Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Meeting Exchange Express 2.0 with Avaya Aura™ Session Manager R5.2 – Issue 1.0

Abstract

These Application Notes present a sample configuration for interoperability between Avaya Meeting Exchange Express and Avaya Aura™ Session Manager. The Avaya Meeting Exchange Express is a standalone single server SIP audio conference solution. A SIP trunk was configured between the Avaya Meeting Exchange Express and Avaya Aura™ Session Manager. The compliance testing covered access to the conferencing services provided by Avaya Meeting Exchange Express. Testing was conducted at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the sample configuration steps required for interoperability between the Avaya Meeting Exchange Express 2.0 Conference Bridge and Avaya Aura™ Session Manager 5.2. The test cases focused on conferencing features which are available via the Telephone User Interface (TUI). Both In-Band and Out-Of-Band DTMF were verified during TUI menu access. All calls to and from Avaya Meeting Exchange Express are routed via the SIP trunk from Avaya Aura™ Session Manager.

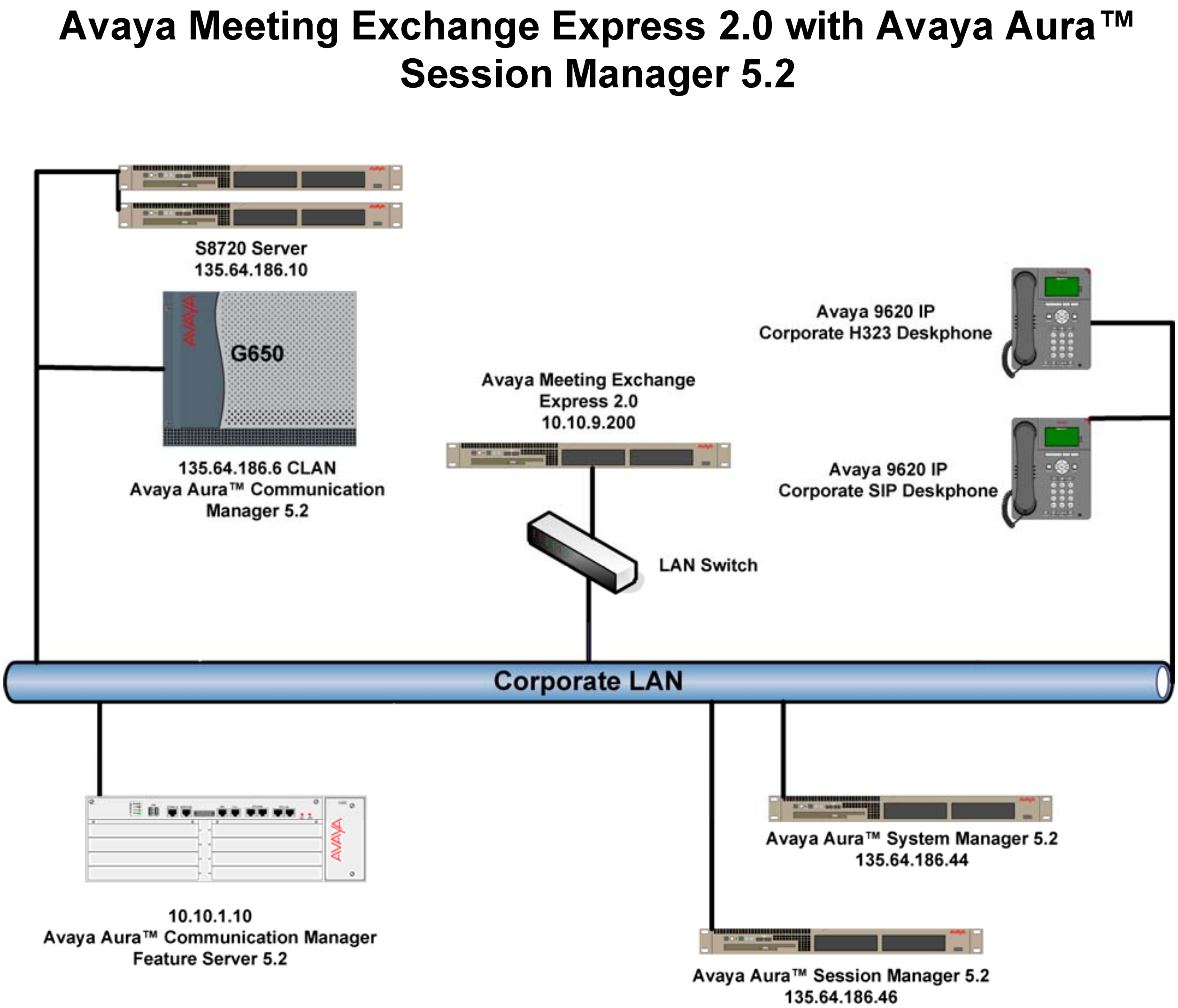


Figure 1: Avaya Meeting Exchange Express and Avaya Aura™ Session Manager

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

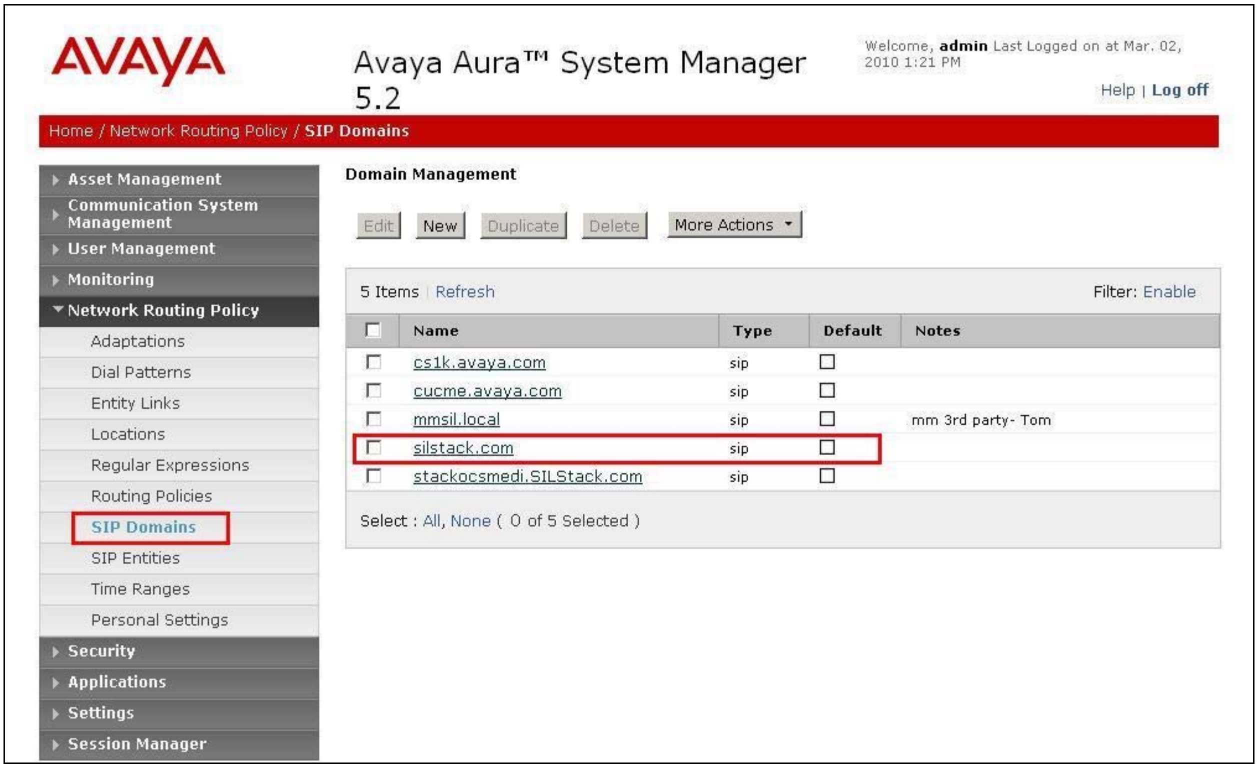
Equipment	Software
Avaya S6100 Server	Avaya Meeting Exchange Express 2.0 (3.0.0.36)
Avaya S8720 Server	Avaya Aura™ Communication Manager Access Element 5.2.1 SP1 02.1.016.4-18111
Avaya G650 Media Gateway	• TN2312BP HW15 FW049 • TN2602AP HW08 FW049 • TN799DP HW01 FW034
Avaya S8510 Server	Avaya Aura™ System Manager R5.2 5.2.7.0 SP 1.1.0.0.8
Avaya S8510 Server	Avaya Aura™ Session Manager R5.2 5.2.0.1.520017 SP0
Avaya Telephones 9620 SIP 9650 H.323 2420 Digital	 R2.6.2.8 R3.1 NA

3. Configure Avaya Aura™ Session Manager

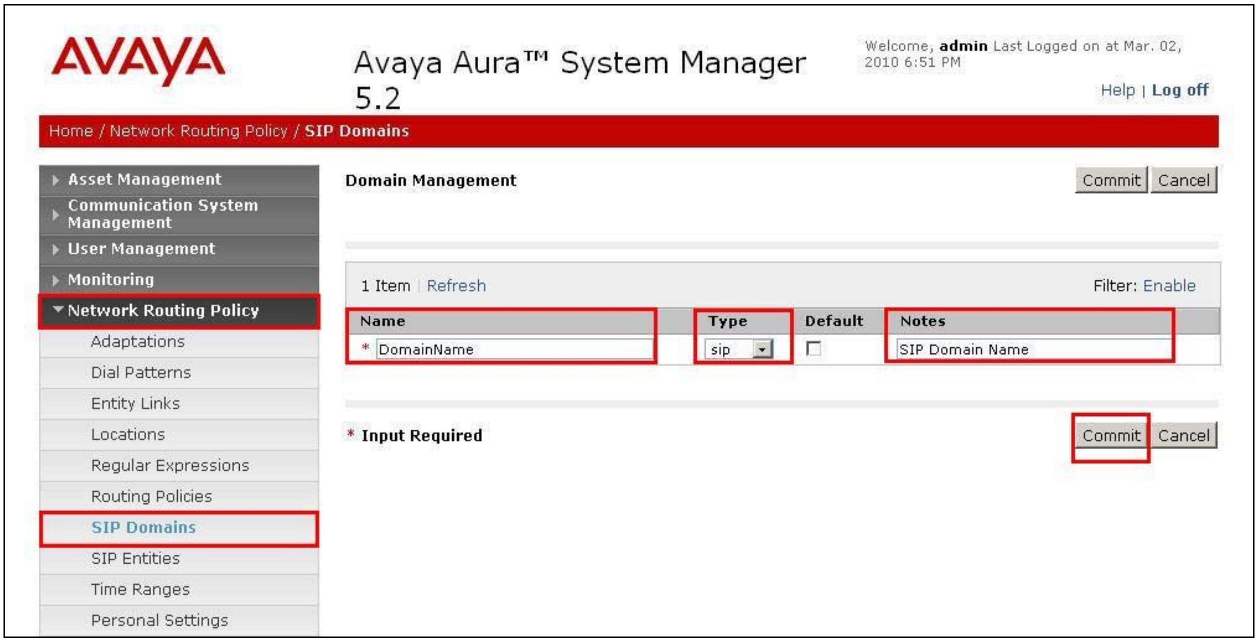
This section details the configuration set-up of Session Manager (SM). Session Manager manages all SIP communications between configured SIP Entities, detailed later in this section. It also manages registration and control of Avaya SIP endpoints using Communication Manager Feature Server. Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR** where **<ip-address>** is the IP address of System Manager. Refer to **Section 8 Reference [1].**

3.1.1. Configure SIP Domain

SIP domains are created as part of the basic configuration. There will be at least one SIP domain for which SM is the authoritative SIP controller. In these sample notes **silstack.com** was used as the main domain. Session Manager can also deal with traffic from other domains, hence the multiple SIP domain entries listed in the image below.

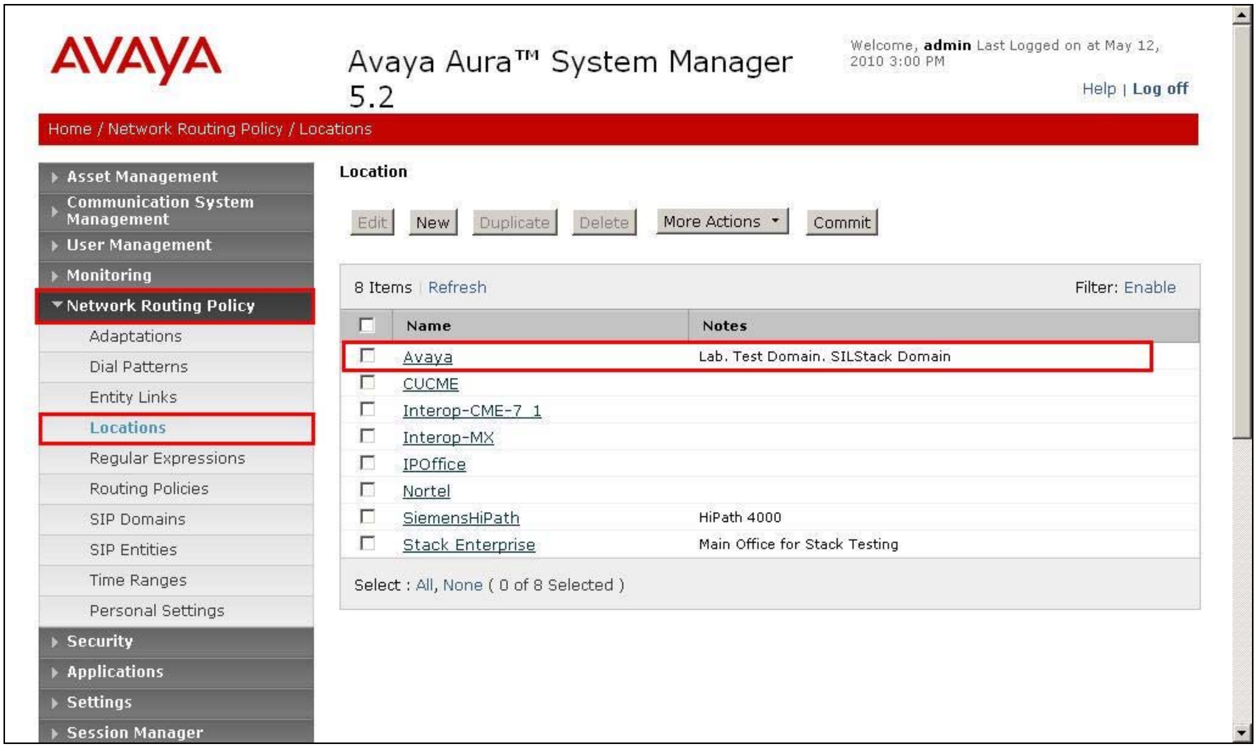


To create a new SIP Domain, expand **Network Routing Policy → SIP Domains.** Click **New**. Under **Name** add a descriptive name. Select **sip** from the **Type** drop down box. Under **Notes**, add a brief description. Click **Commit** to save.

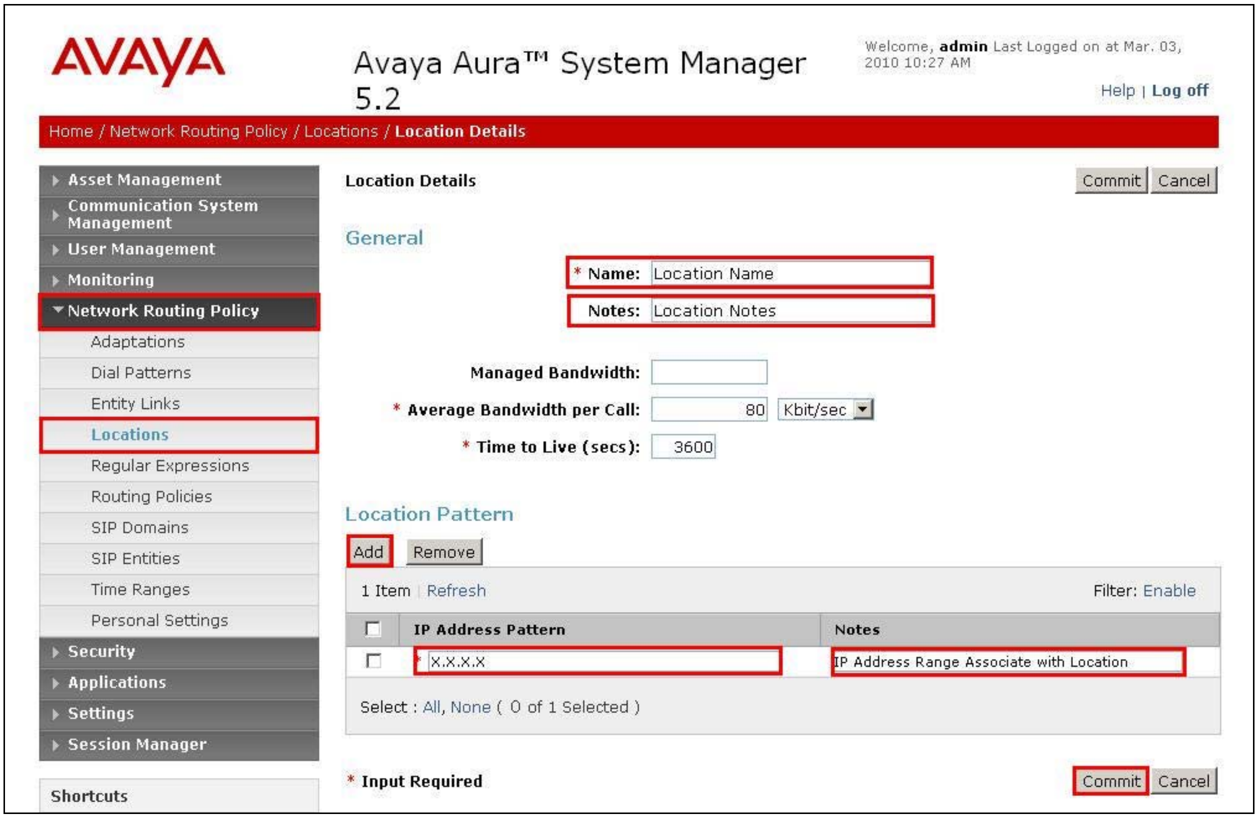


3.1.2. Configure Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management or location-based routing. For this sample configuration, a common location was used for all Avaya equipment

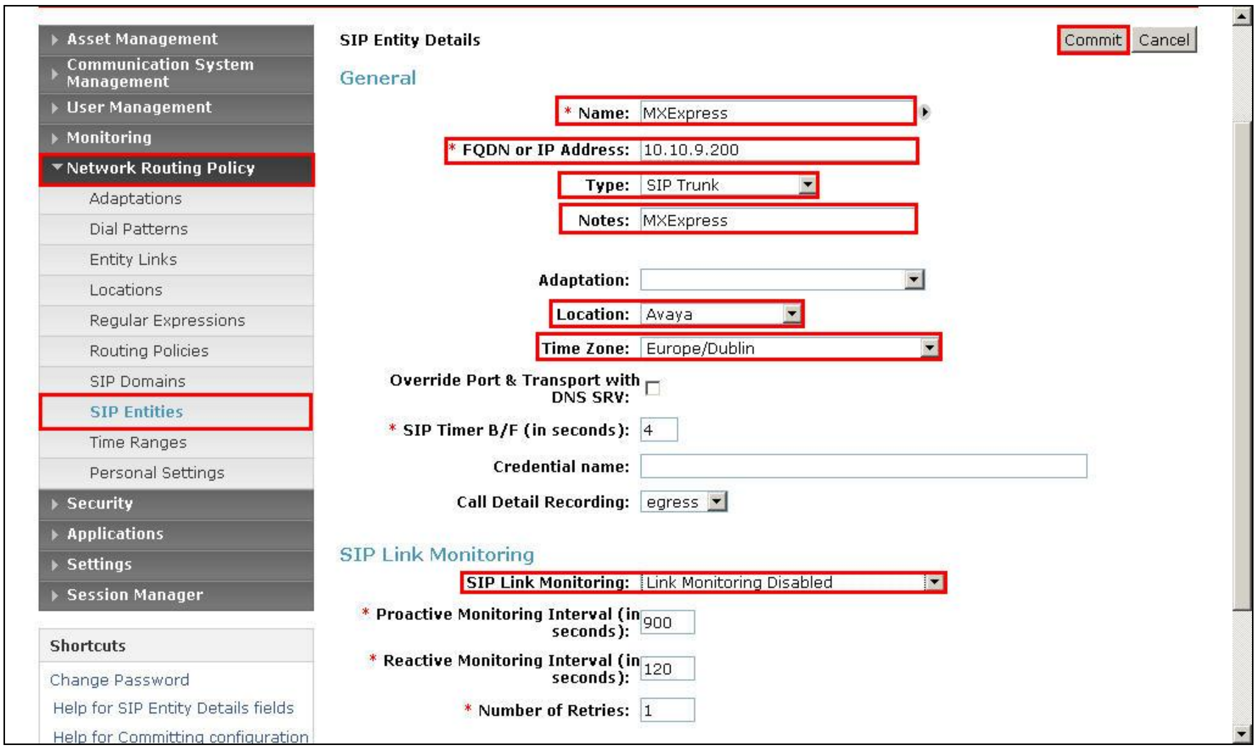


To create a **Location**, expand **Network Routing Policy → Locations**. Click **New.** In the **General** section, under **Name** add a descriptive name. Under **Notes** add a brief description. In the **Location Pattern** section select **Add**, under **IP Address Pattern**, enter an IP address pattern used to logically identify the location. Under **Notes** add a brief description. Click **Commit** to save.



3.1.3. Configure SIP Entity

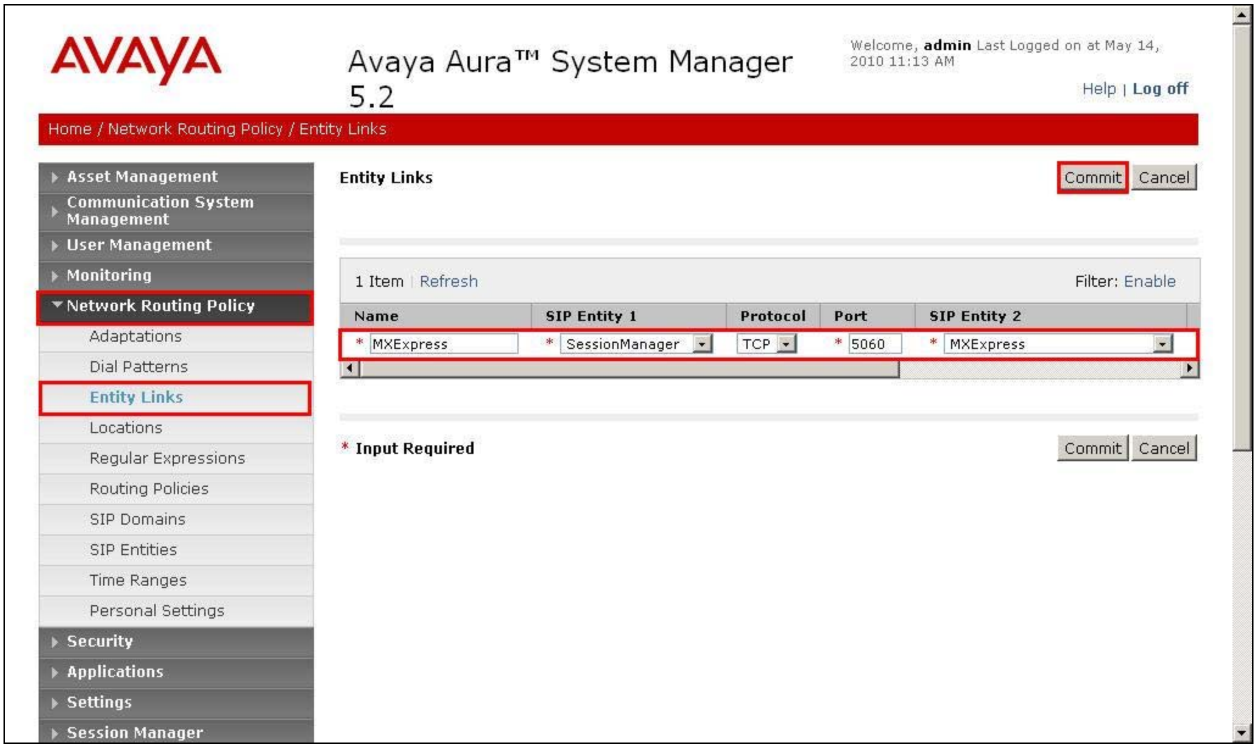
Each SIP device (other than registering devices such as Avaya SIP Phones) that communicates with SM requires a SIP Entity configuration. This section details the steps to create SIP Entities for connection to the Meeting Exchange Express (MXE). Expand **Network Routing Policy** and select **SIP Entities**. Click **New**. In the **General** section, under **Name** add a descriptive name e.g. **MXExpress**. Under **FQDN or IP Address** add the IP Address of the MXE server. From the **Type** drop down box, select **SIP Trunk**. Under **Notes** add a brief description. From the **Location** drop down box, select the appropriate location. From the **Time Zone** drop down box, select the appropriate TZ. From the **SIP Link Monitoring** drop down box select **Link Monitoring Disabled**. Click **Commit** to save.



3.1.4. Configure SIP Entity Links

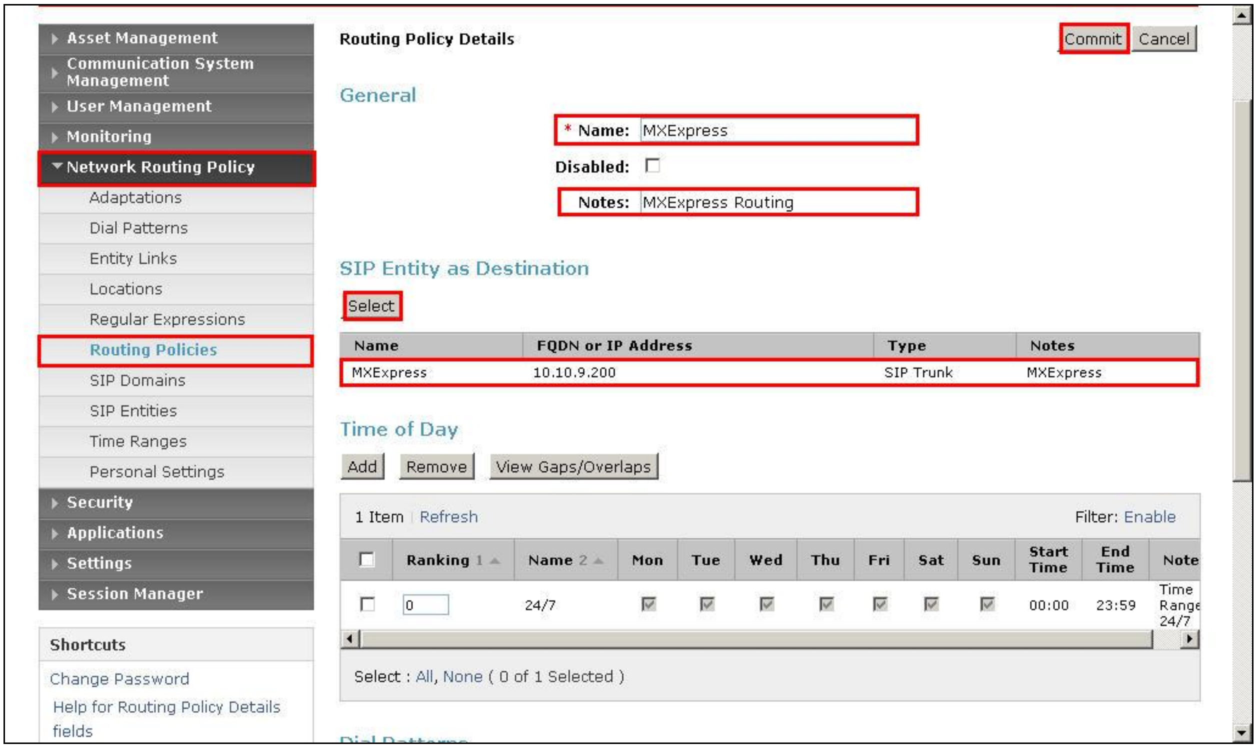
The next step is to create SIP Entity Links, which include the transport parameters to be used for communications between the SM and external SIP devices. Create a SIP Entity Link for MXE. Expand **Network Routing Policy → Entity Links**. Click **New**. Under **Name** enter a suitable identifier e.g. **MXExpress**. Under **SIP Entity 1** drop-down menu select the appropriate SM Entity. Under **Protocol** drop-down menu select **TCP**. Under **Port** enter **5060**. Under **SIP Entity 2** drop-down menu select the SIP Entity added previously, **MXExpress**. (Next fields not shown in screenshot). Under **Port** enter **5060**. Set **Trusted** as ticked. Under **Notes** add a brief description. Click **Commit** to save.

Note: Some of the parameters are not visible in the screenshot below.



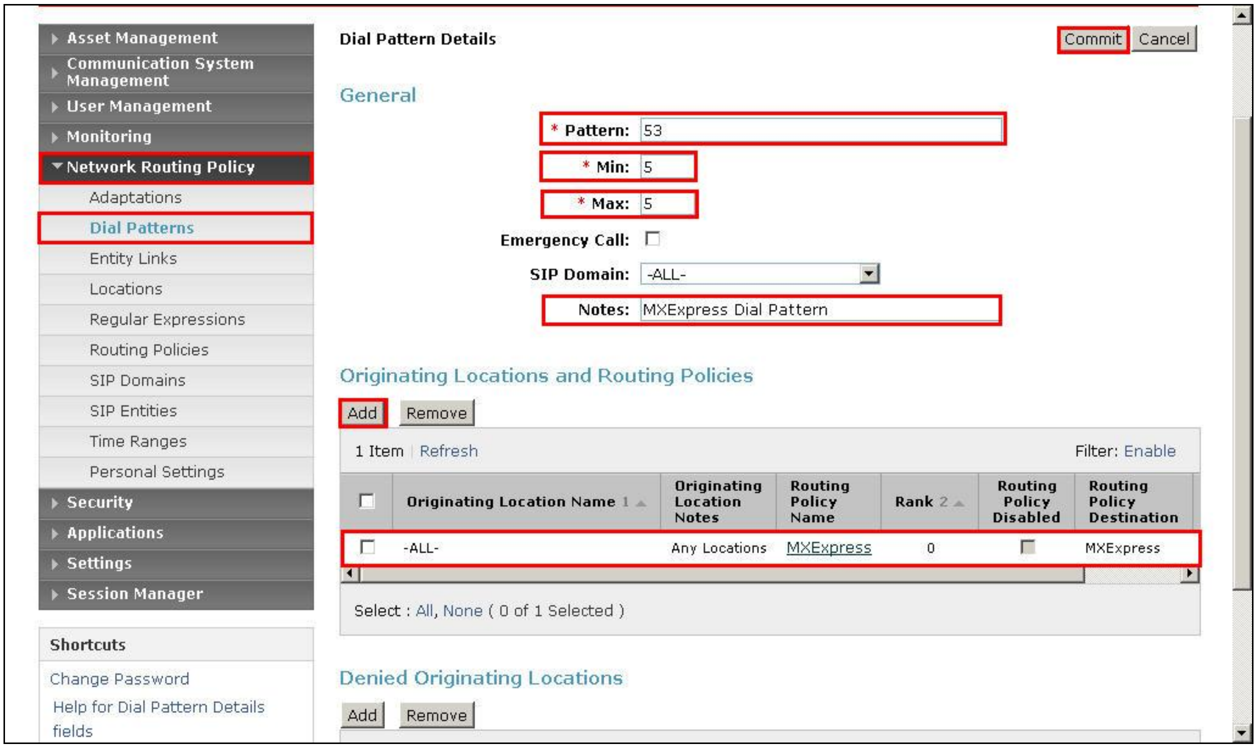
3.1.5. Configure Routing Policy

Create a Routing Policy to route traffic to MXE. Expand **Network Routing Policy**. Select **Routing Policies**. Click **New**. Under **Name** enter a suitable identifier. Under **Notes** enter a suitable description. Under **SIP Entity as Destination** click on **Select**. Choose the appropriate SIP Entity that is to be the call destination. Click **Commit** to save.



3.1.6. Configure Dial Pattern

As one of its main functions, SM routes SIP traffic between connected devices. Dial Patterns are created as part of the configuration to manage SIP traffic routing. Create a Dial Pattern for calls to the MXE. Expand **Network Routing Policy**. Select **Dial Patterns**. Click **New**. Under **Pattern** enter a dial string pattern e.g. **53** (all calls to 5 digit extensions beginning with **53** will be routed to MXE). Under **SIP Domain** drop-down menu select **All**. Under **Notes** enter a suitable description. Under the **Originating Locations and Routing Policies**, click on **Add**. The **Originating Location and Routing Policy List** screen will be displayed. (Screen shot not shown). Select **ALL** as the **Originating Location**, and under **Routing Policies**, select the routing policy created in **Section 3.1.5**. Click **Commit** to save.



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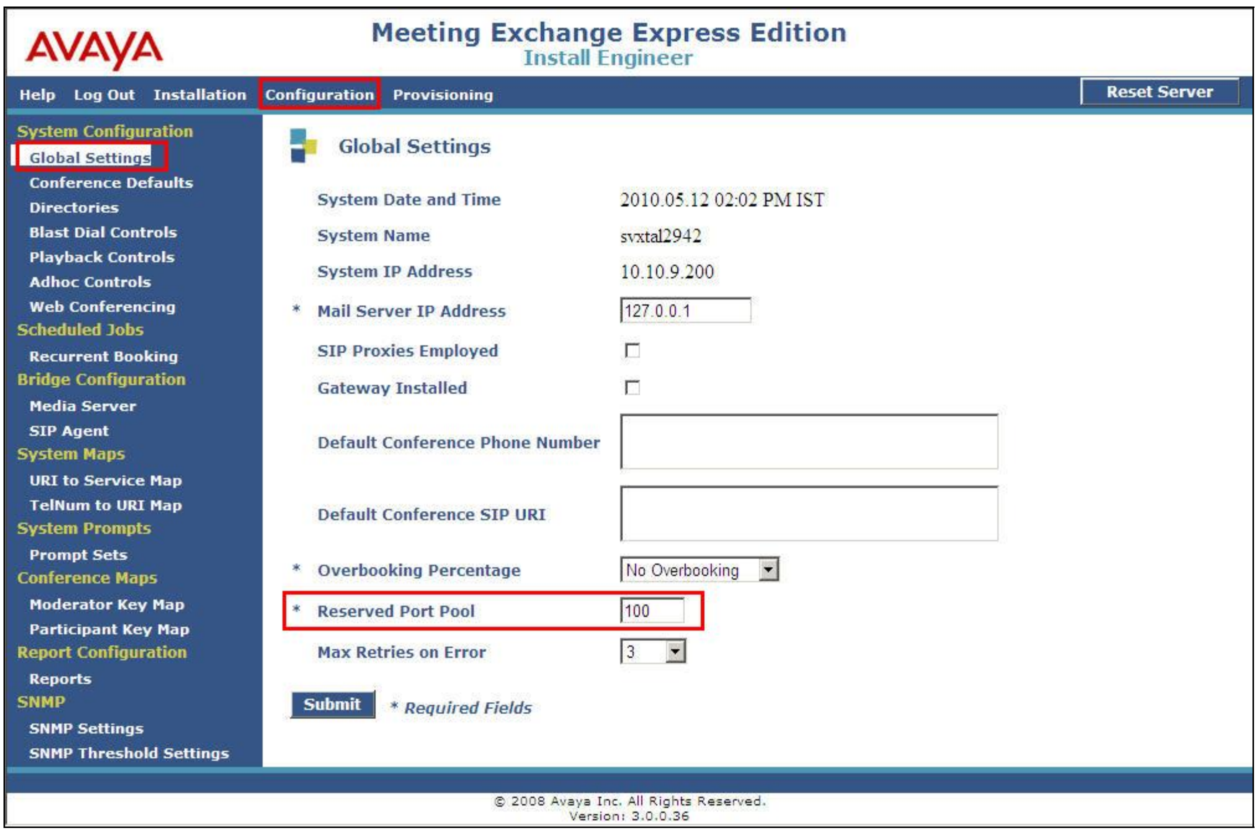
4. Configure Avaya Meeting Exchange Express

This section details the configuration set-up of Meeting Exchange Express. It is assumed that the MXE has been installed and licensed in accordance with Avaya installation procedure, refer to **Section 8 Reference [2]** for more details. Configuration is accomplished by accessing the browser-based GUI of MXE, using the URL **https://<MXE-ip-address>/mx**. Log in using the installer credentials Refer to **Section 8 Reference [2].** (Screen shot not shown).

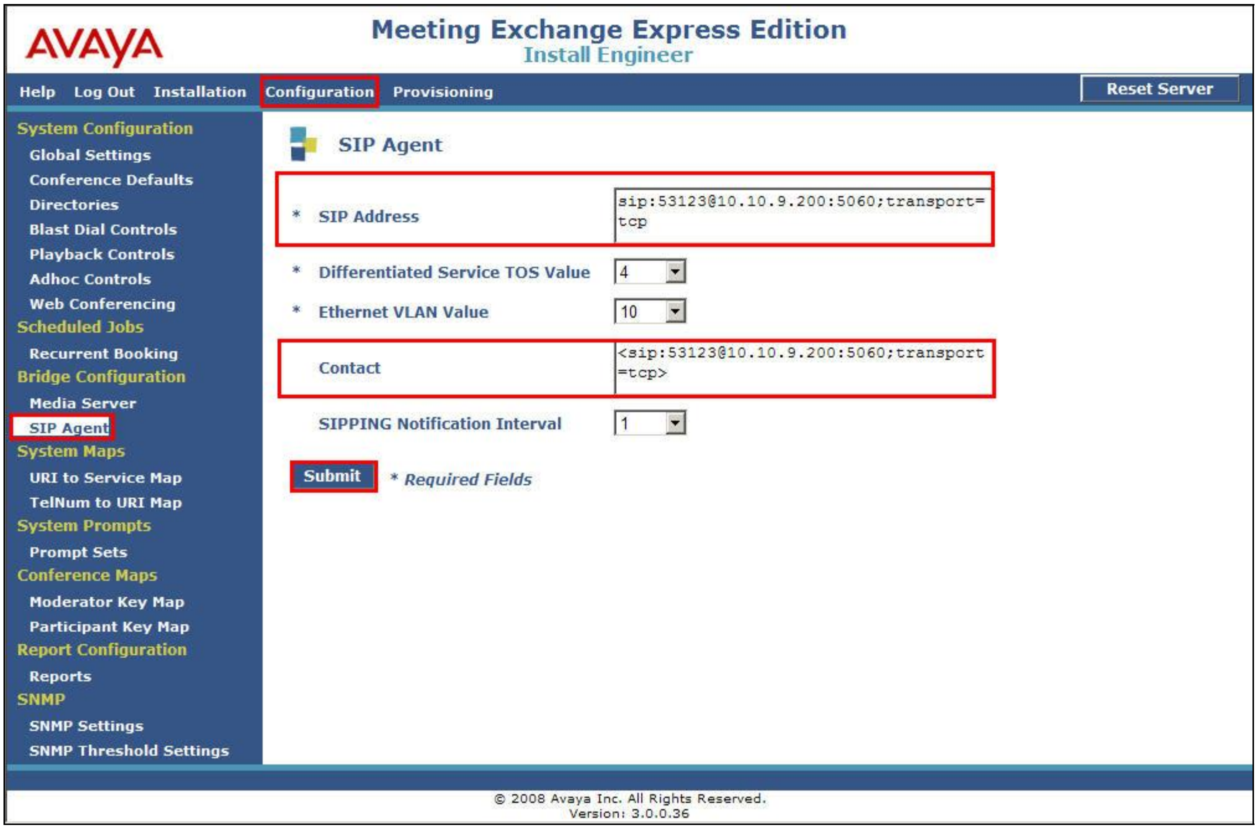
4.1. Configure Conference Pool Resources

This parameter sets the amount of ports which will be available for scheduled conferences. In this sample configuration **100** ports out of **300** are reserved for scheduled conferences. For these application notes, the default demand conferences were used for testing. Select **Configurations** → **Global Settings**. On this page, set the **Reserved Port Pool** value. Click on **Submit** to save changes.



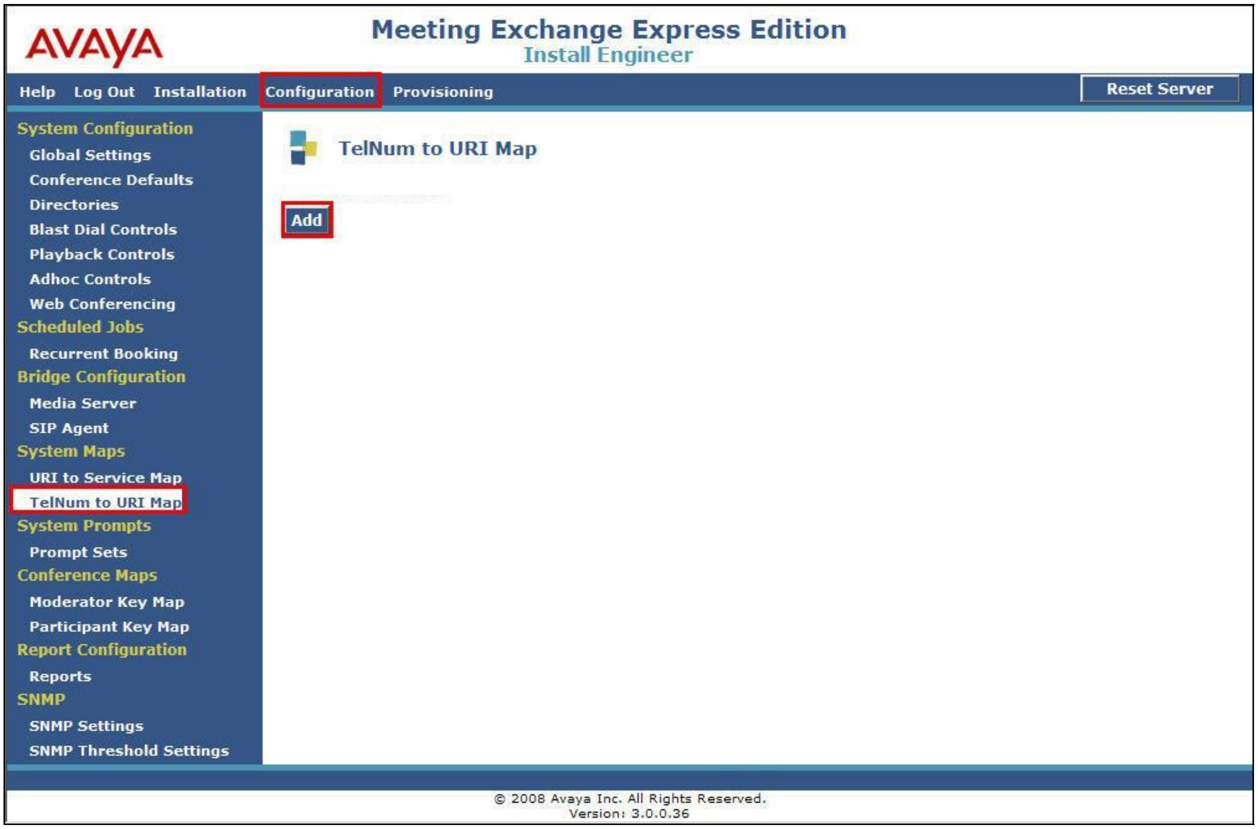
4.2. Configure SIP Agent Contact Details

Select **Configurations →SIP Agent.** For **SIP Address** enter details in SIP URI format e.g. **sip:NUM@MXE-IPaddress:5060;transport=tcp** where **NUM** is the MXE dial in access number, **MXE-IPaddress** is the IP address of the MXE server. Transport is TCP using default SIP port 5060. A similar SIP URI is required as the **Contact**. Use the same URI enclosed with <> brackets. Click on **Submit** to save changes.

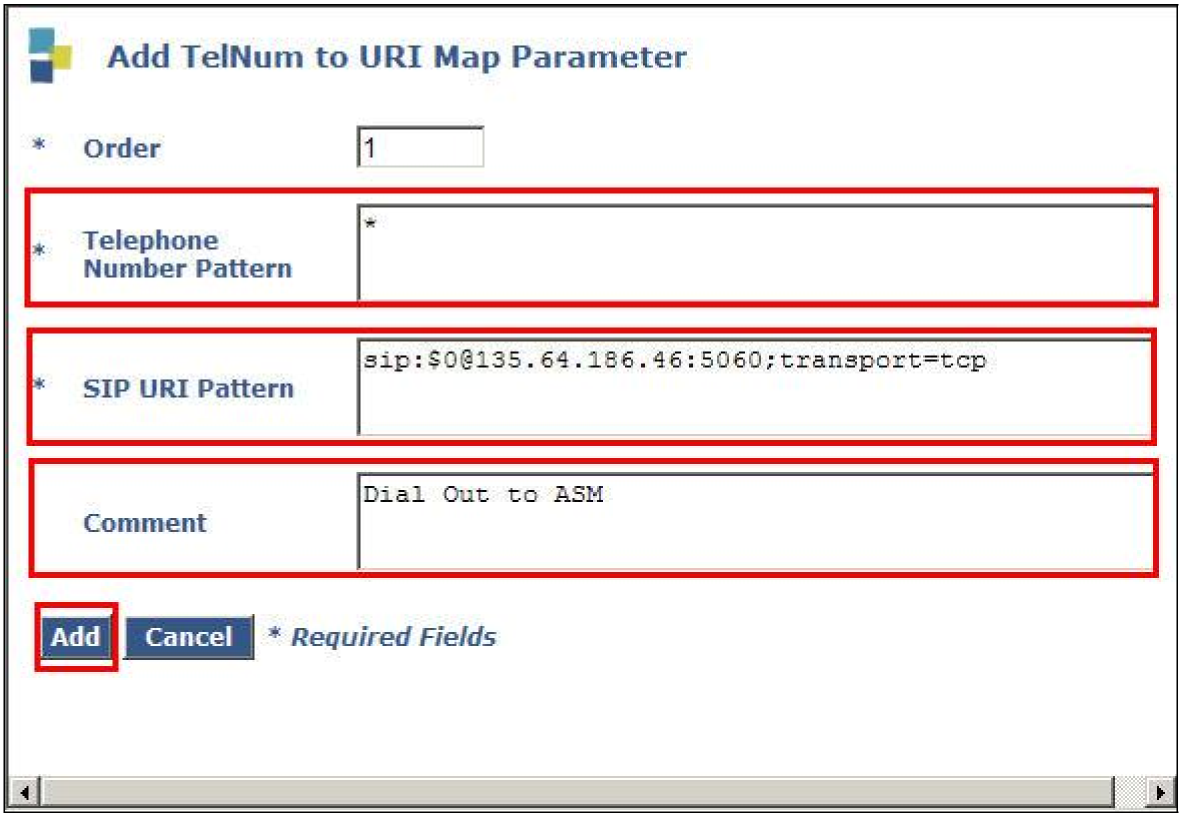


4.3. Configure TelNum To URI Map

For MXE dial-out, the system requires a **TelNum To URI** entry. For this sample configuration, a wildcard dial-out pattern was configured, sending all calls to SM. Select **System Maps → TelNum to URI Map.** Select **Add**.

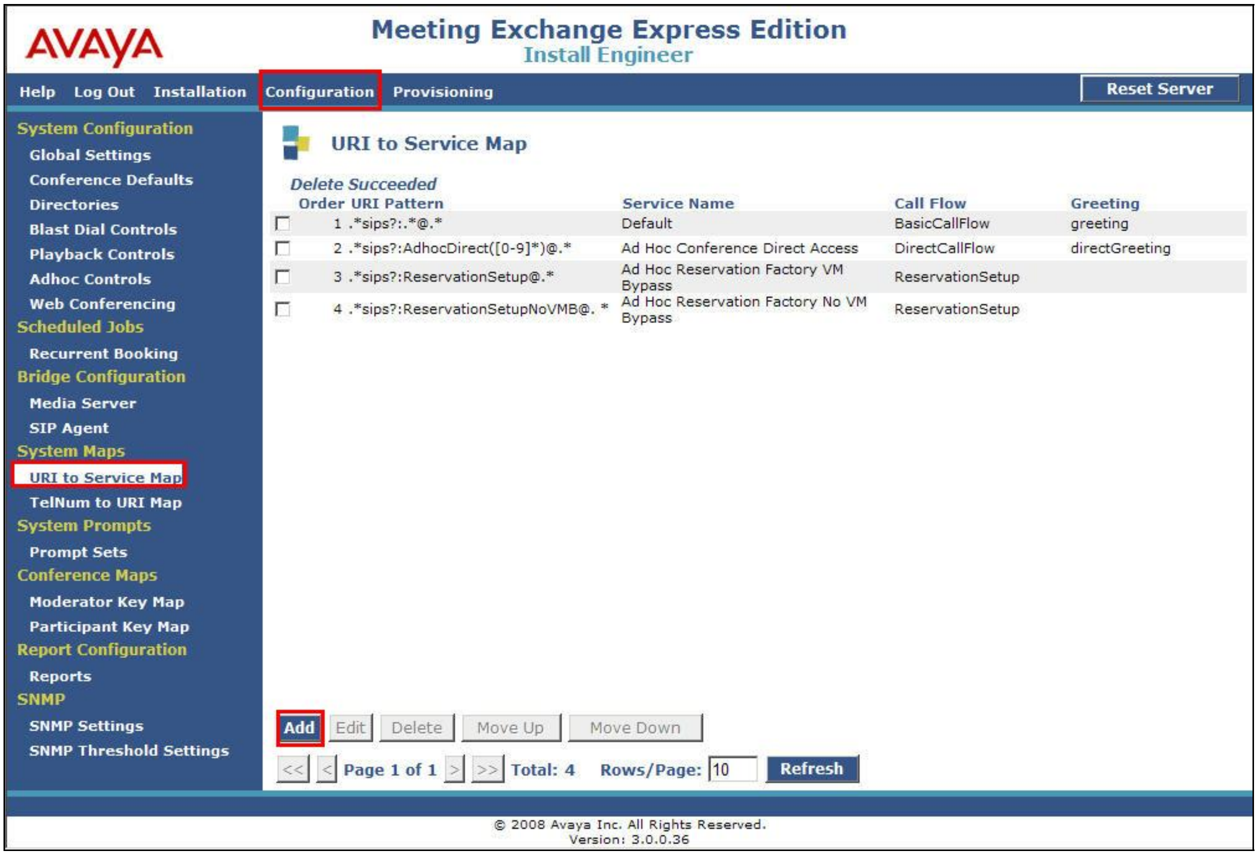


Enter the wildcard pattern ***** as the **Telephone Number Pattern**. In the **SIP URI Pattern** field enter the default dial-out pattern **sip:$0@SM100 IPaddress:5060;transport=tcp**. Where **$0** is a variable parameter, the value is set as the number being dialed. **SM100 IPaddress** is the IP address of the Session Manager SM100 card, followed by the SIP default port, and transport is indicated as **tcp.** Enter a suitable description as the **Comment**. Click on **Add** to apply changes.

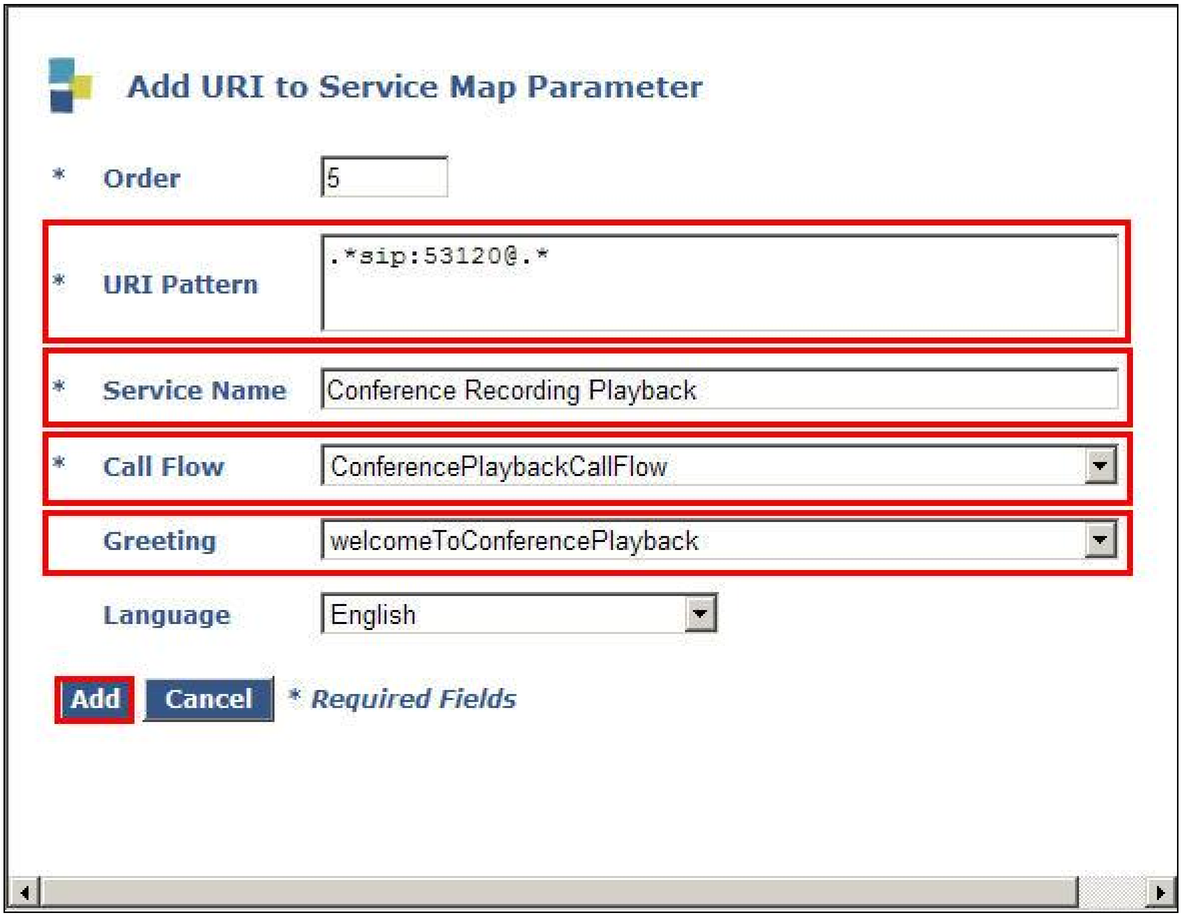


4.4. Configure URI to Service Map

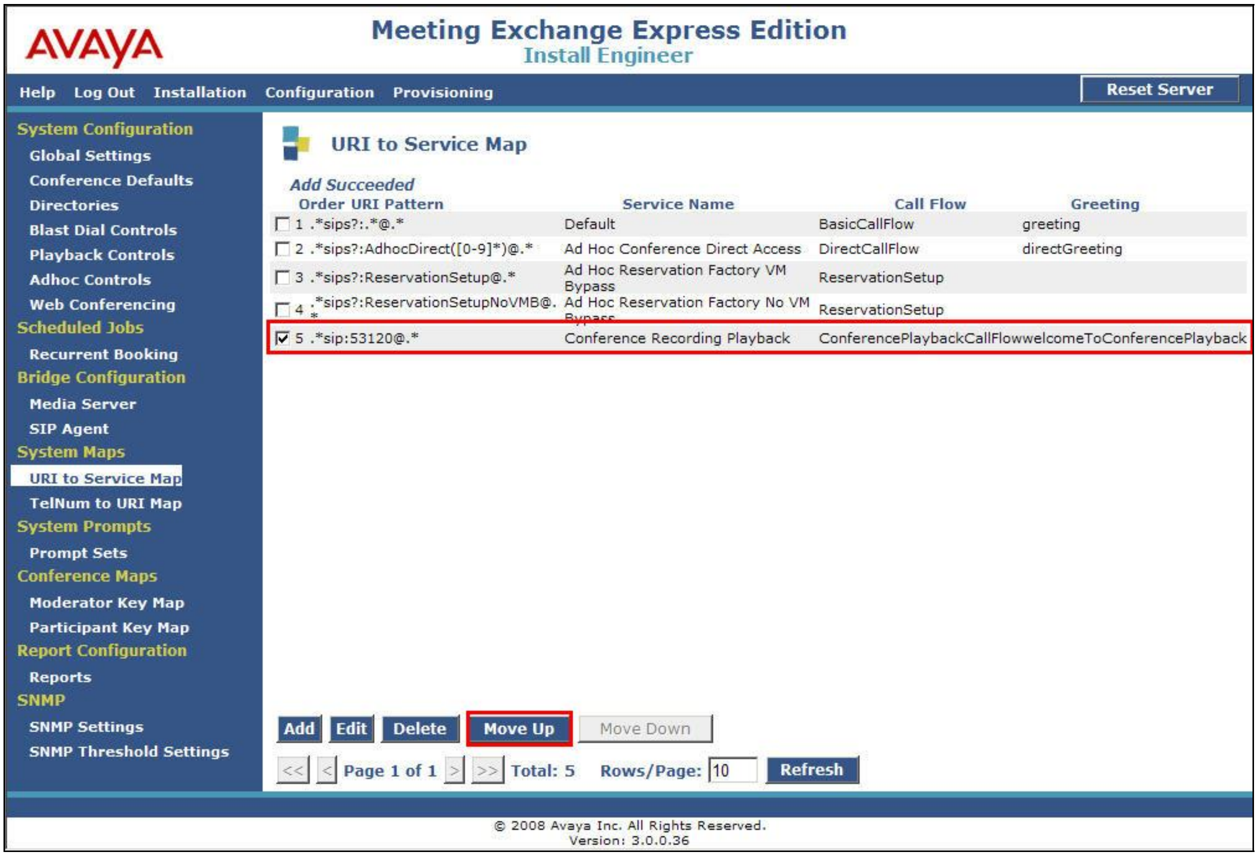
For calls incoming to MXE, some general **URI to Service Map** configurations are required. MXE comes with a default configuration of URI to Service Map entries. For access to the Conference Playback facility, a specific entry was added. Select **Configurations → URI to Service Map.** Select **Add**.



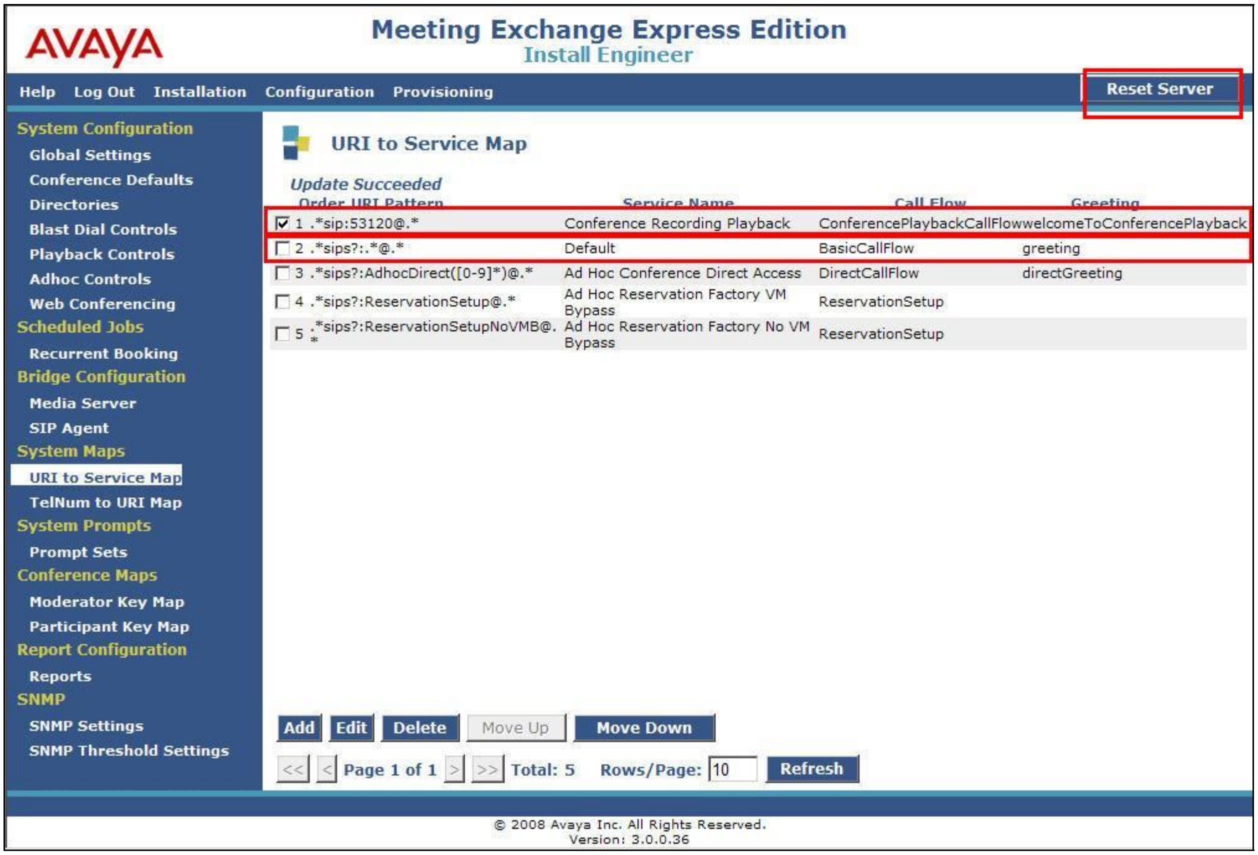
For **URI Pattern** enter **.*sip:CPB_DDI@.*** where **CPB_DDI** is the dial in number for Conference Recording Playback. Enter **Conference Recording Playback** from the **Service Name** drop down box. Select **ConferencePlaybackCallFlow** from the **Call Flow** drop down box. Select **welcomeToConferencePlayback** from the **Greeting** drop down box. Click on **Add** to apply changes.



When MXE refers to the **URI to Service Map** it selects a matching pattern starting at the top of the list. The default **URI to Service Map** entry **.*sips?:.*@.*** will be matched with any incoming call. Therefore the entry for **Conference Recording Playback** was moved to the top of the order list. Select the tick box beside the entry and click on **Move UP**. Repeat until the **Conference Recording Playback** is at the top of the list



The screenshot below illustrates desired order of URI to Service Map entries.



To apply all the changes, click on the **Reset Server** button. Select **Yes**, wait approx 1 minute for the bridge to reset.

5. Verification Steps

This section provides details on how to verify the set-up of the main components of this sample configuration.

Verify MXE server processes by logging into the MXE via ssh terminal. Refer to **Section 8 Reference [2].** Use the command **sudo lc status all** to list the status of the MXE processes.

```
[craft@svxtal2942 ~]$ sudo lc status all
dbx is running.
httpd is running.
ipcb is running.
jboss is running.
notificationCtrlServer is running.
postgresql is running.
[craft@svxtal2942 ~]$
```

Carry out test calls. Dial **53123** for generic conference call access. Verify that the greeting "***Welcome, you've reached the audio conference system***" is heard. Enter a valid conference access code e.g. **22346#**. Verify that the prompt message "***at the tone, please state your name, then press the pound key***" is heard. Follow the menu options and verify successful entry to the conference. Press ***8** to perform a roster playback check. Verify that the system prompts with roster name and the number callers in conference. Dial **53120** for access to the Conference Playback facility; verify the welcome message "***Welcome to the conference playback….***" is heard.

6. Conclusion

These Application Notes describe the steps for configuring Avaya Meeting Exchange Express with Avaya Aura™ Session Manager. Call access to Avaya Meeting Exchange Express including general conference access, conference recording playback, Moderator dial out and DTMF controls were tested. All test cases were completed successfully and the configuration described in these Application Notes has been successfully compliance tested.

Note: The interoperability configuration of the Avaya Meeting Exchange Express *SIP Element* on Avaya Aura™ Session Manager requires *SIP Entity Monitoring* to be disabled as detailed in **Section 3.1.3** of this document.

7. Additional References

Listed below are documents referenced in this Application Notes. These documents are available at the Avaya Support website. Http://support.avaya.com

[1] Configuring Avaya Aura™ Session Manager 5.2 with Avaya Aura™ Communication Manager Access Element, Avaya Voice Portal and Avaya Aura™ Communication Manager Feature Server – Issue 1.0

[2] Avaya Meeting Exchange Express 2.0 Installation and Configuration Guide 04-602593

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