are in-service/idle.

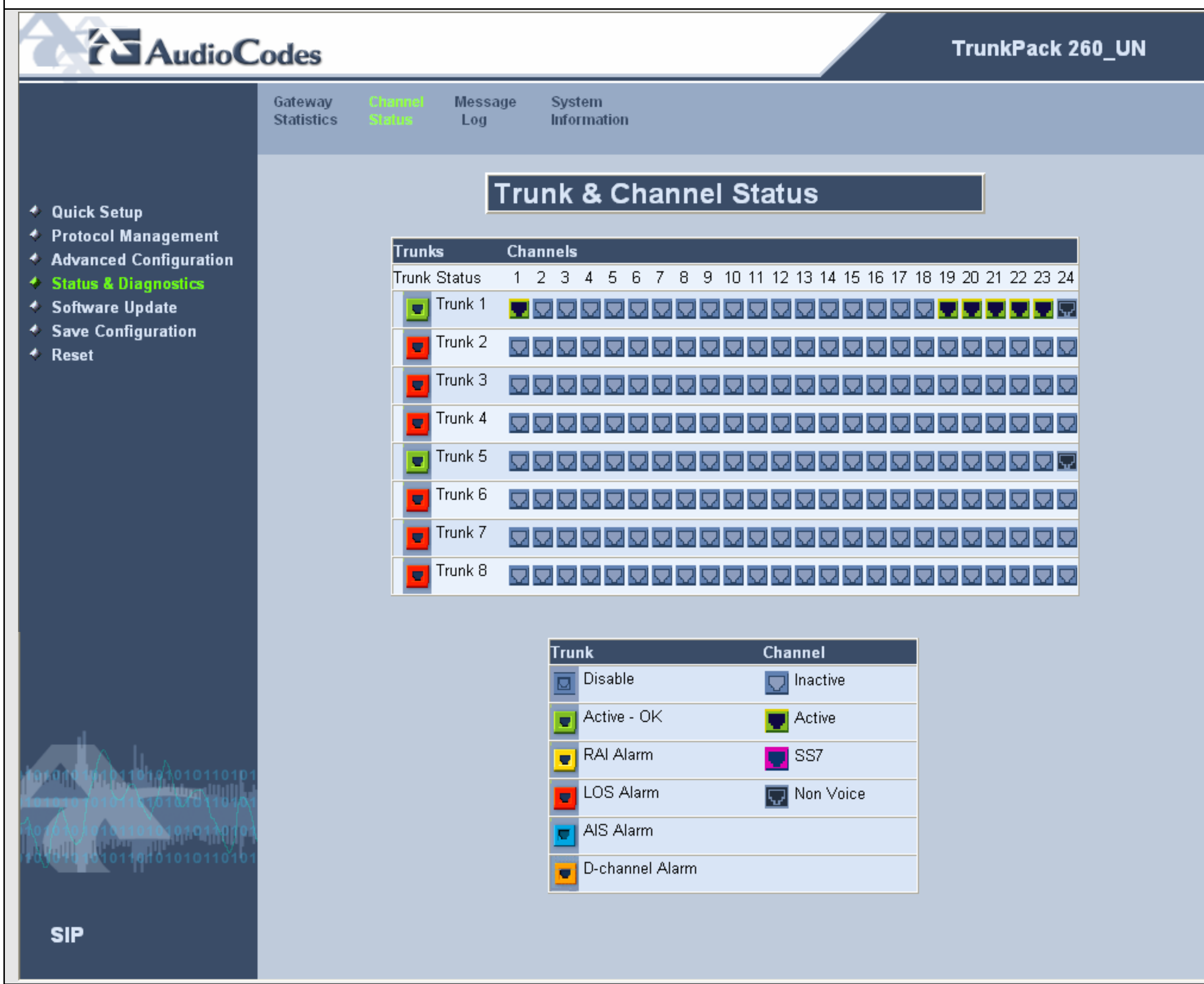
From a SAT session:
Issue the command “status trunk 3”.
All members should return the value in-service/idle.</td></tr>
<tr><td>6.2</td><td>Run the dcbps script to verify all ‘conferencing related’ processes are running on Avaya Meeting Exchange.
Log in to the Avaya Meeting Exchange Server.
cd to /usr/dcb/bin
At the command prompt, run the script: dcbps
Note:
All processes are running.
<pre>S6200> ./dcbps
  1719   FP 101 ?   0:00 bridgeTr
  1718   FP 101 ?   0:00 log
  1676   FP 144 ?   0:01 initdcb
  1720   FP 105 ?   0:00 netservi
  1723   FP 129 ?   0:00 timer
  1724   FP 101 ?   0:00 traffic
  1725   FP 104 ?   0:00 chdbased
  1726   FP 101 ?   0:00 startd
  1727   FP 109 ?   0:00 cdr
  1728   FP 101 ?   0:00 modapid
  1729   FP 101 ?   0:00 schapid
  1730   FP 104 ?   0:00 callhand
  1731   FP 139 ?   0:00 initipcb
  1732   FP 139 ?   0:00 sipagent
  1733   FP 139 ?   0:00 msdispat
  1734   FP 158 ?   0:00 softms
  1574   TS 80 ?   0:00 sqlexecd with 5 children</pre></td></tr>
</table>

Step	Description
6.3	Verify the ISDN trunk group is utilized when a call from a SIP station Dials-In to Avaya Meeting Exchange. From a SAT session: • Issue the command "**list trace tac 103**", where **103** is the TAC defined for the trunk group provisioned in **Step 3.6**. From a SIP station, dial **556** to initiate a DNIS **direct** with Auto Blast call scenario. Note: • The hunt pattern for this trunk is ascending.

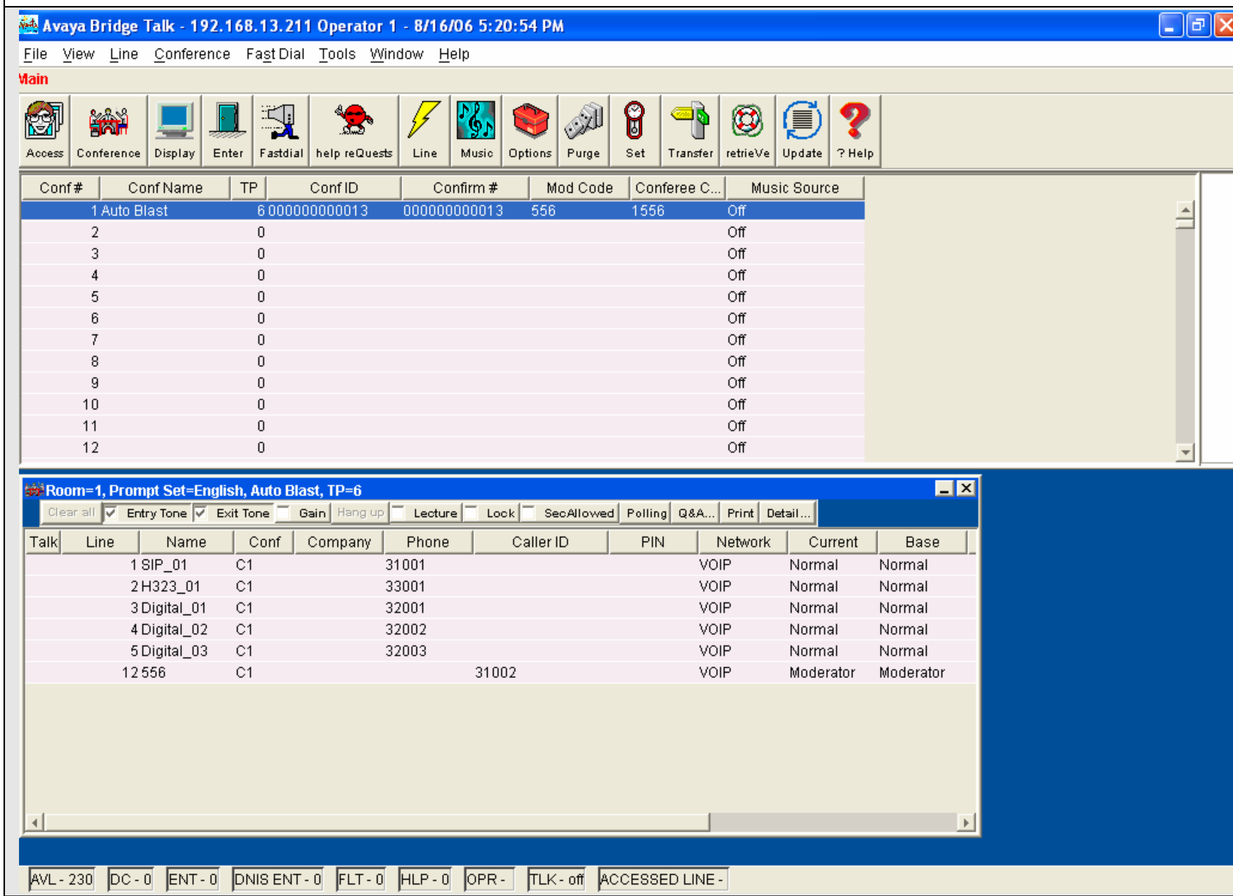
```
list trace tac 103                                                        Page   1

                                LIST TRACE

time            data

17:21:25     dial 556 route:AAR
17:21:25     term trunk-group 3    cid 0x219
17:21:25     dial 556 route:AAR
17:21:25     route-pattern  3 preference 1  cid 0x219
17:21:25     seize trunk-group 3 member 1  cid 0x219
17:21:25     Calling Number & Name 31002 SIP 31002
17:21:25     Proceed trunk-group 3 member 1  cid 0x219
17:21:26     active trunk-group 3 member 1  cid 0x219
```

Step	Description

6.4	Verify ISDN **Trunk & Channel Status** on AudioCodes as follows:
	• Click **Status & Diagnostics**.
	• Click **Channel Status**.
	
	Note:
	• This screen capture depicts the Auto Blast Dial initiated in **Step 6.3**.
	• Status for **Trunk 1** is **Active - OK**.
	• 'Hunt' pattern for 6 **Active** channels on **Trunk 1**.
	    o Ascending Channels are used for Dial-In. For this scenario, **Channel 1** is the moderator.
	    o Channels 19-23 are the conferees that were added to the conference via Auto Blast Dial.



Step	Description
6.5	Verify that calls to and from Avaya Meeting Exchange are managed correctly, e.g., callers are added/removed from conferences. This is verified visually by the following procedures: • **Log In** to Avaya Bridge Talk • **Double-Click** the highlighted **Conf #** to open a **Conference Room** window • Verify callers are added/removed from conferences by observing the Conference Navigator and/or Conference Room windows Note: • This screen capture depicts the Auto Blast Dial initiated in **Step 6.3**.



7. Conclusion

These Application Notes have presented the steps required for configuring the following:

- Dial-In to Avaya Meeting Exchange from Avaya Communication Manager via ISDN-PRI.
- Dial-Out from Avaya Meeting Exchange to Avaya Communication Manager via ISDN-PRI.

8. Additional References

- *Administrator Guide for Avaya Communication Manager*, Doc ID: 03-300509, available at http://support.avaya.com
- *Administration for Network Connectivity for Avaya Communication Manager*, Doc ID: 555-233-504, available at http://support.avaya.com
- *Meeting Exchange Field Service Guide for the S6200, S6500, and S6800 1.2 Media Servers*, Doc ID: 04-300521, available at http://support.avaya.com
- *Bridge Talk User's Guide, Version 4.0*, Doc ID: 81100300, available at http://support.avaya.com
- *TP-260 UN SIP User's Manual Version 4.4*, Document #: LTRT-68002

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