Avaya Solution & Interoperability Test Lab

Configuring SIP Trunking between AT&T IP Flexible Reach and IP Toll Free Services with the Avaya Meeting Exchange S6200 Conferencing Server via Avaya SIP Enablement Services - Issue 1.0

Abstract

These Application Notes present the procedures for configuring SIP Trunking connectivity between the AT&T IP Flexible Reach and IP Toll Free services with the Avaya Meeting Exchange S6200 Conferencing Server via Avaya SIP Enablement Services.

AT&T IP Flexible Reach and IP Toll Free are managed Voice over IP communication solutions using SIP trunks to provide inbound and outbound local, long distance, international and toll free services for U.S. sites.

AT&T is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab in conjunction with remote access to the AT&T Virtual Interoperability Test lab.

1. Introduction

These Application Notes present the procedures for configuring SIP trunking connectivity between the AT&T IP Flexible Reach and IP Toll Free services and the Avaya Meeting Exchange S6200 Conferencing Server via Avaya SIP Enablement Services.

AT&T IP Flexible Reach and IP Toll Free services (AT&T Services) are managed Voice over IP communication solutions using SIP trunks to provide inbound and outbound local, long distance, international and toll free services for U.S. sites.

Avaya Meeting Exchange is an advanced conferencing solution offering reservation-less and scheduled meet-me voice conferencing capabilities using SIP trunking.

The following conferencing features have been verified:

- Dial-In Conferencing:
  - **Dialed Number Identification Service (**DNIS) direct call function, where conference participants enter a conference as moderator without entering a participant access code (passcode).
  - Scan call function, where conference participants enter a conference with a valid passcode.
- Dial-Out Conferencing from Avaya Meeting Exchange:
  - Blast dial
    - Auto, where a conference participant enters a conference via a DNIS direct call function and automatically invokes a Blast dial to a pre-provisioned dial list of one or more participants.
    - Manual, where a conference participant is already in a conference as a moderator and invokes a Blast dial to a pre-provisioned dial list of one or more participants.
  - Originator Dial-Out, where a conference participant is already in a conference as a moderator and invokes a Dial-Out to a single participant.
  - Operator Fast Dial, where an operator can Dial-Out to a pre-provisioned dial list of one or more participants.
- Operator Dial-Out to set up an Audio Path.
- Operator Dial-In to set up an Audio Path.
- Dial-Out for audio recording.
- Line Transfer initiated from Avaya Bridge Talk.
- Conference Transfer initiated from Avaya Bridge Talk.

The following codecs were verified:

- G.711mu
  (Note: The Avaya S6200 Meeting Exchange only supports the G.711mu and G.711a codecs; G.729 is not available.)

Fax is not supported by the Avaya S6200 Meeting Exchange and is not applicable to these tests.

These Application Notes provide the administrative steps for configuring the application as shown in **Figure 1**.

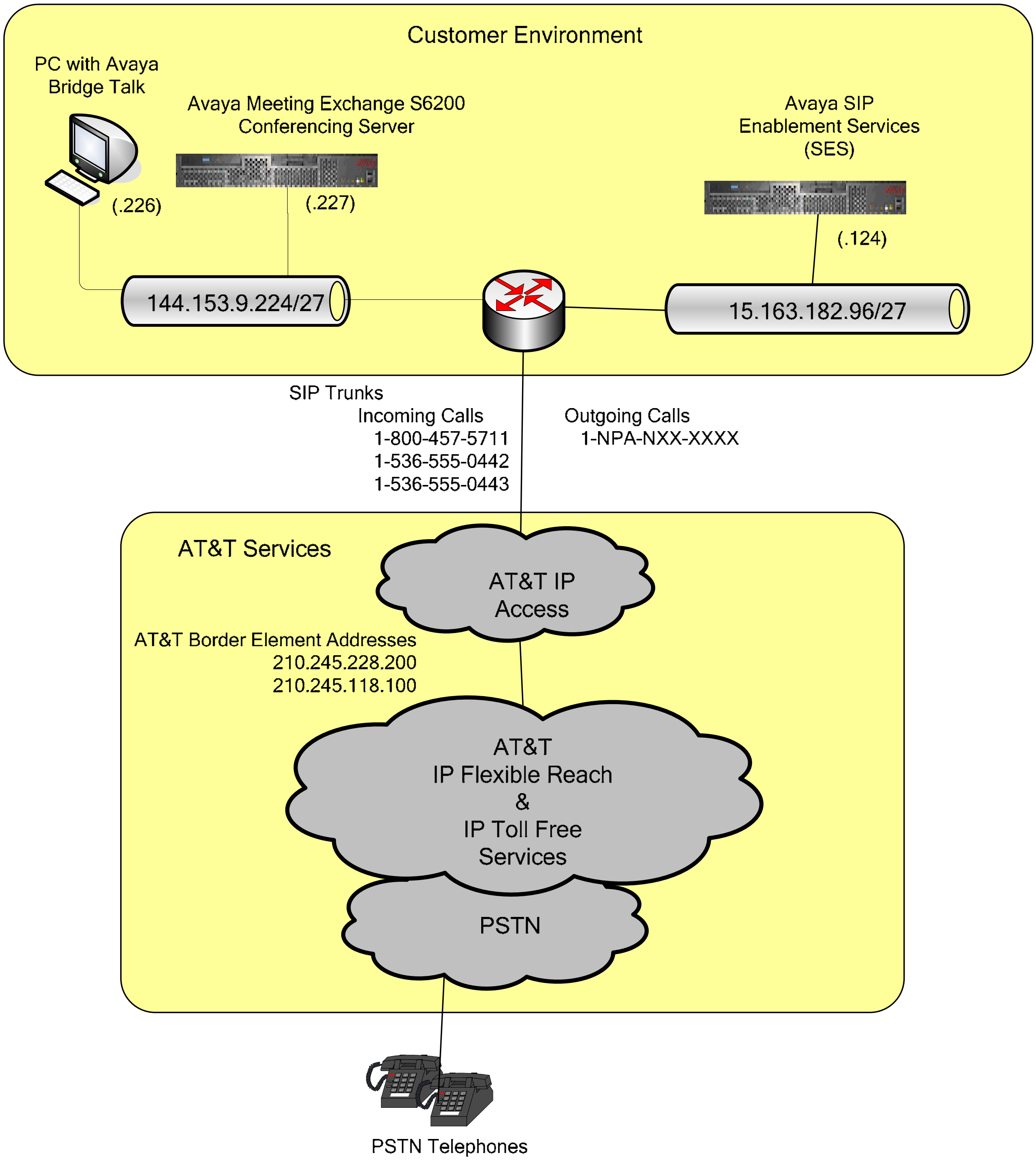


Figure 1: Network Configuration

1.1. AT&T Services Configuration Information

These Application Notes provide **an illustrative example** of how the Avaya S6200 Meeting Exchange is configured with the AT&T IP Flexible Reach and IP Toll Free services.

The specific values provided below are illustrative only and must not be used for customer configurations. *Each customer must obtain the specific values for their configuration from AT&T during service provisioning of their AT&T IP Flexible Reach or IP Toll Free services.*

AT&T Provisioning Information	**Illustrative Values in these Application Notes**
AT&T Border Element Address(es)	210.245.228.200 210.245.118.100
G.711MU Codec Support	Yes
RFC 2833 (DTMF Event) Supported	Yes
Assigned Direct Inward Dial (DID) Numbers	1-536-555-0442 1-536-555-0443
DID Digits Passed in SIP Request URI	5365550442 5365550443
DID Digits Passed in SIP To Header	Same as SIP Request URI
Incoming Toll Free Number	800-457-5711
Incoming Toll Free Digits Passed in SIP Request URI	0000000010
Incoming Toll Free Digits Passed in SIP To Header	0004152100010

2. Equipment and Software Validated

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

Equipment	Software
Avaya Meeting Exchange S6200 Conferencing Server	Release 5.0
Avaya Meeting Exchange Bridge Talk	Release 5.0
Avaya SIP Enablement Services	SES-4.0.0.0-033.6
AT&T IP Flexible Reach and IP Toll Free Services	VNI-9

Table 1 - Hardware and Software Versions

3. Avaya Meeting Exchange Configuration

This section describes the steps for configuring Avaya Meeting Exchange to interoperate with Avaya SIP Enablement Services via secure SIP connectivity utilizing Transport Layer Security (TLS).

Step	Description
1	Log in to the Avaya Meeting Exchange Server console with the appropriate credentials.

2	Configure settings that enable secure SIP connectivity between Avaya Meeting Exchange and other SIP User Agents 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.,
	  o **IPAddress=144.153.9.227**
	• Add a line to populate the From header field in SIP INVITE messages from Avaya Meeting Exchange, e.g.,
	  MyListener=sip:6000@144.153.9.227
	  *The string "**6000**" is arbitrarily chosen*.
	• Add a line to provide User Agents a Contact address to use for acknowledging SIP messages from Avaya Meeting Exchange, e.g.,
	  o **respContact=<sip:6000@144.153.9.227:5061;transport=tls>**
	  ***Note**: The string "**6000**" is arbitrarily chosen*.

Step	Description

3	To associate incoming calls to Avaya Meeting Exchange with different call handling flows, edit the **UriToTelnum.tab** file to extract the Direct Inward Dial / Dialed Number Identification Service (DID/DNIS) and Automatic Number Identification (ANI) values as follows:
	• cd to **/usr/ipcb/config**.
	• Edit the **UriToTelnum.tab** file with a text editor, e.g., vi.
	    o Add a line to match the regular expression pattern of the To and From headers in SIP INVITE messages from the AT&T services. In these Application Notes this line is:
	
	`"<sip:*@*"  $1`
	
	If a match occurs, the $1 variable will contain the DID/DNIS address digits extracted from the To header and the ANI extracted from the From header.
	
	For example, "**5365550442**" is the DID/DNIS value derived from the following To header.
	
	`To: <sip:5365550442@15.163.182.124;user=phone>`
	
	and "**+17358551637**" is the ANI value derived from the following From header.
	
	`From: "John" <sip:+17358551637@210.245.228.200:5060;user=phone>;`
	
	• Enable an undefined caller to receive a prompt for operator assistance by administering for the condition of an unmatched SIP INVITE message by adding a wildcard entry as the last line in this file.
	    o This line is:
	
	`*  $0`
	
	Note*: Entries in this file are read sequentially and the first match used; therefore, the undefined caller line (e.g.,* **`*   $0`** *) must be the last line in the file. Otherwise, all calls to Avaya Meeting Exchange would match the wildcard and thus receive a prompt for operator assistance*.
	
	Specific guidelines for the configuration of this table are discussed in Chapter 3 of Reference [1].

Step	Description
4	To enable Dial-Out from Avaya Meeting Exchange using SIP trunking to the AT&T services via the Avaya SES, 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 route all outbound calls from Avaya Meeting Exchange to Avaya SIP Enablement Services, e.g., *** sip:$0@15.163.182.124:5061;transport=tls default_gateway** In this example: ▪ the pattern “*” is a wild card that matches any dialed digits, ▪ the string “sip:$0@15.163.182.124:5061;transport=tls” is the URI that will be sent to the Avaya SES (at 15.163.182.124) using the tls transport protocol on port 5061. Avaya Meeting Exchange will replace “$0” with the actual dialed digits. ▪ the string “default_gateway” is a comment describing the purpose of the line.

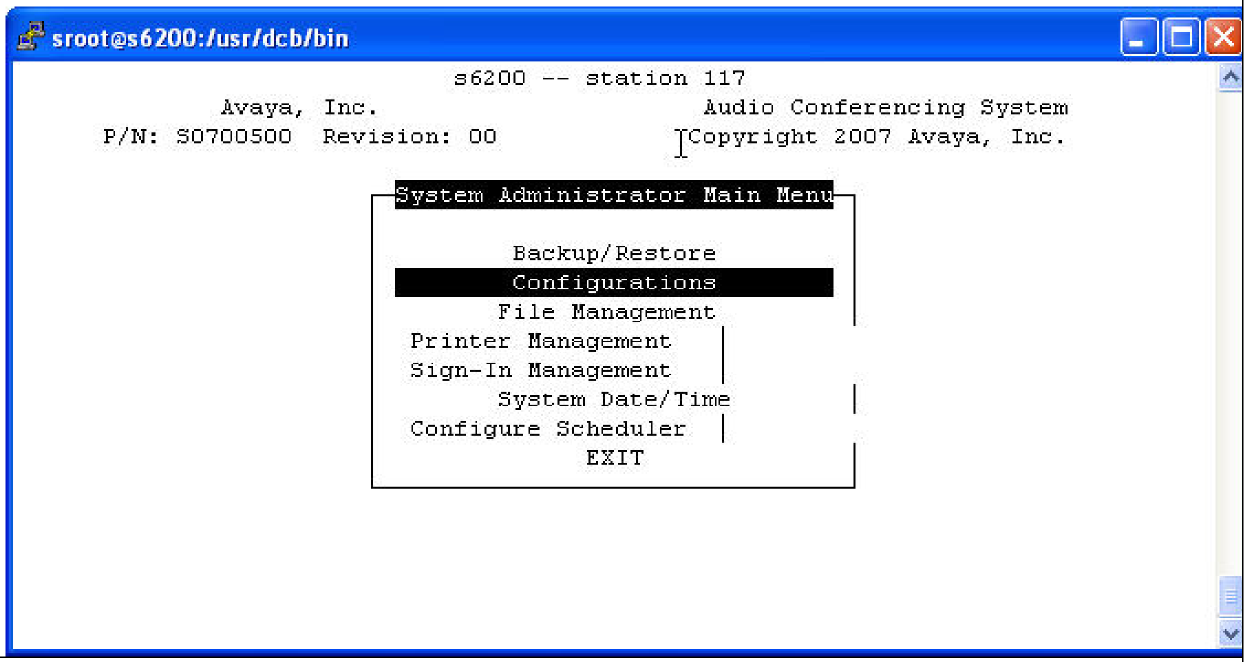
3.1. Call Routing Configuration

The following steps configure the Avaya Meeting Exchange to handle the expected length of the DID/DNIS digits received from AT&T on incoming calls. This prepares the CBUTIL utility (covered in the next section) to perform the appropriate searches in the CBUTIL call branding tables.

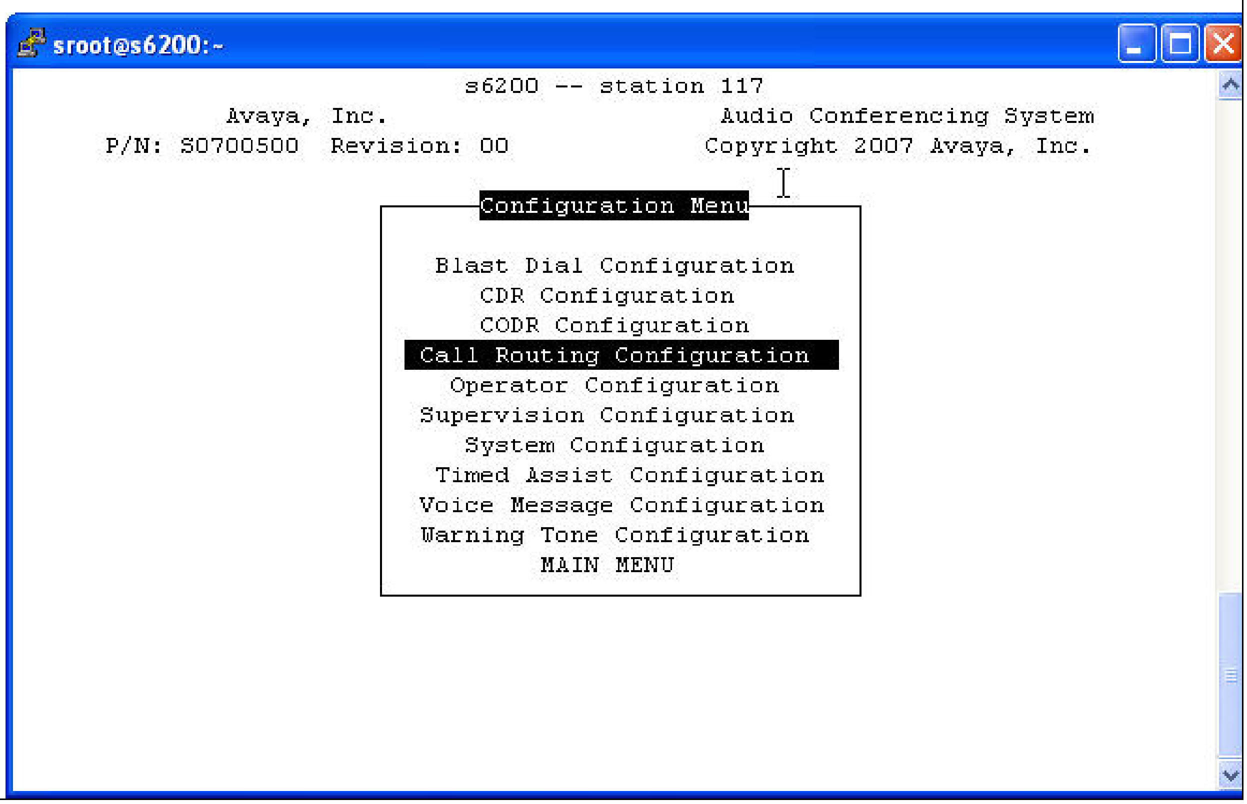
Call Routing Configuration is performed using the System Management Interface.

Step	Description
5	To configure Call Routing Configuration using the System Management Interface: • Log into the Avaya Meeting Exchange Server console with the appropriate credentials. • At the command prompt, enter “**dcbadmin 116”** to invoke the System Management Interface. [s6200]# dcbadmin 116 The System Administrator Main Menu page should appear.

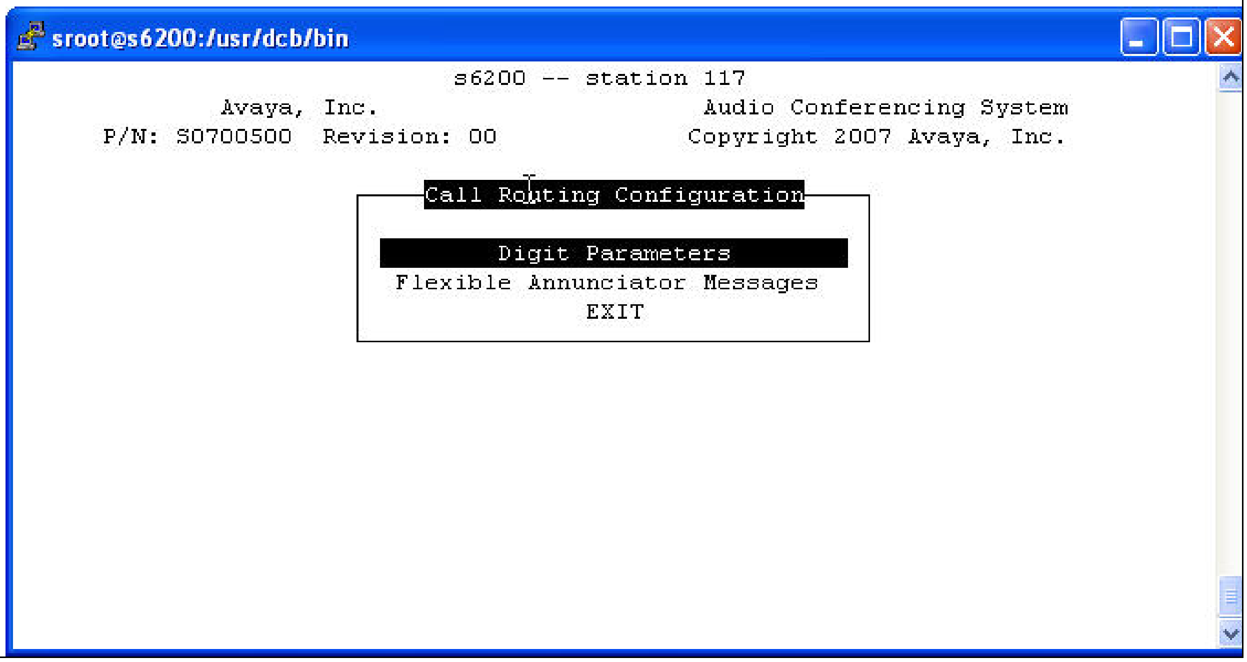
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6</td><td>Select the Configurations menu item and press Enter.



</td></tr>
<tr><td>7</td><td>Select Call Routing Configuration and press Enter.



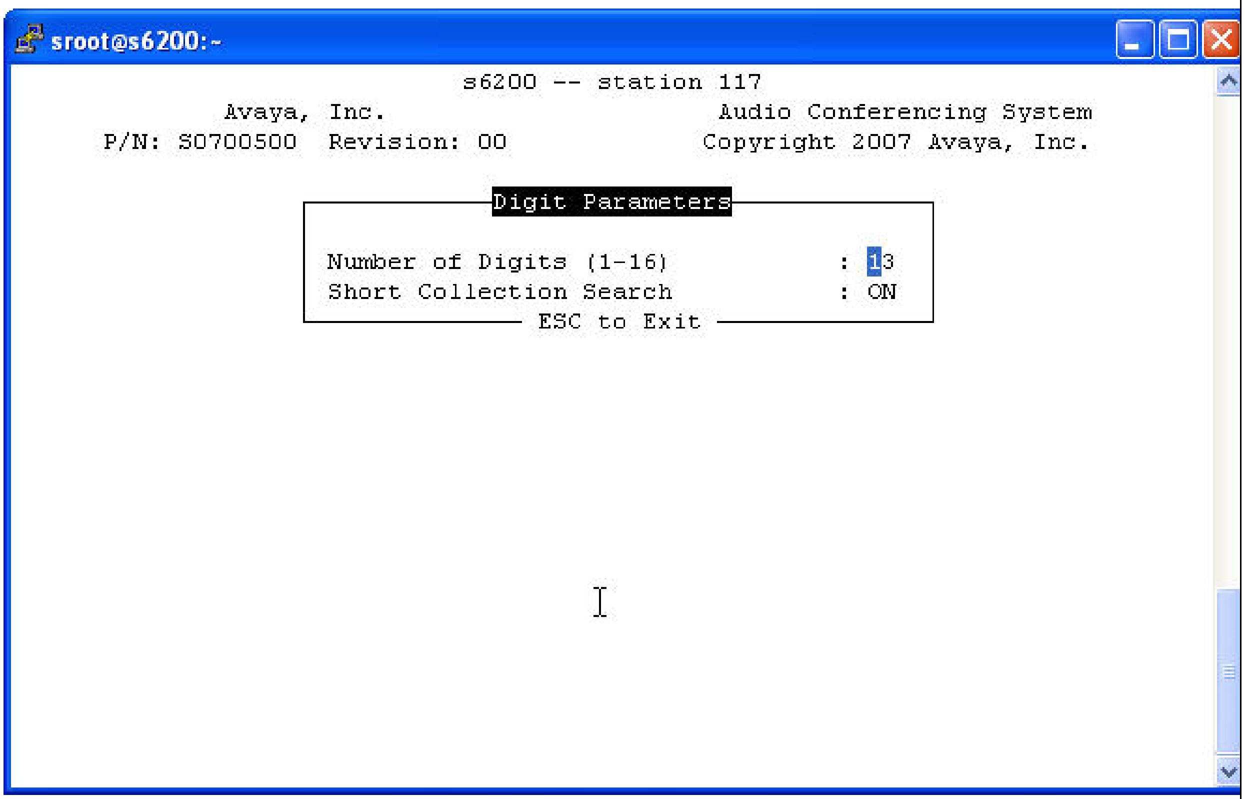
</td></tr>
</table>

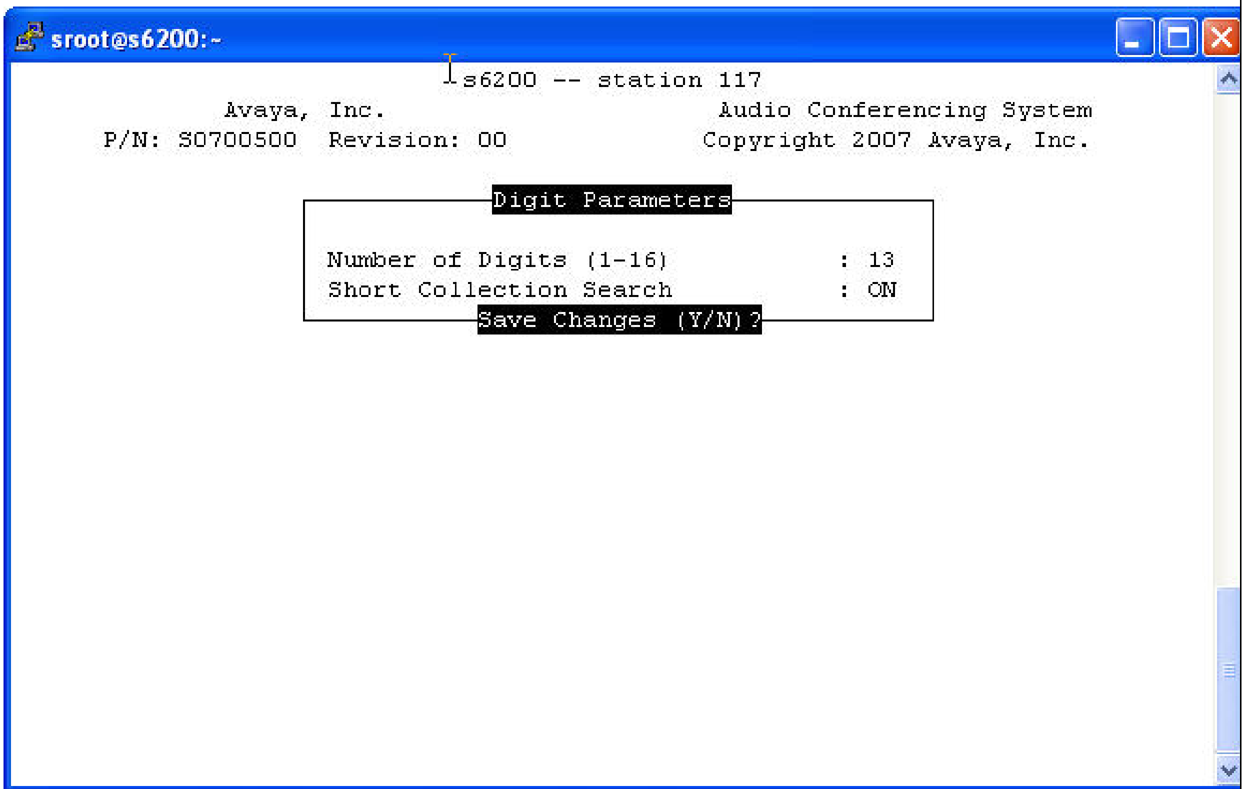
Step	Description
8	Select **Digit Parameters** and press Enter. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>9</td><td>Set the Number of Digits to the maximum length incoming digit string (DNIS) expected.

In these Application Notes the AT&T IP Toll Free service sends 13 digits and the AT&T IP Flexible Reach service sends 10 digits. Thus, set the Number of Digits value to “13”.

Since the Number of Digits varies for the two AT&T services, set the Short Collection Search value to “ON”. This instructs the Avaya Meeting Exchange to attempt a partial match (in right to left order) in the call branding table when fewer then 13 DNIS digits are received.


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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>10</td><td>Press “ESC” and “Y” to save the Digit Parameters changes.

</td></tr>
<tr><td>11</td><td>Reboot Avaya Meeting Exchange for changes to take effect.

Note: Rebooting Avaya Meeting Exchange is service impacting.

<code>[S6200]> init 6</code></td></tr>
</table>

3.2. CBUTIL Utility

The CBUTIL utility enables specific annunciator messages, line name, company name and routing function to be assigned to each DID/DNIS patterns. These assignments are stored in the call branding table. The DID/DNIS values are obtained from the To Header of the SIP INVITE messages according to the rules specified in Step 3. *Note that the values in the To header may not match the digits found in the SIP Request URI.*

The routing functions used in these Application Notes are:

- ENTER – places the incoming call matching the corresponding DID/DNIS pattern into an Avaya Meeting Exchange ENTER queue for handling by an operator. The operator will screen the call and place the caller into the proper conference using the BridgeTalk application.

- DIRECT – places the incoming calls matching the DID/DNIS pattern directly into an assigned conference without operator screening or caller entered access codes.
- SCAN – prompts caller to enter a conference access code before for placing them into the conference matching the DID/DNIS patterns. Failed attempts are routed to the ENTER queue for operator handling.

In these Application Notes the AT&T Services DID/DNIS numbers are assigned the Avaya Meeting Exchange routing function as shown in **Table 2**.

Dialed PSTN Number	**Digits Received in the SIP Request URI**	**DID/DNIS Digits Received in SIP To Header**	**Avaya Meeting Exchange Assigned Routing Function**
1-800-457-5711	0000000010	0004152100010	SCAN
1-536-555-0442	5365550442	5365550442	SCAN
1-536-555-0443	5365550443	5365550443	DIRECT
Other		Other Unrecognized	ENTER

Table 2 - Call Branding Routing Function Assignments

Step	**Description**
12	To provide the call branding treatment defined in **Table 2** using the DID/DNIS values obtained by the rule defined in Step 3, run the **cbutil** utility as follows: • Log in to the Avaya Meeting Exchange Server console with the appropriate credentials. • At the command prompt, run the **cbutil** utility to administer DNIS entries provisioned on Avaya Meeting Exchange. Note that entering **cbutil** without an additional command argument displays the cbutil help. Entering **cbutil list** will list all existing entries in the call branding table.

```
[craft@s6200 ~]$ cbutil
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

Usage:  <command> [command-specific args...]
 where <command> may be one of:
  add           Add an entry to the Call Branding table
  remove        Remove an entry from the Call Branding table
  update        Update an entry in the Call Branding table
  lookup        Display an entry in the Call Branding table
  count         Display the number of entries in the Call Branding table
  list          List entries in the Call Branding table
  dnissize      Set system configured max dnis length (1-16)
   Note:  This command should only be used when the bridge is not running.
Use "<command> -help" to get help on a specific command

[craft@s6200 ~]$
```

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>13</td><td>
Enable dial-in access (via passcode) to conferences using the AT&T Toll Free DID/DNIS value “0004152100010” with the following command. These conferences use the SCAN call routing function.

cbutil add <dnis> <rg> <msg> <ps> <ucps> <func> [-l <ln> -c <cn>], where the variables for add command are defined as follows:

<dnis> DNIS
<rg> Reservation Group
<msg> Annunciator message number
<ps> Prompt Set number (0-20)
<ucps> Use Conference Prompt Set (y/n)
<func> One of: DIRECT/SCAN/ENTER/HANGUP/AUTOVL/FLEX
-l <"ln"> Optional line name to associate with caller
-c <"cn"> Optional company name to associate with caller

<pre>
[craft@s6200 ~]$ cbutil add 0004152100010 0 1 1 n scan
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

[craft@s6200 ~]$
</pre>
</td></tr>
<tr><td>14</td><td>
In a similar manner, enable dial-in access (via passcode) to conferences using the AT&T IP Flexible Reach “5365550442” DID/DNIS value. These conferences use the SCAN call routing function.

Note that the <dnis> value must be padded using the “?” wild card character to the full 13 digit length previously set in Section 3.1.
<pre>
[craft@s6200 ~]$ cbutil add ???5365550442 0 1 1 n scan
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

[craft@s6200 ~]$
</pre>
</td></tr>
<tr><td>15</td><td>
Enable dial-in access (without entering a passcode) using the AT&T IP Flexible Reach “5365550443” DID/DNIS value. These conferences use the DIRECT call routing function.

Note that the <dnis> value must be padded using the “?” wild card character to the full 13 digit length previously set in Section 3.1.
<pre>
[craft@s6200 ~]$ cbutil add ???5365550443 0 301 1 n direct
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

[craft@s6200 ~]$
</pre>
</td></tr>
</table>

Step	Description
16	At the command prompt, enter **cbutil list** to verify the DNIS entries were administered correctly. ***Note****: The last entry in the call brand table is the wild card entry “****???****”. This entry captures any wrong number (e.g., unmatched* ***DID/DNIS*** *values) and places the call into the Enter queue for operator assistance.*

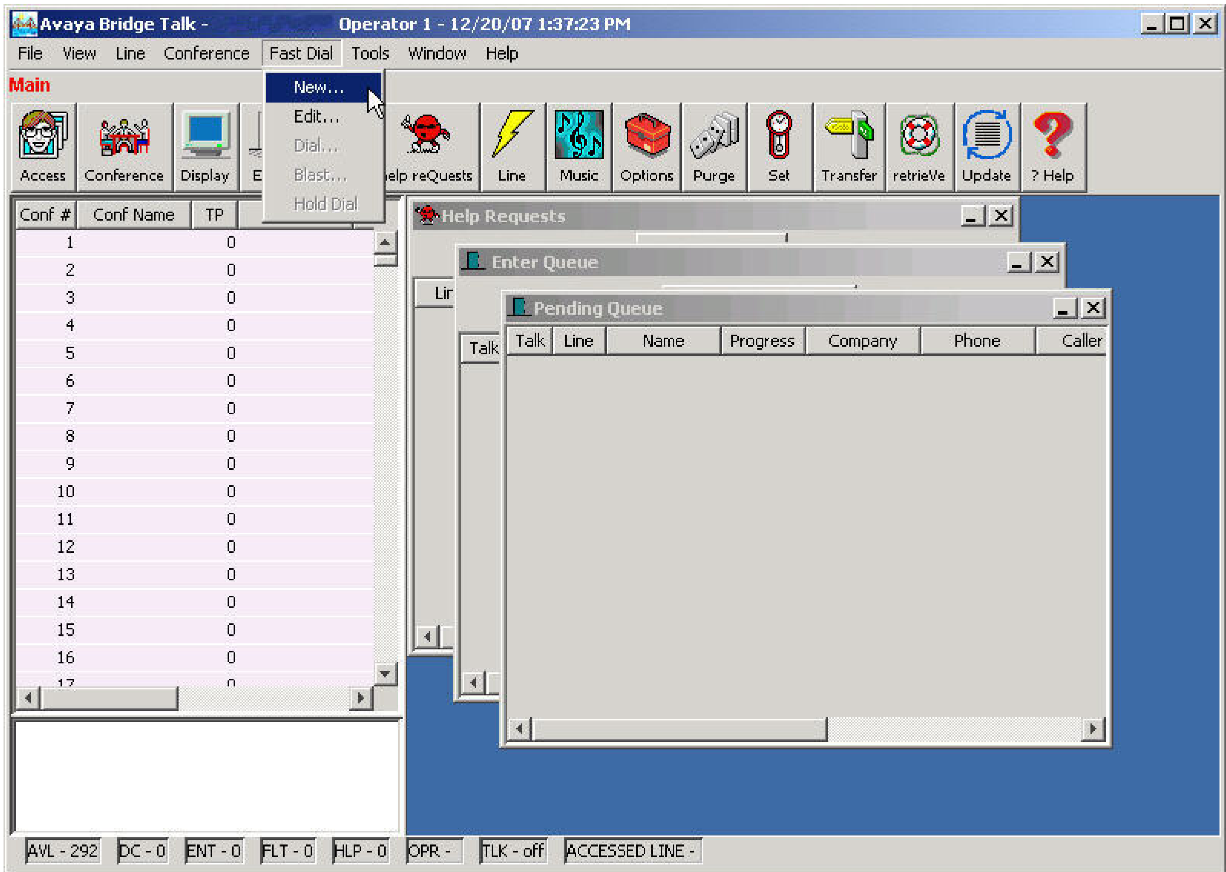
```
[craft@s6200 ~]$ cbutil list
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

DNIS             Grp Msg PS  CP Function Line Name            Company Name
---------------- --- --- --- -- -------- -------------------- -----------------
0004152100010    0   1   1   N  SCAN
???5365550442    0   1   1   N  SCAN
???5365550443    0   301 1   N  DIRECT
?????????????    0   208 1   N  ENTER
[craft@s6200 ~]$
```

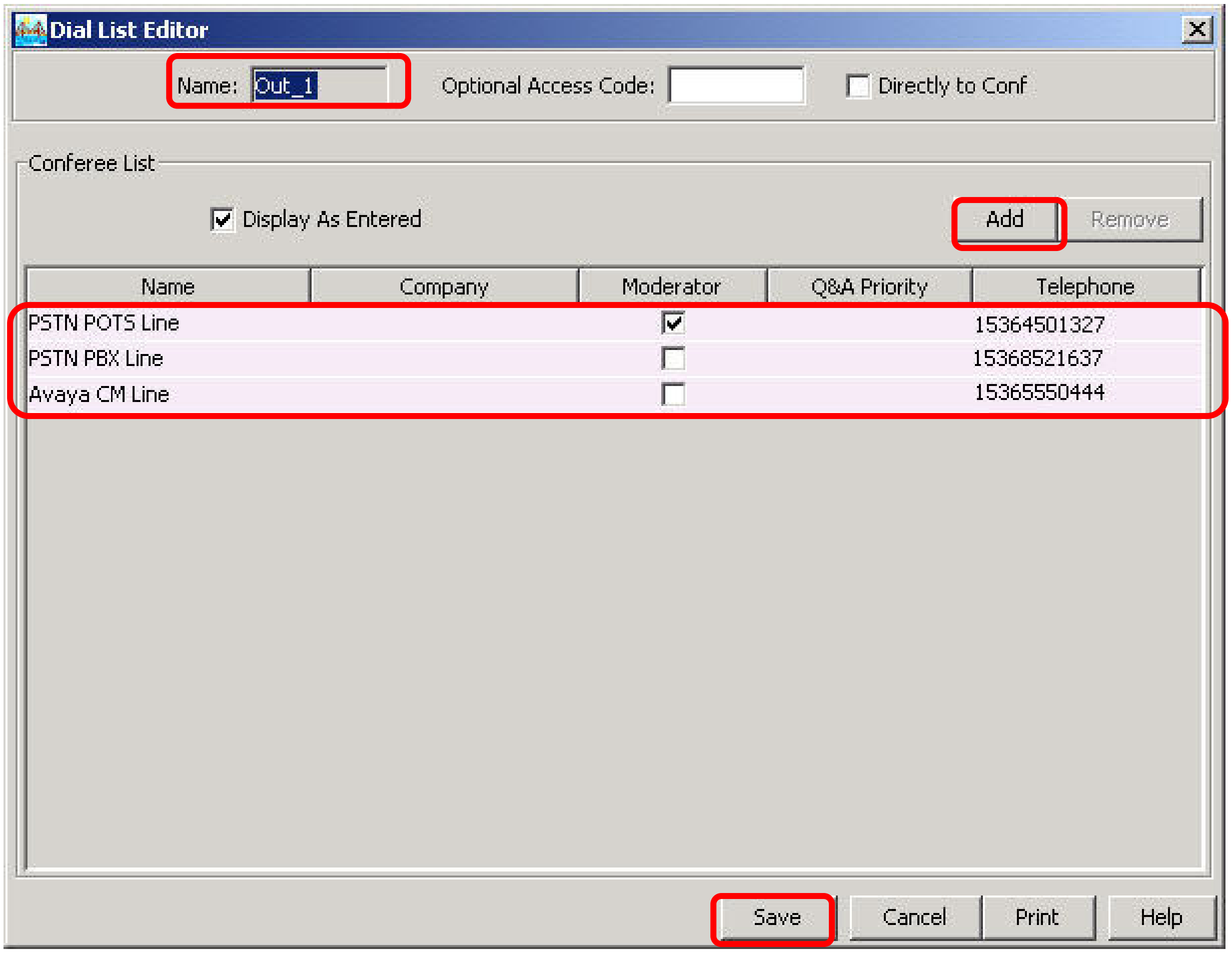
3.3. Bridge Talk

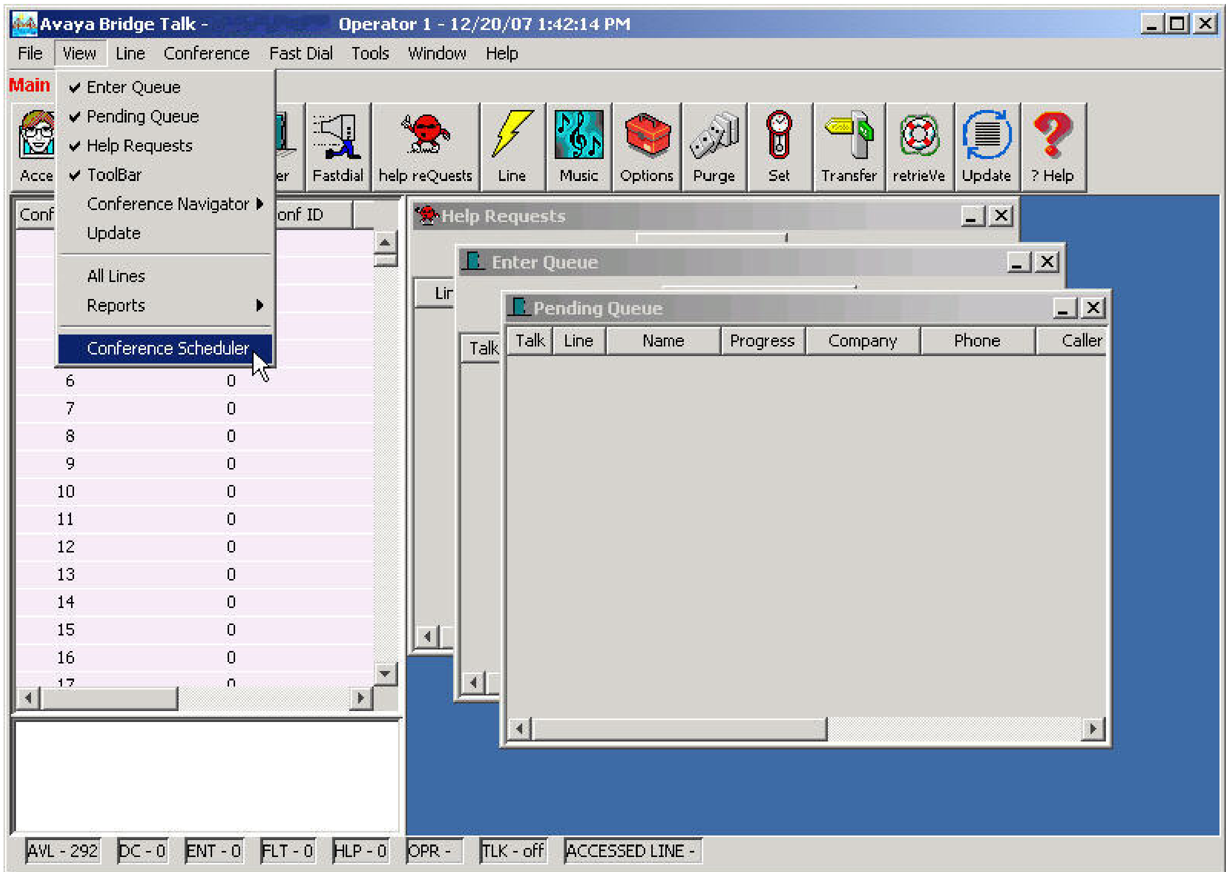
The following steps provide an example of how to provision a conference on Avaya Meeting Exchange from the Avaya Bridge Talk application. This sample conference is utilized in conjunction with the Direct and Scan call functions (provisioned in the previous section) to enable both Dial-In and Dial-Out access to audio conferencing for endpoints associated with Avaya Communication Manager.

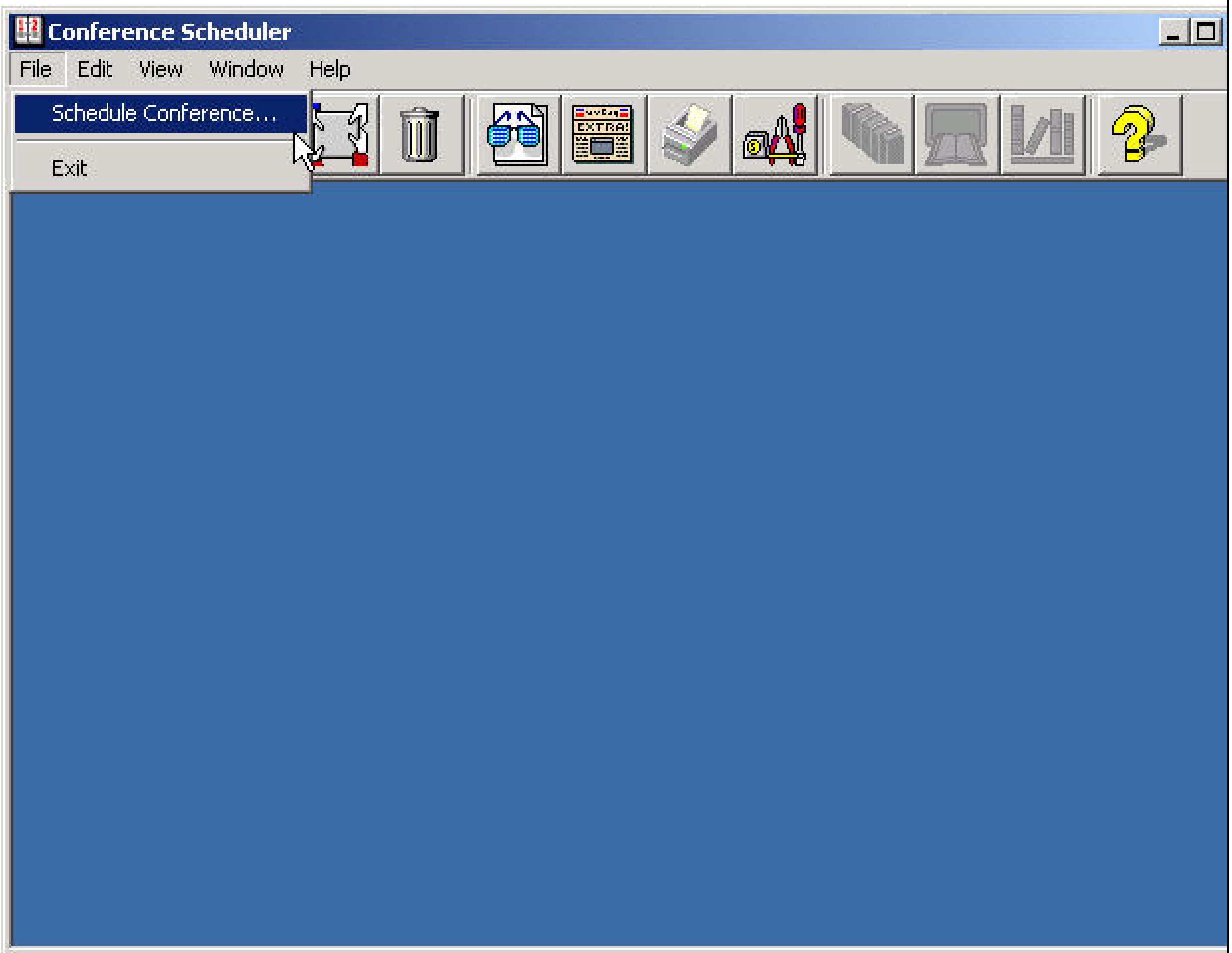
Note: *If any of the features shown in the following Avaya Bridge Talk screen captures are not present, contact an authorized Avaya sales representative to make the appropriate changes.*

Step	Description
17	Open the Avaya Bridge Talk application and log in to Avaya Meeting Exchange with the appropriate credentials. 

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

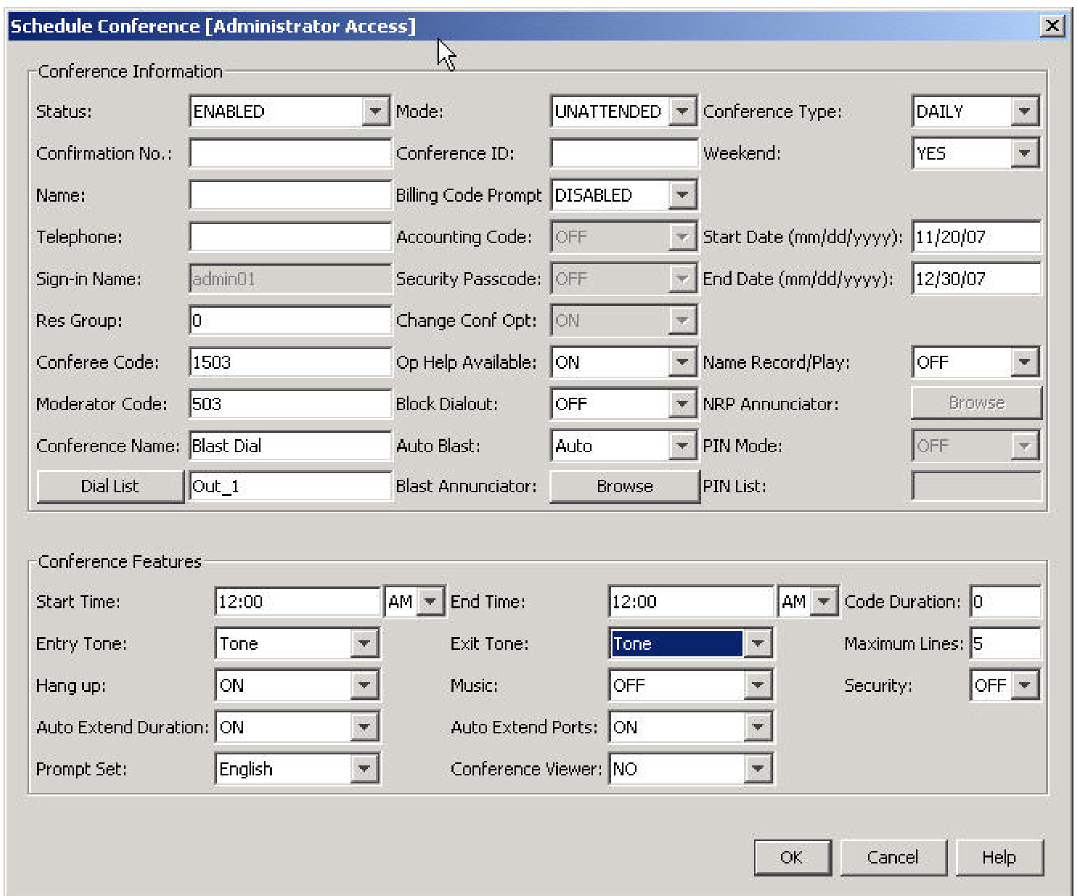
Step	Description
19	From the **New Dial List** window that is displayed: • Enter a descriptive name for the **Name** field. • Add entries to the dial list by clicking the **Add** button for each entry. o Assign moderator privileges to a conference participant by checking the **Moderator** box. • See **Reference 3** in **Section 8** for provisioning of the remaining entries in this screen. • When finished, click the **Save** button on the bottom of the screen. 

Step	Description
20	Provision a conference with Auto Blast enabled. From the Avaya Bridge Talk menu bar, click **View ➔ Conference Scheduler**. 

Step	Description
21	From the **Conference Scheduler** window that is displayed, click **File → Schedule Conference**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>22</td><td>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.
Enter a descriptive name for the Conference Name field.
Administer settings to enable an Auto Blast dial by setting Auto Blast to Auto and selecting the dial list provisioned in Step 19.
[Not Shown] Select a dial list by clicking the Dial List button, then selecting a dial list from the Create, Select or Edit Dial List window that is displayed and clicking the Select button.
See Reference 3 in Section 8 for provisioning of the remaining entries in this screen.
When finished, click the OK button on the bottom of the screen.



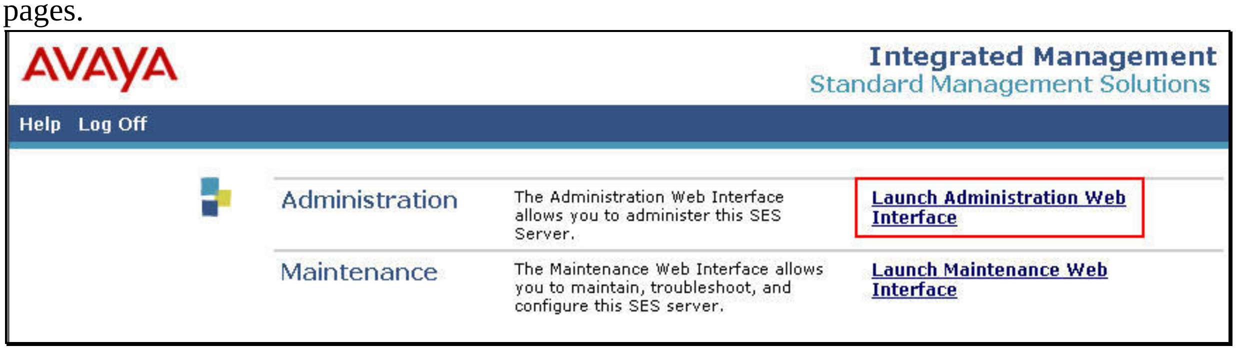
</td></tr>
</table>

4. Avaya SIP Enablement Services Configuration

This section describes the steps for configuring Avaya SIP Enablement Services to serve as a SIP proxy between the Avaya Meeting Exchange and the AT&T SIP trunking services.

4.1. Initial Avaya SES Setup

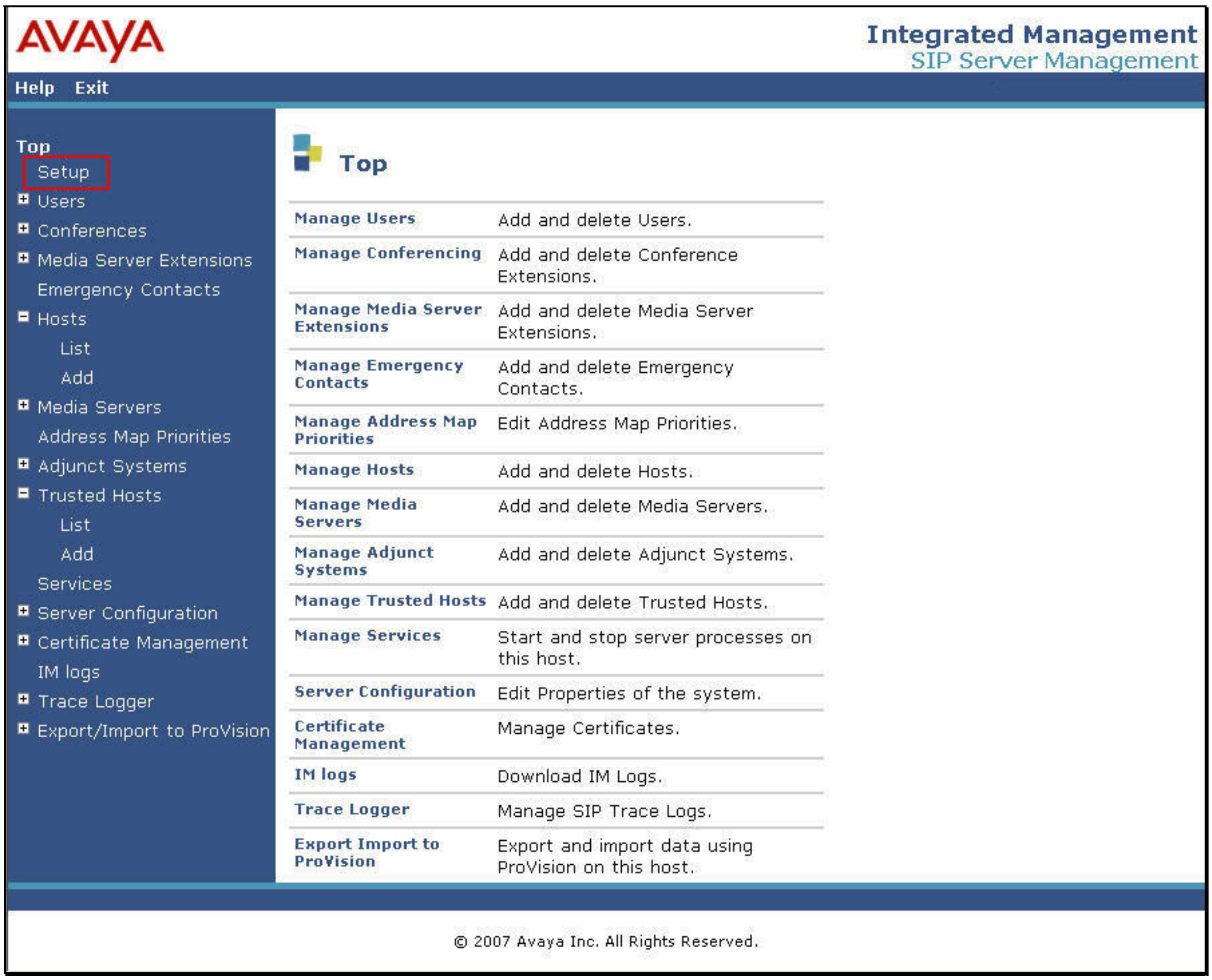
These Application Notes assume the Avaya SES is configured immediately following the initial installation (using the “initial_setup” script). The SIP Server Management page “setup” steps may vary slightly if the Avaya SES has been previously administered. If so, the left hand navigation menu of the SIP Server Management pages may be used to locate the referenced pages as appropriate.

Step	Description
1	Administer settings for Avaya SIP Enablement Services as follows: • Open a web browser and enter the following URL: **https://<IP address of Avaya SIP Enablement Services>/admin** • Log in to Avaya SIP Enablement Services with the appropriate credentials. 
2	Click the **Launch Administration Web Interface** link to enter the SIP Server Management pages. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3</td><td>
From the main SIP Server Management page:

- Click on the **Setup** link to start the initial configuration.

  Note: These Application Notes assume the Avaya SES is being configured immediately following installation. In this case, the **Setup** link will be present. If not, in the following steps access the corresponding page directly using the left hand navigation menus.

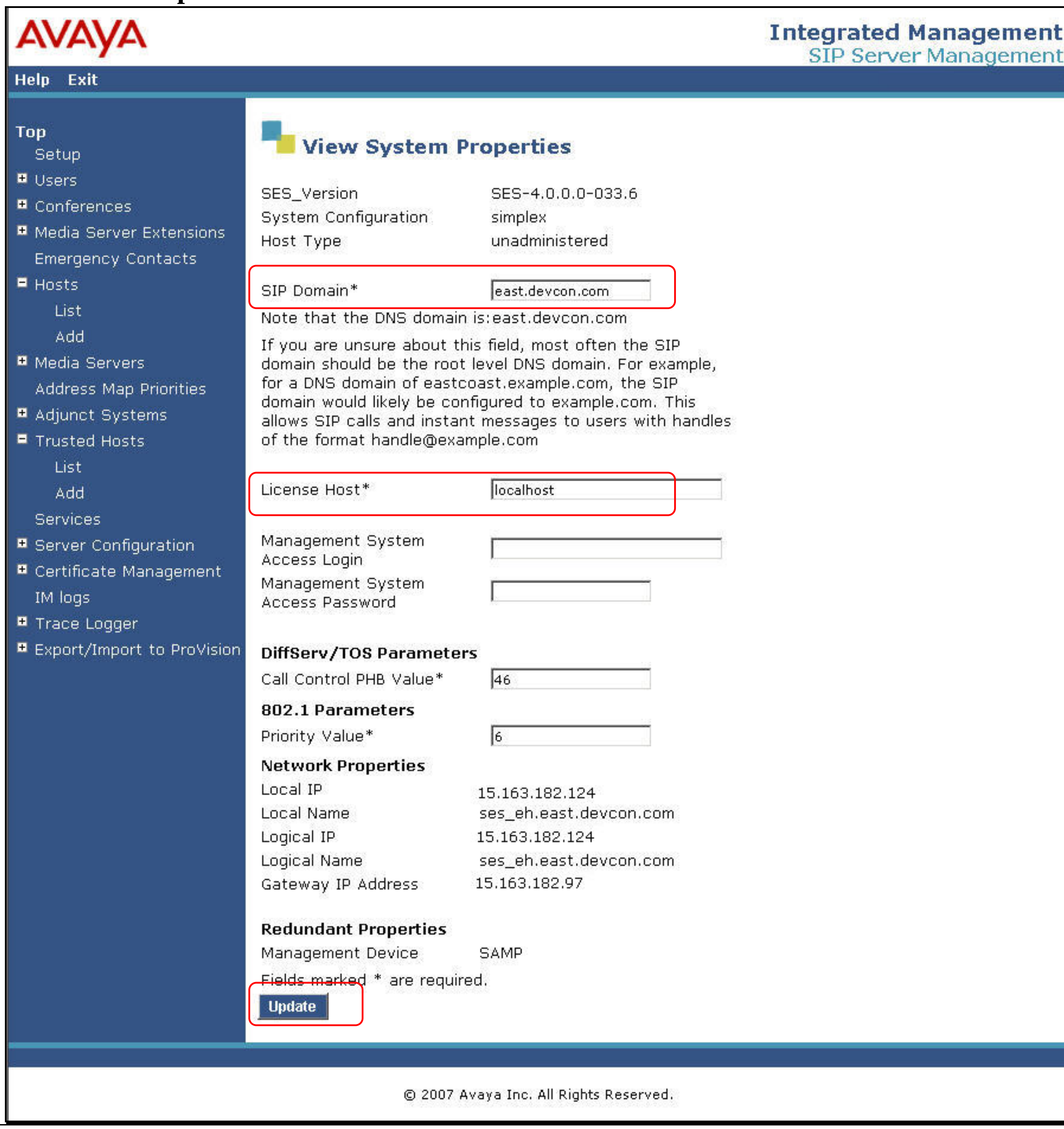


</td></tr>
</table>

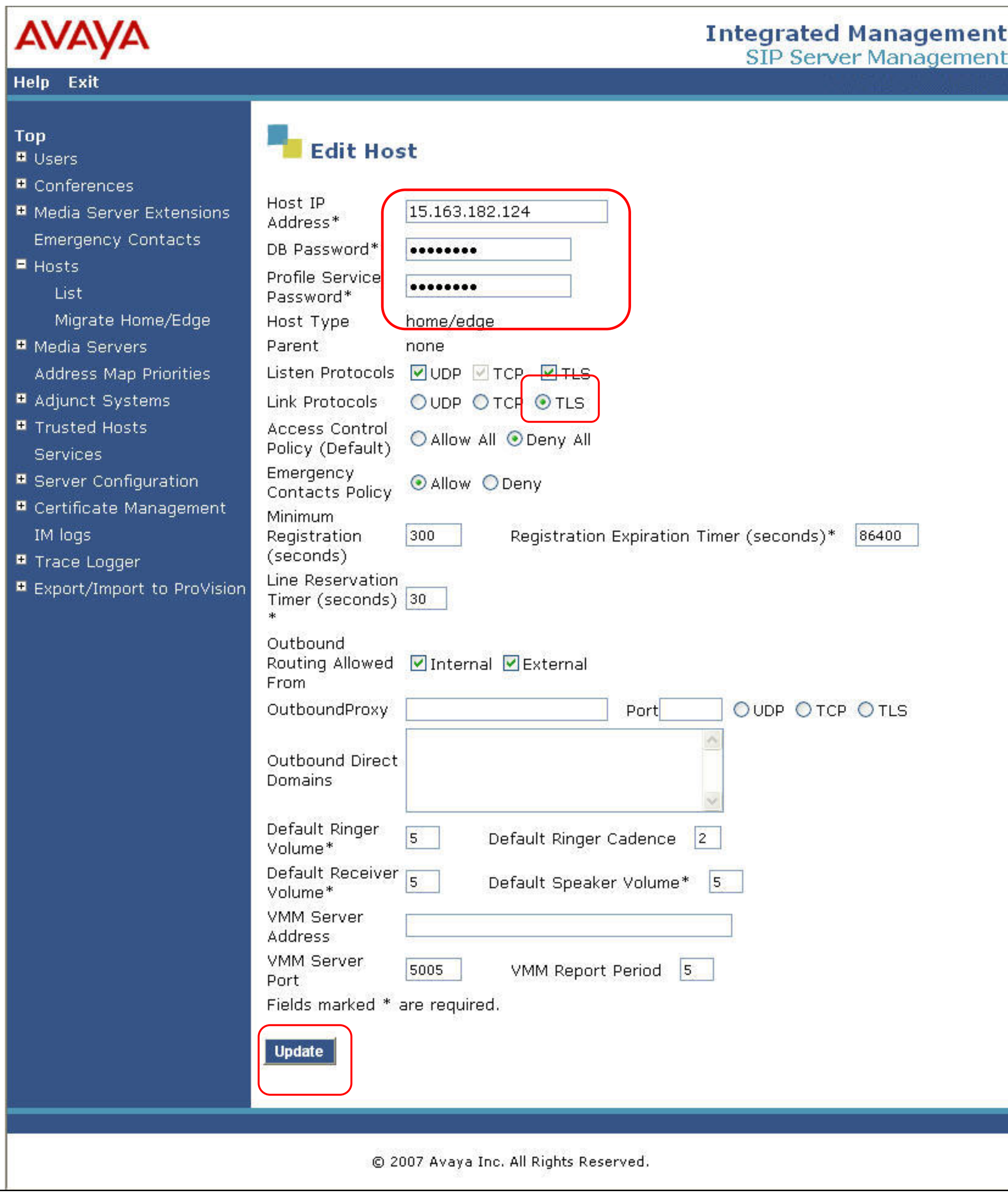
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4</td><td>

Follow the setup links to reach the View System Properties page.

- Enter the customer's **SIP Domain** and **License Host** values as shown (if not previously configured).
- Click **Update** when done.



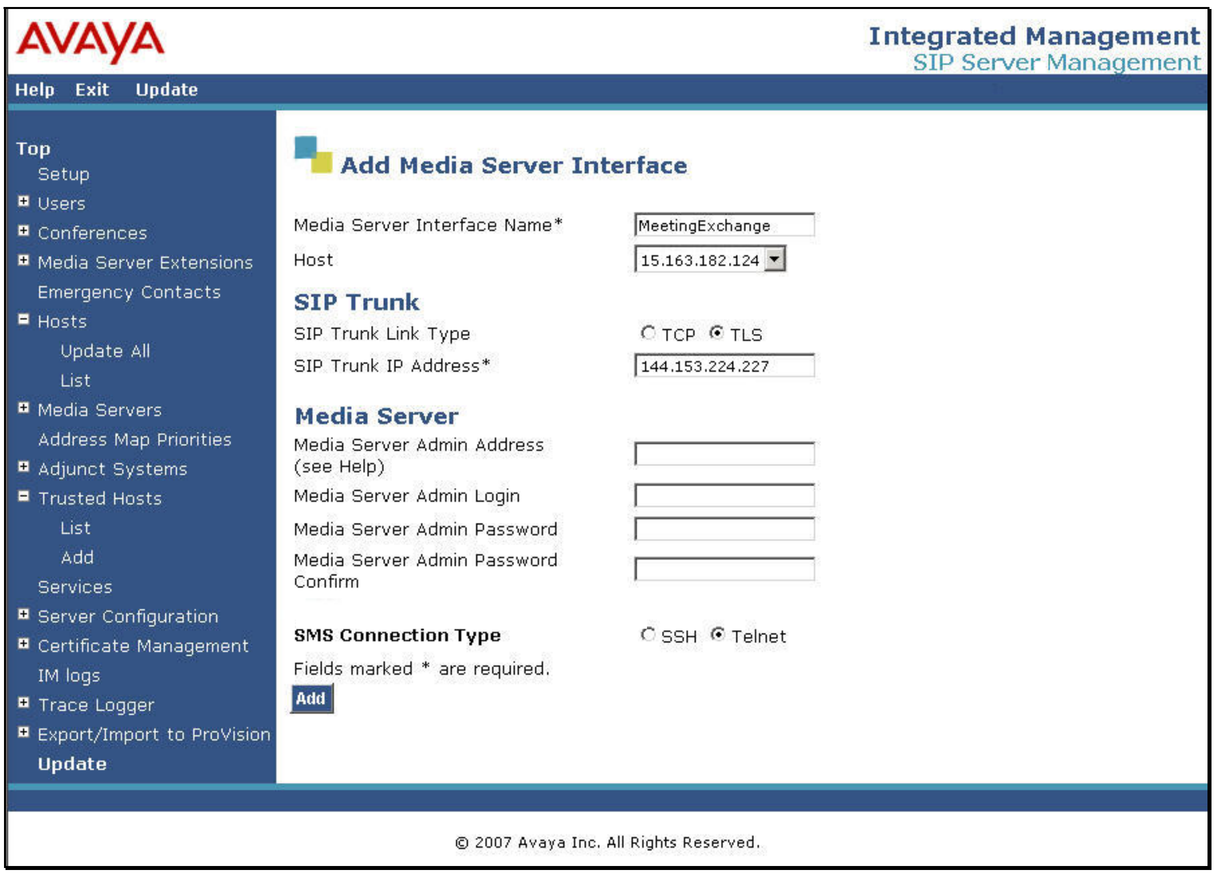
</td></tr>
</table>

Step	Description
5	Follow the links to reach the Add Host page. • Enter the Avaya SES IP address in the **Host IP Address** field. • Enter the Avaya SES database password (assigned when the “initial_setup” installation script was run) in the **DB Password** field. • Enter a password that uniquely identifies the Avaya SES for intra (and inter) proxy communications in the **Profile Service Password** field. • Select **“**TLS” from the available **Link Protocols**, which is consistent with the “system.cfg” file configured for Avaya Meeting Exchange in **Section 3 Step 2**. • Leave all remaining fields at their default settings as shown. • Click the **Add** button when done. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6</td><td>
Follow the links to reach the Add Media Server Interface page.

To configure the Avaya Meeting Exchange as a media server:

Enter a descriptive name for the Avaya Meeting Exchange in the Media Server Interface Name field.
Select “TLS” as the SIP Trunk Link Type consistent with the Avaya Meeting Exchange “system.cfg” file configuration done in Section 3 Step 2.
Enter the IP address of the Avaya Meeting Exchange in the SIP Trunk IP Address field.
Leave the Media Server Admin fields blank.
Select “Telnet” in the SMS Connection Type field.
Click the Add button when done.



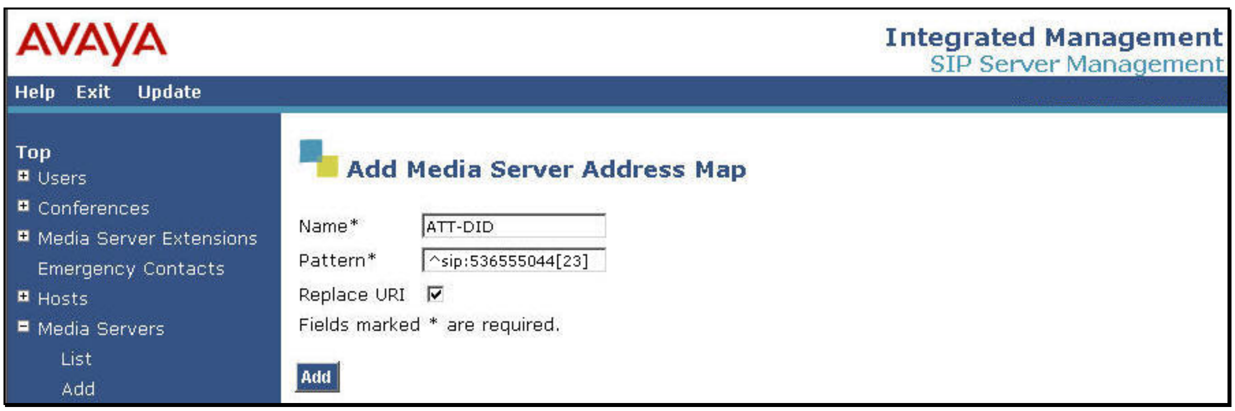
</td></tr>
</table>

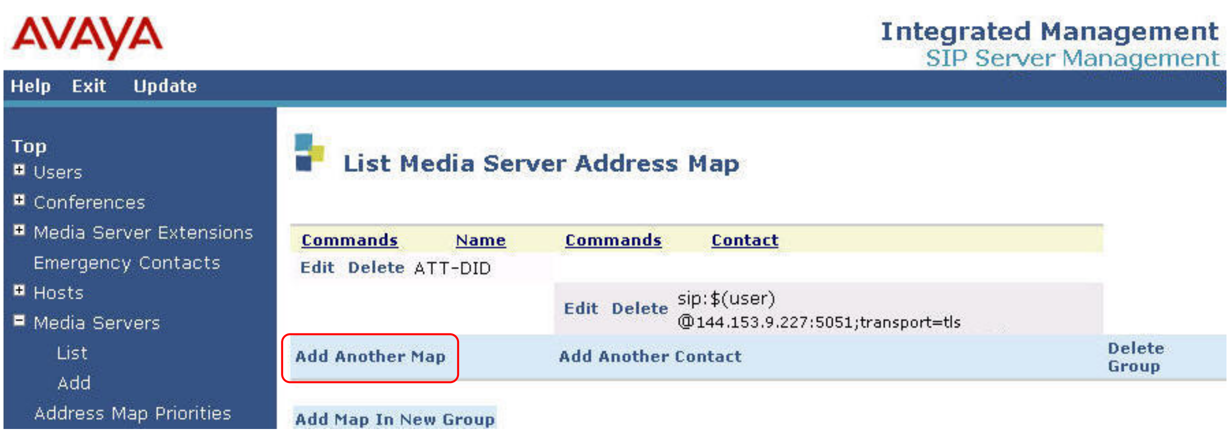
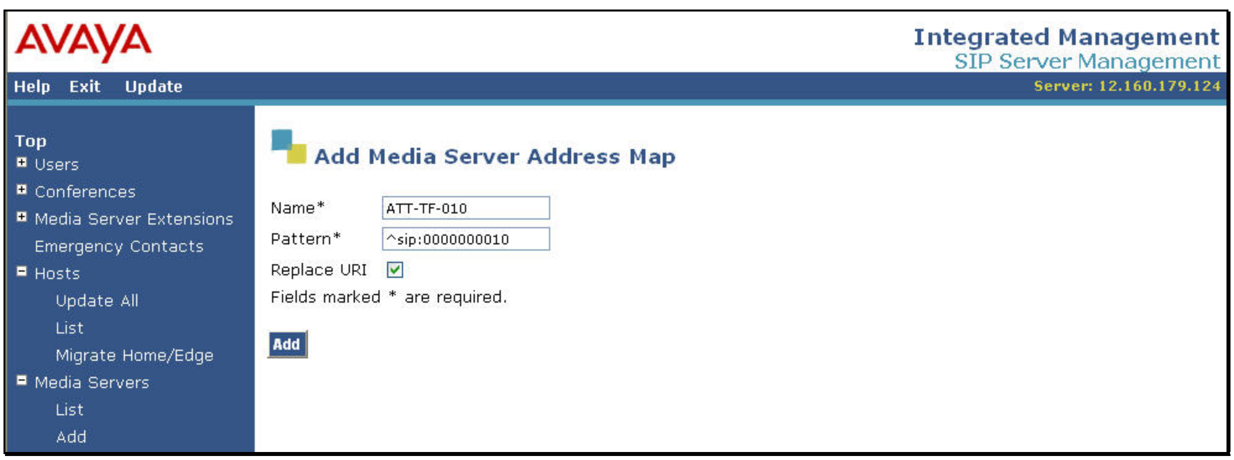
4.2. Dial-In to the Avaya Meeting Exchange

The following steps configure the Avaya SES to route incoming calls to the Avaya Meeting Exchange. In these Application Notes, two Direct Inward Dialed numbers (1-536-555-0442, 1-

536-555-0443) and one Toll Free number (1-800-457-5711 with the incoming DNIS digits of “0000000010”) have been provided by the AT&T Services. (See **Table 2**)

Step	Description
7	On the List Media Servers page: • Click the **Map** link to add the address maps that will route the incoming calls from the AT&T Services to the Avaya Meeting Exchange. 
8	On the List Media Server Address Map page: • Click the **Add Map In New Group** link. 

Step	Description
9	On the Add Media Server Address Map page: • Enter a descriptive name for the incoming number in the **Name** field. Here, "ATT-DID" indicates that this map is used for the incoming Direct Inward Dialed numbers assigned by the AT&T IP Flexible Reach Service. • Enter in the **Pattern** field the regular expression pattern matching the incoming call digits received from the AT&T Services in the SIP INVITE message. In these Application Notes the pattern "^sip:536555044[23]" is used. This indicates that a Request URI beginning with "sip:" followed by the 9 digits "536555044" and either "2" or "3" will be matched and routed to the Avaya Meeting Exchange SIP Trunk IP address defined in **Section 4.1 Step 6**. The corresponding portion (shown in bold) of a Request URI that this pattern matches is: "INVITE **sip:5365550442**@15.163.182.124:5060 SIP/2.0". • Check the **Replace URI** field. • Click the **Add** button when done.  Note: The regular expression patterns may be designed to match more than one incoming DID number by using additional types of matching patterns. The further use of regular expression patterns is described in Appendix B.

Step	Description
10	On the List Media Server Address Map page: • Click the **Add Another Map** link to create a second incoming call address map for the AT&T Services toll free number. 
11	Enter the information for the AT&T Services toll free number as was done in Step 9. Note in this case the AT&T DNIS digits of "0000000010" are what is received in the incoming Request URI. 

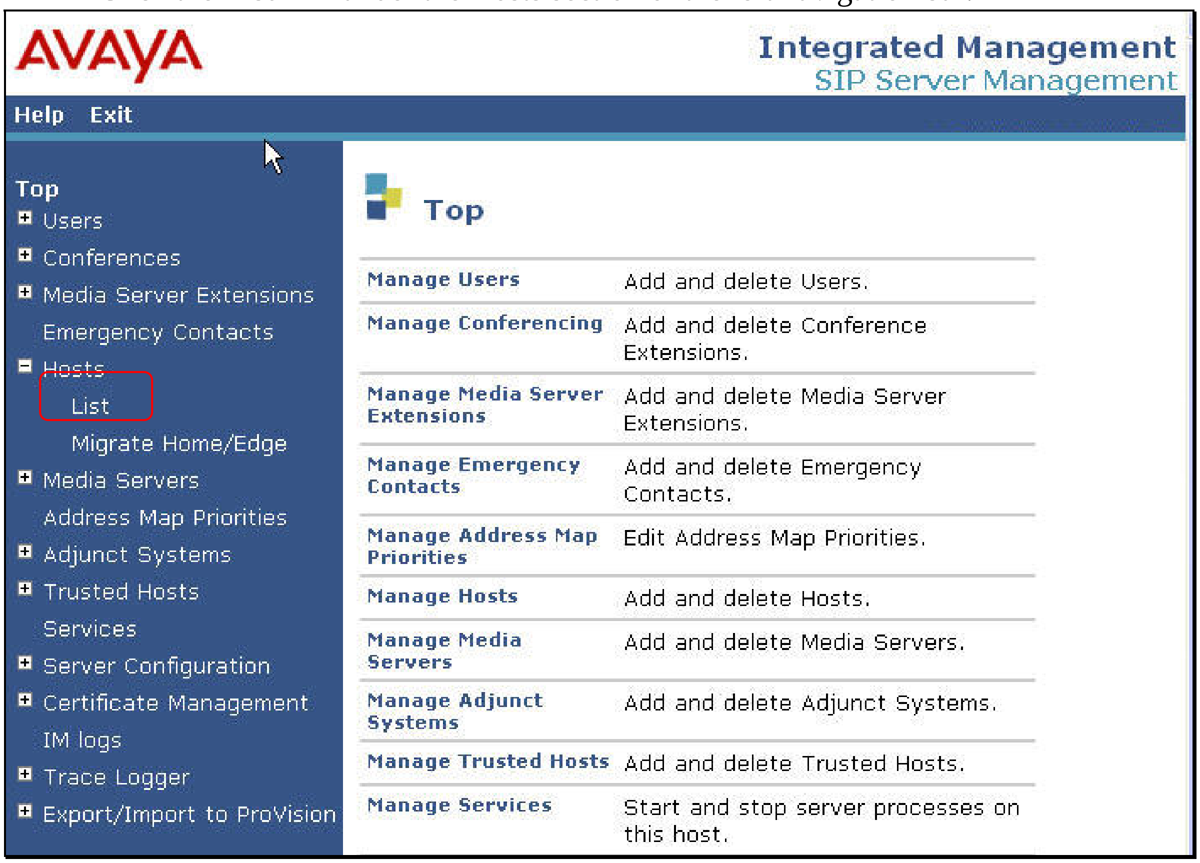
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>12</td><td>The resulting List Media Server Address Map page displays the completed incoming call address routing configuration. Note that the Contact information displays the Request URI that will be used to communicate with the Avaya Meeting Exchange.

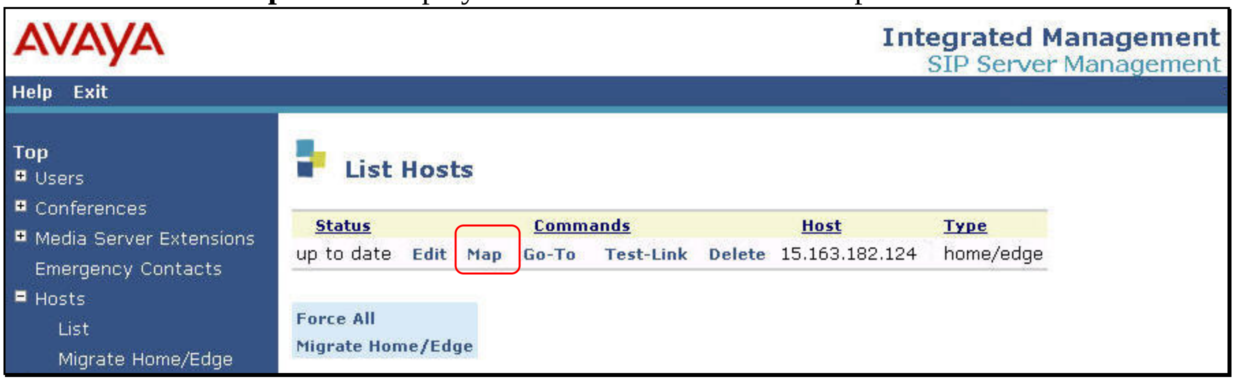
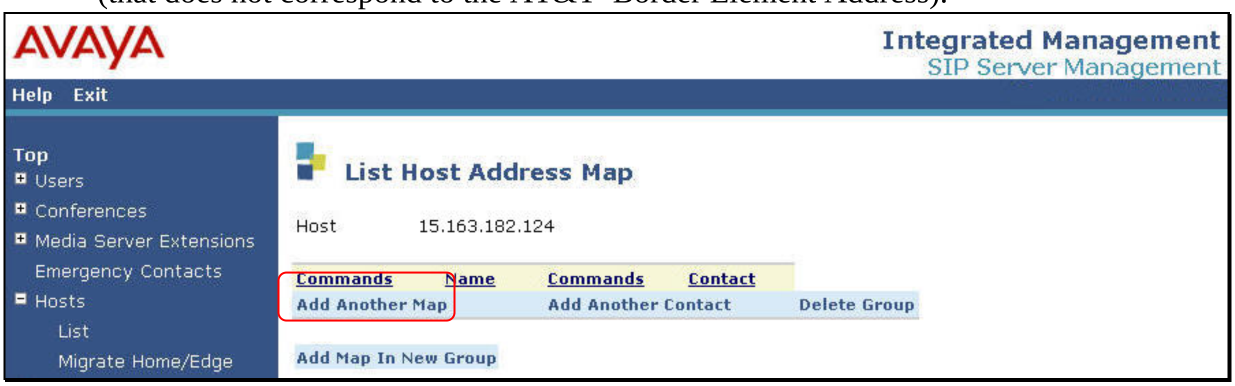
</td></tr>
</table>

4.3. Dial-Out from the Avaya Meeting Exchange

The following steps configure the Avaya SES to route outbound calls from the Avaya Meeting Exchange to the AT&T IP Flexible Reach Service for completion to a Public Switched Telephone Number (PSTN) telephone number.

Step	Description
13	From any SIP Server Management page: • Click the **List** link under the Hosts section of the left navigation bar.



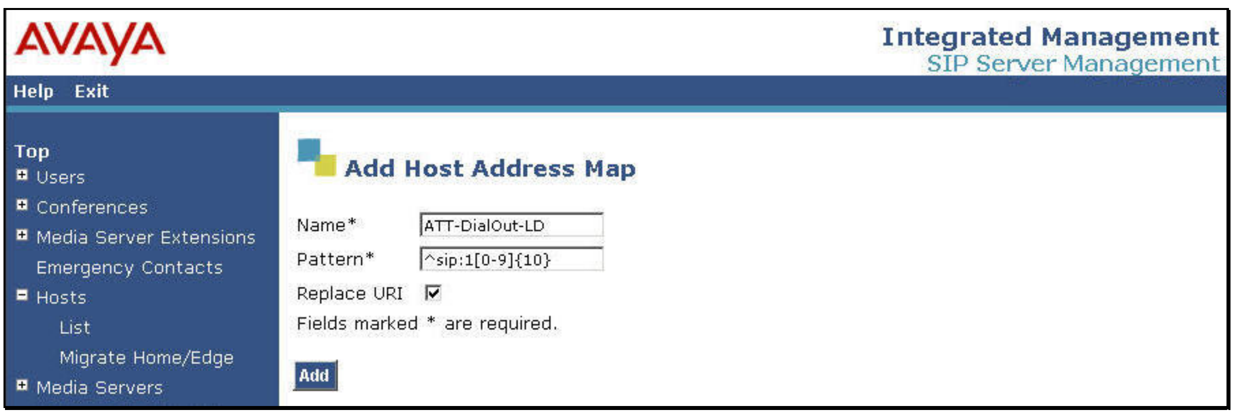
Step	Description
14	From the List Hosts page: • Click the **Map** link to display the current Host Address Maps. 
15	From the List Host Address Map page: • Click the **Add Another Map** link. Note: Use the **Add Map in New Group** link if there is a Contact previously defined (that does not correspond to the AT&T Border Element Address). 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>16</td><td>From the Add Host Address Map page:

Enter a descriptive name for the outbound address map in the Name field. Here, “ATT-DialOut-LD” indicates this will be used for 1+10 digits PSTN calls routed via the AT&T IP Flexible Reach Service.
Enter in the Pattern field the regular expression pattern matching the outgoing call digits being sent to the AT&T IP Flexible Reach Service in the SIP INVITE message.

In these Application Notes the pattern “^sip:1[0-9]{10}” is used. This indicates that a Request URI beginning with “sip:” plus “1” plus any 10 digits will be matched and routed to the associated Contact to be defined in the next step.

The corresponding portion (shown in bold) of the Request URI that this pattern matches is “INVITE sip:17358551637@210.245.228.200:5060;transport=udp SIP/2.0”.
Check the Replace URI field.
Click the Add button when done.



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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>17</td><td>From the List Host Address Map page:
Click the Add Another Contact link corresponding to the ATT-DialOut-LD address map created in the previous step.

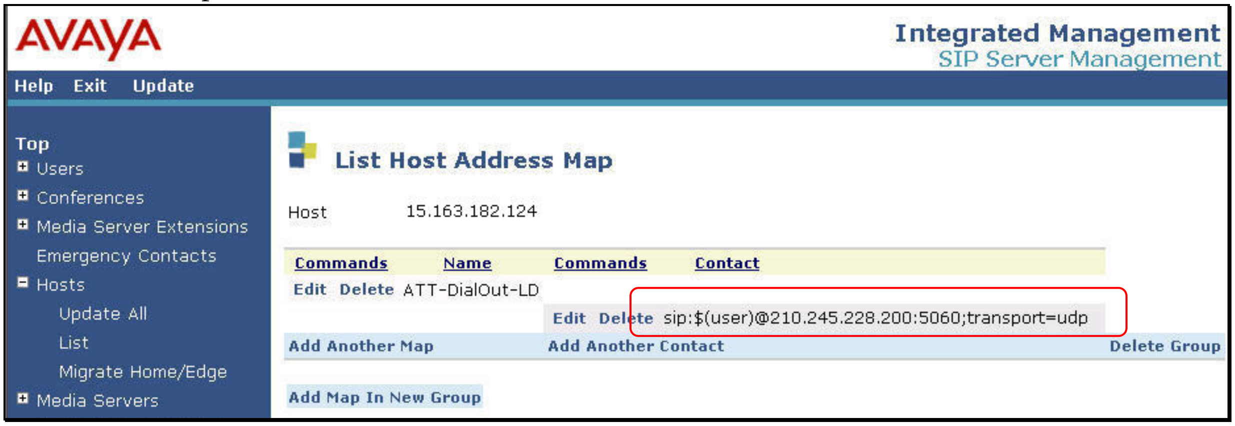

</td></tr>
<tr><td>18</td><td>From the Add Host Contact page:
Enter the required SIP Request URI that will be sent to the AT&T IP Flexible Reach Service. In the case, the specific entry (without the double quotes) is:

“sip:$(user)@210.245.228.200:5060;transport=udp”

This indicates that the SIP INVITE will be sent to the AT&T Border Element Address (202.242.225.200) using port 5060 and the udp transport method. The “$(user)” is a variable used to substitute the specific dialed address (e.g., 17358551637) used for each call.

This information will vary for individual customers and must be obtained from AT&T as part of the AT&T Services provisioning process.
Click the Add button when done.

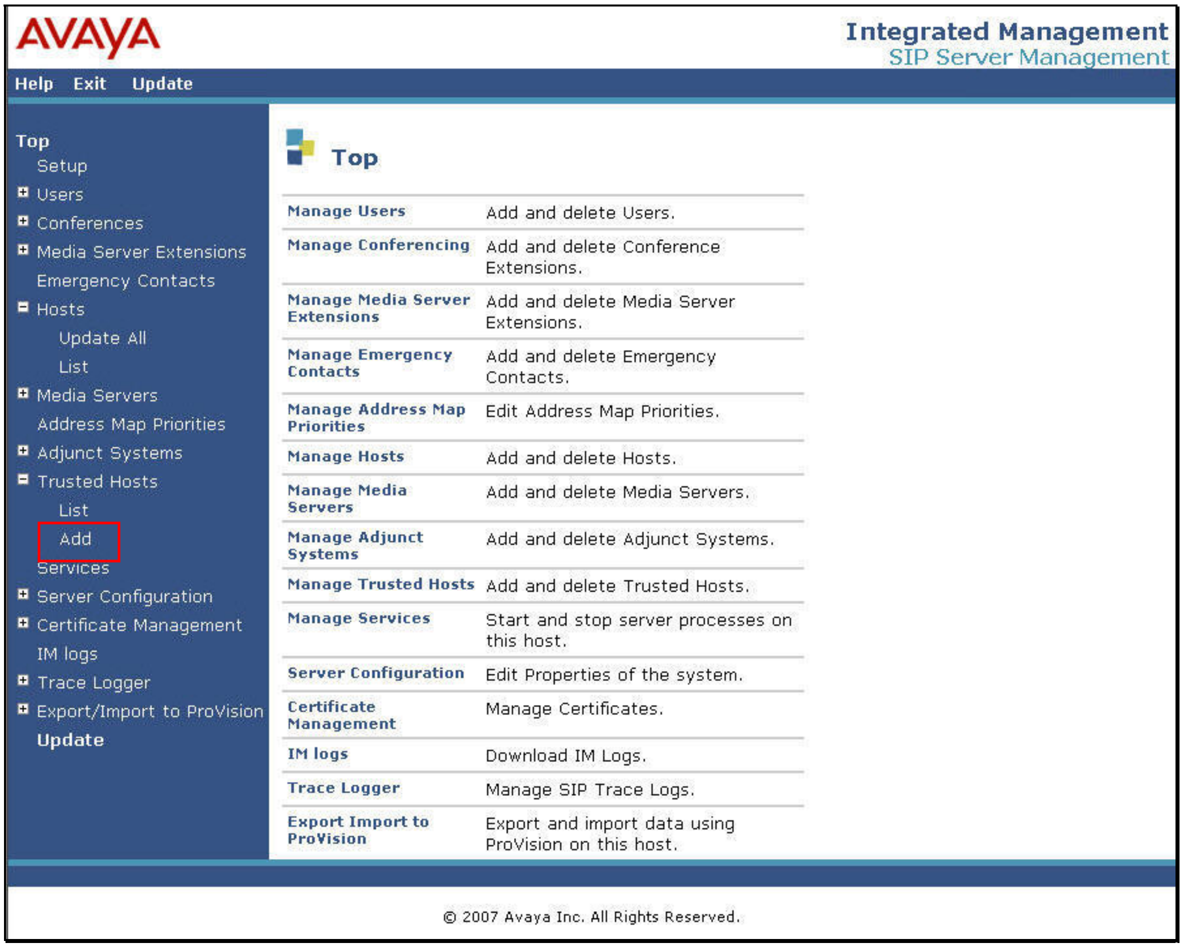

</td></tr>
</table>

Step	Description
19	The List Host Address Map page is displayed. • Verify the **Contact** information is properly entered and assocated with the correct address pattern **Name.** 

4.4. AT&T Services as a Trusted Host

This section designates that the AT&T Border Element Addresses are trusted hosts to the Avaya SES. This prevents the Avaya SES from challenging the AT&T Services for SIP authentication when SIP messages are received from the AT&T Services.

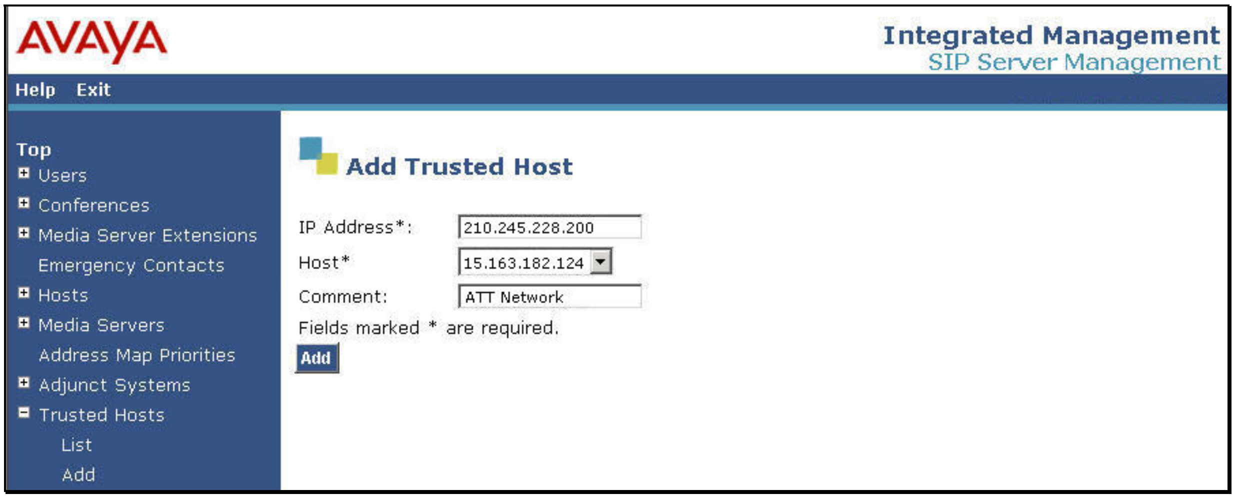
20	Configure the Avaya SES trusted hosts by clicking the **Add** link under Trusted Hosts.



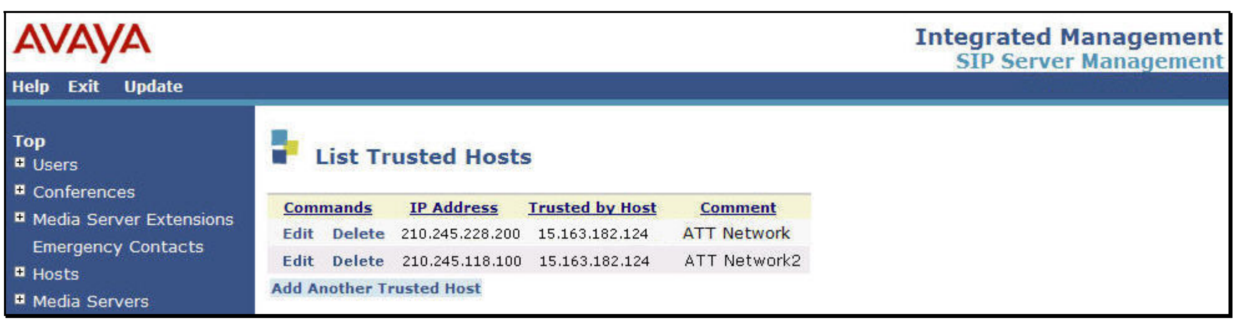
<table>
<tr><th></th><th></th></tr>
<tr><td>21</td><td>The Add Trusted Host page is displayed.

To add the AT&T Services as a trusted host:

Enter the AT&T Border Element Address in the IP Address field.
Set the Host field to the Avaya SES IP address.
Enter a descriptive phrase into the Comment field.
Click the Add button when finished.



</td></tr>
<tr><td>22</td><td>Repeat for any other AT&T Border Element Addresses provided.</td></tr>
<tr><td>23</td><td>Confirm the entries on the List Trusted Hosts page.



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

4.5. Commit Avaya SES Administrative Changes

The various Avaya SES administrative changes performed above will not take effect until the update action is performed.

Step	Description
24	To perform the Avaya SES update action: • Click on either **Update** link found any SIP Server Management page.

5. Verification Steps

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

5.1. Verification Tests

This section provides steps that may be performed to verify the operation of the SIP trunking configuration described in the Application Notes.

- Incoming Calls – Verify that calls placed from a PSTN telephone using the AT&T provided DID or toll free telephone number assigned are properly routed via the SIP trunk to the Meeting Exchange. The expected Avaya Meeting Exchange announcement should be heard. Verify that the conference PIN is accepted and/or calls are routed to the Avaya Bridge Talk operator queue. Verify the talk-path exists in both directions, among all various conference participants and that calls remain stable for several minutes and disconnect properly.

- Outbound Calls – Verify that an Avaya Bridge Talk operator or conference moderator can place outbound calls to a PSTN destination via the AT&T Services. Verify that the talk-path exists in both directions, among all various conference participants and that calls remain stable and disconnect properly.

- Using Avaya Bridge Talk verify participants in conferences, operator ability to monitor and enter conferences, and the ability of the operator to add and disconnect conference parties.

5.2. Troubleshooting Tools

The "Trace Logger" function within the Avava SES Administration Web Interface may be used to capture SIP traces between Avaya SES and the AT&T Services. These traces can be instrumental in understanding SIP protocol issues resulting from configuration problems.

If port monitoring is available, a SIP protocol analyzer such as WireShark (a.k.a., Ethereal) to monitor the SIP messaging between the SES and the AT&T Services. Note that SIP messaging between Avaya Meeting Exchange and Avaya SES uses TLS encryption and cannot be viewed using WireShark.

6. Support

AT&T customers may obtain support for the AT&T IP Flexible Reach Service by calling 1-877-288-8362. Support for the AT&T IP Toll Free Service should be directed to 1-800-325-5555.

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. The "Connect with Avaya" section provides the worldwide support directory. In the United States, 1-866-GO-AVAYA (1-866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on support.avaya.com) to directly access specific support and consultation services based upon their Avaya support agreements.

7. Conclusion

These Application Notes provide administrators with the procedures to configure SIP trunking connectivity between AT&T IP Flexible Reach and IP Toll Free services with the Avaya Meeting Exchange S6200 Conferencing Server via Avaya SIP Enablement Services.

8. Additional References

The following Avaya references are available at http://support.avaya.com.

1. *Meeting Exchange 5.0 Administration and Maintenance S6200/S6800 Media Server*, Issue 1, Doc ID 04-602167, August 2007.
2. *Avaya Meeting Exchange Groupware Edition Version 4.1 User's Guide for Bridge Talk*, Issue 2, Doc ID 04-600878, July 2006.
3. *SIP Enablement Services Implementation Guide*, Issue 4, Doc ID: 16-300140, May 2007.

APPENDIX A: Sample SIP INVITE Messages

This section displays the format of typical SIP INVITE messages sent between AT&T and the Avaya SES. These INVITE messages may be used for comparison and troubleshooting purposes. Differences in these messages may indicate that different configuration options were selected.

Sample SIP INVITE Message from the AT&T services to the Avaya SES:

```
INVITE sip:5365550442@144.153.9.227:5060;transport=tcp SIP/2.0
Accept: application/sdp,application/isup,application/dtmf,application/dtmf-relay,multipart/mixed
Accept-Language: en;q=0.0
Call-ID: SD5ah8801-0f2fa65682480bc12b825764c9126f03-fms3e43
CSeq: 1 INVITE
From: "OUT_OF_AREA" <sip:+17358551637@210.245.228.200:5060;user=phone>;tag=SD5ah8801-ds84f142e2
To: <sip:5365550442@15.163.182.124;user=phone>
Via: SIP/2.0/TCP 15.163.182.124:5060;branch=z9hG4bK263535C644441353534558.0,SIP/2.0/UDP
210.245.228.200:5060;psrrposn=1;received=210.245.228.200;branch=z9hG4bKbl156u1010g0lb0i17g1.1
Content-Length: 275
Content-Type: application/sdp
Contact: <sip:+17358551637@210.245.228.200:5060;transport=udp>
Max-Forwards: 67
Allow: INVITE,ACK,CANCEL,BYE,INFO,PRACK
Content-Disposition: session;handling=required
P-Asserted-Identity: <sip:7358551637@210.245.228.200:5060>
Record-Route: <sip:15.163.182.124:5060;transport=tcp;lr>

v=0
o=Sonus_UAC 5764 1479 IN IP4 210.245.228.200
s=SIP Media Capabilities
c=IN IP4 210.245.228.200
t=0 0
m=audio 17076 RTP/AVP 2 18 0 96
a=rtpmap:2 G726-32/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=sendrecv
```

Sample SIP INVITE Message from Avaya SES to the AT&T services:

```
INVITE sip:17358551637@210.245.228.200:5060;transport=udp SIP/2.0
Call-ID: 7CC40F2A@144.153.9.2275060
CSeq: 833742147 INVITE
Expires: 180
From: <sip:6000@144.153.9.227>;tag=144.153.9.2275060+1+13d0000+b4c5d8d5
To: sip:17358551637@15.163.182.124:5060;transport=tcp
Via: SIP/2.0/UDP 15.163.182.124:5060;branch=z9hG4bKB2731363344336341342l8.0,SIP/2.0/TCP
144.153.9.227:5060;psrrposn=1;received=144.153.9.227;branch=z9hG4bK+6c07642345e3508f8cf868e049f50
bb9+144.153.9.2275060+1
Content-Length: 218
Content-Type: application/sdp
Contact: <sip:S6200@144.153.9.227:5060;transport=tcp>
Max-Forwards: 69
Supported: timer
Min-SE: 900
Session-Expires: 900
Record-Route: <sip:15.163.182.124:5060;lr>

v=0
o=- 534363200 534363200 IN IP4 144.153.9.227
s=-
c=IN IP4 144.153.9.227
t=0 0
m=audio 42044 RTP/AVP 0 8 101
a=rtpmap:8 pcma/8000/1
a=rtpmap:0 pcmu/8000/1
a=fmtp:101 0-15
a=rtpmap:101 telephone-event/8000
```


APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within the Avaya SES is a Linux regular expression used to match against the URI string found in the SIP INVITE message.

Regular expressions are a way to describe text through pattern matching. The regular expression is a string containing a combination of normal text characters, which match themselves, and special *metacharacters*, which may represent items like quantity, location or types of character(s).

In the pattern matching string used in the Avaya SES:

- Normal text characters and numbers match themselves.
- Common metacharacters used are:
    - A period **.** matches any character once (and only once).
    - An asterisk **`*`** matches zero or more of the preceding characters.
    - Square brackets enclose a list of any character to be matched. Ranges are designated by using a hyphen. Thus the expression **`[12345]`** or **`[1-5]`** both describe a pattern that will match any single digit between 1 and 5.
    - Curly brackets containing an integer 'n' indicate that the preceding character must be matched exactly 'n' times. Thus **`5{3}`** matches '555' and **`[0-9]{10}`** indicates any 10 digit number.
    - The circumflex character **^** as the first character in the pattern indicates that the string must begin with the character following the circumflex.

Putting these constructs together as used in this document, the pattern to match the SIP INVITE string for any valid 1+ 10 digit number in the North American dial plan would be:

`^sip:1[0-9]{10}`

This reads as: "Strings that begin with exactly **sip:1** and having any 10 digits following will match.

A typical INVITE request below uses the shaded portion to illustrate the matching pattern.

```
INVITE sip:17325551638@20.1.1.54:5060;transport=udp SIP/2.0
```

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.