Avaya Solution & Interoperability Test Lab

Configuring Secure SIP Connectivity Utilizing Transport Layer Security (TLS) between Avaya MultiVantage® Express and Avaya Meeting Exchange Express Edition - Issue 1.0

Abstract

These Application Notes present the procedures for configuring secure SIP connectivity between Avaya MultiVantage® Express and Avaya Meeting Exchange Express Edition. Secure SIP connectivity is enabled by utilizing the Transaction Layer Security (TLS) authentication and encryption standard providing customers with a secure standards based solution. This configuration leverages the flexibility offered by Avaya MultiVantage Express to support a rich set of audio conferencing options provided by Avaya Meeting Exchange Express Edition.

1. Introduction

These Application Notes present the procedures for configuring secure SIP connectivity between Avaya MultiVantage Express and Avaya Meeting Exchange Express Edition. Secure SIP connectivity is enabled by utilizing the Transaction Layer Security (TLS) authentication and encryption standard providing customers with a secure standards based solution. This configuration leverages the flexibility offered by Avaya MultiVantage Express to support a rich set of audio conferencing options provided by Avaya Meeting Exchange Express Edition.

Figure 1 illustrates the sample network configuration utilized for this compliance tested solution. For this sample configuration, Avaya MultiVantage Express provided endpoint aggregation and media gateway functionality. For example, any station or trunk type associated with Avaya MultiVantage Express can interoperate with Avaya Meeting Exchange Express Edition via secure SIP connectivity. Avaya MultiVantage Express also provided feature functionality for stations interoperating with Avaya Meeting Exchange Express Edition, e.g., call hold, call transfer, three-way conference.

Avaya Meeting Exchange Express Edition was provisioned to accept calls from Avaya MultiVantage Express via either direct or basic call flows. A direct call flow allows access to conference(s) provisioned on Avaya Meeting Exchange Express Edition without entering a passcode. Conversely, to enter a conference via a basic call flow requires a passcode. Avaya Meeting Exchange Express Edition was also administered for outbound calling, which enabled call origination from Avaya Meeting Exchange Express Edition to add participant(s) to a conference.

Signaling connectivity between Avaya MultiVantage Express and Avaya Meeting Exchange Express Edition was SIP/TLS and traversed the path depicted by the blue dashed line. Media connectivity was RTP and traversed the path represented by the red dotted line. Note that the media exchange between IP stations registered to either Avaya MultiVantage Express or Avaya SIP Enablement Services, and Avaya Meeting Exchange Express Edition is via direct IP-to-IP audio connectivity.

For this sample configuration, Avaya SIP Enablement Services was utilized to register SIP stations only, e.g., it was not involved in signaling between Avaya MultiVantage Express and Avaya Meeting Exchange Express Edition.

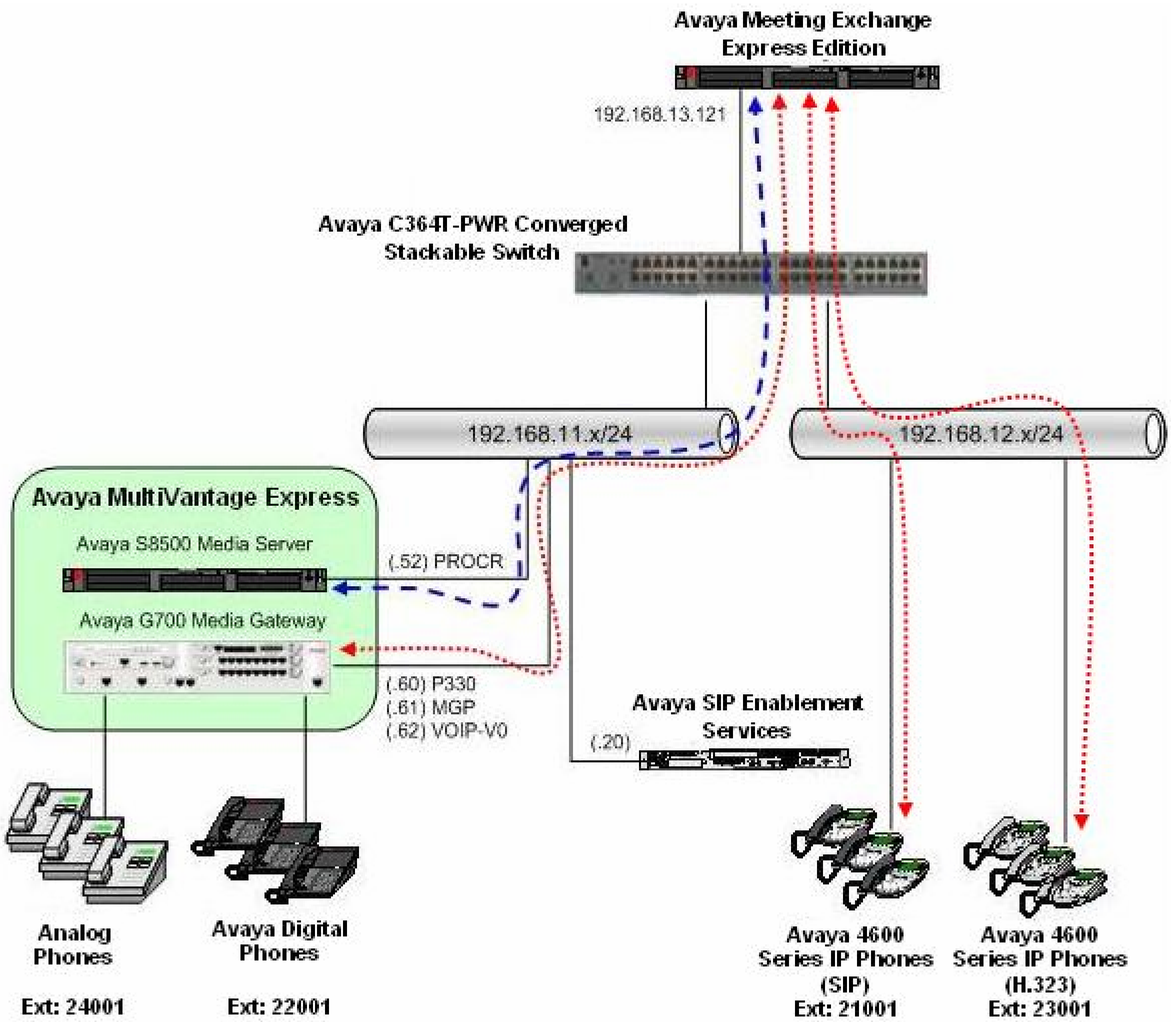


Figure 1: Sample Network Configuration

1.1. Avaya MultiVantage Express

Avaya MultiVantage Express is a packaged business solution for 100 to 500 users. The solution resides on a single Avaya S8500-class server and consists of:

- Avaya Communication Manager.
- INTUITY AUDIX® 770 (IA770) Messaging.
- Application Enablement Services (AE Services).
- An Avaya G700 Media Gateway.

For this sample configuration, only the Avaya Communication Manager component of Avaya MultiVantage Express was utilized.

1.2. Avaya Meeting Exchange Express Edition

The Avaya Meeting Exchange Express Edition (Avaya Meeting Exchange) is a SIP-based voice conferencing solution that runs on an S6100 server and provides mid-market enterprise customers with an IP based audio conferencing system. Avaya Meeting Exchange is administered and maintained using a standard web browser over a secure connection.

2. Equipment and Software Validated

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

Equipment	Software
Avaya MultiVantage Express • Avaya S8500 Media Server • Avaya G700 Media Gateway o MM711AP o MM712AP o MM710AP o MM710AP	 Avaya Communication Manager 3.1.2 (R013x.01.2.632.1) HW28 FW068 HW07 FW007 HW05 FW018 HW05 FW015
Avaya Meeting Exchange Express Edition	S6100-2.5.21.0
Avaya SIP Enablement Services	SES-3.1.1.0-114.0
Avaya C364T-PWR Converged Stackable Switch	4.5.14
Avaya 4600 Series IP Telephones	2.8 (H.323)
Avaya 4600 Series IP Telephones	2.2.2 (SIP)
Avaya 6408D+ Digital Telephones	--
Analog Telephones	--

Table 1: Hardware and Software Versions

3. Avaya MultiVantage Express Configuration

This section describes the steps for configuring Avaya MultiVantage Express to interoperate with Avaya Meeting Exchange. Recall, that Avaya MultiVantage Express is running Avaya Communication Manager. Also, only the Avaya Communication Manager component of Avaya MultiVantage Express was utilized for this sample configuration. Therefore, provisioning Avaya MultiVantage Express is analogous to provisioning Avaya Communication Manager.

In these Application Notes, Avaya MultiVantage Express administrative software screens are shown with a gray shaded background. These screens are also referred to as System Access Terminal (SAT) screens. In some instances, the information from the original screen has been edited or annotated for brevity or clarity in presentation. For example, entries and/or fields in the SAT screens that were either modified or were required for these Application Notes are displayed with boldface type. After completion of the configuration in this section, perform a **save translation** command to make the changes permanent.

The administrative steps in this section have been divided into the following sub-sections:

- Verifying licensing on Avaya MultiVantage Express that is required to support the configuration displayed in these Application Notes.
- Configuring connectivity to enable signaling (secure SIP/TLS) and media connectivity between Avaya MultiVantage Express and Avaya Meeting Exchange.
- Configuring call routing from Avaya MultiVantage Express to Avaya Meeting Exchange.

3.1. Verify Licensing

The following steps show procedures to verify licensing on Avaya MultiVantage Express that is required to support the configuration displayed in these Application Notes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.1.1</td><td>Verify customer options.

Issue the command “display system-parameters customer-options”, and proceed to Page 2 on the system-parameters customer-options form. Verify that the Maximum Administered SIP Trunks supported by Avaya MultiVantage Express is sufficient.

Note: Each call between two SIP stations (whether internal or external) requires two SIP trunks for the duration of the call. For this sample configuration, Avaya Meeting Exchange is treated as an external SIP station. Thus, a call from a SIP station registered to Avaya SIP Enablement Services as an off-pbx-telephone on Avaya MultiVantage Express to Avaya Meeting Exchange will use two SIP trunks. A call between a non-SIP station and Avaya Meeting Exchange will use only one SIP trunk.

<pre>
display system-parameters customer-options   Page   2 of 11

                                OPTIONAL FEATURES

IP PORT CAPACITIES   USED
                     Maximum Administered H.323 Trunks: 150   0
           Maximum Concurrently Registered IP Stations: 500   0
             Maximum Administered Remote Office Trunks: 0   0
Maximum Concurrently Registered Remote Office Stations: 0   0
              Maximum Concurrently Registered IP eCons: 5   0
  Max Concur Registered Unauthenticated H.323 Stations: 0   0
                  Maximum Video Capable H.323 Stations: 2000 0
                   Maximum Video Capable IP Softphones: 2000 0
                       Maximum Administered SIP Trunks: 150   0

   Maximum Number of DS1 Boards with Echo Cancellation: 0   0
                             Maximum TN2501 VAL Boards: 10   0
                    Maximum G250/G350/G700 VAL Sources: 10   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0   0

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

Step	Description
3.1.2	Proceed to Page 3 on the system-parameters customer-options form and verify that the system is licensed for **ARS/AAR Dialing without FAC**. ***Note****: Automatic Alternate Routing (AAR) without Feature Access Code (FAC) allows direct access to the AAR digit analysis table (see* ***Step 3.3.3****) upon matching a dialed string in the dial plan analysis table (see* ***Step 3.3.1****).*

```
display system-parameters customer-options                      Page   3 of  11

                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? n           Audible Message Waiting? y
        Access Security Gateway (ASG)? n               Authorization Codes? n
        Analog Trunk Incoming Call ID? n Backup Cluster Automatic Takeover? n
 A/D Grp/Sys List Dialing Start at 01? n                        CAS Branch? n
Answer Supervision by Call Classifier? n                          CAS Main? n
                                 ARS? y                  Change COR by FAC? n
                 ARS/AAR Partitioning? y  Computer Telephony Adjunct Links? y
          ARS/AAR Dialing without FAC? y   Cvg Of Calls Redirected Off-net? n
          ASAI Link Core Capabilities? n                       DCS (Basic)? n
          ASAI Link Plus Capabilities? n                 DCS Call Coverage? n
       Async. Transfer Mode (ATM) PNC? n                DCS with Rerouting? n
  Async. Transfer Mode (ATM) Trunking? n
              ATM WAN Spare Processor? n    Digital Loss Plan Modification? n
                                ATMS? n                            DS1 MSP? n
                  Attendant Vectoring? y             DS1 Echo Cancellation? n



        (NOTE: You must logoff & login to effect the permission changes.)
```

3.2. Configure Connectivity

The following steps show procedures to enable signaling (secure SIP/TLS) and media connectivity between Avaya MultiVantage Express and Avaya Meeting Exchange.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.2.1</td><td>Configure an IP codec set.

Issue the command "change ip-codec-set <n>", where n is the number of an available codec set. Add entry(s) for audio codec(s) that are supported on Avaya Meeting Exchange. For this sample configuration, an entry for G.711MU was added as displayed.
<pre>
change ip-codec-set 1   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:
</pre></td></tr>
<tr><td>3.2.2</td><td>Proceed to page 2 on the ip-codec-set form and administer as displayed.

Note: The Maximum Call Rate for Direct-IP Multimedia is the combined audio and video transmit or receive rate. This setting may be used to limit the amount of bandwidth required for calls that utilize this codec set. For this sample configuration, only audio media is applicable.
<pre>
change ip-codec-set 1   Page   2 of   2

                          IP Codec Set

                              Allow Direct-IP Multimedia? y
              Maximum Call Rate for Direct-IP Multimedia: 64:Kbits

                    Mode   Redundancy
    FAX   relay   0
    Modem   off   0
    TDD/TTY   US   3
    Clear-channel   n   0
</pre></td></tr>
</table>

Step	Description
3.2.3	Configure an IP network region. Issue the command “**change ip-network-region <n>**”, where **n** is the number of an available IP network region, and administer settings as displayed. • Enter the number of the IP codec set provisioned in **Step 3.2.1** in the **Codec Set** field. • Verify that the **Inter-region IP-IP Direct Audio** field is set to **yes**. This will allow direct IP-to-IP audio connectivity between IP stations registered to either Avaya MultiVantage Express or Avaya SIP Enablement Services, and Avaya Meeting Exchange. For this sample configuration; the C-LAN, and all IP stations registered to either Avaya MultiVantage Express or Avaya SIP Enablement Services are in IP network region 1 and Avaya Meeting Exchange is in IP network region 12. • Leave remaining fields at default values. ***Note:*** *To enable direct IP-to-IP audio connectivity, the following must be administered:* • *[**Not Shown**] Direct IP-to-IP audio connectivity must be enabled at the system-level on the **system-parameters features** form by setting the **Direct IP-IP Audio Connections** field to **y**.* • *Direct IP-to-IP audio connectivity must be enabled on the signaling group associated with this IP network region (see **Step 3.2.6**).* • *[**Not Shown**] Direct IP-to-IP audio connectivity must be enabled on the **station** form by setting the **Direct IP-IP Audio Connections** field to **y**.*

```
change ip-network-region 12                                    Page   1 of  19

                               IP NETWORK REGION
  Region: 12
Location:         Authoritative Domain:
    Name: Meeting Exchange
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? n
   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
3.2.4	Proceed to Page 3 of the ip-network-region form and enable inter-region connectivity between IP network regions 12 and 1 by entering the IP codec set provisioned in **Step 3.2.1** in the **codec set** field as displayed.

```
change ip-network-region 12                                     Page   3 of  19

                   Inter Network Region Connection Management

 src dst codec direct   Total           Video                         Dyn
 rgn rgn  set   WAN  WAN-BW-limits  WAN-BW-limits  Intervening-regions  CAC IGAR
 12  1    1     y        :NoLimit       :NoLimit                              n
 12  2
 12  3
 12  4
 12  5
 12  6
 12  7
 12  8
 12  9
 12  10
 12  11
 12  12   1
```

Step	Description
3.2.5	Configure IP node names. Issue the command "**change node-names ip**", and administer settings as displayed. • Add an entry to assign a **Name** corresponding to the **IP Address** (**192.168.13.121**) for Avaya Meeting Exchange. • Verify that an entry exists (from system configuration procedures not shown in these Application Notes) for the **procr**. ***Note**: The **procr** is equivalent to the CLAN on an Avaya G650 Media Server*.

```
change node-names ip                                            Page   1 of   1

                                 IP NODE NAMES
    Name              IP Address
001s6100            192.168.13 .121
IA770               192.168.11 .56
MV_BCMS             192.168.11 .53
MV_CDR              192.168.11 .53
aeserver1           192.168.11 .54
default             0   .0   .0   .0
procr               192.168.11 .52
ses                 192.168.11 .20
```

Step	Description

3.2.6	Configure a signaling group.
	
	Issue the command "**add signaling-group <n>**", where **n** is the number of an unallocated signaling group, and administer settings as displayed.
	• To enable secure SIP connectivity utilizing TLS, configure the following:
	  o Set the **Group Type** field to **sip**.
	  o Set the **Transport Method** field to **tls**.
	  o Set the **Far-end Listen Port** field to **5061**.
	  o Leave the **Near-end Listen Port** field at the default value (**5061**).
	• Enter the IP node name for the procr (see **Step 3.2.5**) in the **Near-end Node Name** field.
	• Enter the IP node name of Avaya Meeting Exchange provisioned in **Step 3.2.5** in the **Far-end Node Name** field.
	• Enter the number of the IP network region provisioned in **Step 3.2.3** in the **Far-end Network Region** field.
	• Set the **Direct IP-IP Audio Connections** field to **y** to enable direct IP-to-IP audio connectivity for IP stations utilizing this signaling group.
	• Leave remaining fields at default values.

```
add signaling-group 12                                      Page   1 of   1

                                SIGNALING GROUP

 Group Number: 12            Group Type: sip
                       Transport Method: tls






   Near-end Node Name: procr                 Far-end Node Name: 001s6100
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 12
       Far-end Domain:

                                     Bypass If IP Threshold Exceeded? n

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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.2.7</td><td>
Configure a trunk group.

Issue the command “add trunk-group <n>”, where n is the number of an unallocated trunk group, and administer settings as displayed.

Set the Group Type field to sip, which is consistent with the signaling group provisioned in Step 3.2.6.
Set the Trunk Access Code (TAC) field to a number that is consistent with the existing dial plan (see Step 3.3.1).
Set the Service Type field to tie.
Enter the number of the signaling group provisioned in Step 3.2.6 in the Signaling Group field.
Assign a value in the Number of Members field to define the capacity of this trunk group. As mentioned in Step 3.1.1, each call between two SIP stations (whether internal or external) requires two SIP trunks for the duration of the call. For this sample configuration, Avaya Meeting Exchange is treated as an external SIP station. Thus, a call from a SIP station registered to Avaya SIP Enablement Services as an off-pbx-telephone on Avaya MultiVantage Express to Avaya Meeting Exchange will use two SIP trunks. A call between a non-SIP station and Avaya Meeting Exchange will use only one SIP trunk.
Leave remaining fields at default values.

<pre>
add trunk-group 12   Page   1 of 21

                                TRUNK GROUP

Group Number: 12   Group Type: sip   CDR Reports: y
  Group Name: 001s6100 SIP   COR: 1   TN: 1   TAC: 812
   Direction: two-way   Outgoing Display? n
 Dial Access? n   Night Service:
Queue Length: 0
Service Type: tie   Auth Code? n

                                                      Signaling Group: 12
                                                    Number of Members: 25
</pre>
</td></tr>
</table>

3.3. Configure Call Routing

The following steps show procedures to enable call routing from Avaya MultiVantage Express to Avaya Meeting Exchange. For this sample configuration, AAR Dialing without FAC is utilized (in conjunction with a route pattern) to route calls over the secure SIP trunk group provisioned in **Step 3.2.7** to Avaya Meeting Exchange. Note that other forms of call routing may be utilized. Refer to [**2**] in **Section 8** for definitions regarding fields in the forms displayed in this section.

Step	Description
3.3.1	Configure the dial plan analysis table. Issue the command "**change dialplan analysis**", and add an entry in the table to access the AAR digit analysis table provisioned in **Step 3.3.3** as displayed. ***Note****: There is an entry that has been previously provisioned that corresponds to extensions (the* ***Call Type*** *field is* ***ext****) utilized by all stations in this sample configuration (see* ***Section 1****,* ***Figure 1****).*

```
change dialplan analysis                                        Page   1 of  12

                             DIAL PLAN ANALYSIS TABLE
                                                            Percent Full:    2

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

Step	Description
3.3.2	Configure a route pattern. Issue the command "**change route-pattern <n>**", where **n** is the number of the route pattern to be administered, and add an entry in the table to utilize the trunk group provisioned in **Step 3.2.7**.

```
change route-pattern 12                                         Page   1 of   3

                    Pattern Number: 12  Pattern Name: 001s6100 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: 12   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
```

Step	Description
3.3.3	Configure the AAR digit analysis table. Issue the command "**change aar analysis <n>**", where **n** is the leading digit of the digit string to be administered, and add an entry in the table to utilize the route pattern provisioned in **Step 3.3.2**.

```
change aar analysis 4                                           Page   1 of   2

                             AAR DIGIT ANALYSIS TABLE
                                                            Percent Full:    2

          Dialed           Total     Route    Call   Node  ANI
          String         Min  Max  Pattern   Type   Num   Reqd
    4                     3    3     12       aar          n
```

4. Avaya Meeting Exchange Configuration

This section describes the steps for configuring Avaya Meeting Exchange to interoperate with Avaya MultiVantage Express.

The administrative steps in this section have been divided into the following sub-sections:

- Verifying licensing on Avaya Meeting Exchange that is required to support the configuration displayed in these Application Notes.
- Configuring connectivity to enable secure SIP/TLS trunking between Avaya Meeting Exchange and Avaya MultiVantage Express.
- Configuring call routing to enable both call mapping on Avaya Meeting Exchange, and outbound dialing from Avaya Meeting Exchange to Avaya MultiVantage Express.
- Provisioning accounts to provide end users access to Avaya Meeting Exchange.

4.1. Verify Licensing

The following steps show procedures to verify licensing on Avaya Meeting Exchange that is required to support the configuration displayed in these Application Notes.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.1.1</td><td>
Avaya Meeting Exchange uses Avaya Web License Manager (WebLM) for licensing. WebLM is a web-based license manager that runs on both Microsoft Windows and UNIX systems. The WebLM server provides a user interface for license administration that can be accessed from a standard web browser over a secure SSL link.

Open a web browser and enter the following URL:
https://<IP Address of Avaya Meeting Exchange>/WebLM
Log in to the WebLM server with the appropriate credentials and click Express_Edition under Licensed Products.
Verify that the Number of Meeting Exchange Express Edition Ports supported by Avaya Meeting Exchange is sufficient.

Note: Each participant in a conference on Avaya Meeting Exchange requires one port for the duration they are in conference. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

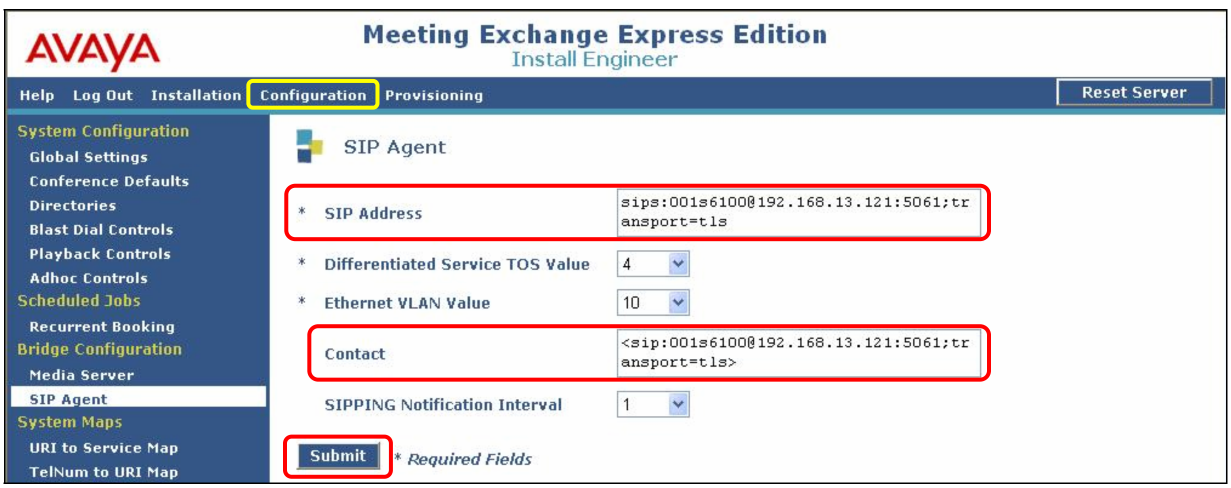
</td></tr>
</table>

4.2. Configure Connectivity

The following steps show procedures to enable secure SIP/TLS connectivity between Avaya Meeting Exchange and Avaya MultiVantage Express.

Avaya Meeting Exchange is administered and maintained using a standard web browser over a secure connection. Administer settings for Avaya Meeting Exchange as follows:

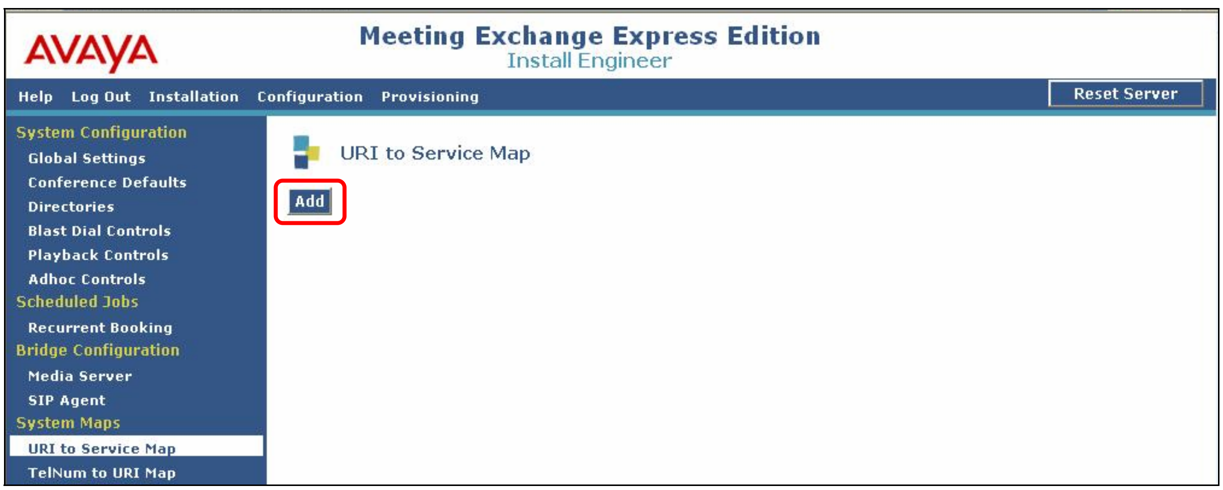
- Open a web browser and enter the following URL:
  https://<IP address of Avaya Meeting Exchange>/mx
- Log in to Avaya Meeting Exchange with the appropriate credentials.

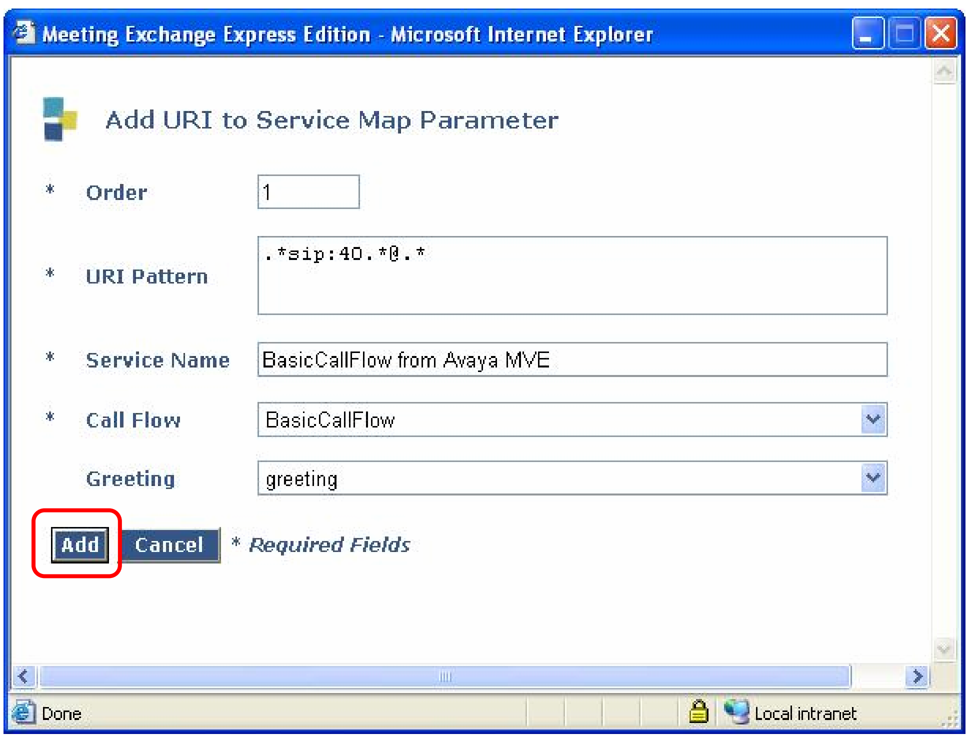
Step	Description
4.2.1	Configure SIP agent settings. Administer settings that enable secure SIP connectivity between Avaya Meeting Exchange and other SIP User Agents as follows: • Click **Configuration** from the web interface toolbar. • Click **SIP Agent** under **Bridge Configuration**. • Enter a SIP URI for Avaya Meeting Exchange that conforms to SIP standards in the **SIP Address** field. This field is used to populate the From header field in SIP INVITE messages from Avaya Meeting Exchange. To enable secure SIP connectivity utilizing TLS, this entry must contain **sips**, **5061** and **transport=tls**. • Enter a Contact header for Avaya Meeting Exchange that conforms to SIP standards in the **Contact** field. This entry provides SIP User Agents a contact to use for acknowledging SIP messages from Avaya Meeting Exchange. To enable secure SIP connectivity utilizing TLS, the entry must contain **5061** and **transport=tls**. • Leave remaining fields at default values. • When finished, click the **Submit** button to add the configuration to the database. 

4.3. Configure Call Routing

The following steps show procedures to enable call routing for Avaya Meeting Exchange. On Avaya Meeting Exchange, call routing is defined by Service Maps as follows:

- For inbound calls (to Avaya Meeting Exchange), URI to service map(s) are utilized. These maps associate calls to Avaya Meeting Exchange with corresponding call flows, thus allowing for specific treatment for a conference based on incoming SIP URI(s).
- For outbound calls (from Avaya Meeting Exchange), TelNum to URI map(s) are utilized. These maps associate a telephone number pattern with a corresponding SIP URI, thus allowing call origination from Avaya Meeting Exchange to add participant(s) to a conference.

Step	Description
4.3.1	Add a URI to service map entry. To associate incoming calls to Avaya Meeting Exchange with a call flow, add a URI to service map entry as follows: • Click **URI to Service Map** under **System Maps**. • Click the **Add** button. 

Step	Description
4.3.2	From the **Add URI to Service Map Parameter** screen, administer settings for a basic call flow as follows: • Leave the **Order** field at the default value. *__Note__: Avaya Meeting Exchange parses URI to service map entries for pattern matches in descending order, terminating the search once a pattern is matched. For this sample configuration, order is irrelevant as the patterns for basic call flow and direct call flow (see __Step 4.3.3__) are mutually exclusive.* • Enter a rule in the **URI Pattern** field to match the pattern of incoming Request URIs in SIP INVITE messages from Avaya MultiVantage Express. Regular expression metacharacters such as **.** (matches any one character) or **.*** (matches any series of zero or more characters) may be utilized. o For example, assume Avaya MultiVantage Express sends the following URI: **sip:401@192.168.13.121:5061;transport=tls SIP/2.0**. The **URI Pattern .*sip:40.*@.*** would match **sip:40** then zero or more characters followed by **@** then zero or more characters. • Select a **Call Flow** from the drop down menu. To allow access to conferences via passcode, **BasicCallFlow** was selected. • Enter a descriptive name for this map in the **Service Name** field. • Select a **Greeting** from the drop down menu. • When finished, click the **Add** button to add the map to the database. 

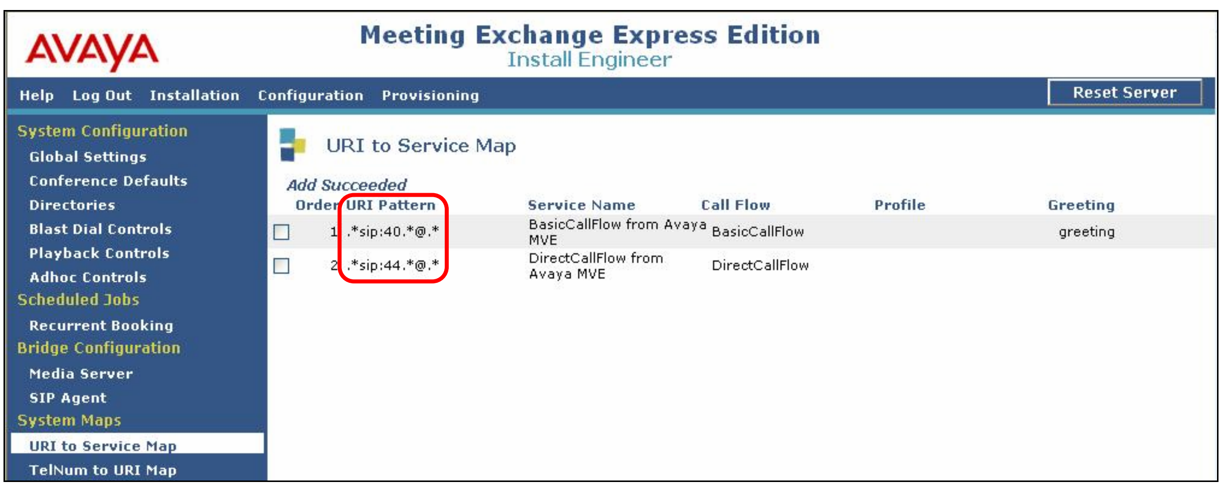
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.3.3</td><td>
To associate incoming calls to Avaya Meeting Exchange with a direct call flow, repeat Steps 4.3.1 - 4.3.2 to add a URI to service map entry for a direct call flow with the following parameters:

Leave the Order field at the default value.
Enter .*sip:44.*@.* in the URI Pattern field to match the pattern of incoming Request URIs in SIP INVITE messages from Avaya MultiVantage Express.
Note: To enable access to Avaya Meeting Exchange via a direct call flow, the following are required:

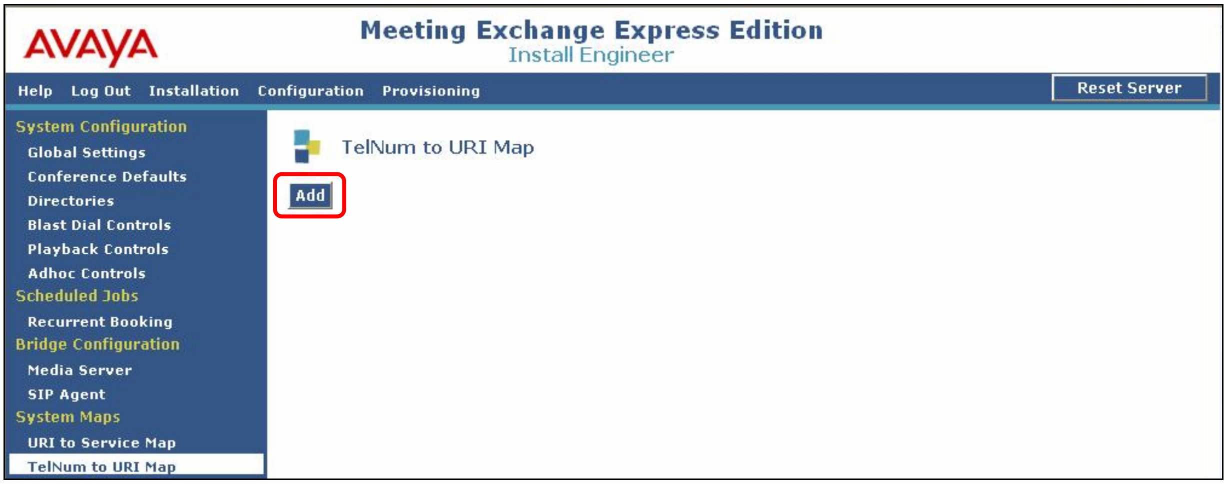
Call routing on Avaya MultiVantage Express must be provisioned (see Section 3.3) to send digits that match the URI Pattern field for the direct call flow.
A conference must be provisioned with a moderator passcode (see Step 4.4.4) that matches the URI Pattern for the direct call flow.

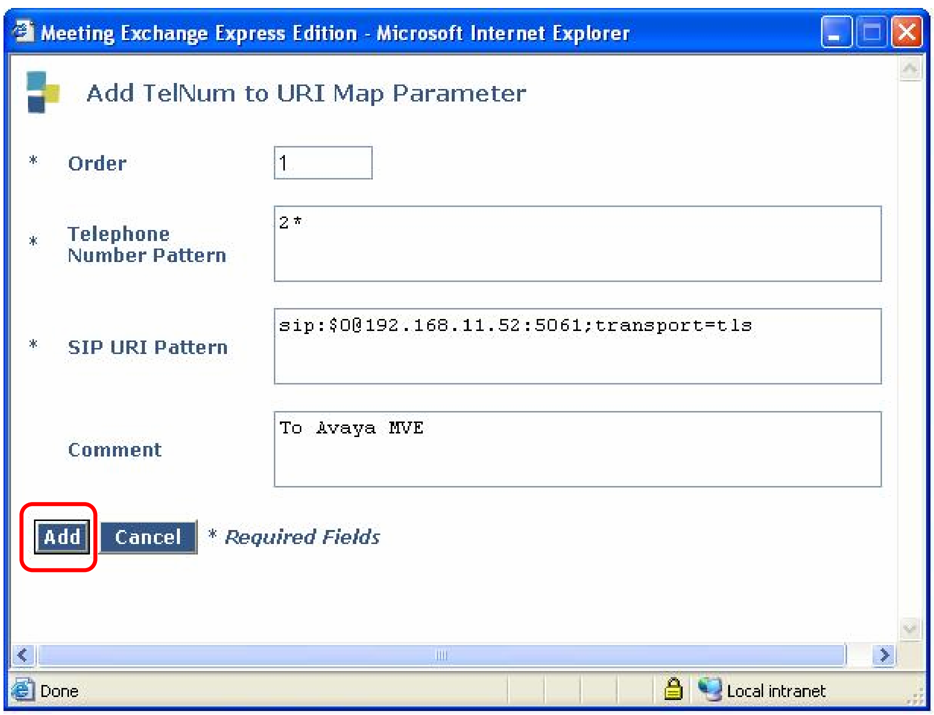
Select DirectCallFlow from the Call Flow drop down menu to allow access to a conference as moderator, without entering a passcode.
Enter a descriptive name for this map in the Service Name field.
The resulting URI to service map list is displayed below.

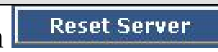
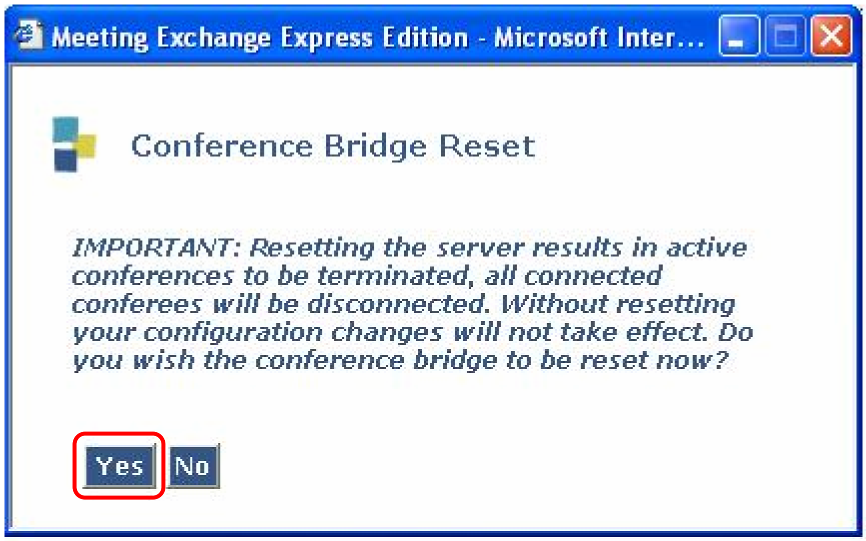
Note: Setting a greeting for a direct call flow is not necessary. A call to Avaya Meeting Exchange that matches a conference associated with a direct call flow will inherit that conference's settings (including a greeting). Also, note that the provisioning for the URI Pattern fields for both the direct call flow and basic call flow (see Step 4.3.2) utilize wild cards that make both fields mutually exclusive while maximizing the breadth of the pattern match. For example, the URI Pattern field for the direct call flow is .*sip:44.*@.*. This aligns with the provisioning for call routing on Avaya MultiVantage Express in Section 3.3 and allows 44X (where X can be any digit) to match this direct call flow.



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

Step	Description
4.3.4	Add a TelNum to URI map entry. To enable routing of outbound calls from Avaya Meeting Exchange, add a TelNum to URI map entry as follows: • Click **TelNum to URI Map** under **System Maps**. • Click the **Add** button. 

Step	Description
4.3.5	From the **Add TelNum to URI Map Parameter** screen, administer settings to enable outbound calling to Avaya MultiVantage Express as follows: • Leave the **Order** field at the default value. *__Note__: Avaya Meeting Exchange parses TelNum to URI map entries for pattern matches in descending order, terminating the search once a pattern is matched. For this sample configuration, order is irrelevant as there is only one entry in the database.* • Enter a rule in the **Telephone Number Pattern** field that matches the administration on Avaya MultiVantage Express regarding station extensions (see **Step 3.3.1**). Shell pattern matching metacharacters such as * (refers to a character string) or ? (refers to a single character) may be utilized. • Enter a rule in the **SIP URI Pattern** field that conforms to SIP standards to allow outbound calling from Avaya Meeting Exchange. To enable secure SIP connectivity utilizing TLS for outbound calls to Avaya MultiVantage Express, the pattern must contain **5061** and **transport=tls**. The shell metacharacter, **$0** is replaced by the entire **Telephone Number Pattern** at the location of **$0** in the **SIP URI Pattern**. o For example, if **21001** is the dialed string, Avaya Meeting Exchange will send a SIP INVITE message with a SIP URI and To header field formatted as follows: **sip:21001@192.168.11.52:5061;transport=tls** • Enter a descriptive name for this map in the **Comment** field. • When finished, click the **Add** button to add the map to the database. 

Step	Description
4.3.6	Apply the configuration by clicking the **Reset Serve**r button  located on the right hand side of the web interface toolbar. Confirm this action by clicking **Yes** in the pop up window. 

4.4. Provision Accounts

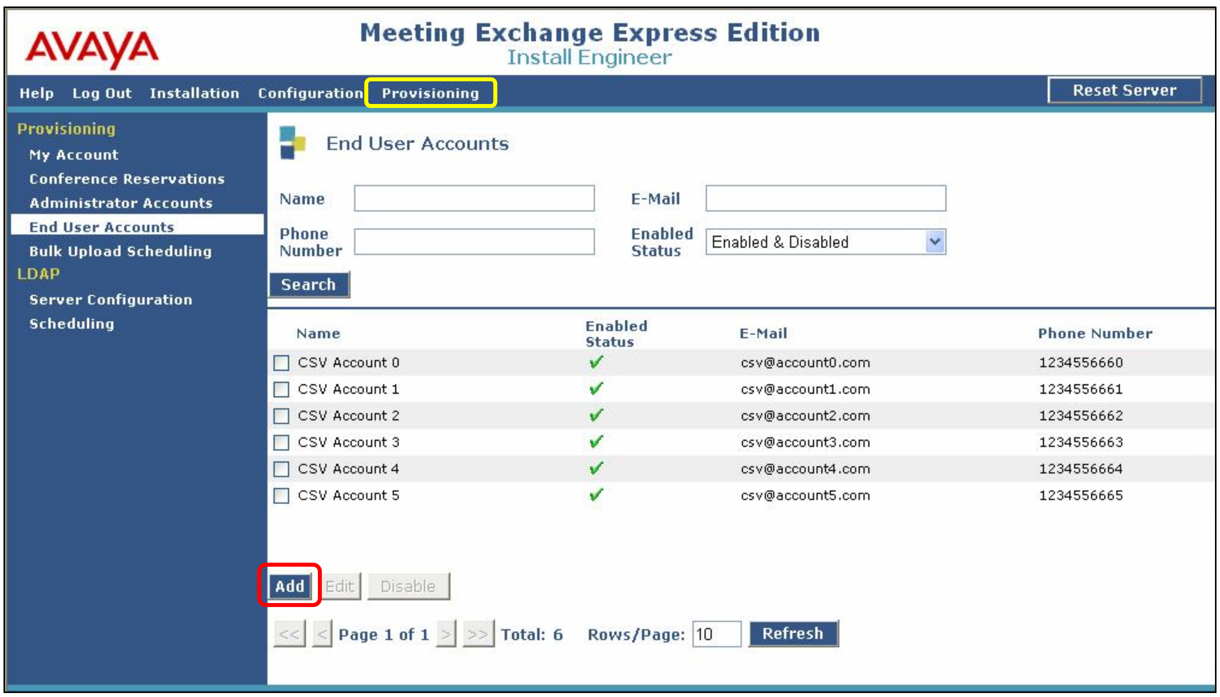
The following steps present an example of provisioning an end user account and associated conference reservation on Avaya Meeting Exchange.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.4.1</td><td>Add an end user account.

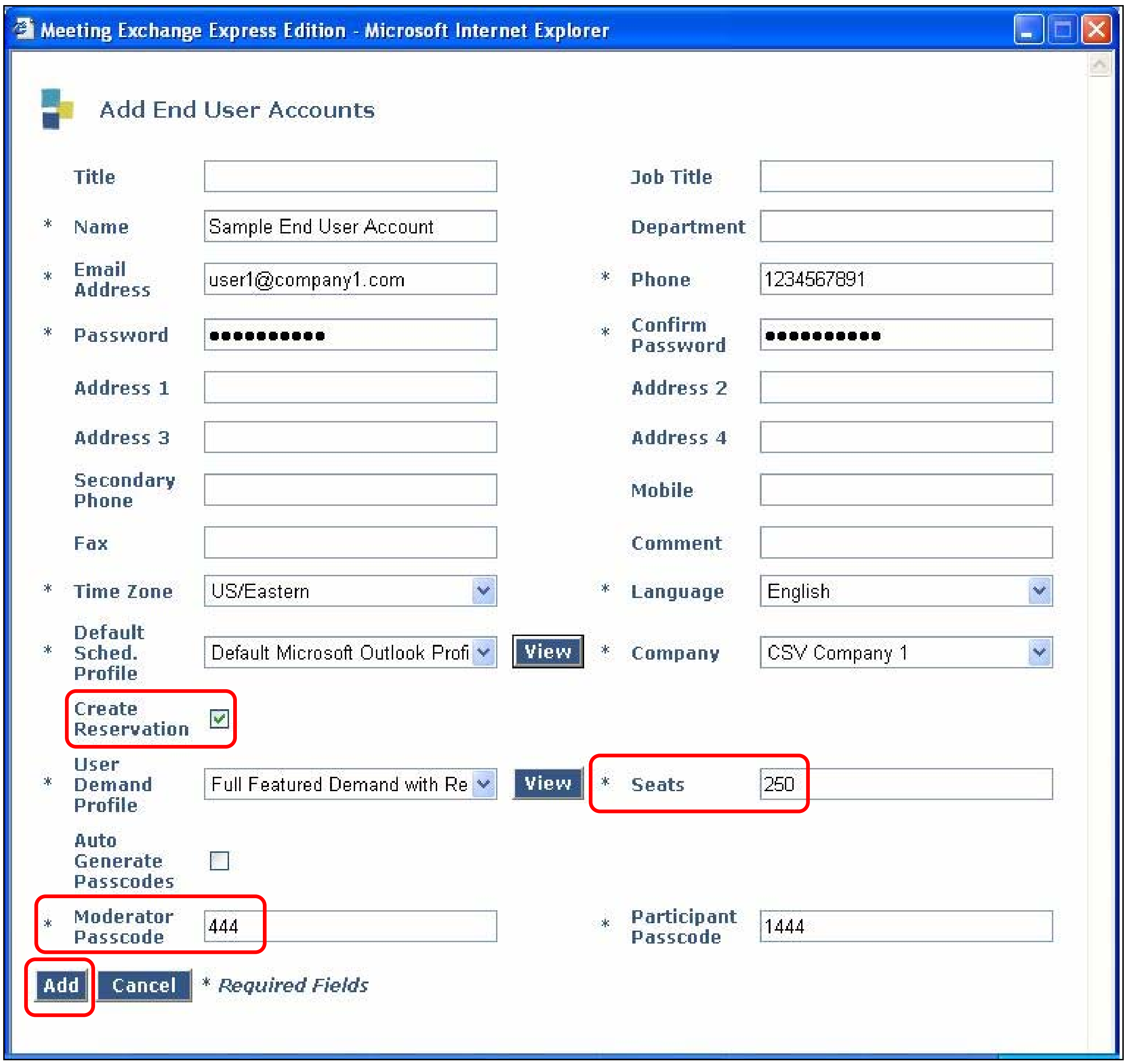
To provide end users access to the conferencing features available on Avaya Meeting Exchange, add an end user account as follows:

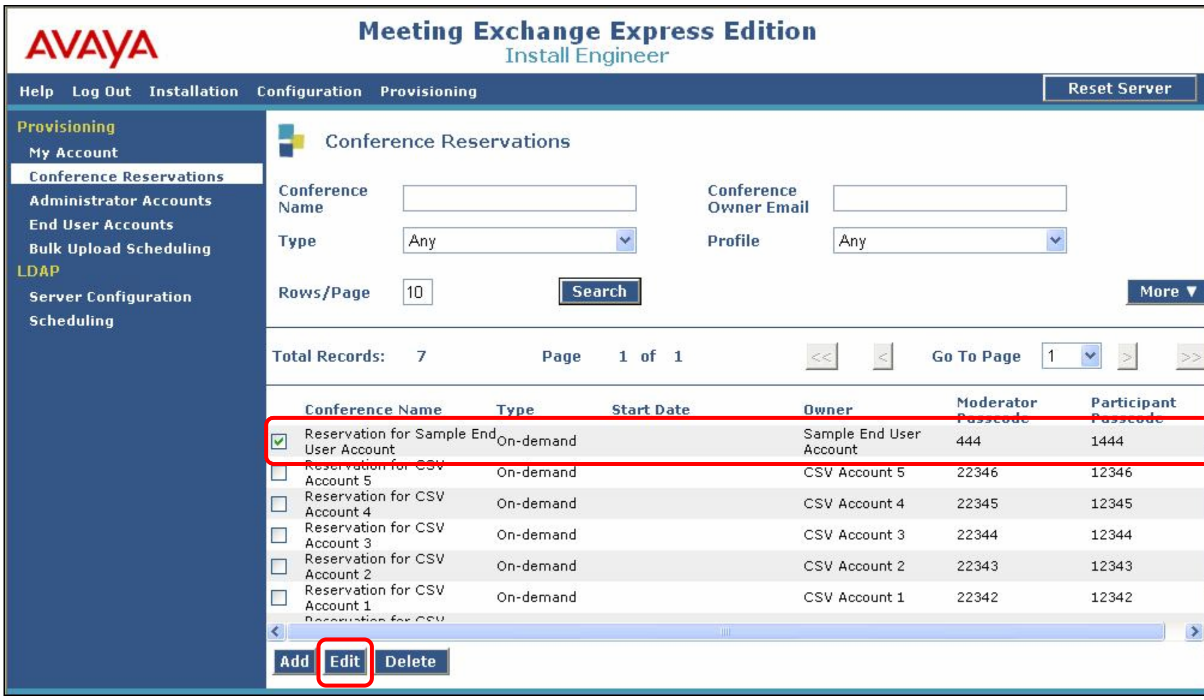
Click Provisioning from the web interface toolbar.
Click End User Accounts under Provisioning.
Click the Add button.

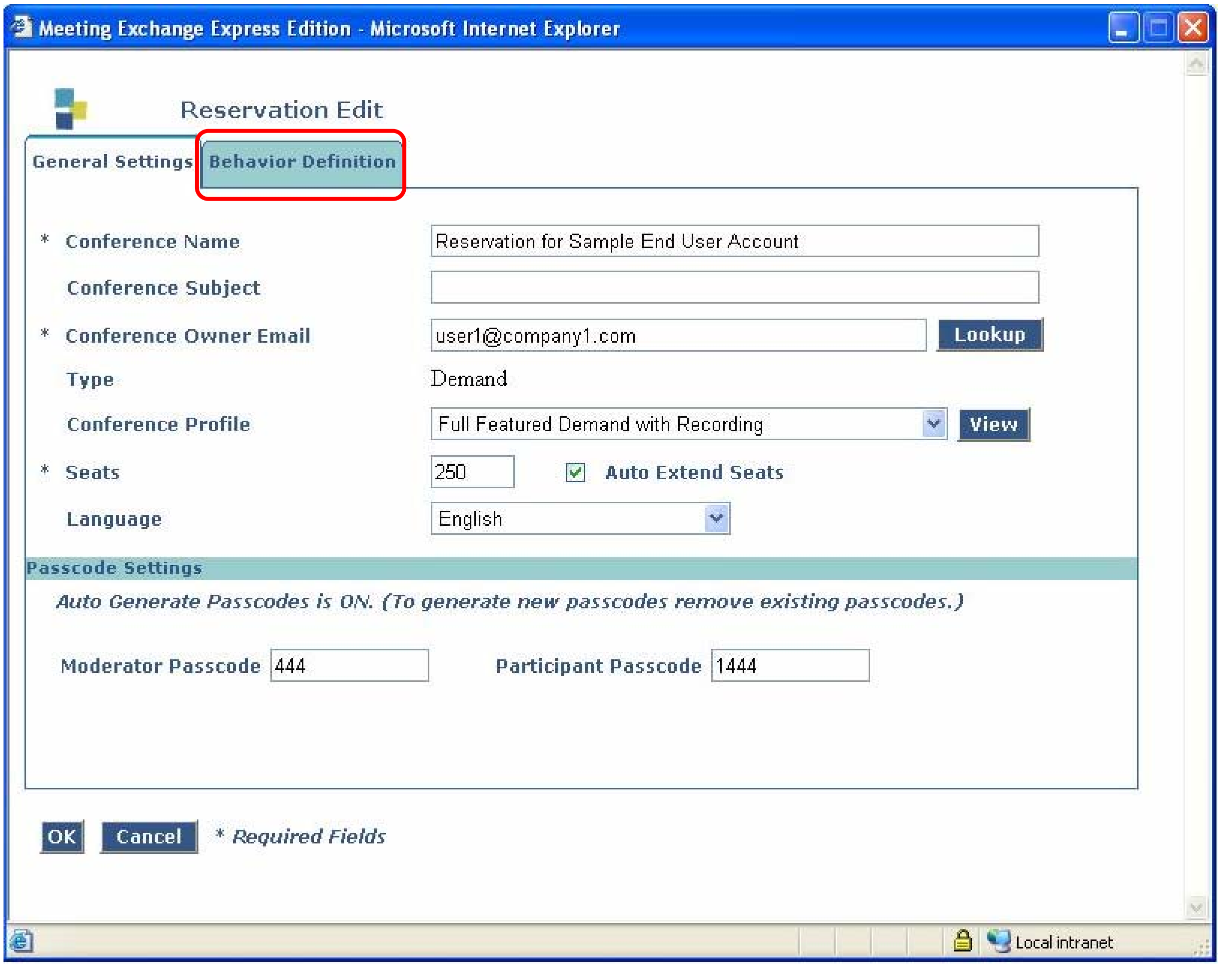
Note: Avaya Meeting Exchange comes with pre-provisioned accounts: CSV Account 0 - CSV Account 5.

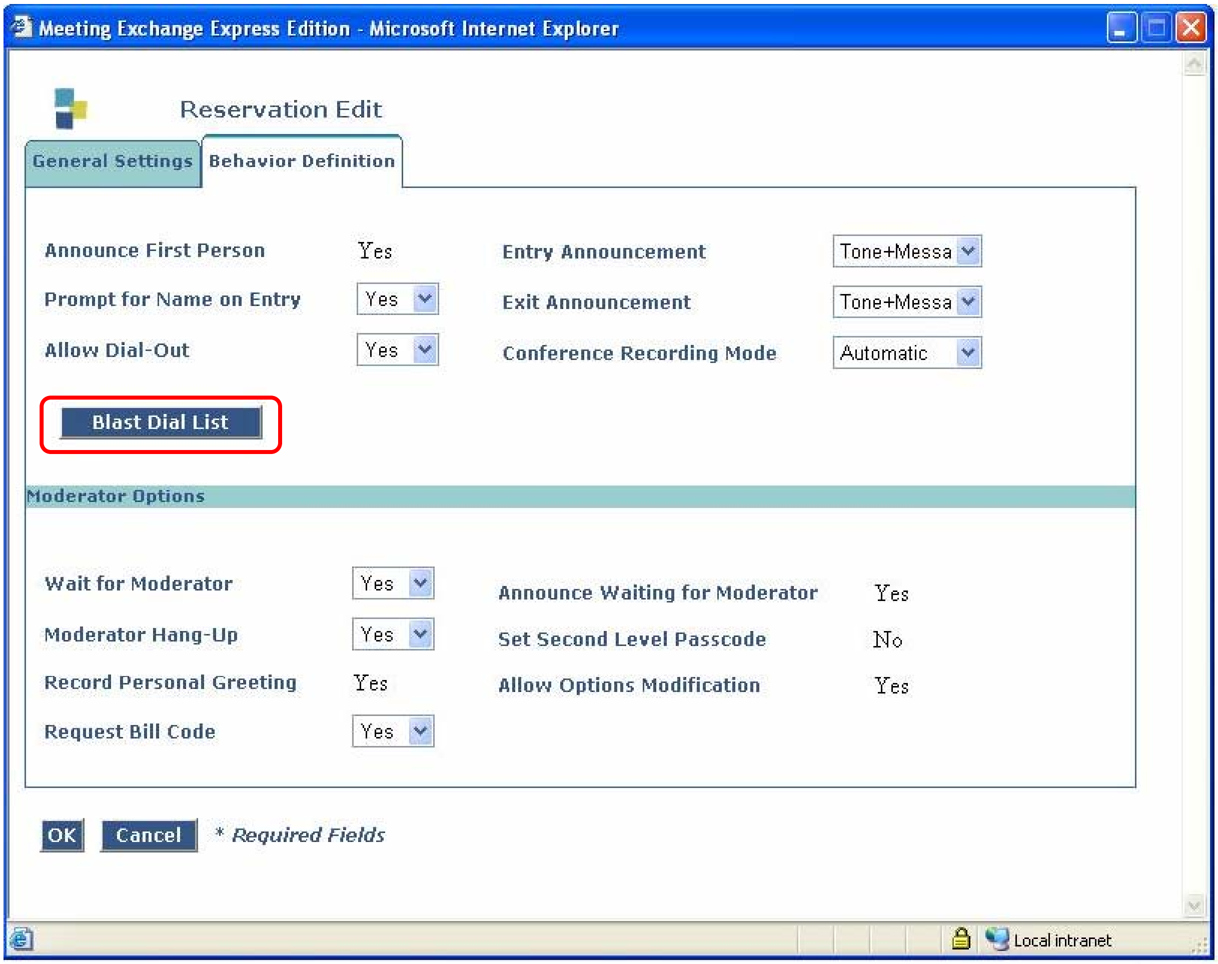


</td></tr>
</table>

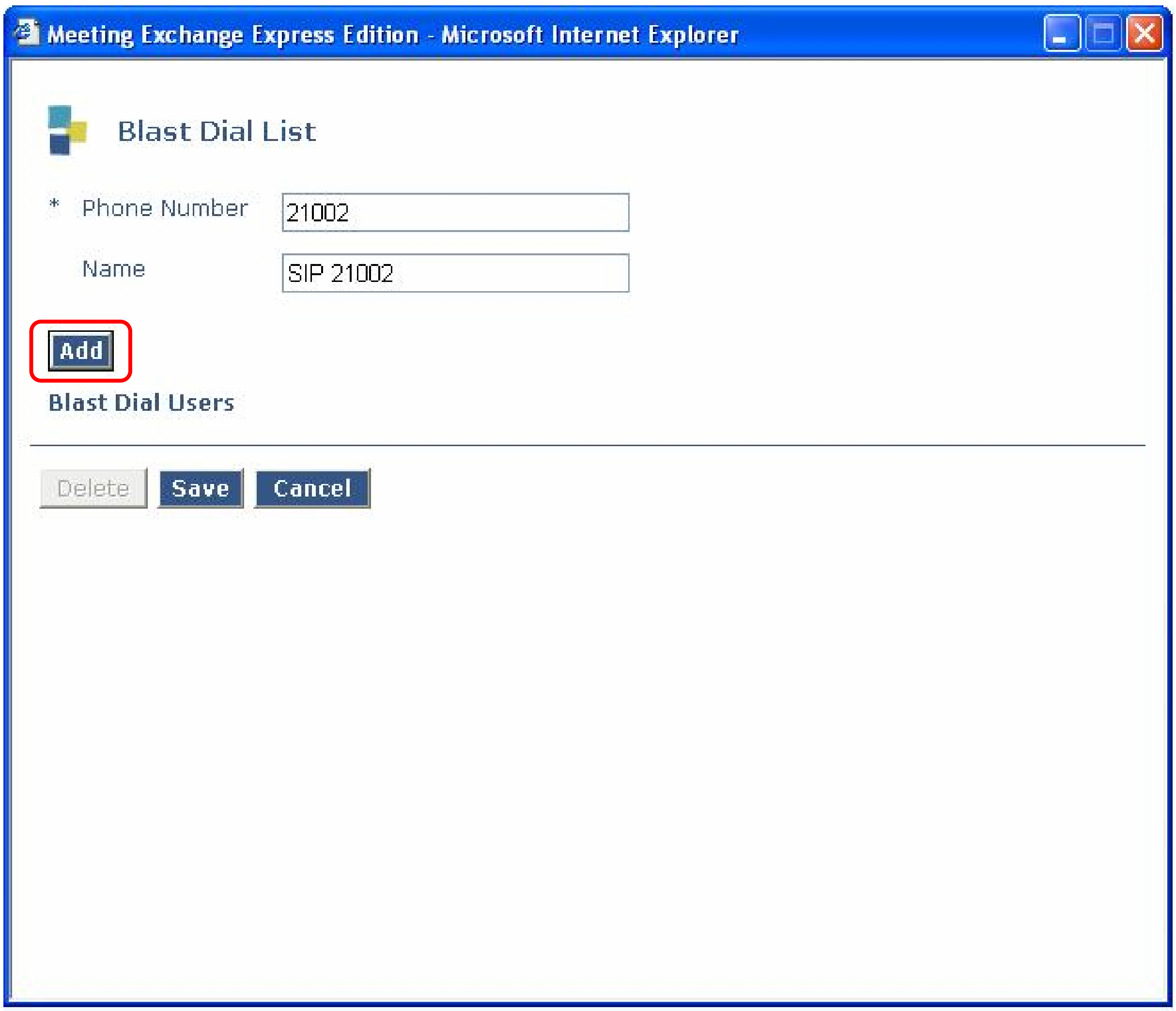
Step	Description
4.4.2	From the **Add End User Accounts** screen, provision an end user account as follows: • Enter a passcode in the **Moderator Passcode** field that corresponds to the direct call flow provisioned in **Step 4.3.3**. • Enter the number of ports assigned to this conference in the **Seats** field. • Refer to **Section 8** in [**3**] for definitions regarding the required fields on this screen. • When finished, click the **Add** button to add the account to the database. ***Note**: If the **Create Reservation** field is checked, a demand conference reservation is automatically generated (see **Step 4.4.3**) and associated with this end user account.* 

Step	Description
4.4.3	Modify the conference reservation corresponding to the end user account provisioned in **Step 4.4.2** as follows: • Click **Conference Reservations** under **Provisioning**. • Check the conference reservation corresponding to the end user account provisioned in **Step 4.4.2**. • Click the **Edit** button. 

Step	Description
4.4.4	The configuration displayed in the **General Settings** tab for this conference reservation is correlated with the configuration entered for the end user account provisioned in **Step 4.4.2**. Any updates made in this screen will be reflected in the corresponding end user account and vice-versa. To modify parameters associated with this conference reservation, click the **Behavior Definition** tab. 

Step	Description
4.4.5	The configuration displayed in the **Behavior Definition** tab may be modified to suit the requirements for this conference. For this sample configuration, a blast dial list was provisioned. To configure a blast dial list, click the **Blast Dial List** button. 

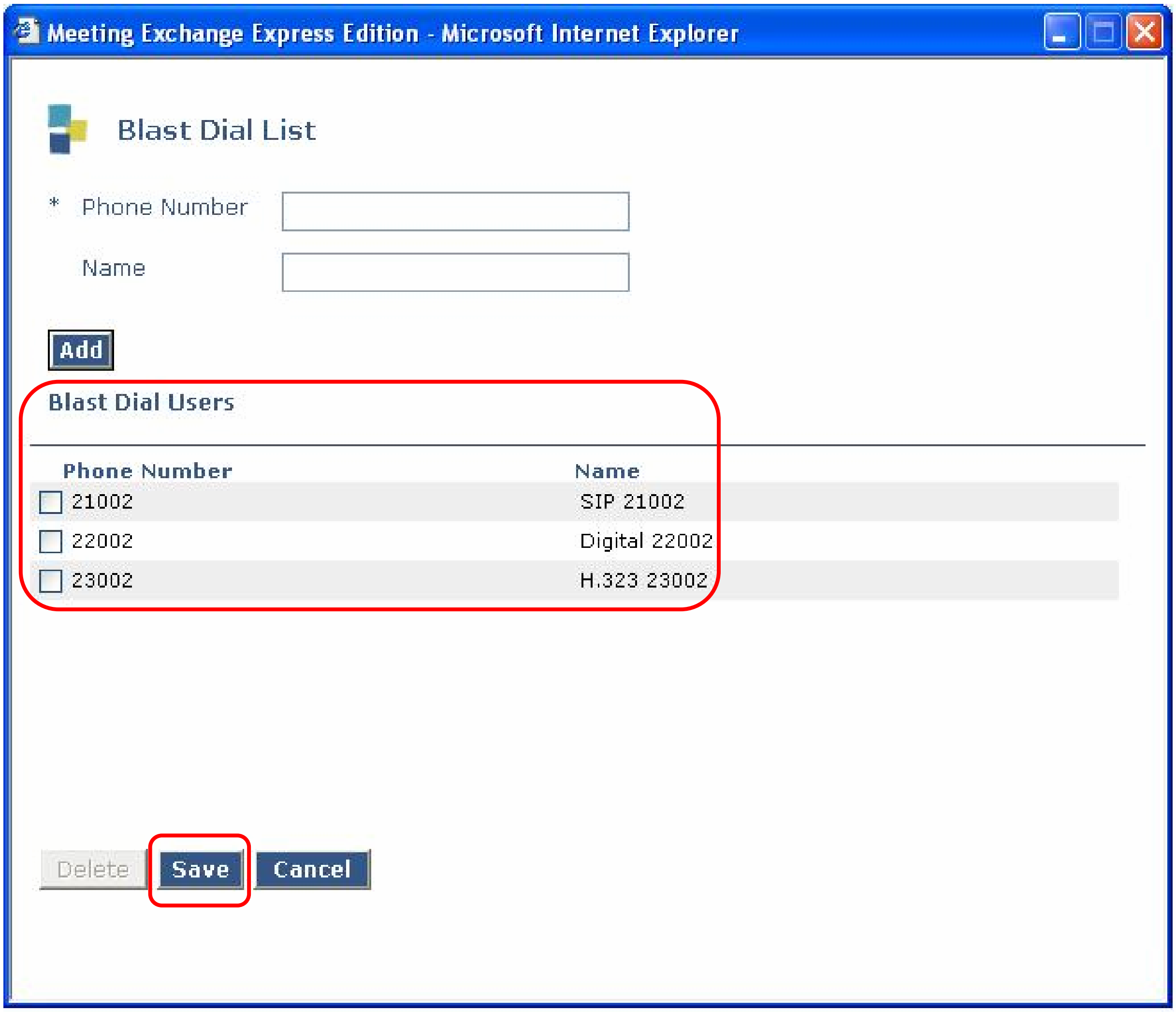
Step	Description

4.4.6	From the **Blast Dial List** screen, add entries to the blast dial list as follows:
	
	• Enter a number in the **Phone Number** field that is associated with both:
	    o The telephone number pattern provisioned for the TelNum to URI map in **Step 4.3.5**.
	    o Telephones registered to either Avaya MultiVantage Express or Avaya SIP Enablement Services.
	• Enter a descriptive name for this phone number in the **Name** field.
	• Click the **Add** button to add entries to this blast dial list.
	
	

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.4.7</td><td>Repeat Step 4.4.6 to add additional phone numbers to the blast dial list.</td></tr>
<tr><td>4.4.8</td><td>The resultant blast dial list is displayed below.

Click the save button to save and associate this blast dial list with this conference.
[Not Shown] Click the OK button 

(displayed in the lower left hand corner of the Behavior Definition tab in Step 4.4.5) to save the modifications to this conference in the database.



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

5. Interoperability Compliance Testing

5.1. General Test Approach

The general test approach was to place calls between Avaya MultiVantage Express and Avaya Meeting Exchange utilizing the sample network configuration displayed in **Section 1**, **Figure 1**. The main objectives were to verify the following:

- Inbound calling from Avaya MultiVantage Express to both scheduled and demand conferences provisioned on Avaya Meeting Exchange via:
  - Direct call flow (<u>without</u> participant-access-code).
  - Basic call flow (<u>with</u> participant-access-code).
- Outbound calling from Avaya Meeting Exchange to stations registered to either Avaya MultiVantage Express or Avaya SIP Enablement Services via:
  - Blast dial to a pre-provisioned blast dial list.
  - Originator dial-out.
- The following feature testing was executed:
  - Conference features (as defined in [**3**] in **Section 8**) for both moderators and participants accessed during a conference call via touchtone commands.
  - Avaya MultiVantage Express related features, e.g., call hold, call transfer, three-way conference.
- Shuffling or direct IP-to-IP audio connectivity for IP stations, e.g., SIP, H.323.

  Note*: Shuffling is defined as the rerouting of the audio connection between two IP stations. 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 stations*.
- The following transport methods for signaling were tested between Avaya Meeting Exchange and Avaya MultiVantage Express:
  - SIP/TCP.
  - SIP/TLS.
- The following audio codec(s) were tested:
  - G711MU.
- Voice quality was verified as follows:
  - Subjectively, utilizing stations registered to either Avaya MultiVantage Express or Avaya SIP Enablement Services participating in a conference.
  - Empirically, utilizing Perceptual Evaluation of Speech Quality (PESQ) scores obtained from emulated SIP stations on an Empirix FX-IP test set registered to Avaya SIP Enablement Services as an off-pbx-telephone on Avaya MultiVantage Express.

5.2. Test Results

- All test cases, as defined by the general test approach, passed.

6. Verification Steps

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

- The SIP trunking between Avaya MultiVantage Express and Avaya Meeting Exchange configured in **Section 3.2** and **Section 4.2** respectively (verified in **Step 6.1.1**).
- The signaling and media connectivity between Avaya MultiVantage Express and Avaya Meeting Exchange:
    - Verified for inbound calls to Avaya Meeting Exchange from stations registered to either Avaya MultiVantage Express or Avaya SIP Enablement Services in **Steps 6.1.2 - 6.1.3**.
    - Verified for outbound calls from Avaya Meeting Exchange to stations registered to either Avaya MultiVantage Express or Avaya SIP Enablement Services in **Steps 6.1.4 - 6.1.5**.

Step	Description
6.1.1	Verify all members for the SIP trunk group provisioned in **Step 3.2.7** are **in-service/idle**. From a SAT session: • Issue the command “**status trunk <n>**”, where **n** is the number of the trunk group to verify. • Verify that all members in the trunk group are **in-service/idle**.

Step	Description
6.1.2	Validate signaling and media connectivity for inbound calls to Avaya Meeting Exchange from Avaya MultiVantage Express. This is accomplished by verifying that the SIP trunk provisioned in **Step 3.2.7** is utilized when a call from a phone registered to either Avaya MultiVantage Express or Avaya SIP Enablement Services dials in to a conference provisioned on Avaya Meeting Exchange. From a SAT session: • Issue the command "**list trace tac <n>**", where **n** is the TAC defined for the trunk group provisioned in **Step 3.2.7**. • From a SIP station registered to Avaya SIP Enablement Services as an off-pbx-telephone on Avaya MultiVantage Express, dial **444** to enter the conference provisioned in **Section 4.4** as moderator via the direct call flow provisioned in **Step 4.3.3**. ***Note***: *The trace below shows a SIP station (**21001**) dialing in to Avaya Meeting Exchange as moderator via a direct call flow (**444**). This trace also shows direct audio connectivity between the SIP station (**192.168.12.201**) and Avaya Meeting Exchange (**192.168.13.121**) utilizing **G.711MU**. A SIP station was arbitrarily selected to place the call, as the configuration presented in these Application Notes allows any station or trunk type (e.g., SIP, H.323, Digital or Analog) on Avaya MultiVantage Express access to Avaya Meeting Exchange via secure SIP connectivity*.

```
list trace tac 812                                                    Page   1

                                LIST TRACE

time            data

15:10:17     Calling party station    21001 cid 0x139
15:10:17     Calling Number & Name 21001 SIP 21001
15:10:17     dial 444 route:AAR
15:10:17     term trunk-group 12    cid 0x139
15:10:17     dial 444 route:AAR
15:10:17     route-pattern  12 preference 1  cid 0x139
15:10:17     seize trunk-group 12 member 8  cid 0x139
15:10:17     Calling Number & Name NO-CPNumber NO-CPName
15:10:17     Setup digits 444
15:10:17     Calling Number & Name NO-CPNumber SIP 21001
15:10:17     Proceed trunk-group 12 member 8  cid 0x139
15:10:17     active trunk-group 12 member 8  cid 0x139
15:10:17     G711MU ss:off ps:20 rn:12/1 192.168.13.121:41426 192.168.11.62:2050
15:10:17     xoip: fax:Relay modem:off tty:US 192.168.11.62:2050 uid:0x5002d
15:10:17     G711MU ss:off ps:20 rn:12/1 192.168.13.121:41426 192.168.12.201:34008
15:10:17     G711MU ss:off ps:20 rn:1/12 192.168.12.201:34008 192.168.13.121:41426
```

Step	Description

6.1.3	For additional information regarding the active call to Avaya Meeting Exchange (initiated in **Step 6.1.2**), status the trunk group member obtained from the trace in **Step 6.1.2**.
	
	From a SAT session:
	• Issue the command "**status trunk t/m**", where **t** is the trunk group and **m** is the trunk group member. Note the following:
	  o The **Audio Connection Type** field returns **ip-direct**. This indicated a direct audio connection for this trunk group member.
	  o The signaling connection is between the procr (**192.168.11.52**) on Avaya MultiVantage Express and Avaya Meeting Exchange (**192.168.13.121**).
	  o The audio connection utilizes **G.711MU**, and is between a SIP station registered to Avaya SIP Enablement Services as an off-pbx-telephone on Avaya MultiVantage Express (**192.168.12.201**) and Avaya Meeting Exchange (**192.168.13.121**).
	
	***Note**: An **Audio Connection Type** field returning **ip-tdm** would indicate that direct IP-to-IP audio connectivity is <u>not</u> enabled.*

```
status trunk 12/8                                                 Page   1 of   2

                              TRUNK STATUS

 Trunk Group/Member: 0012/008             Service State: in-service/active
              Port: T00045             Maintenance Busy? no
 Signaling Group ID:



    Connected Ports: T00011




                  Port     Near-end IP Addr : Port    Far-end IP Addr : Port
        Signaling: 01A0017  192.168. 11. 52  : 5060    192.168. 13.121 : 5060

G.711MU     Audio:          192.168. 12.201  : 34008   192.168. 13.121 : 41426
            Video:
      Video Codec:
                                              Authentication Type: None
    Audio Connection Type: ip-direct
```

Step	Description
6.1.4	Validate signaling and media connectivity for outbound calls from Avaya Meeting Exchange to Avaya MultiVantage Express. This is accomplished by verifying that the SIP trunk provisioned in **Step 3.2.7** is utilized when a call from a phone in conference on Avaya Meeting Exchange adds a participant registered to either Avaya MultiVantage Express or Avaya SIP Enablement Services to the conference via dial-out procedures. From a SAT session: • Issue the command "**list trace tac <n>**", where **n** is the TAC defined for the trunk group provisioned in **Step 3.2.7**. • From a station in a conference on Avaya Meeting Exchange, enter the appropriate touchtone command (for this sample configuration *9) to invoke a blast dial to the blast dial list provisioned in **Section 4.4**. ***Note**: The trace below shows a call (originating from Avaya Meeting Exchange) to a SIP station registered to Avaya SIP Enablement Services as an off-pbx-telephone on Avaya MultiVantage Express. This trace also shows direct audio connectivity between the SIP station (**192.168.12.202**) and Avaya Meeting Exchange (**192.168.13.121**) utilizing **G.711MU**.*

```
list trace tac 812                                                      Page   1

                                LIST TRACE

time            data

15:15:35     Calling party trunk-group 12 member 1  cid 0x13c
15:15:35     Calling Number & Name NO-CPNumber NO-CPName
15:15:35     active trunk-group 12 member 1  cid 0x13c
15:15:35     G711MU ss:off ps:20 rn:12/1 192.168.13.121:41430 192.168.11.62:2058
15:15:35     xoip: fax:Relay modem:off tty:US 192.168.11.62:2058 uid:0x50024
15:15:35     dial 21002
15:15:35     term station    21002 cid 0x13c
15:15:37     active station    21002 cid 0x13c
15:15:37     G711MU ss:off ps:20 rn:12/1 192.168.13.121:41430 192.168.12.202:340
15:15:37     G711MU ss:off ps:20 rn:1/12 192.168.12.202:34008 192.168.13.121:41430
```

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.1.5</td><td>For additional information regarding the active call from Avaya Meeting Exchange initiated in Step 6.1.4, status the trunk group member obtained from the trace in Step 6.1.4.

From a SAT session:
Issue the command “status trunk t/m”, where t is the trunk group and m is the trunk group member. Note the following:
The Audio Connection Type is ip-direct.
The signaling connection is between procr (192.168.11.52) on Avaya MultiVantage Express and Avaya Meeting Exchange (192.168.13.121).
The audio connection utilizes G.711MU and is between the SIP station registered to Avaya SIP Enablement Services as an off-pbx-telephone on Avaya MultiVantage Express (192.168.12.202) and Avaya Meeting Exchange (192.168.13.121).
<pre>
status trunk 12/1   Page   1 of   2

                              TRUNK STATUS

 Trunk Group/Member: 0012/001   Service State: in-service/active
              Port: T00036   Maintenance Busy? no
 Signaling Group ID:

    Connected Ports: T00027

                   Port   Near-end IP Addr : Port   Far-end IP Addr : Port
        Signaling: 01A0017 192.168. 11. 52 : 5060   192.168. 13.121 : 5060

G.711MU   Audio: 192.168. 12.202 : 34008   192.168. 13.121 : 41430
            Video:
      Video Codec:
                                             Authentication Type: None
    Audio Connection Type: ip-direct
</pre></td></tr>
<tr><td>6.1.6</td><td>To verify direct IP-to-IP audio connectivity between H.323 stations registered to Avaya MultiVantage express and Avaya Meeting Exchange repeat Steps 6.1.2 - 6.1.5.</td></tr>
</table>

Step	Description
6.1.7	Verify that calls to and from Avaya Meeting Exchange are managed correctly, e.g., callers are added/removed from conferences. This is verified by the following procedures: • Log in to the Avaya Meeting Exchange server console with the appropriate credentials. • At the command prompt, enter the command: **watch -t -n 5 -d "ipinfo -l |egrep -ci active"** o This command provides a real time, continuous update of port utilization on Avaya Meeting Exchange.

7. Conclusion

These Application Notes presented a compliance-tested solution comprised of Avaya MultiVantage Express and Avaya Meeting Exchange Express Edition. This solution enables connectivity between Avaya MultiVantage Express and Avaya Meeting Exchange Express Edition utilizing secure standards based SIP connectivity via TLS/TCP.

8. Additional References

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

[1] *Installing and Maintaining Avaya MultiVantage Express*, Issue 2, Doc ID: 03-601204, November 2006.

[2] *Administrator Guide for Avaya Communication Manager*, Issue 2.1, Doc ID: 03-300509, May 2006.

[3] *Avaya Meeting Exchange Express Edition Release 1.5 Administration and Maintenance Guide*, Issue 1, Doc ID: 04-601909, March 2007.

[4] *Avaya Meeting Exchange Express Edition Release 1.5 Installation and Configuration Guide*, Issue 1, Doc ID: 04-601898, March 2007.

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