Avaya Solution & Interoperability Test Lab

Configuring secure SIP connectivity utilizing Transport Layer Security (TLS) between Avaya Communication Manager and Avaya Meeting Exchange (S6200) - Issue 1.0

Abstract

These Application Notes present the provisioning required for configuring secure SIP 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. 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 secure SIP 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. 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 securely interoperate with Avaya Meeting Exchange via TLS. TLS is a protocol that ensures privacy between communicating applications and their users on the Internet. When a server and client communicate, TLS ensures that no third party may eavesdrop or tamper with any message. 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.

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

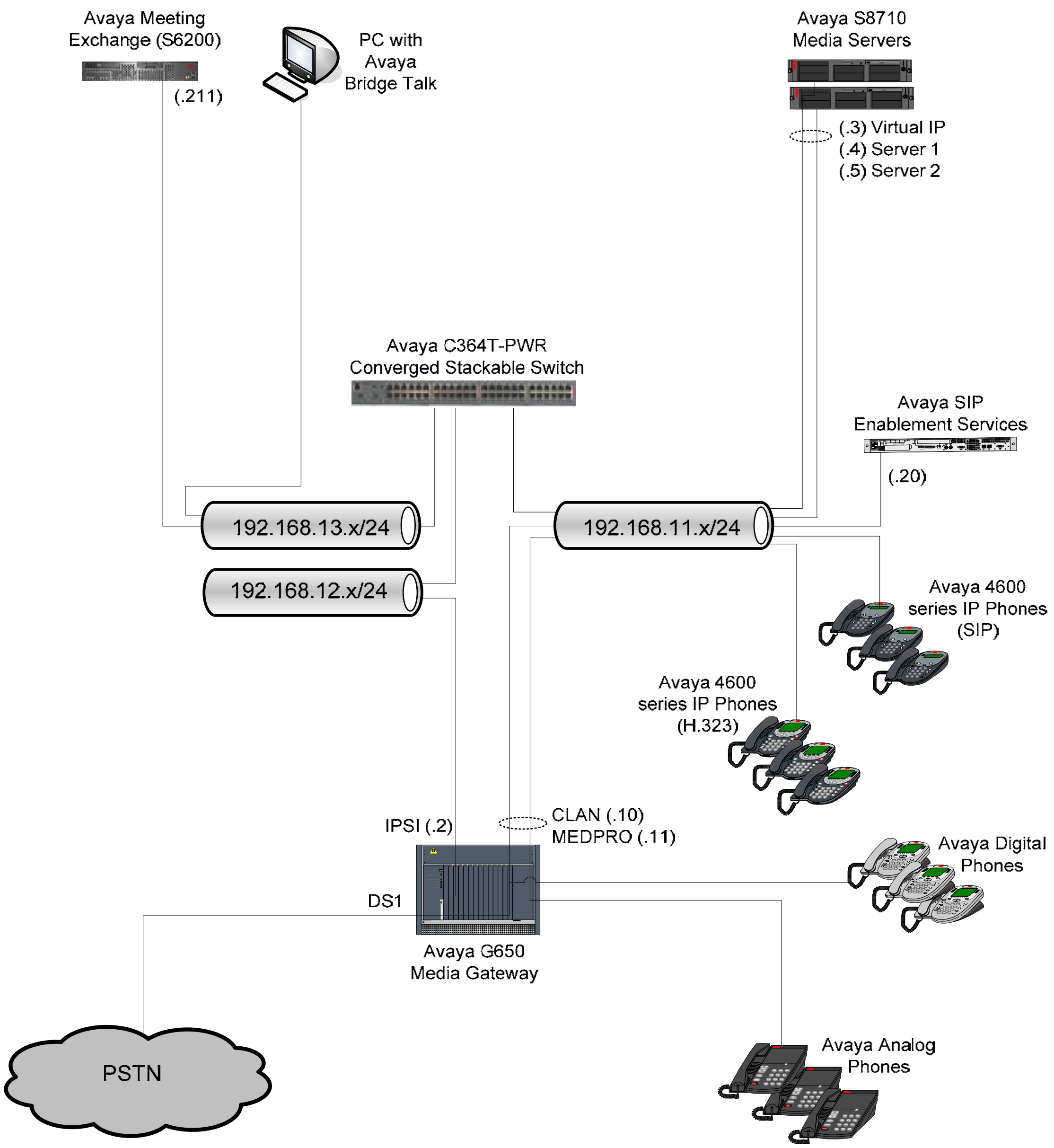


Figure 1: Network Configuration

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)	40002n
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.

Step	Description
3.1	**Log In** to the S8710 Virtual IP and open a SAT session.
3.2	Verify **Licensing** for **OPTIONAL FEATURES** Issue the command "**display system-parameters customer-options**", and verify system is licensed for SIP Trunks.

```
Page 2 of 10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                      USED
                     Maximum Administered H.323 Trunks: 1000  0
           Maximum Concurrently Registered IP Stations: 100   0
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                   Maximum Video Capable H.323 Stations: 0     0
                    Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 1000  0

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 1     0
                     Maximum G250/G350/G700 VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 0     0
          Maximum TN2602 Boards with 320 VoIP Channels: 0     0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

Step	Description
3.3	Configure an **IP Codec Set**. Issue the command "**change ip-codec-set <number>**" (for these Application Notes, **number = 1**), and administer settings as per below. Note: • Configure an **Audio Codec** that is supported on Avaya Meeting Exchange; either **G.711MU**, or **G.711A**.
	(screen below)

```
Page 1 of 2
                          IP Codec Set

    Codec Set: 1

    Audio         Silence      Frames   Packet
    Codec         Suppression  Per Pkt  Size(ms)
 1: G.711MU            n         2        20
 2:
 3:
 4:
 5:
 6:
 7:
```

Step	Description
3.4	Configure an **IP NETWORK REGION**. Issue the command "**change ip-network-region <number>**" (for these Application Notes, **number = 2**), and administer settings as per below. Note: • **Codec Set (**from **Step 3.3): 1**.

```
Page 1 of 19
                               IP NETWORK REGION
  Region: 2
Location:         Authoritative Domain:
    Name: S6200
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3327
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Step	Description
3.5	Proceed to Page 3 and administer the common codec sets on the **Inter Network Region Connection Management** screen as per below. Note: • **codec set** = **1** (from **Step 3.3**) added to enable inter-region connectivity to IP network region 1.

```
Page 3 of 19
                   Inter Network Region Connection Management

 src dst  codec  direct                                      Dynamic CAC
 rgn rgn   set    WAN    WAN-BW-limits  Intervening-regions    Gateway    IGAR
 2   1     1       y           :NoLimit                                    n
 2   2     1
 2   3
 2   4
 2   5
 2   6
 2   7
 2   8
 2   9
 2   10
 2   11
 2   12
 2   13
 2   14
 2   15
```

Step	Description
3.6	Configure **IP NODE NAMES**. Issue the command "**change node-names ip**", and administer settings as per below. Note: • Add a node name for Avaya Meeting Exchange. • Verify **CLAN** and **MEDPRO** are present.

```
Page 1 of 1
                                  IP NODE NAMES
    Name                IP Address
CLAN-1A02             192.168.11 .10
MEDPRO-1A03           192.168.11 .11
S6200                 192.168.13 .211
SES                   192.168.11 .20
```

Step	Description
3.7	Configure a SIP **SIGNALING GROUP**. Issue the command "**add signaling-group <number>**" (for these Application Notes, **number = 2**), and administer settings as per below. Note: • **Near-end Node Name (**from **Step 3.6): CLAN-1A02**. • **Far-end Node Name (**from **Step 3.6): S6200**. • **Far-end Network Region (**from **Step 3.4): 2**.

```
Page 1 of 1
                                SIGNALING GROUP

 Group Number: 2              Group Type: sip
                        Transport Method: tls






   Near-end Node Name: CLAN-1A02             Far-end Node Name: S6200
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 2
       Far-end Domain:

                                             Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? y
 Session Establishment Timer(min): 120
```

Step	Description
3.8	Configure a SIP **TRUNK GROUP**. Issue the command “**add trunk-group <number>**” (for these Application Notes, **number = 2**), and administer settings as per below. Note: • **Signaling Group (**from **Step 3.7): 2**. • The **Number of Members** added to this trunk group can be provisioned to a maximum value = **255**, but can not exceed the ‘licensed’ value for **Maximum Administered SIP Trunks** (from **Step 3.2**).

```
Page 1 of 21
                                TRUNK GROUP

Group Number: 2                   Group Type: sip           CDR Reports: y
  Group Name: S6200 SIP                  COR: 1        TN: 1        TAC: 102
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                         Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                      Signaling Group: 2
                                                    Number of Members: 50
```

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 2**. Issue the command "**change aar analysis 5**", and administer settings as per below. Note: • Dialed String **502** 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      2        aar          n
    556                  3    3      2        aar          n
```

Step	Description
3.11	Configure a **ROUTE PATTERN** to 'route' to **Grp No 2** (Trunk Group 2, see **Step 3.8**). Issue the command "**change route-pattern 2**", and administer settings as per below.

```
Page 1 of 3
                    Pattern Number: 2   Pattern Name: S6200 SIP
                             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: 2    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. • To support TLS, from the command prompt, enter command: **sum /usr/local/ssl/certs/*.pem** • The values for the certificates should come back as follows: **11666** 4 /usr/local/ssl/certs/**CAcert.pem** **41653** 9 /usr/local/ssl/certs/**CliCert1.pem** **12729** 2 /usr/local/ssl/certs/**CliKey1.pem** **41827** 9 /usr/local/ssl/certs/**ServCert1.pem** **12985** 2 /usr/local/ssl/certs/**ServKey1.pem** • If the values for the certifications come back different, contact Avaya Services.

Step	Description
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=sips:conf-bridge@192.168.13.211:5061;transport=tls** Note: The name **conf-bridge** is a label. • Add a line to overwrite the contact field for SIP responses, e.g., **respContact=<sip:conf-bridge@192.168.13.211:5061;transport=tls>**
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:*@*" $1** where **""*"*<sip:*@*"** will match the incoming SIP URI and **$1** will utilize the variable contained in the first ***** as the DDI value for the call Note: The variable contained in **$1** is utilized by Avaya Meeting Exchange to display the station **Name**, (as configured in **Step 5.3)** in a conference room on Avaya Bridge Talk (see **Step 5.6**). $3 could also have been used, however, this would <u>not</u> capture the station **Name** variable. • 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@192.168.11.10:5061;transport=tls** where ***** will allow any dialed digits to be sent to the default gateway **192.168.11.10 (**where 192.168.11.10 is the IP Address of the CLAN on Avaya Communication Manager). Therefore, if 123 were dialed, the SIP URI would be defined as: sip:123@192.168.11.10:5061;transport=tls.

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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.5</td><td>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.</td></tr>
<tr><td>4.6</td><td>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.
<pre>S6200>cbutil list
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

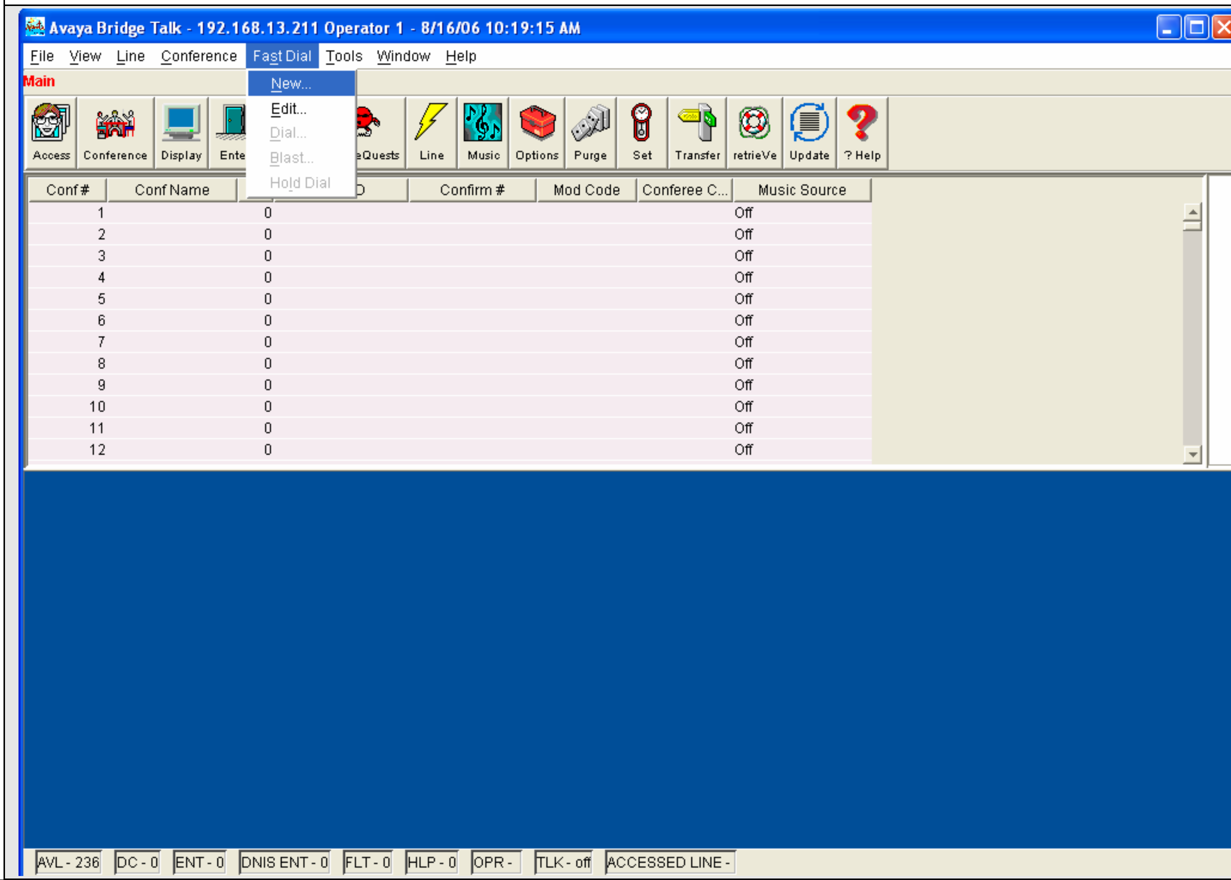
DNIS   Msg PS Function Line Name   Company Name
---------------- --- --- -------- -------------------- --------------------
???   208 1   ENTER</pre></td></tr>
<tr><td>4.7</td><td>At the command prompt enter cbutil add to add a DNIS entry for a scan function for DNIS 502.
<pre>S6200>cbutil add 502 1 1 scan
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.</pre></td></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>
</table>

Step	Description
4.9	• At the command prompt enter **cbutil list** to verify DNIS entries provisioned on Avaya Meeting exchange.

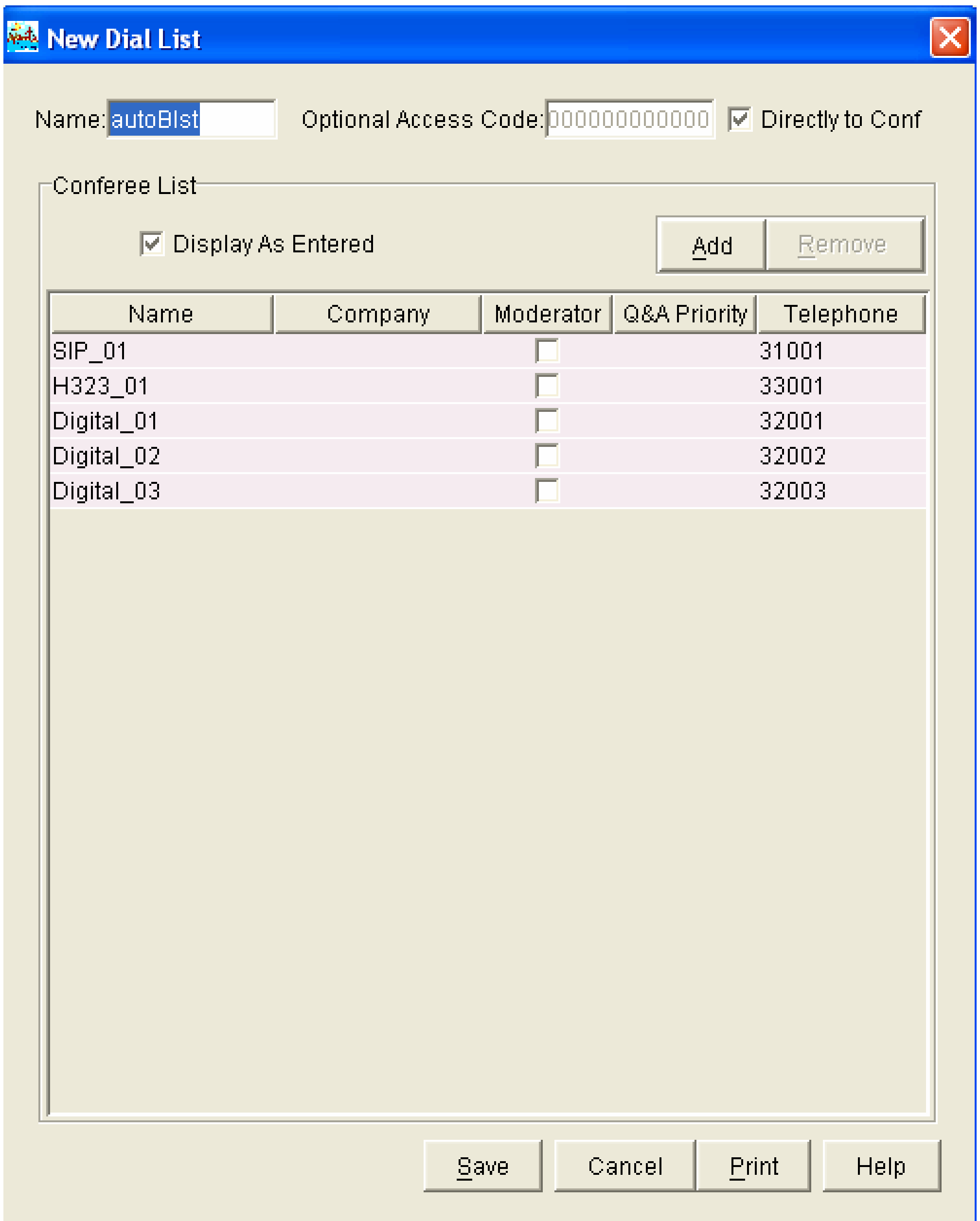
```
S6200>cbutil list
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

DNIS             Msg PS  Function Line Name            Company Name
---------------- --- --- -------- -------------------- --------------------
502              1   1   SCAN
555              0   1   DIRECT
556              0   1   DIRECT
???              208 1   ENTER
```

Step	Description
4.10	**Reboot** Avaya Meeting Exchange to make change take effect.

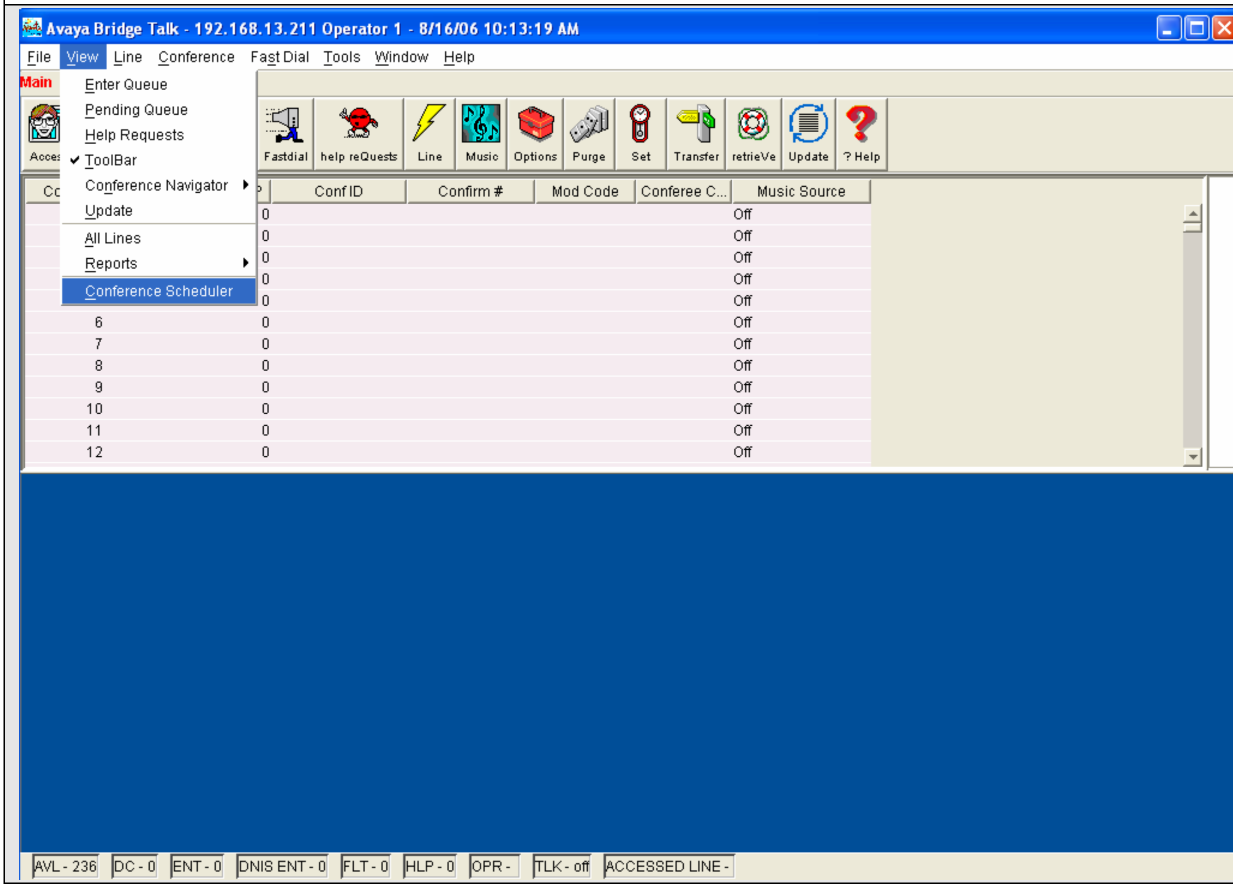
Step	Description
4.11	To provision conferences on Avaya Meeting Exchange: • Open the Avaya Bridge Talk Application and **Log in**. Note: • The following steps will detail how to provision an Auto Blast dial conference using DNIS entry **556** (see **Step 4.9**).
4.12	From the Avaya Bridge Talk **Menu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

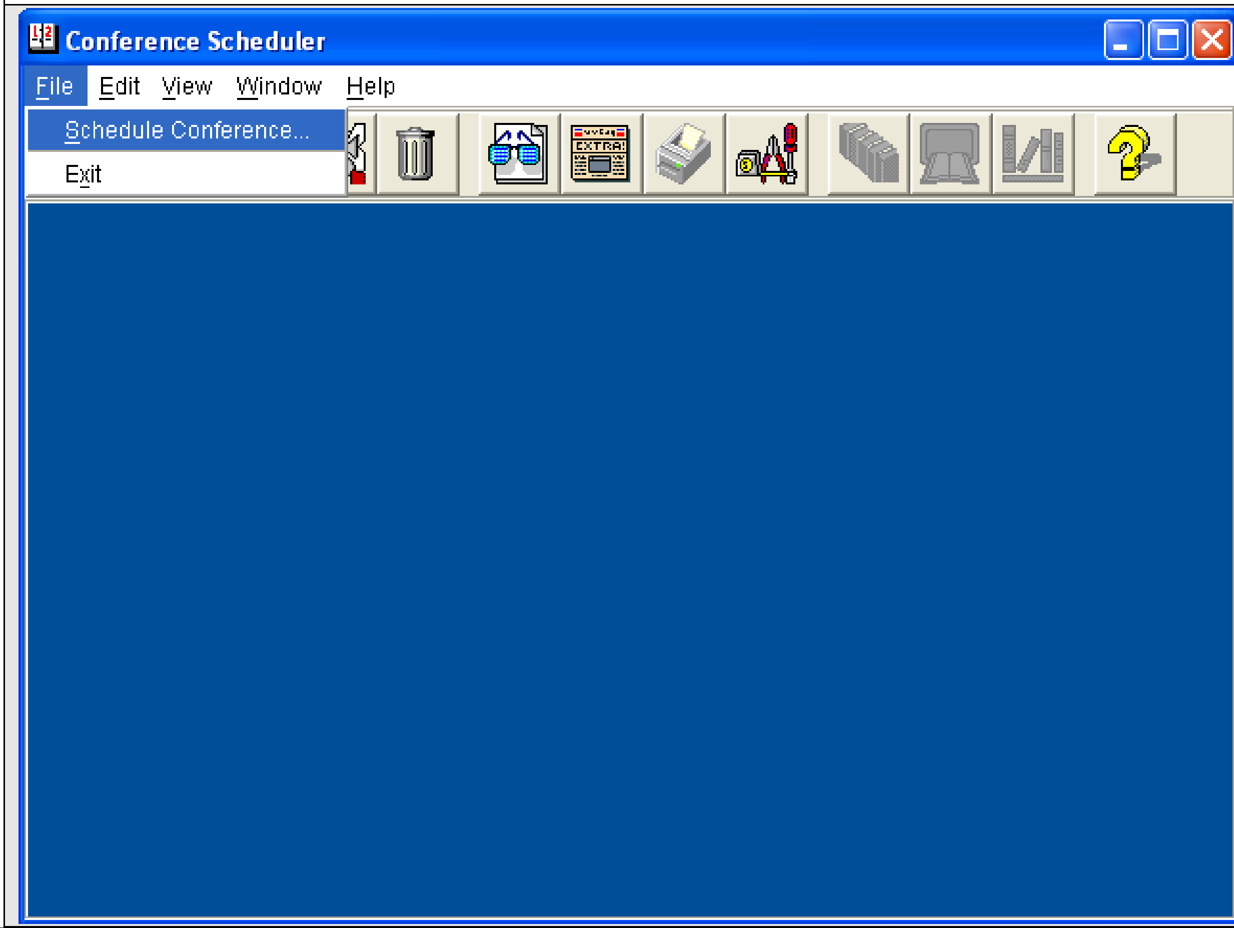


Step	Description
4.13	From the **New Dial List** window: • Check the **Directly to Conf** box to allow conferees to enter a conference without a passcode. • Add conferees to 'Blast Dial' by clicking the **Add** button for each entry. ○ Give moderator privileges to a conferee by checking the Moderator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

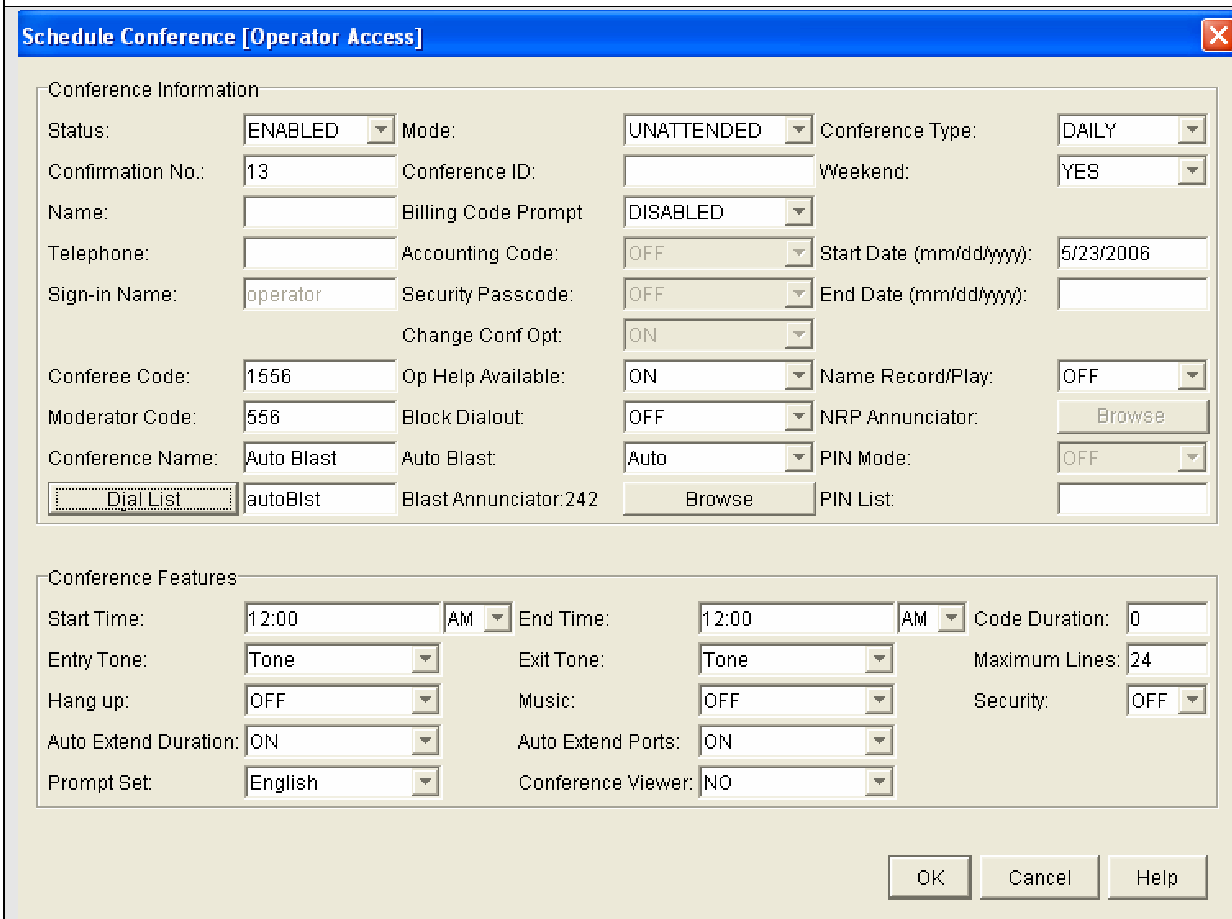


Step	Description
4.14	From the Avaya Bridge Talk **Menu Bar**: • Click **View**, **Conference Scheduler**.



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

Step	Description
4.16	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**).



5. 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>5.1</td><td>Verify all members for the SIP trunk group are in-service/idle.

From a SAT session:
Issue the command “status trunk 2”.
All members should return the value in-service/idle.</td></tr>
<tr><td>5.2</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
5.3	Configure a **STATION** Issue the command “**add station <extension>**”, and administer settings as per below. Note: • The station **Name** is a label field and is utilized by Avaya Meeting Exchange to display information regarding the caller on Avaya Bridge Talk (see **Step 5.6**). • Other stations utilized in these Verification Steps were configured in a similar fashion, and will not be depicted for these Application Notes.

```
Page 1 of 3
                                    STATION

Extension: 31002                          Lock Messages? n          BCC: 0
     Type: 4602+                          Security Code:             TN: 1
     Port: S00004                       Coverage Path 1:            COR: 1
     Name: SIP 31002                    Coverage Path 2:            COS: 1
                                        Hunt-to Station:

STATION OPTIONS
              Loss Group: 19            Personalized Ringing Pattern: 1
                                                    Message Lamp Ext: 31002
            Speakerphone: 1-way                  Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal                 Media Complex Ext:
   Survivable Trunk Dest? y                             IP SoftPhone? n
```

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.4</td><td>Verify the SIP 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 102”, where 102 is the TAC defined for the trunk group provisioned in Step 3.8.

From the SIP station configured in Step 5.3, dial 556 to initiate a DNIS direct with Auto Blast call scenario.</td></tr>
</table>

```
list trace tac 102                                                       Page   1

                                LIST TRACE

time            data

11:54:02     dial 556 route:AAR
11:54:02     term trunk-group 2    cid 0x259
11:54:02     dial 556 route:AAR
11:54:02     route-pattern  2 preference 1  cid 0x259
11:54:02     seize trunk-group 2 member 29  cid 0x259
11:54:02     Calling Number & Name 31002 SIP 31002
11:54:02     Proceed trunk-group 2 member 29  cid 0x259
11:54:02     active trunk-group 2 member 29  cid 0x259
11:54:02     G711MU ss:off ps:20 rn:2/1 192.168.13.211:42010 192.168.11.11:2276
11:54:02     xoip: fax:Relay modem:off tty:US 192.168.11.11:2276 uid:0x5004f
11:54:03     G711MU ss:off ps:20 rn:2/1 192.168.13.211:42010 192.168.12.11:34008
11:54:03     G711MU ss:off ps:20 rn:1/2 192.168.12.11:34008 192.168.13.211:42010
```

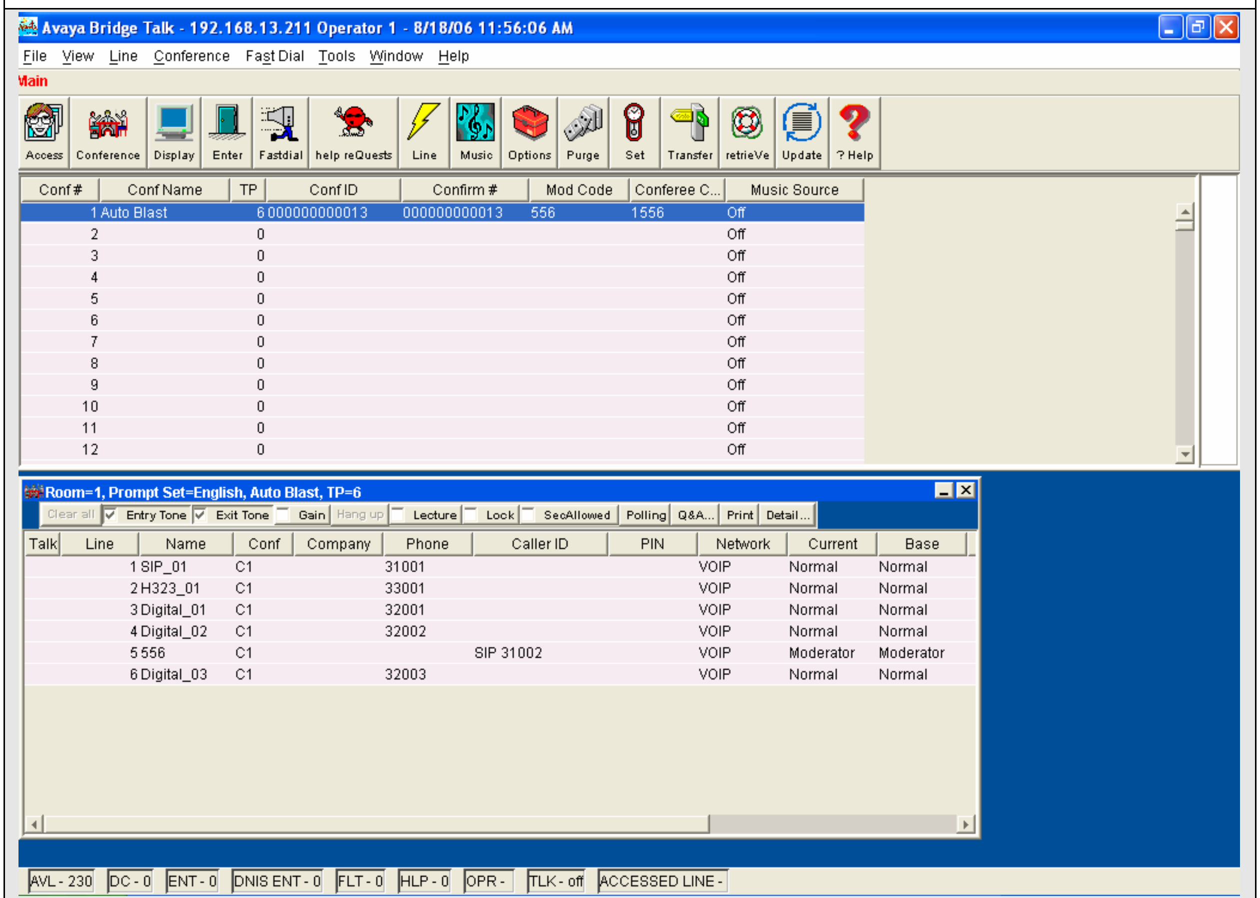
Step	Description
5.5	Verify shuffling for the SIP station Dialing-In to Avaya Meeting Exchange, where shuffling is defined as: • Rerouting the audio channel connecting two IP endpoints. After shuffling, the audio which previously was carried in a mixed connection of IP signaling and TDM bus signaling, goes directly through the LAN or WAN between the two IP endpoints. • Shuffling also can mean reversing this process if an endpoint requests a resource to support a feature, such as conferencing that requires the TDM bus. From a SAT session: • Issue the command "**status trunk 2/29** (where **2/29** is obtained from **Step 5.4**)". • The **Audio Connection Type = ip-direct** shows that shuffling is enabled for this endpoint.

```
status trunk 2/29                                               Page   1 of   2
                              TRUNK STATUS

 Trunk Group/Member: 0002/029             Service State: in-service/active
               Port: T00079            Maintenance Busy? no
 Signaling Group ID:



    Connected Ports: T00032




                    Port     Near-end IP Addr : Port    Far-end IP Addr : Port
        Signaling: 01A0217  192.168. 11. 10  : 5061    192.168. 13.211 : 5061

G.711MU     Audio:          192.168. 12. 11  : 34008   192.168. 13.211 : 42010
            Video:
      Video Codec:
                                            Authentication Type: None
    Audio Connection Type: ip-direct
```

Note:

- An **Audio Connection Type = ip-tdm** would indicate that shuffling is not enabled for an endpoint.
- The following results for shuffling were obtained by utilizing the **status trunk** command.
  - Shuffling is enabled for SIP stations for both Dial-In and Dial-Out.
  - Shuffling is not enabled for H.323 stations.

Step	Description
5.6	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 5.4**.



6. Conclusion

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

- Dial-In to Avaya Meeting Exchange from Avaya Communication Manager via SIP utilizing TLS.
- Dial-Out from Avaya Meeting Exchange to Avaya Communication Manager via SIP utilizing TLS.

7. 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

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