Avaya Solution & Interoperability Test Lab

Application Notes for Interoperability Testing of AudioCodes Mediant 2000 Gateway to Provide Connectivity between the Public Switched Telephone Network (PSTN), Avaya Aura™ Session Manager and Avaya Aura™ Conferencing Standard Edition – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate AudioCodes Mediant 2000 Gateway to provide connectivity between the Public Switch Telephone Network, Avaya Aura™ Session Manager and Aura™ Aura Conferencing Standard Edition. This configuration provides a rich set of conferencing options available on the Avaya Aura™ Conferencing Standard Edition to participants associated with the Public Switched Telephone Network.

1. Introduction

These Application Notes present a sample configuration for a network that uses Avaya Aura™ Session Manager to connect Avaya Aura™ Conferencing Standard Edition and AudioCodes Mediant 2000 Gateway using SIP trunks. SIP trunks connect Avaya Aura™ Conferencing Standard Edition and AudioCodes Mediant 2000 Gateway to Avaya Aura™ Session Manager, using its SM-100 (Security Module) network interface. All inter-system calls are carried over these SIP trunks. Avaya Aura™ Session Manager is managed by Avaya Aura™ System Manager via the management network interface.

The AudioCodes Mediant 2000 Gateway serves as a gateway between TDM and IP networks. AudioCodes Mediant 2000 Gateway supports multiple hardware interfaces and control protocols. Capacity can be scaled upward by adding additional interface modules. During compliance testing, AudioCodes Mediant 2000 Gateway was configured as a SIP to line E1 gateway.

Avaya Aura™ Conferencing Standard Edition is a fully integrated audio and data conferencing solution. Avaya Aura™ Conferencing Standard Edition consists of a number of components which provide booking engines, account management utilities, data sharing functionality, billing outputs, directory server integration capabilities, and audio management for all calls. In Avaya Aura™ Conferencing Standard Edition, the media server and the application server reside on a single server. Avaya Aura™ Conferencing Standard Edition is managed by either Avaya Aura™ Conferencing Manager or Avaya Aura™ System Manager, if one already exists. These Application Notes focus on TCP connectivity and alternative methods such as TLS is not covered in these Application Notes. These Application Notes do not describe how to install or license Avaya Aura™ Conferencing Standard Edition, installation and licensing details can be found in reference **[1]**. Ensure the Avaya Aura™ Conferencing Standard Edition has the latest released patches installed, details can be found in reference **[4]**. Using Avaya Aura™ Conferencing Manager or Avaya Aura™ System Manager the IP addresses of the Conferencing virtual machines need to be specified and connections between the virtual machines need to be established, details can be found in **Chapter 3** of reference **[1]**. These Application Notes do not describe how to schedule a conference by Client Registration Server Front End, installation details can be found in reference **[2]**.

The sample configuration shown in **Figure 1** was used to compliance test AudioCodes Mediant 2000 Gateway, Avaya Aura™ Session Manager and Avaya Aura™ Conferencing Standard Edition.

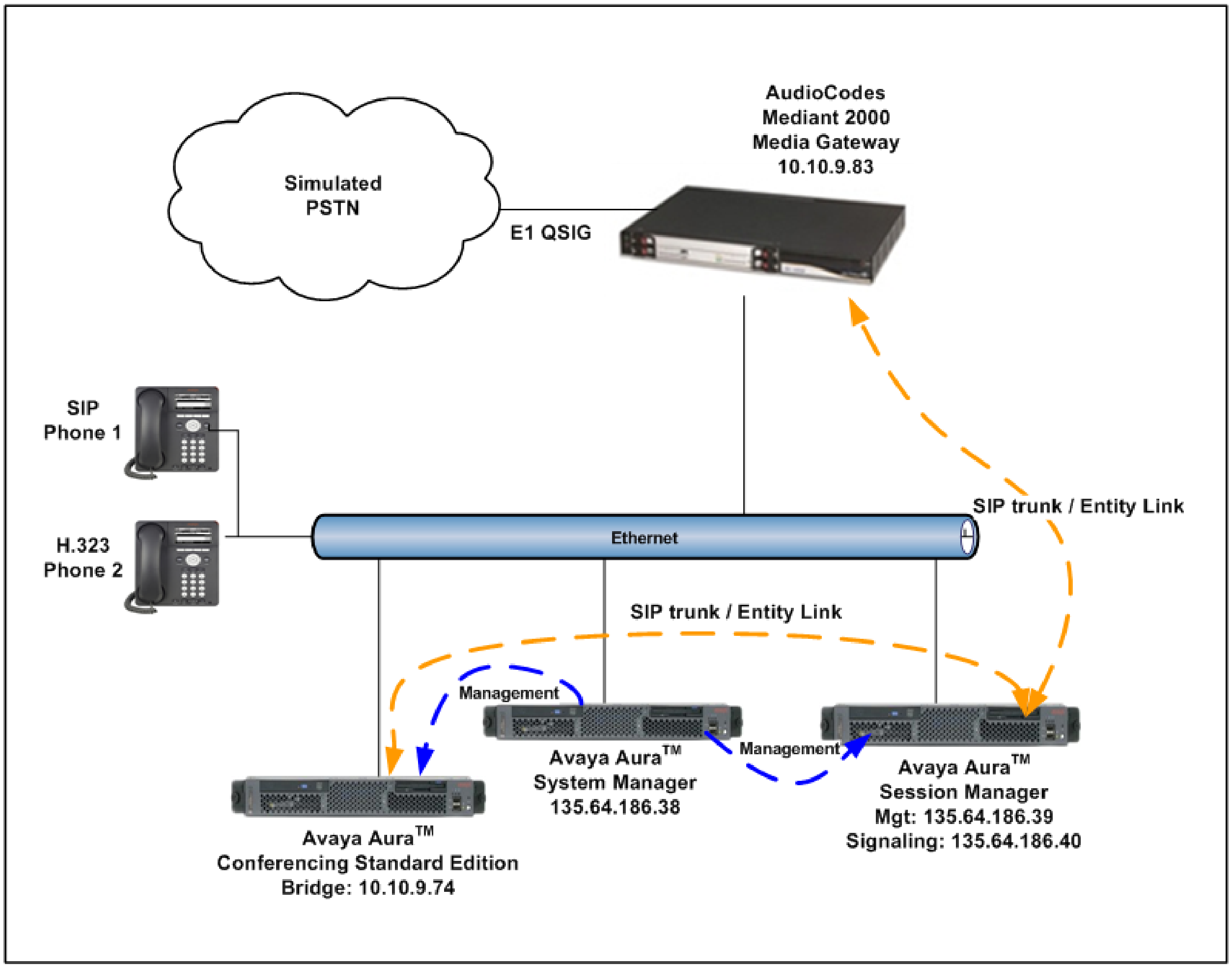


Figure 1 – Test Configuration used in these Application Notes

2. Equipment and Software Validated

The following equipment and software versions were used for the sample configuration provided in these Application Notes.

Equipment	Software
S8510 Server	Avaya Aura™ Session Manager 6.0, Load 600020
	Avaya Aura™ System Manager 6.0, Load 600020
Avaya Aura™ Conferencing Standard Edition Server (S8800)	Avaya Aura™ Conferencing Standard Edition Server 6.0.0.0.262 + Release Patches
AudioCodes Mediant 2000 Gateway	5.80A.039.005
Avaya 9620 IP Telephone (SIP)	2.5.5.18
Avaya 9630 IP Telephone (H.323)	3.10

Table 1: Hardware and Software Versions

The solution was tested with the GA versions of the products shown in **Table 1**. However, a pre-GA build of System Manager was used to capture screens. Therefore, screen captures shown in these Application Notes may not precisely match the final version of the product. Known differences in screens will be noted in the text accompanying the screen capture.

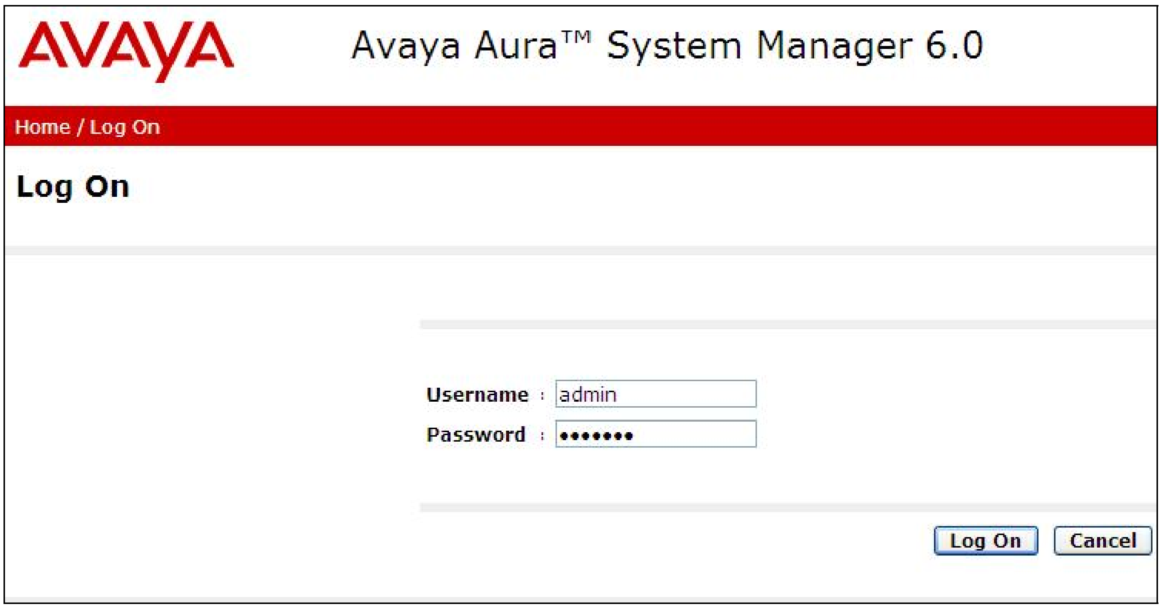
3. Configure Avaya Aura™ Conferencing Standard Edition

This section describes the procedure for configuring the Conferencing Standard Edition to interoperate with Session Manager via SIP trunking. The procedures include the following areas:

- Log in to Avaya Aura™ System Manager
- Configure SIP Connectivity
- Configure Dialout
- Map DNIS Entries

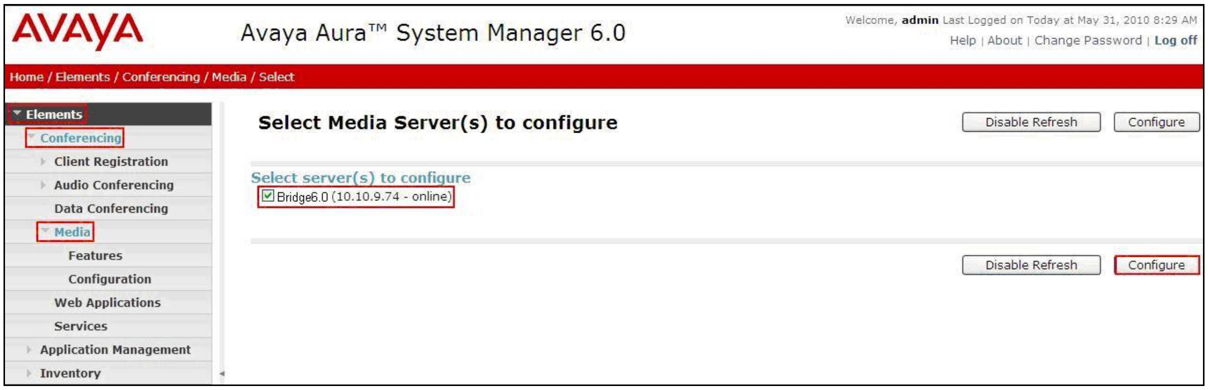
3.1. Log in to Avaya Aura™ System Manager

Access the System Manager using a Web Browser and entering ***https://<ip-address>/smgr***, where <ip-address> is the IP address of System Manager. Log in using appropriate credentials and accept the subsequent Copyright Legal Notice.

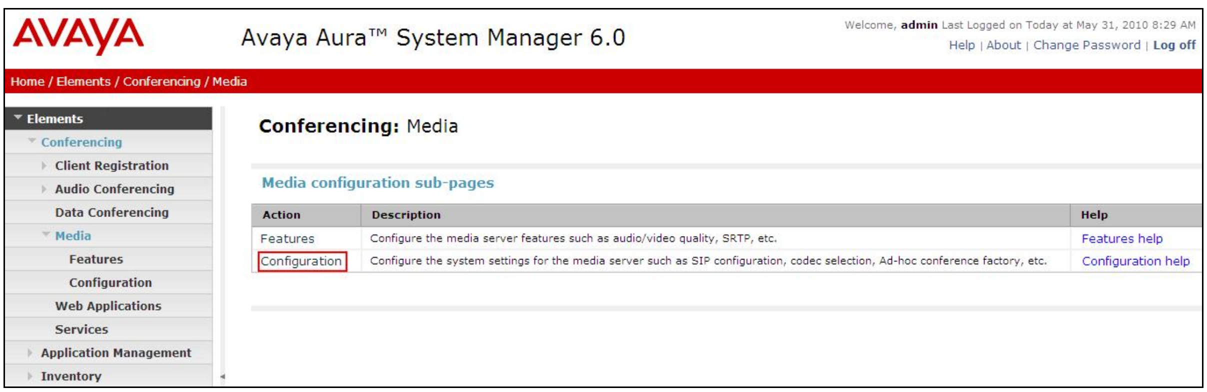


3.2. Configuring SIP Connectivity

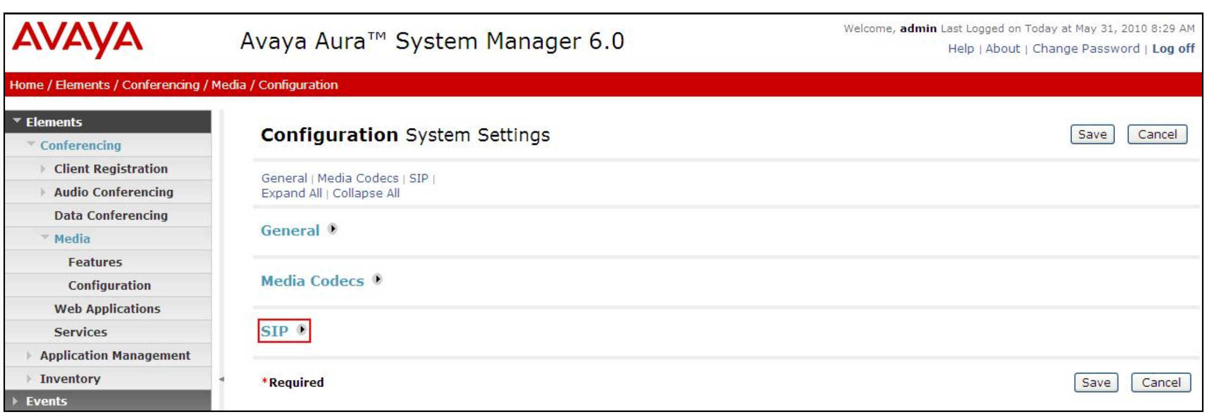
Configure settings that enable SIP connectivity between the Conferencing bridge and other devices by configuring the SIP System Settings by selecting **Elements → Conferencing → Media** on the left panel menu. From the right panel menu select the media server to configure by selecting the tick box and select **Configure**.



From the right panel menu select **Configuration**.



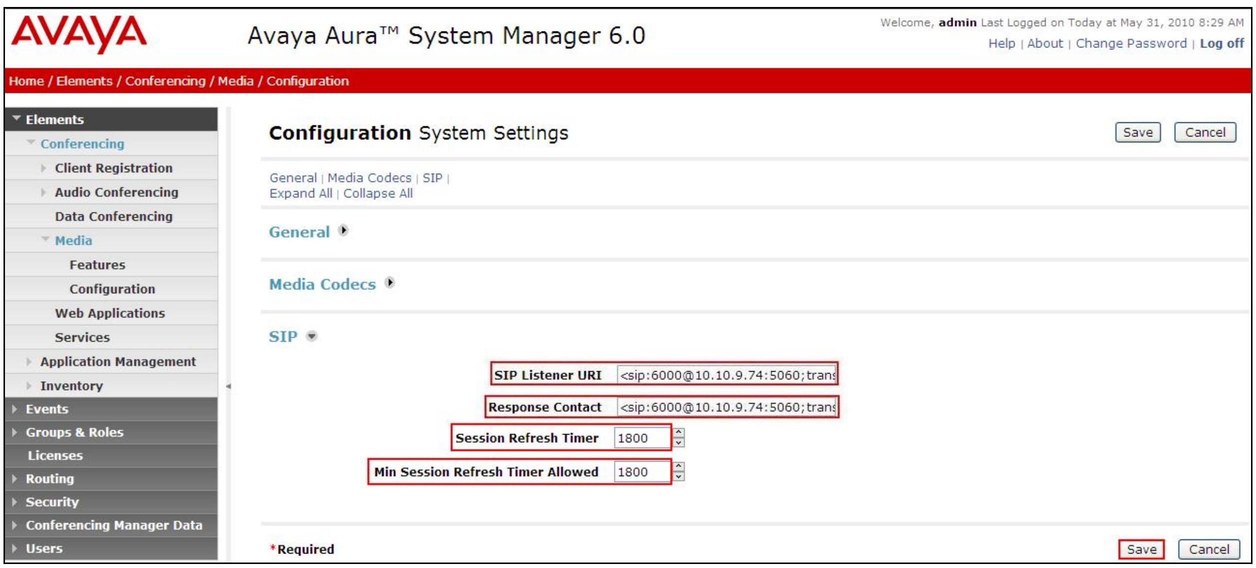
From the right panel menu select **SIP**.



From the **SIP** menu on the right panel menu verify the following options:

- **SIP Listener URI** **<sip:6000@10.10.9.74:5060;transport=tcp>**
  Depending on the SIP signalling protocol, TCP or UDP, configure the following line to populate the From Header Field in SIP INVITE messages:
  Note: The user field 6000, defined for this SIP URI must conform to RFC 3261. For consistency, it is selected to match the user field provisioned for the **Response Contact** entry (see below).
- **Response Contact** **<sip:6000@10.10.9.74:5060;transport=tcp>**
  Depending on the SIP signalling protocol, TCP or UDP, configure the following line to provide SIP Device Contact address to use for acknowledging SIP messages from the Enterprise Standard Edition:
- **Session Refresh Timer** **1800**
- **Min Session Refresh Timer Allowed** **1800**

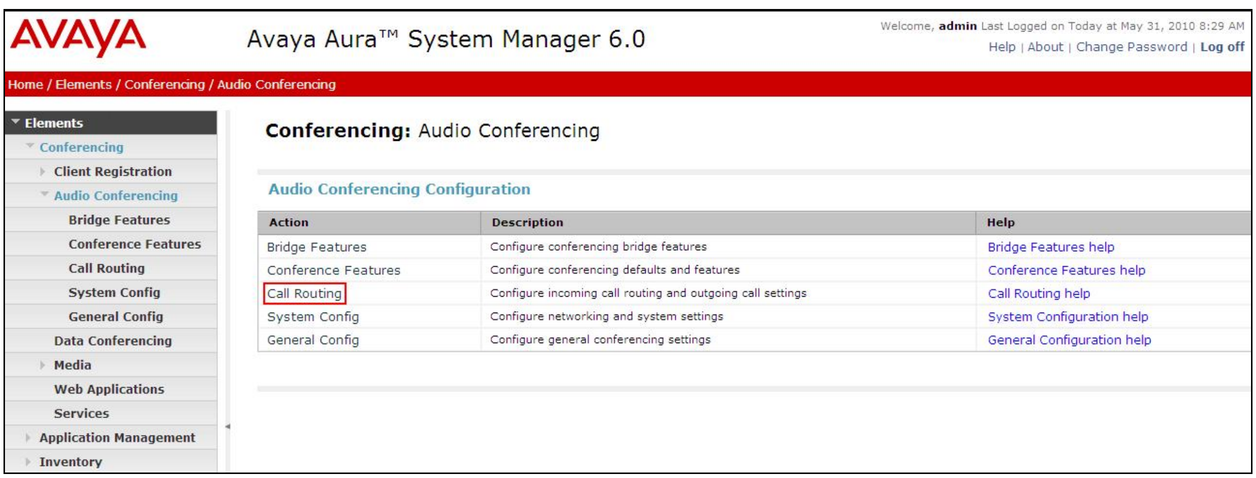
Click the **Save** button.



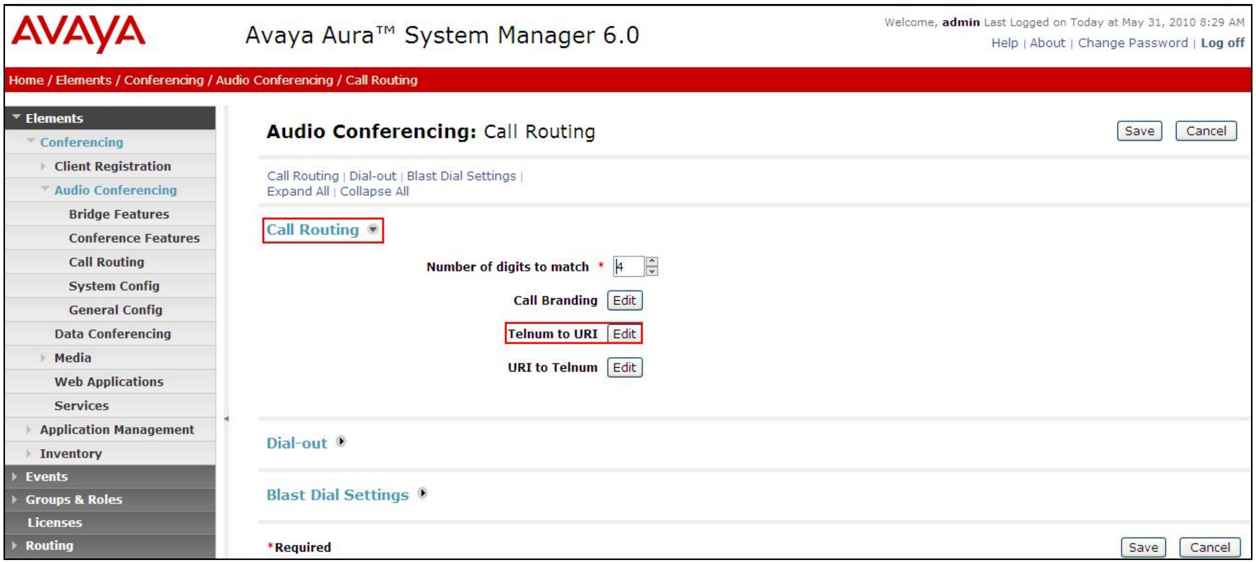
3.3. Configure Dialout

To enable Dial-Out from the Conferencing Bridge to the Session Manager, configure the **telnumToUri** by selecting **Elements → Conferencing → Audio Conferencing** on the left panel menu. From the right panel menu select the conferencing server to configure by selecting the tick box and select **Configure**.

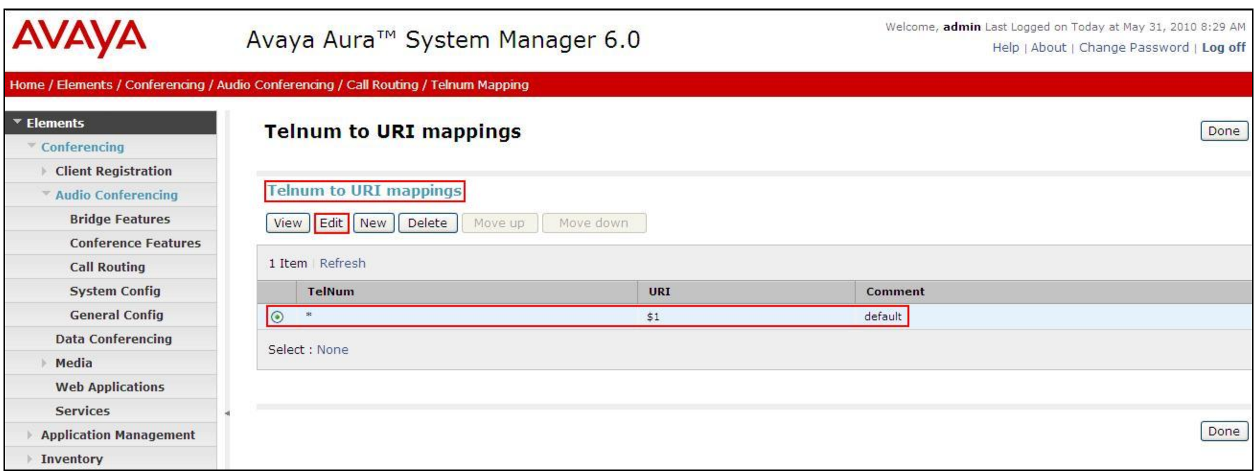
From the right panel menu select **Call Routing**.



From the **Call Routing** menu on the right panel menu select the **Edit** button for **Telnum to URI** option.



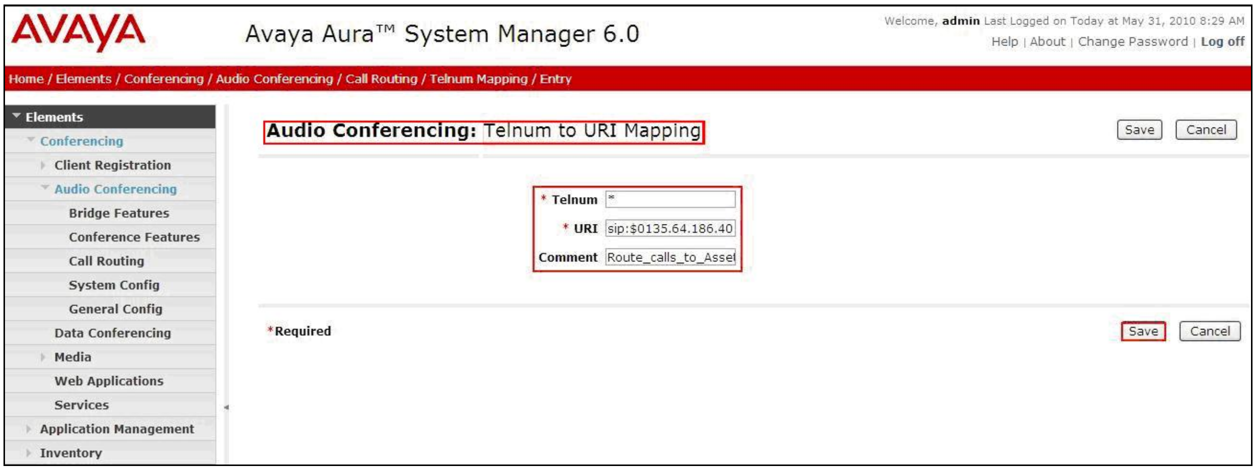
From the right panel menu select the default **Telnum to URI mappings** and select **Edit**.



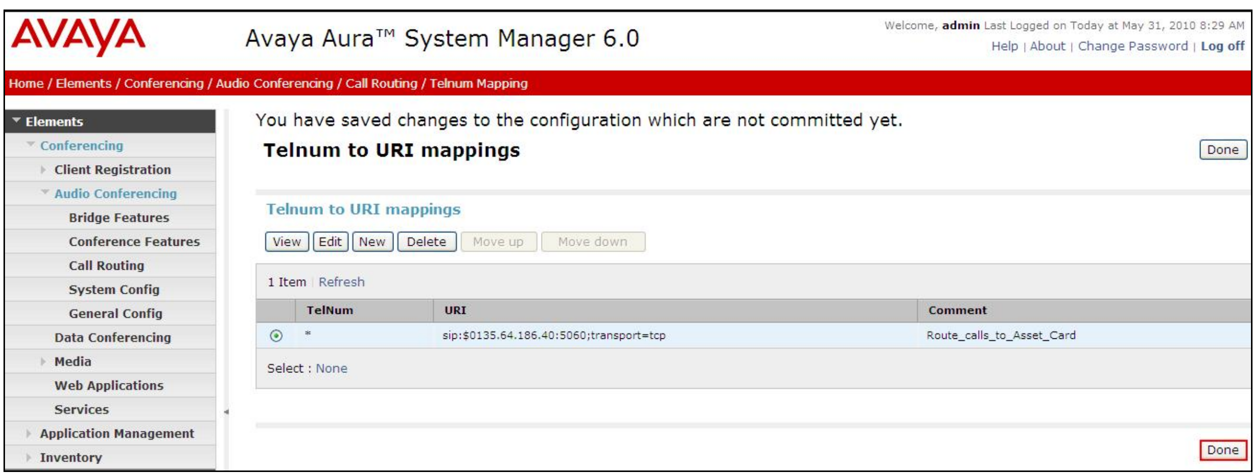
From the right panel menu complete the following options; under **Audio Conferencing: Telnum to URI Mapping**.

- **Telnum** *****
- **URI** **sip:$0@135.64.186.40:5060;transport=tcp**
  To route outbound calls from the Conferencing to the Software Asset Card.
- **Comment** A descriptive comment

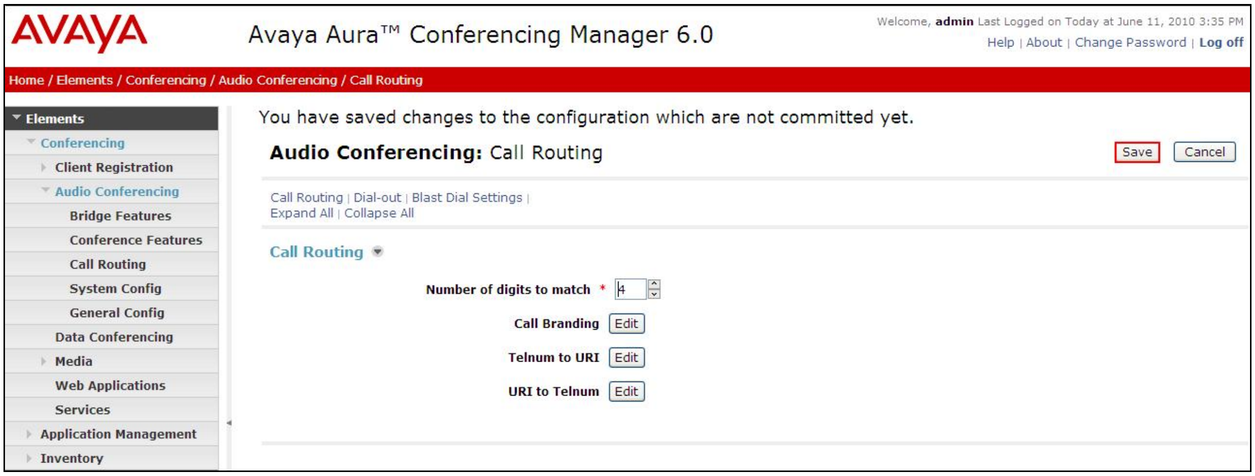
Click the **Save** button.



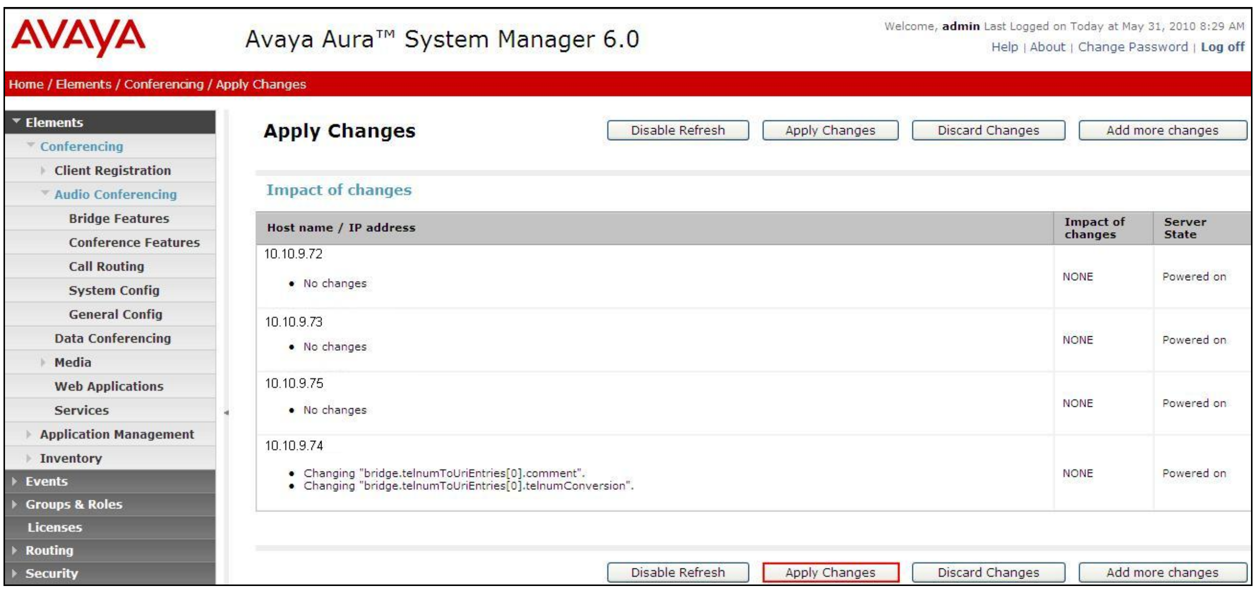
From the right panel menu select **Done**.



From the right panel menu select **Save**.



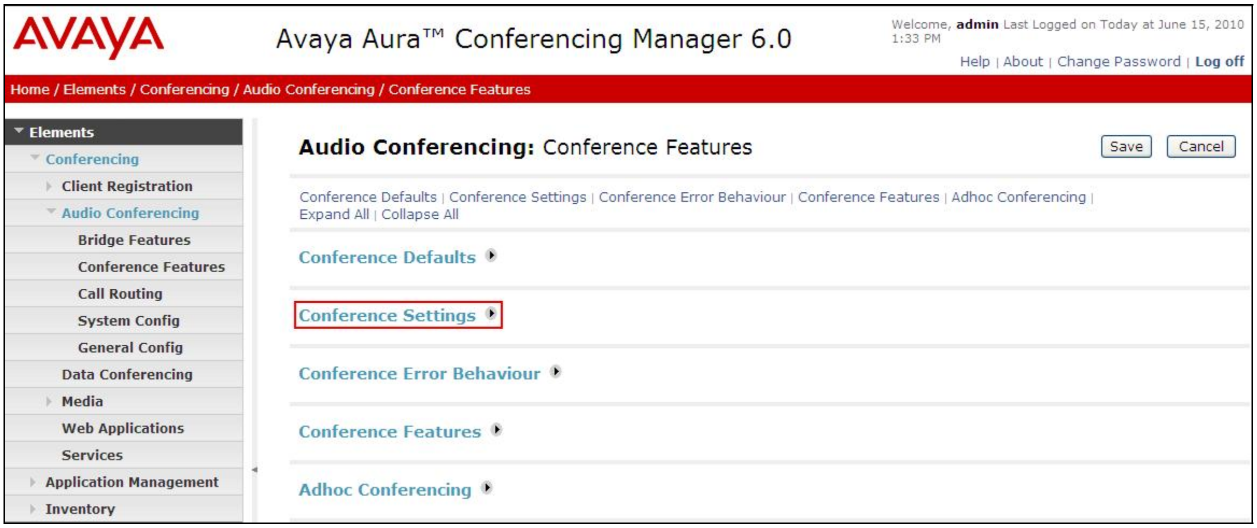
From the right panel menu select **Apply Changes**.



To enable Dial-Out from the Conferencing Bridge to the Session Manager, configure the **Originator Dial Out** by selecting **Elements → Conferencing → Audio Conferencing** on the left panel menu. From the right panel menu select the conferencing server to configure by selecting the tick box and select **Configure**.

From the right panel menu select **Conference Features**.

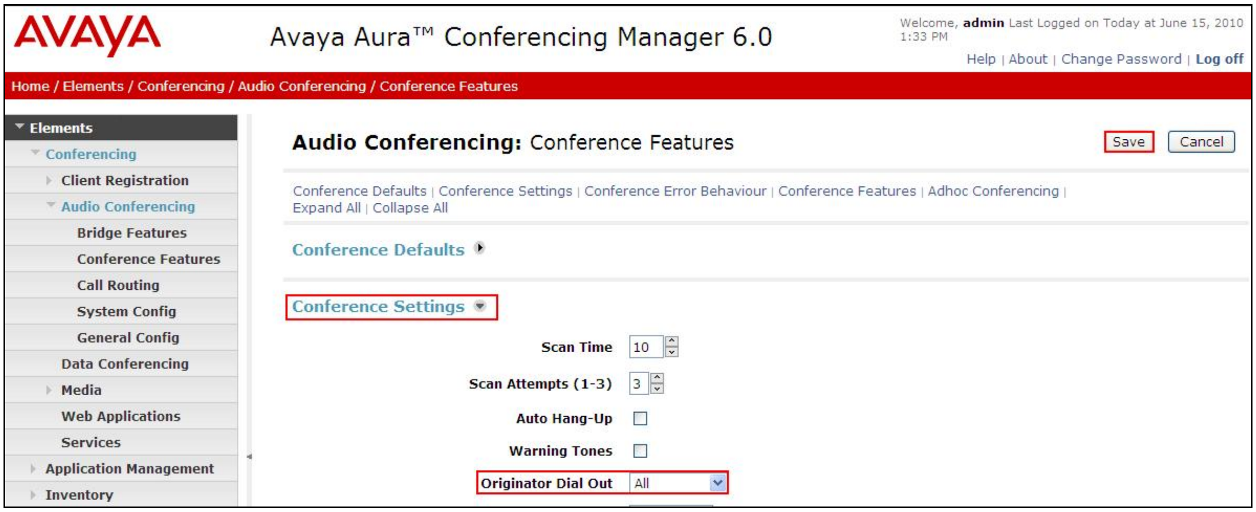
From the right panel menu select **Conference Settings**.



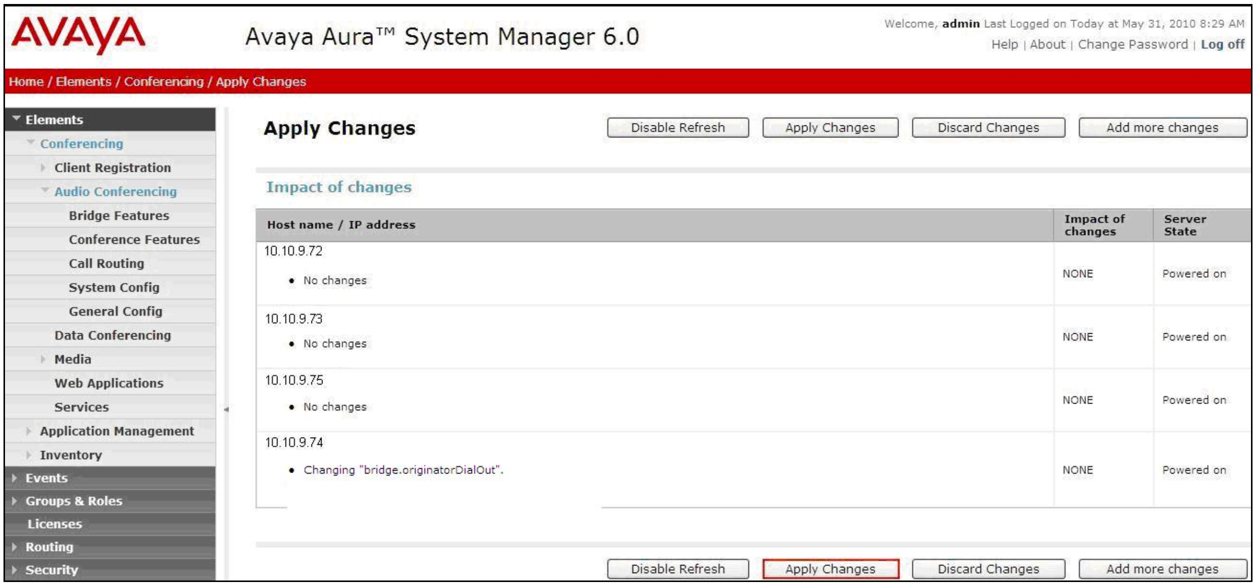
From the **Conference Settings** drop down menu on the right panel menu select the following parameter, leaving the remaining parameters at their default values.

- **Originator Dial Out** Select **All**

Click the **Save** button.



From the right panel menu select **Apply Changes**.

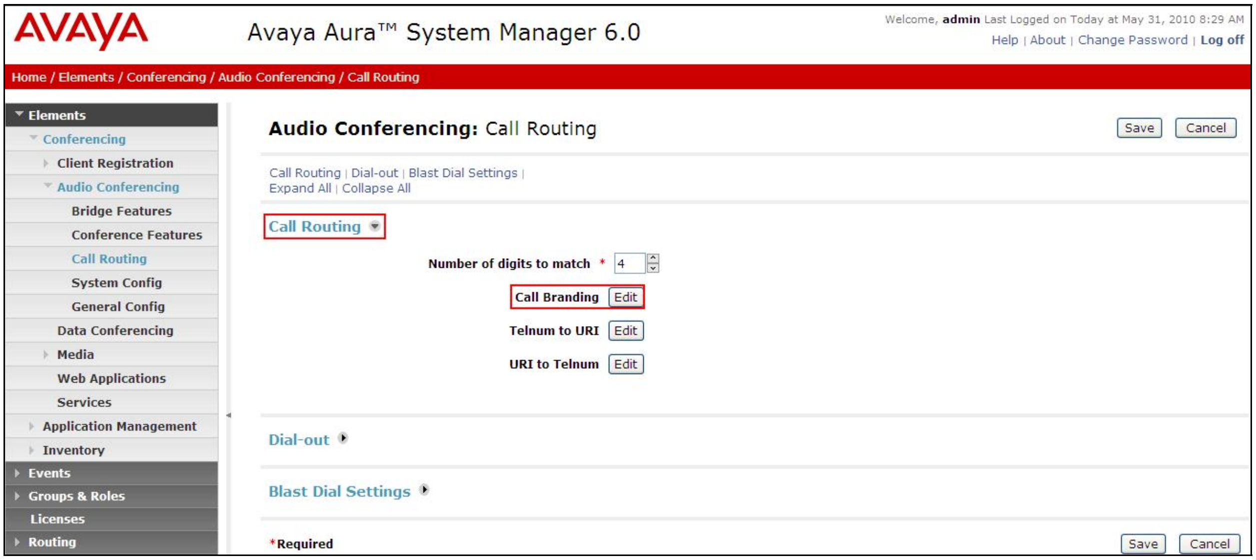


3.4. Map DNIS Entries

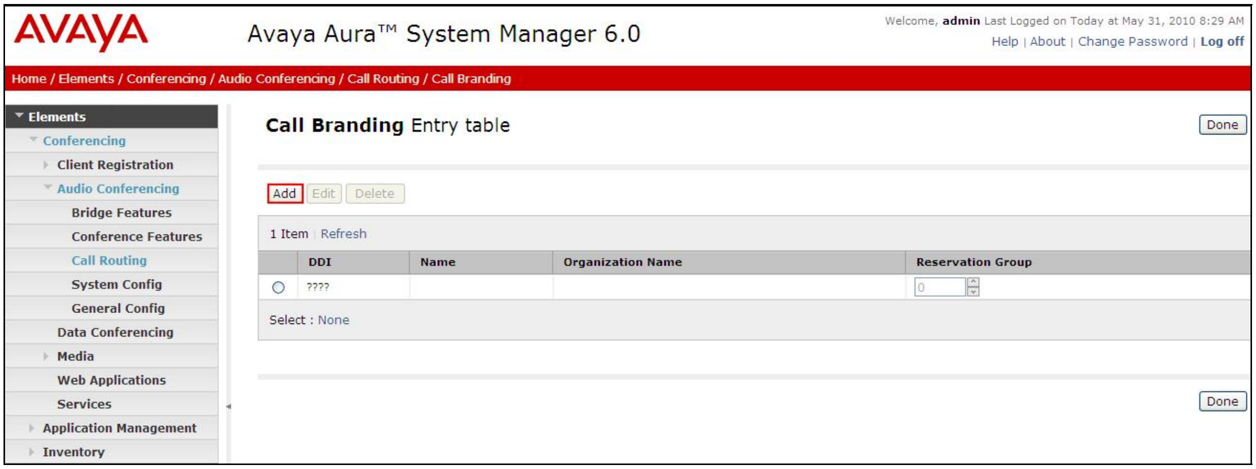
To map DNIS entries, run the Call Branding utility by selecting **Elements → Conferencing → Audio Conferencing** on the left panel menu. From the right panel menu select the conferencing server to configure by selecting the tick box and select **Configure**.

From the right panel menu select **Call Routing**.

From the **Call Routing** menu on the right pane select the **Edit** button for **Call Branding** option.



From the right panel menu select the **Add** button to create a new call branding entry.

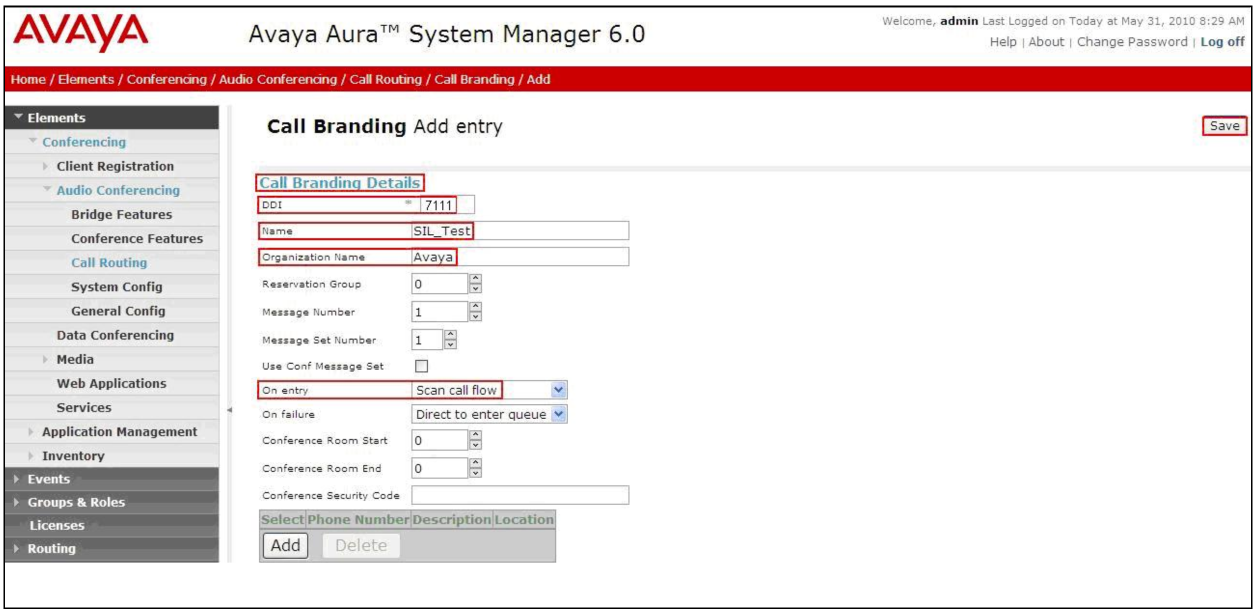


In this sample configuration for **Call Branding Details** select the following parameters, leaving the remaining parameters at their default values.

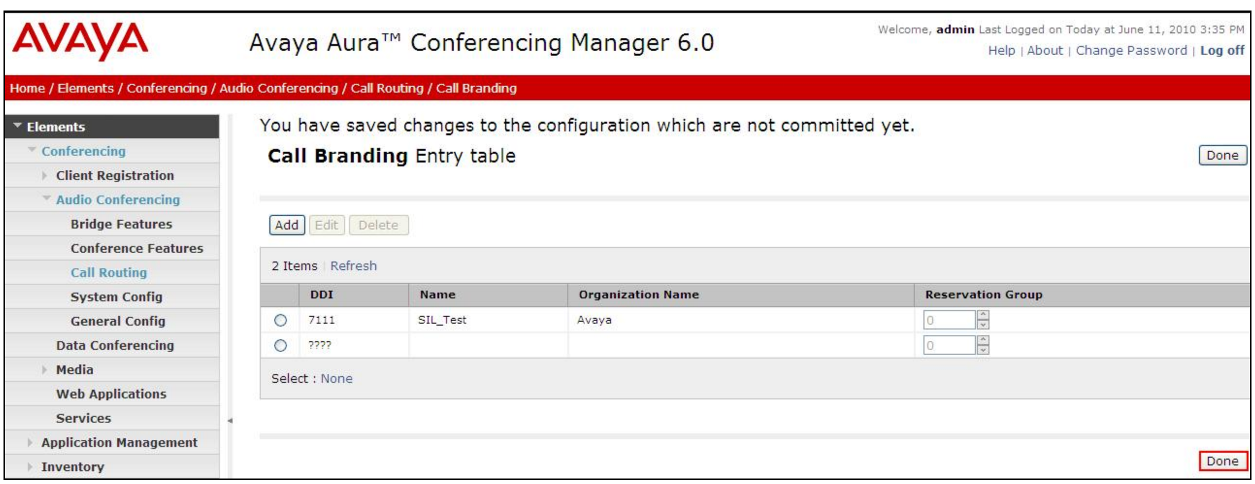
Under **Call Branding Details**

- **DDI** **7111**
- **Name** A descriptive name
- **Organisation Name** A descriptive name
- **On Entry** Select **Scan call flow** from the drop down menu.

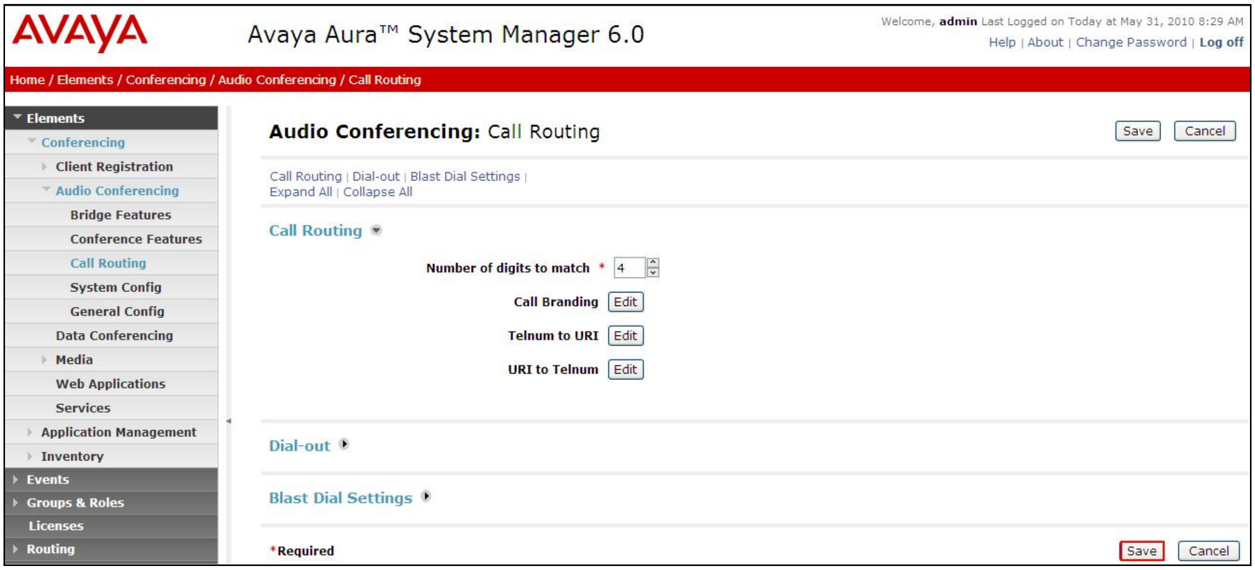
Click the **Save** button.



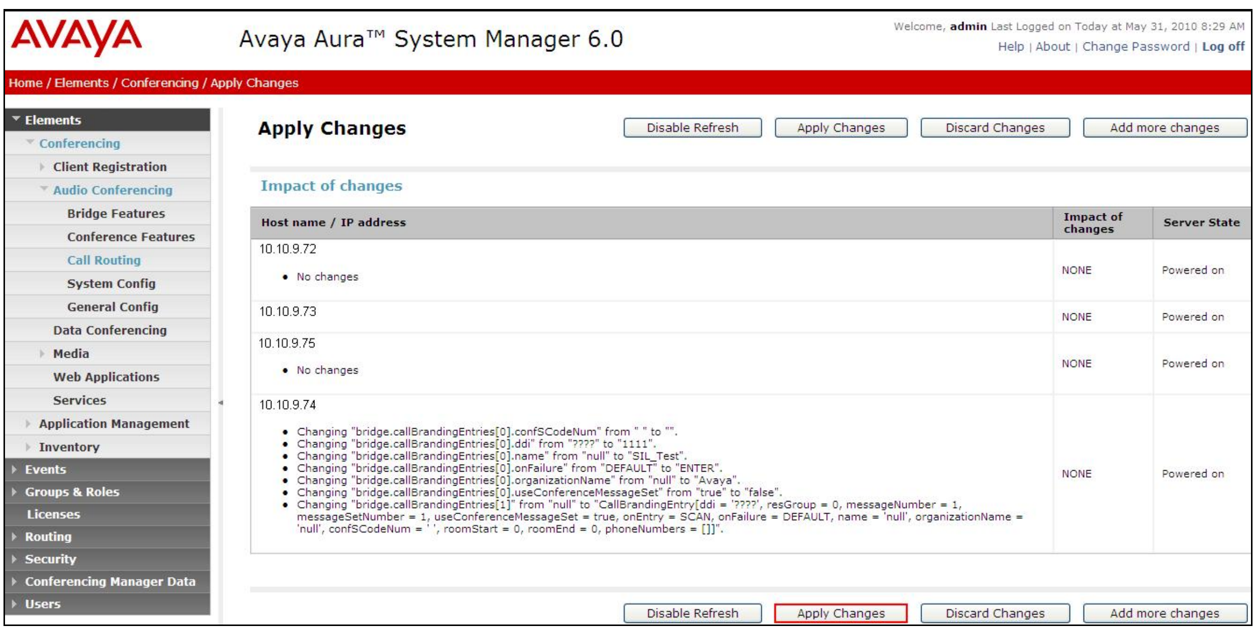
From the right panel menu select **Done**.



From the right panel menu select **Save**.



From the right panel menu select **Apply Changes**.



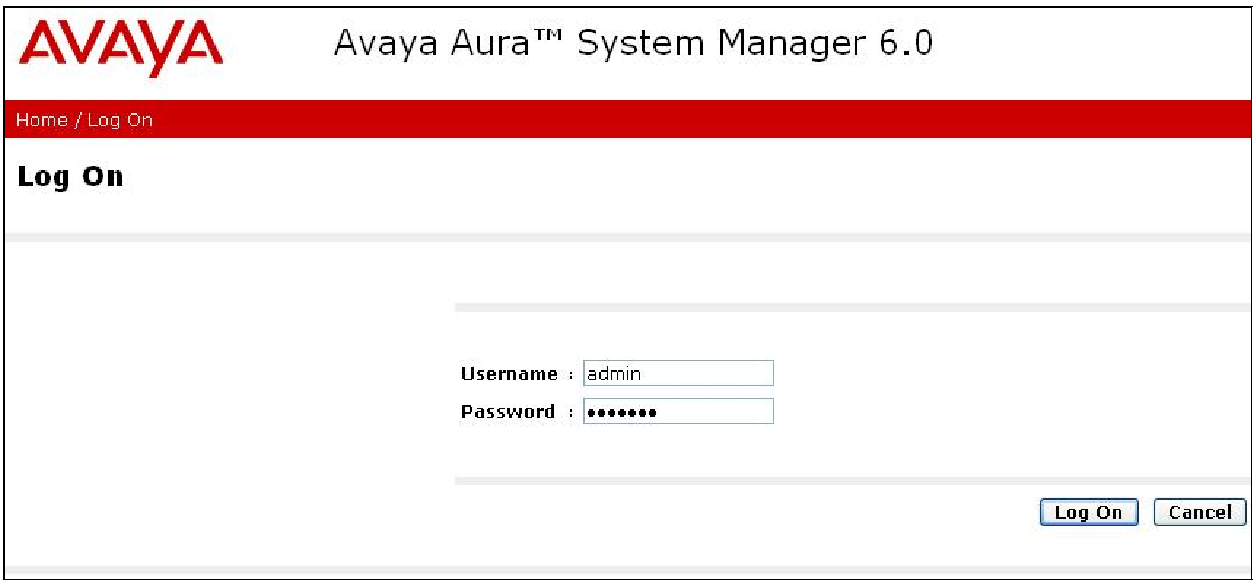
4. Configure Avaya Aura™ Session Manager

This section provides the procedures for configuring Session Manager. Session Manager is configured via the System Manager. The procedures include the following areas:

- Log in to Avaya Aura™ System Manager
- Administer SIP domain
- Administer SIP Entities
- Administer Entity Links
- Administer Time Ranges
- Administer Routing Policies
- Administer Dial Patterns
- Administer Session Manager

4.1. Log in to Avaya Aura™ System Manager

Access Avaya Aura™ System Manager using a Web Browser and enter **https://<ip-address>/SMGR**, where <ip-address> is the IP address of System Manager. Log in using appropriate credentials and accept the subsequent Copyright Legal Notice.

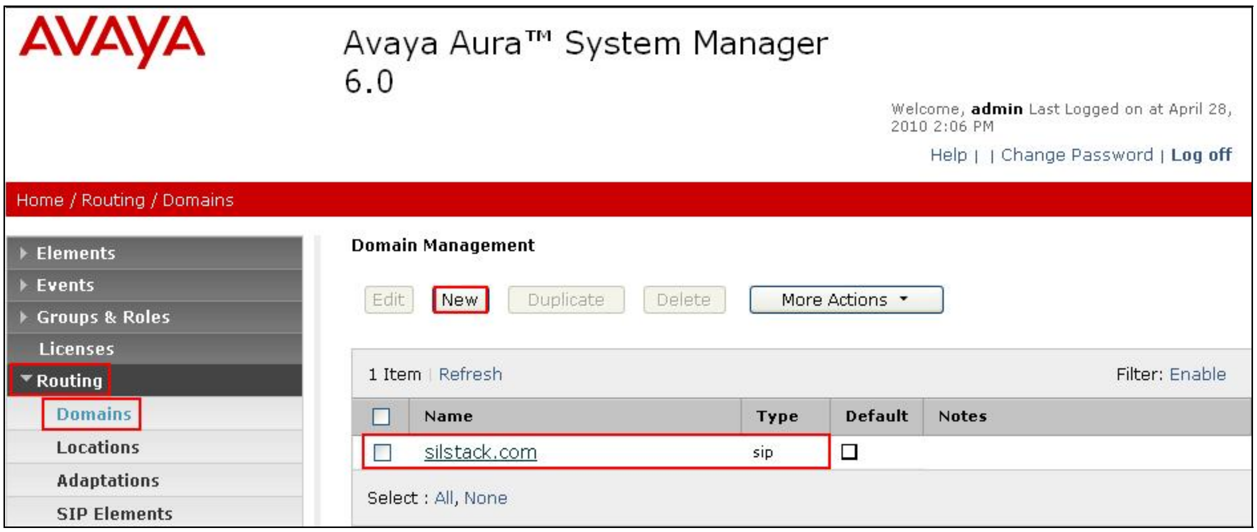


4.2. Administer Domains

Add the SIP authoritative domain for the communications infrastructure by selecting **Routing → Domains** on the left panel menu and click **New** to create a new domain entry. Select the following parameters, leaving the remaining parameters at their default values.

- **Name** The authoritative domain name (e.g., **silstack.com**)
- **Type** Select **sip**
- **Notes** Description for the domain (optional)

Click **Commit** (not shown) to save changes.



4.3. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management. Locations are added to the configuration for both Mediant 2000 and Conferencing Standard Edition. To add a location, select **Routing → Locations** on the left panel menu and click **New** (not shown). Select the following parameters, leaving the remaining parameters at their default values.

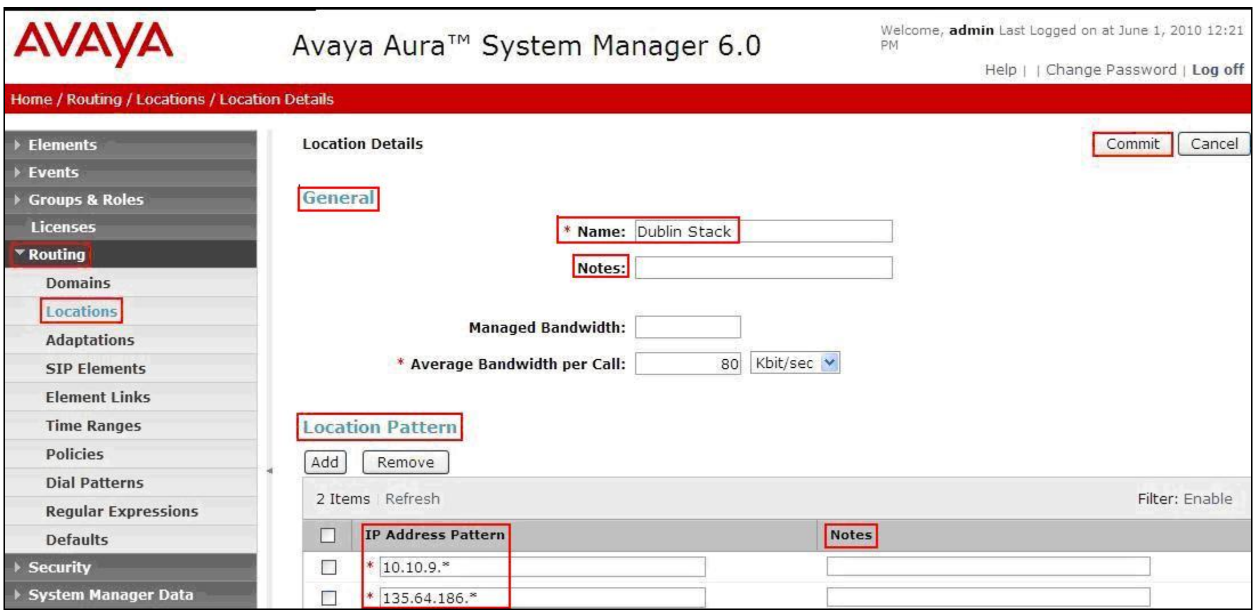
Under **General**:

- **Name:** A descriptive name (e.g., **Dublin Stack**)
- **Notes:** Descriptive text (optional**)**

Under **Location Pattern**:

- **IP Address Pattern:** A pattern used to logically identify the location (e.g.,**10.10.9.*** and **135.64.186.***)
- **Notes:** Descriptive text (optional)

Click **Commit** to save changes.



4.4. Add SIP Elements

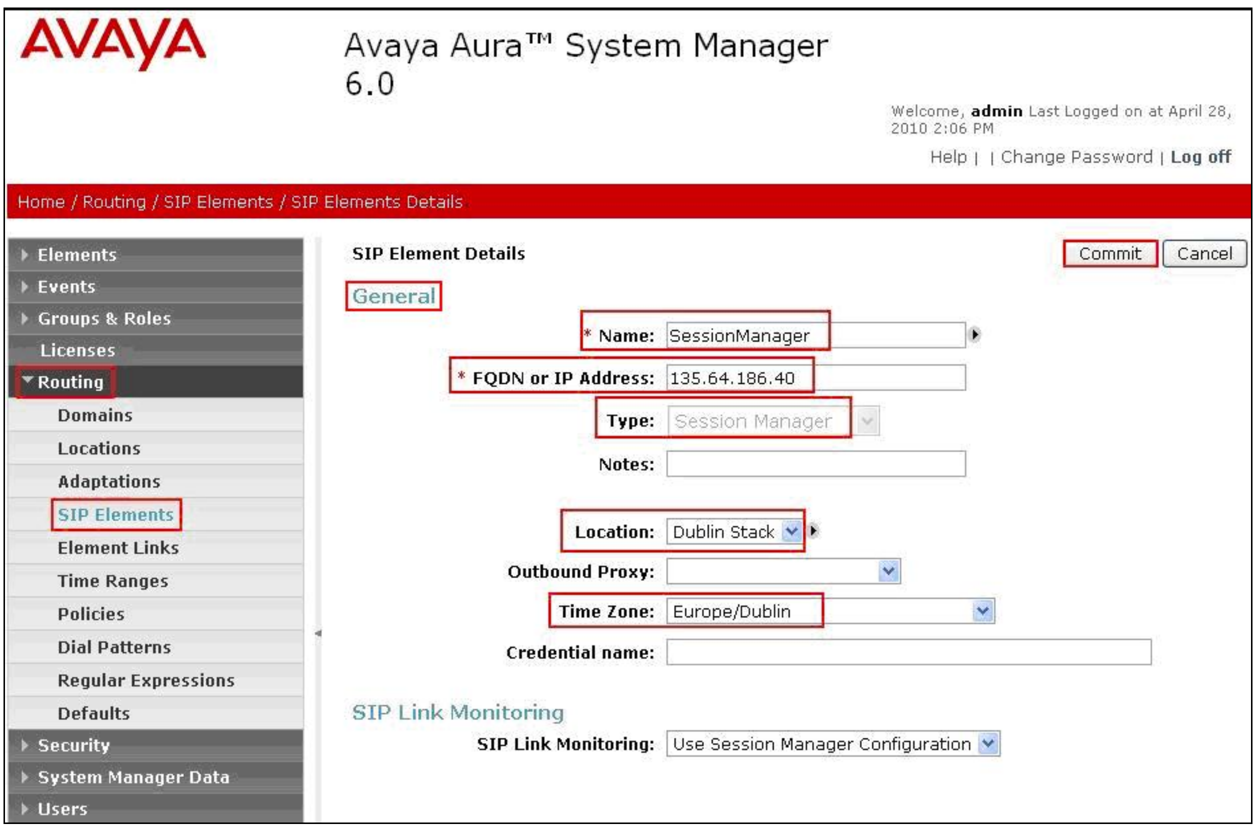
Note that the "SIP Elements" menu option shown in the screen below was changed to "SIP Entities" in the GA release. For the purposes of these Application Notes, the terms "Element" and "Entity" are interchangeable. SIP Elements must be added for Session Manager and for each SIP-based telephony system supported by it using SIP trunks. In the sample configuration, a SIP Element is added for Session Manager and Mediant 2000. To add a SIP Element, select **Routing → SIP Elements** on the left panel menu and click **New** (not shown). Select the following parameters, leaving the remaining parameters at their default values.

Under **General**:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the Session Manager or the signaling interface on the telephony system.
- **Type:** Select between **SessionManager** for Session Manager, **Gateway** for Mediant 2000 and **SIP Trunk** for Conferencing
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

The following screen shows addition of Session Manager. The IP address used is that of the Software Asset Card.

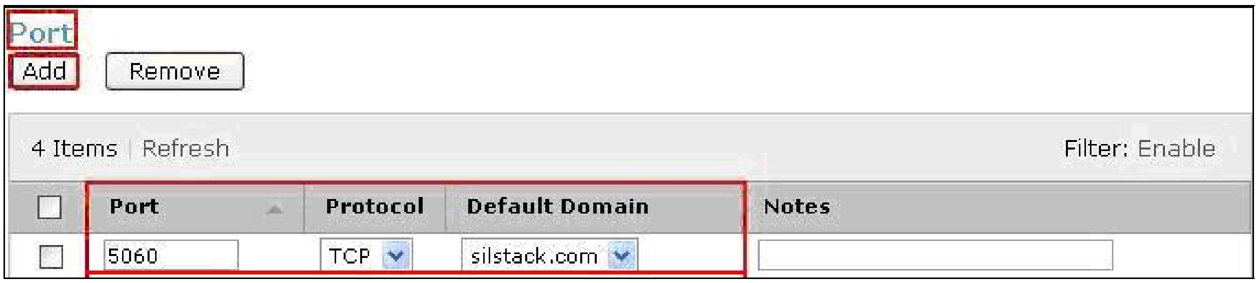
Click **Commit** to save changes.



Under **Port**, click **Add**, select the following parameters, leaving the remaining parameters at their default values. Note that the adding of ports only applies when the SIP Element is a Session Manager.

- **Port** Port number on which the system listens for SIP requests.
- **Protocol** Transport protocol to be used to send SIP requests.
- **Default Domain** The domain used for the enterprise (e.g., **silstack.com**).

Click **Commit** (not shown) to save changes.

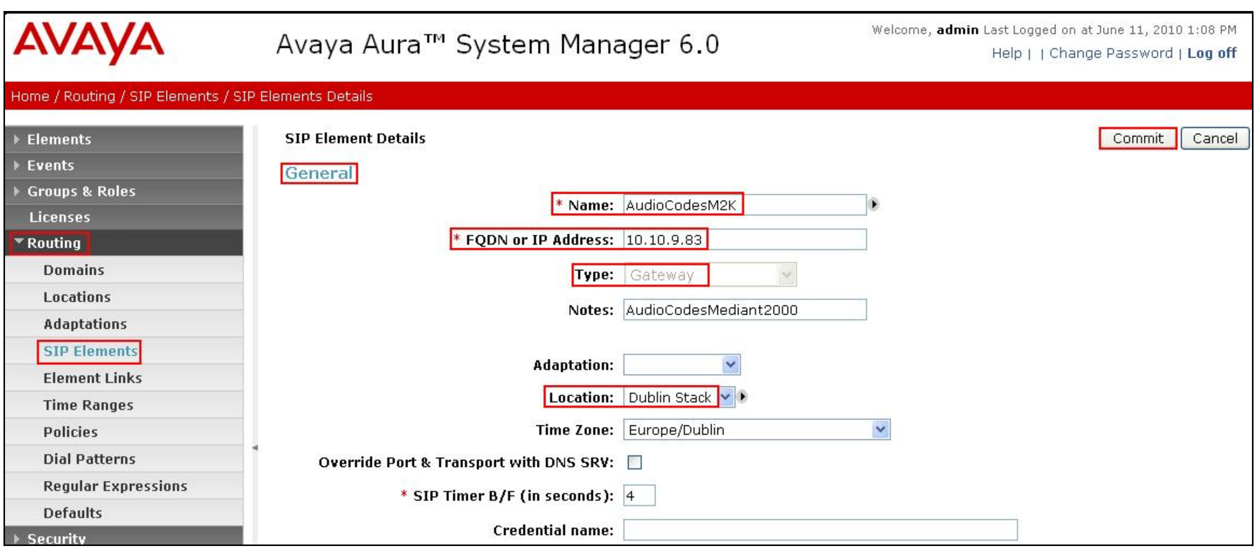


The following screen shows addition of Mediant 2000. Select the following parameters, leaving the remaining parameters at their default values.

Under **General**:

- **Name:** A descriptive name
- **FQDN or IP Address:** IP address of the Mediant 2000
- **Type:** Select **Gateway** for Mediant 2000
- **Location:** Select one of the locations defined previously

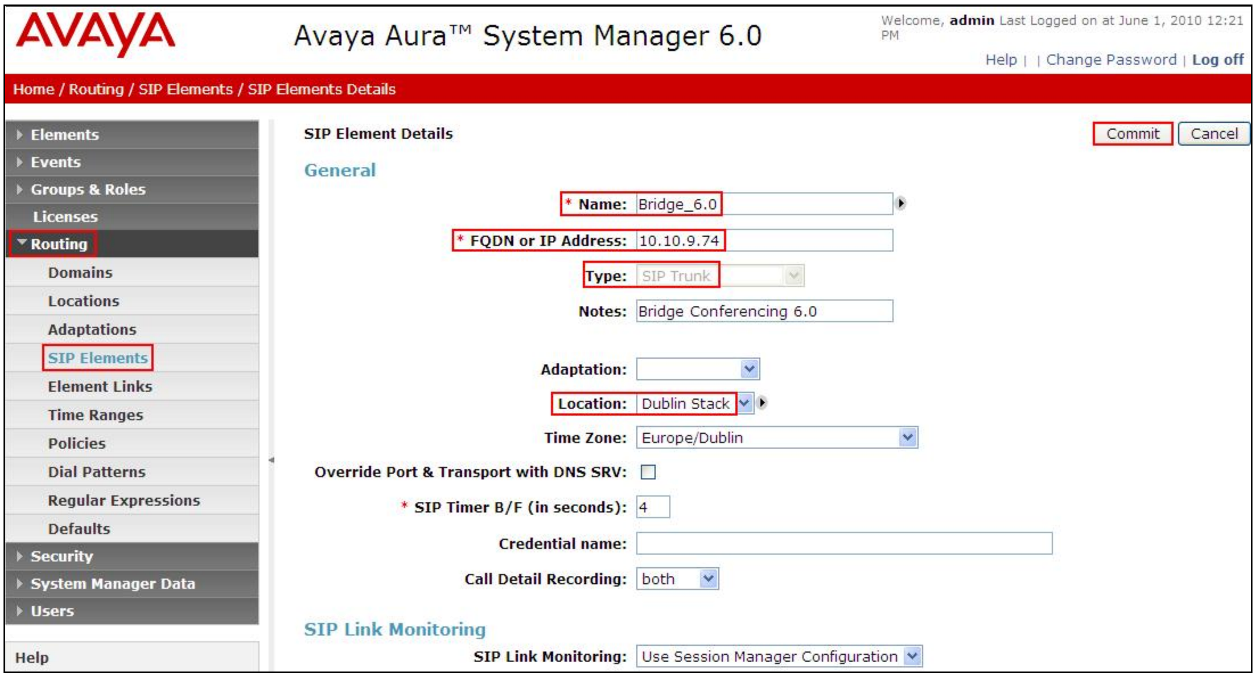
Click **Commit** to save changes.



The following screen shows addition of Conferencing Standard Edition (**Bridge_6.0**). Select the following parameters, leaving the remaining parameters at their default values.
Under **General**:

- **Name:** A descriptive name
- **FQDN or IP Address:** IP address of the Conferencing Bridge
- **Type:** Select **SIP Trunk** for the Conferencing Bridge
- **Location:** Select one of the locations defined previously

Click **Commit** to save changes.

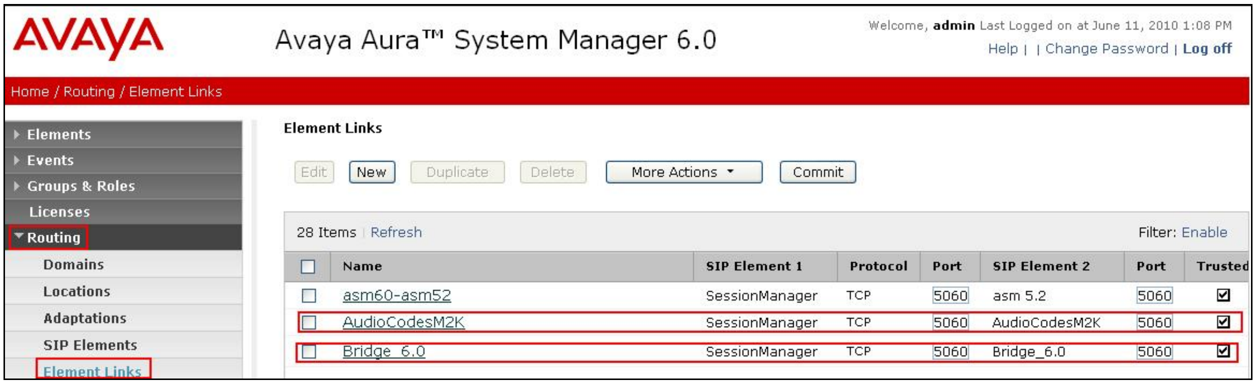


4.5. Add Element Links

Note that the “Element Links” menu option shown in the screen below was changed to “Entity Links” in the GA release. For the purposes of these Application Notes, the terms “Element” and “Entity” are interchangeable. A SIP trunk between a Session Manager and a telephony system is described by an Element Link. To add an Element Link, select **Routing → Element Links** on the left panel menu and click **New**. Select the following parameters in the rows that are displayed:

- **Name** An informative name
- **SIP Element 1** Select **SessionManager**
- **Protocol** Transport protocol to be used to send SIP requests
- **Port** Port number to which the other system sends its SIP requests
- **SIP Element 2** The other SIP Element for this link, created in **Section 4.4**
- **Port** Port number to which the other system expects to receive SIP requests
- **Trusted** Whether to trust the other system

Click **Commit** to save changes. The following screen shows the Element Links used in the sample network.

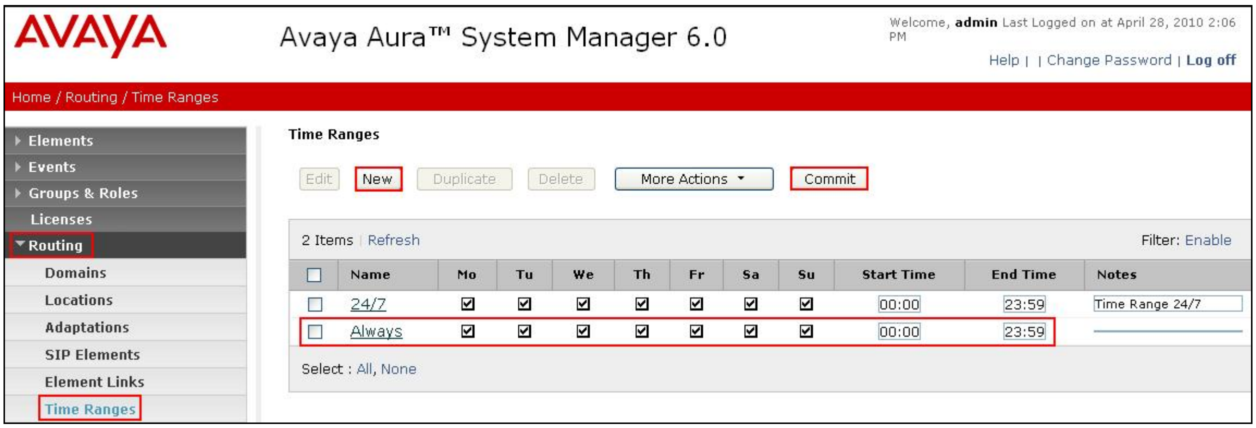


4.6. Administer Time Ranges

Before adding routing policies (see next section), time ranges must be defined during which the policies will be active. In the sample configuration, one policy was defined that would allow routing to occur at any time. To add this time range, select **Routing → Time Ranges** on the left panel menu, then click **New**. Select the following parameters, leaving the remaining parameters at their default values.

- **Name:** A descriptive name (e.g. **Always**)
- **Mo** through **Su** Check the box under each of these headings
- **Start Time** Enter **00:00**
- **End Time** Enter **23:59**

Click **Commit** to save this time range.



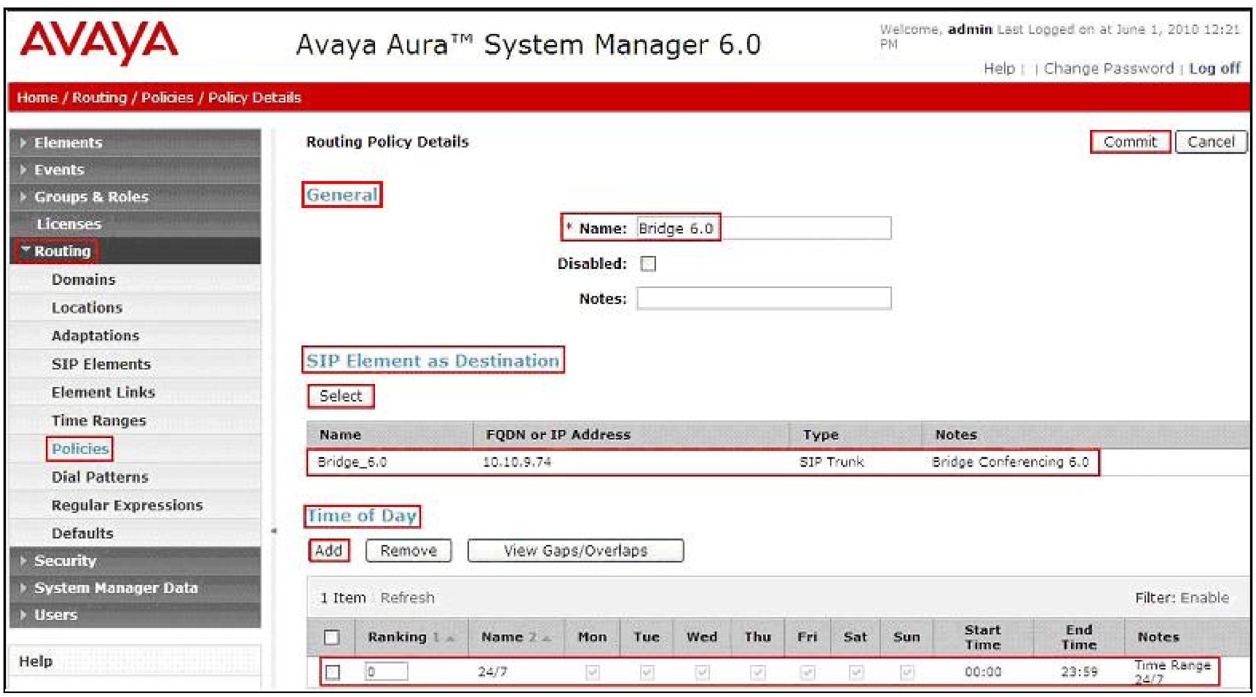
4.7. Administer Routing Policies

A routing policy must be created to direct how calls will be routed to a system. Note that the "Policies" menu option shown in the screen below was changed to "Routing Policies" in the GA release. To add a routing policy, select **Routing → Policies** on the left panel menu and then click **New** (not shown). Select the following parameters, leaving the remaining parameters at their default values.

Under **General**:

- **Name** An informative name (e.g., **Bridge 6.0)**

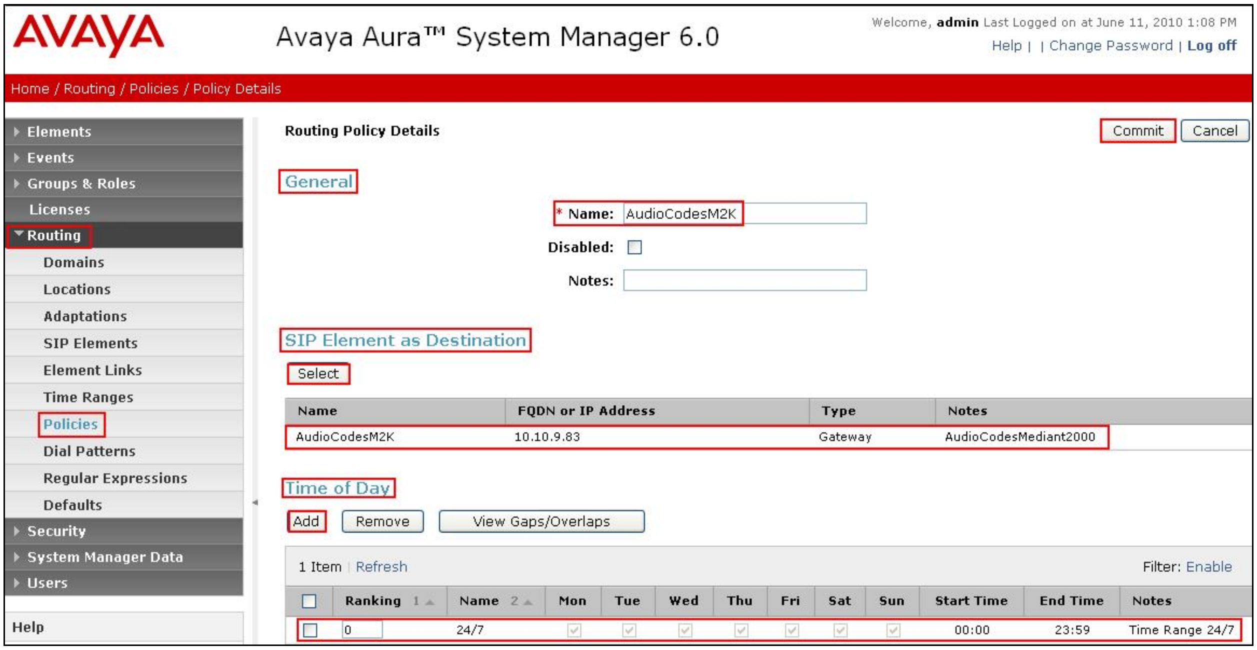
Note that the phrase "SIP Element as Destination" shown in the screen below was changed to "SIP Entity as Destination" in the GA release. For the purposes of these Application Notes, the terms "Element" and "Entity" are interchangeable. Under **SIP Element as Destination,** click **Select**, and then select the appropriate SIP Element to which this routing policy applies. Under **Time of Day**, click **Add**, and then select the time range configured in the previous step. The following screen shows the **Routing Policy Details** for Conferencing. Click **Commit** to save changes.



Select the following parameters, leaving the remaining parameters at their default values.
Under **General**:

- **Name** An informative name (e.g., **AudioCodesM2K**)

Under **SIP Element as Destination,** click **Select**, and then select the appropriate SIP Element to which this routing policy applies. Under **Time of Day**, click **Add**, and then select the time range configured in the previous step. The following screen shows the **Routing Policy Details** for Mediant 2000. Click **Commit** to save changes.



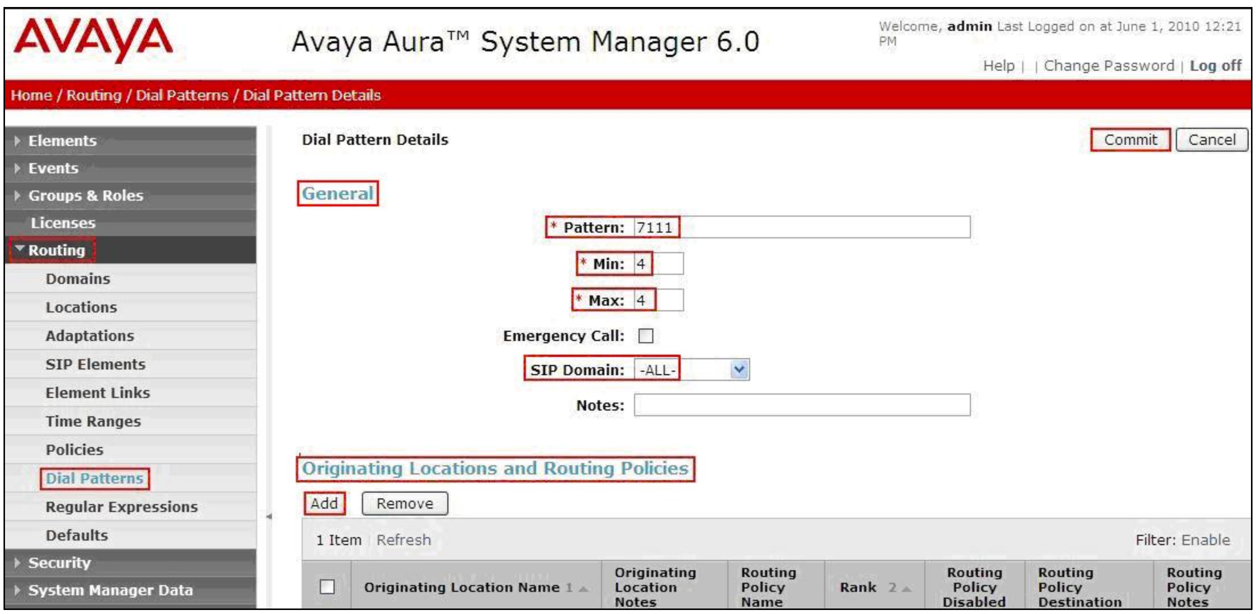
4.8. Administer Dial Patterns

A dial pattern must be defined that will direct calls to the appropriate telephony system. In the sample network, the 4-digit extension **7111** will be used as the number that resides on Conferencing. Select **Routing → Dial Patterns** on the left panel menu and then click **New** (not shown). Select the following parameters, leaving the remaining parameters at their default values.

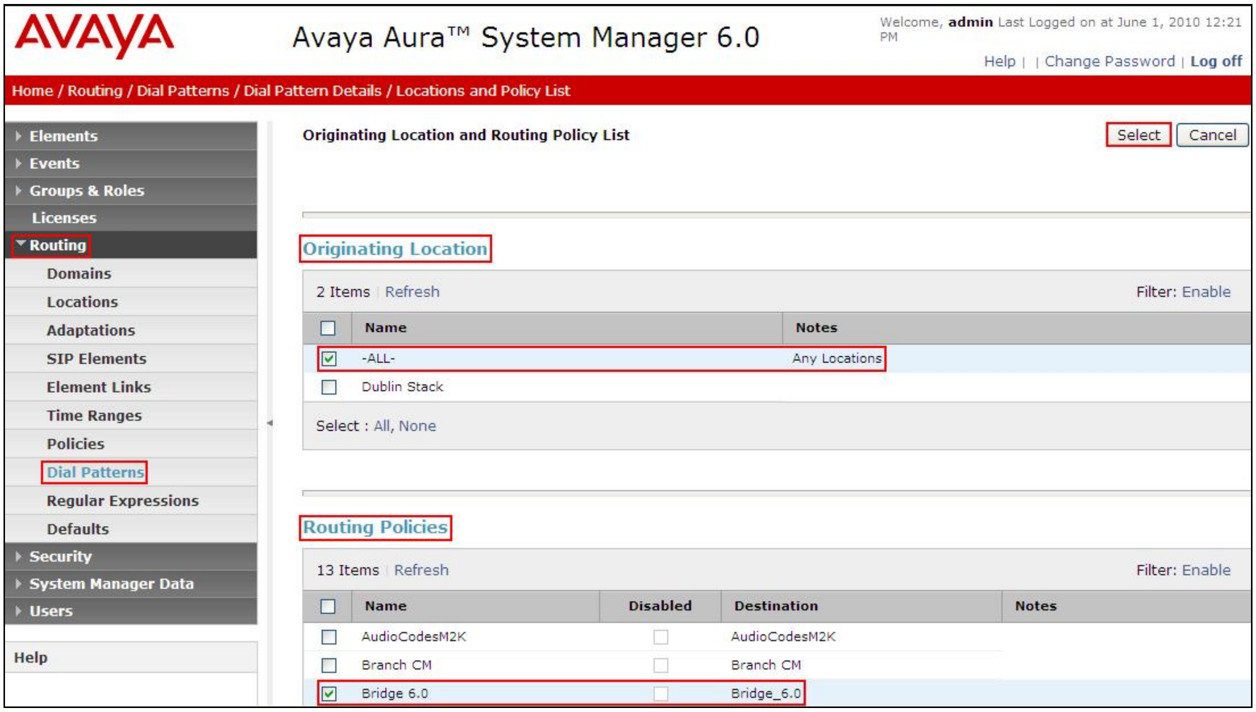
Under **General**

- **Pattern** Dialed number or prefix i.e. **7111**
- **Min** Minimum length of the dialed number i.e. **4**
- **Max** Maximum length of the dialed number i.e. **4**
- **SIP Domain** Select **ALL**
- **Notes** Comment on purpose of dial pattern

Navigate to **Originating Locations and Routing Policies** and select **Add**.



Under **Originating Location** select all locations by checking the box next to **ALL** and under **Routing Policies** select the Routing Policy created in **Section 4.7**. Click **Select** to confirm the chosen options and return to the Dial Pattern screen (shown above). Click **Commit** to save changes shown in the previous screen.



4.9. Administer Avaya Aura™ Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Select **Elements → Session Manager → Session Manager Administration** on the left panel menu. Then click **Add** (not shown) and fill in the following parameters, leaving the remaining parameters at their default values.

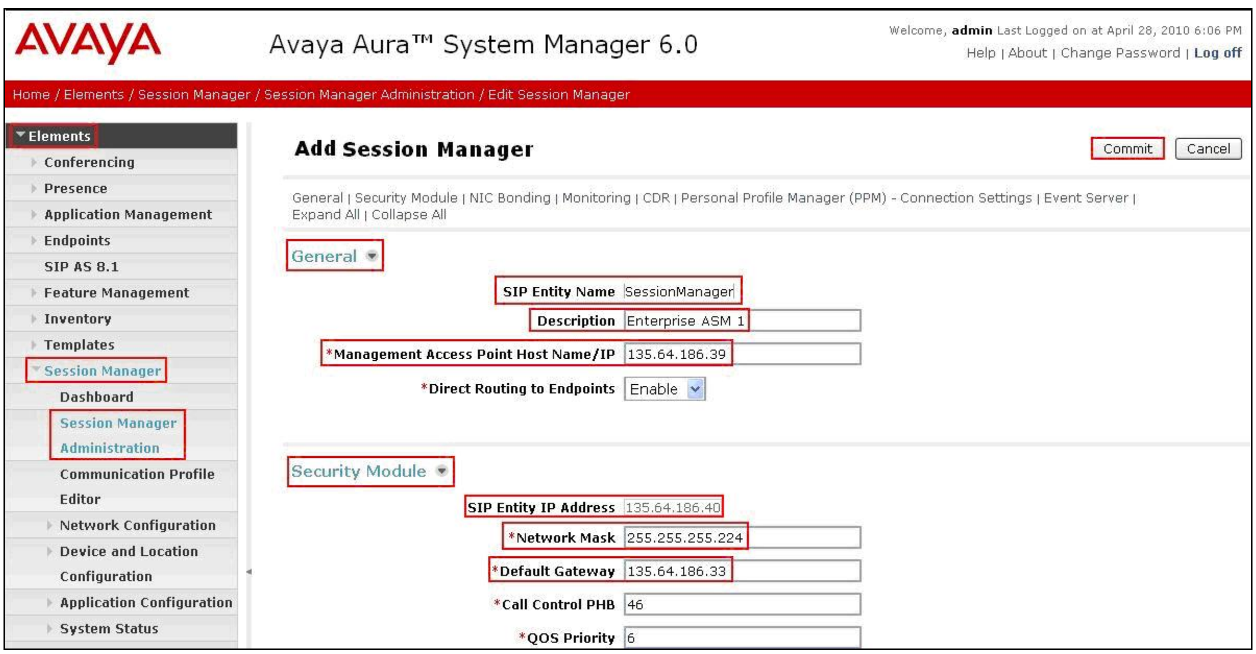
Under **General:**

- **SIP Entity Name** Select the name of the SIP Entity added for Session Manager
- **Description** Descriptive comment (optional)
- **Management Access Point Host Name/IP**
  Enter the IP address of the Session Manager management interface

Under **Security Module:**

- **SIP Entity IP Address** IP Address of Software Asset card
- **Network Mask** Enter the network mask corresponding to the IP address of Session Manager
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager

Click **Commit** to add this Session Manager.



4.10. Add Avaya Aura™ Communication Manager as a Feature Server

In order for Communication Manager to provide configuration and Feature Server support to SIP phones when they register to Session Manager, Communication Manager must be added as an application.

4.10.1. Create an Application Entity

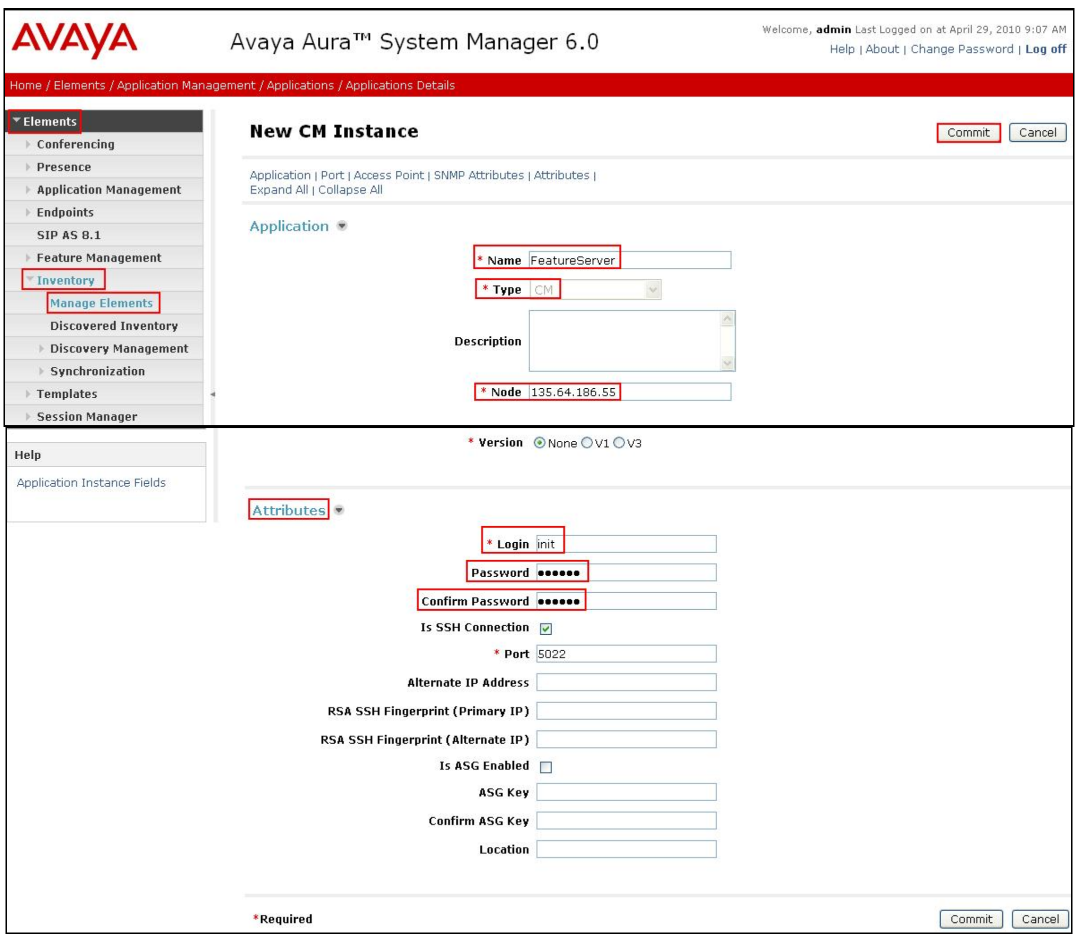
Select **Elements → Inventory → Manage Elements** on the left panel menu. Click on **New** (not shown). Select the following parameters, leaving the remaining parameters at their default values.

- **Name** A descriptive name i.e. **FeatureServer**
- **Type** Select **CM**
- **Node** Enter the IP address for CM SAT access

Navigate to the **Attributes** section and enter the following:

- **Login** Login used for SAT access
- **Password** Password used for SAT access
- **Confirm Password** Password used for SAT access

Click on **Commit** to save.

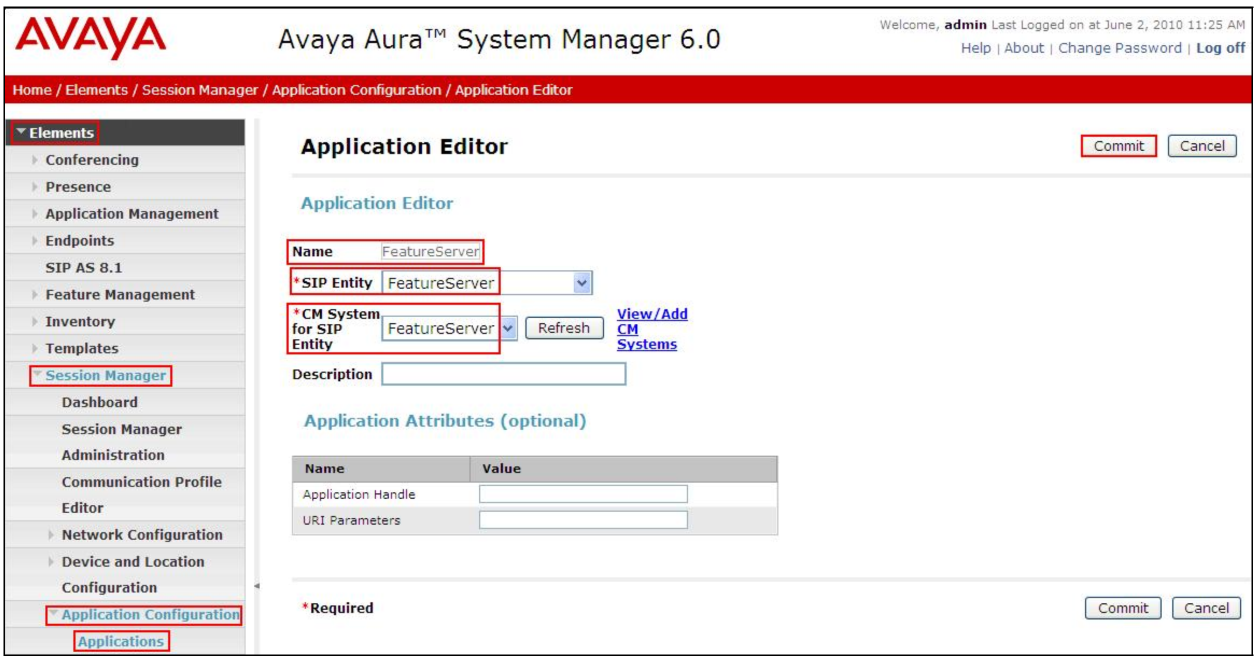


4.10.2. Create a Feature Server Application

Select **Elements → Session Manager → Application Configuration → Applications** on the left panel menu. Click on **New** (not shown). Select the following parameters, leaving the remaining parameters at their default values.

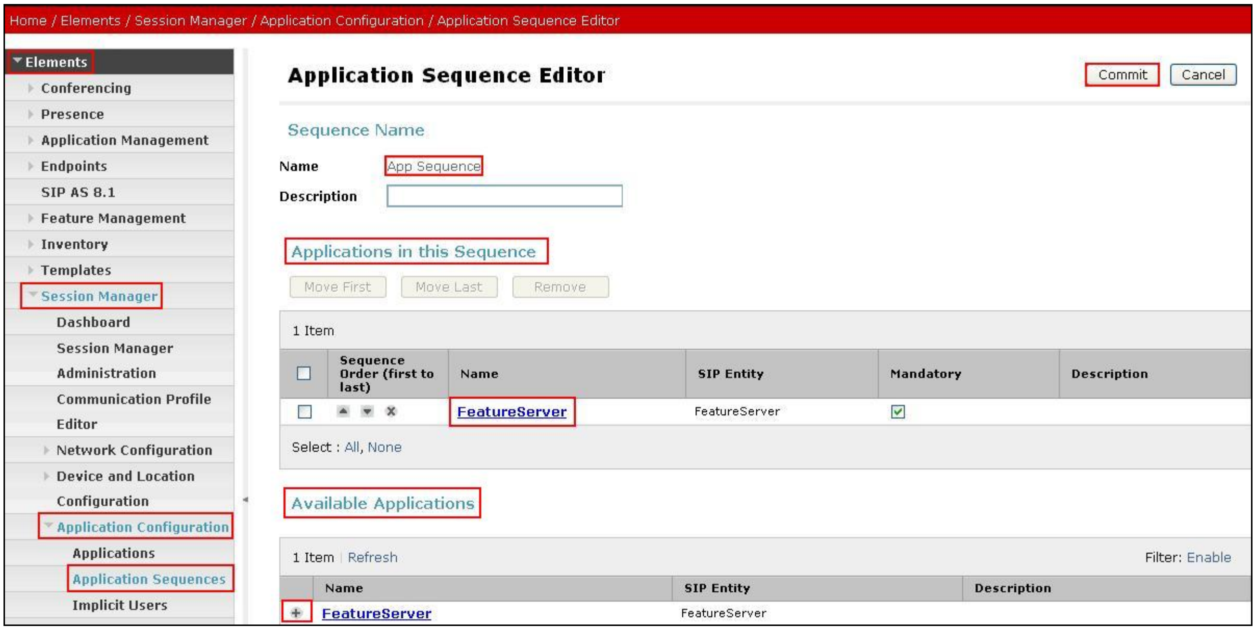
- **Name** A descriptive name
- **SIP Entity** Select the CM Application Entity defined in **Section 4.10.1**
- **CM System for SIP Entity** Select the CM Application Entity defined in **Section 4.10.1**

Click on **Commit** to save.



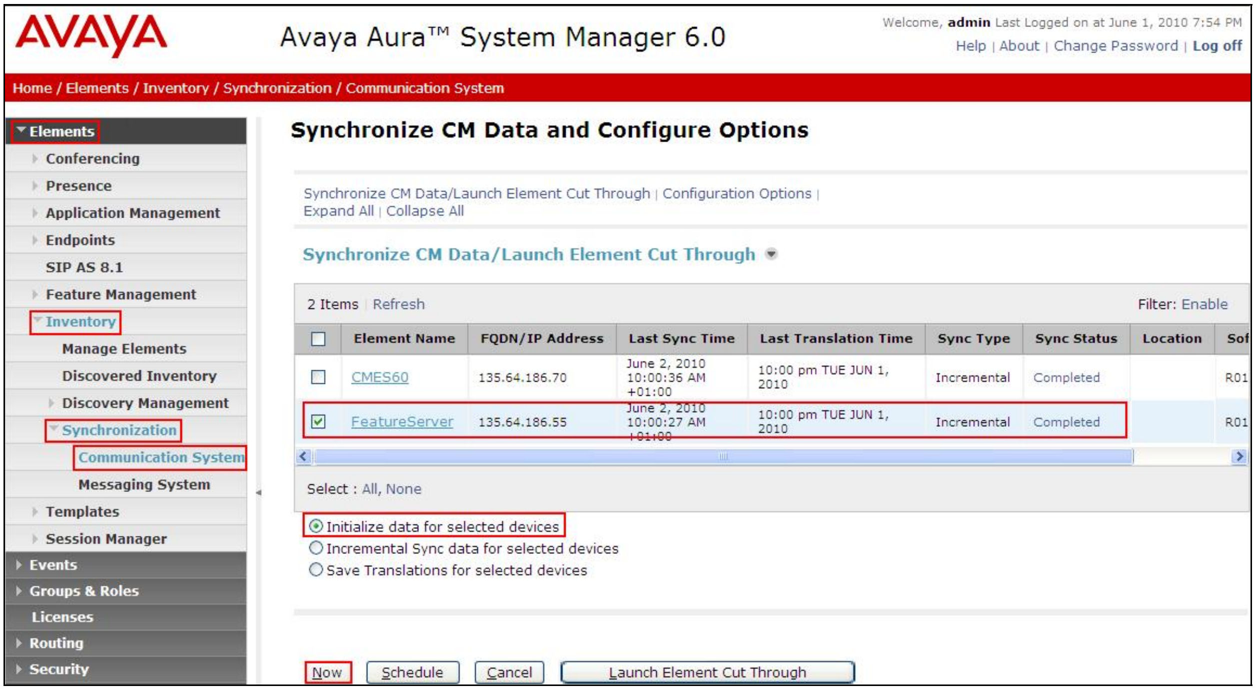
4.10.3. Create a Feature Server Application Sequence

Select **Elements → Session Manager → Application Configuration → Application Sequences** on the left panel menu. Click on **New** (not shown). Enter a descriptive name in the **Name** field. Click on the **+** sign next to the appropriate **Available Applications** and they will move up to the **Applications in this Sequence** section. Click on **Commit** to save.



4.10.4. Synchronize Avaya Aura™ Communication Manager Data

Select **Elements → Inventory → Synchronization → Communication System** on the left panel menu. Select the appropriate **Element Name** from the list. Check the **Initialize data for selected devices** box. Then click on **Now**. This may take some time.

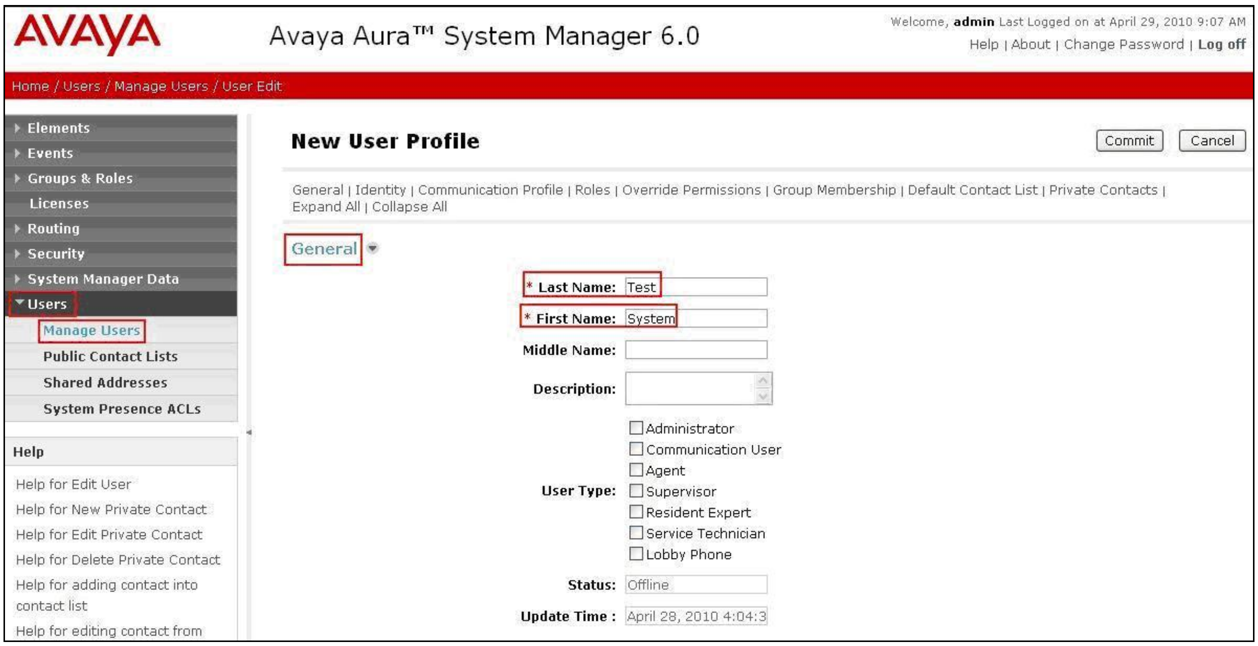


4.11. Add Users for SIP Phones

Users must be added via Session Manager and the details will be updated on Communication Manager. Select **Users → Manage Users** on the left panel menu. Then click on **New** (not shown). Select the following parameters, leaving the remaining parameters at their default values.

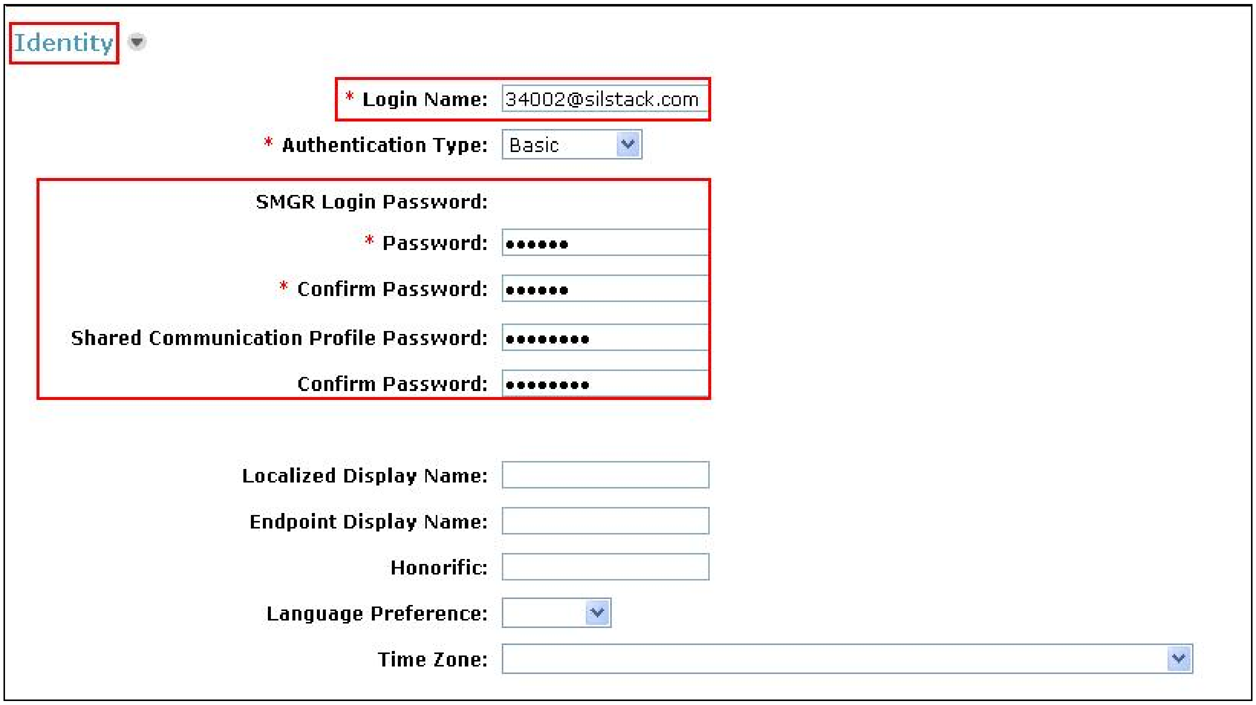
Under **General**:

- **Last Name** Any name
- **First Name** Any name



Navigate to the **Identity** section, select the following parameters, leaving the remaining parameters at their default values.

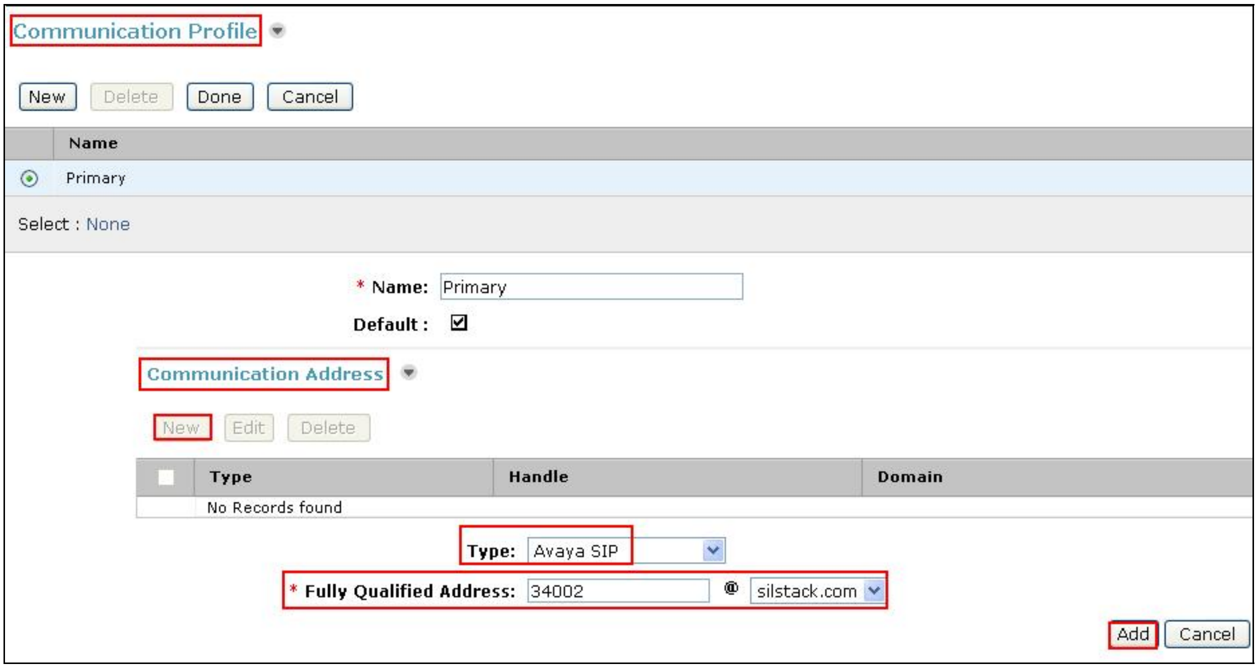
- **Login Name** The desired phone-extension-number@domain where domain was defined in **Section 4.2**
- **Password** Password for user to log into SMGR
- **Shared Communication Profile Password**
  Password to be entered by the user when logging into the phone



Navigate to and click on **Communication Profile** section to expand that section, use the default values. Then click on **Communication Address** to expand that section, click **New** and enter the following:

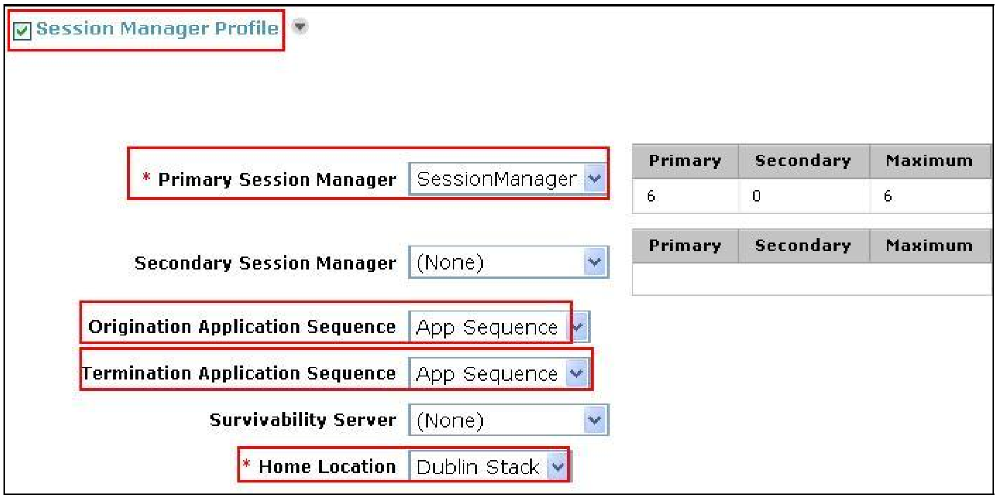
- **Type** Select **Avaya SIP**
- **Fully Qualified Address** Enter the extension-number@domain

Click on **Add**.



Navigate to and click on **Session Manager Profile** section to expand. Select the following parameters, leaving the remaining parameters at their default values.

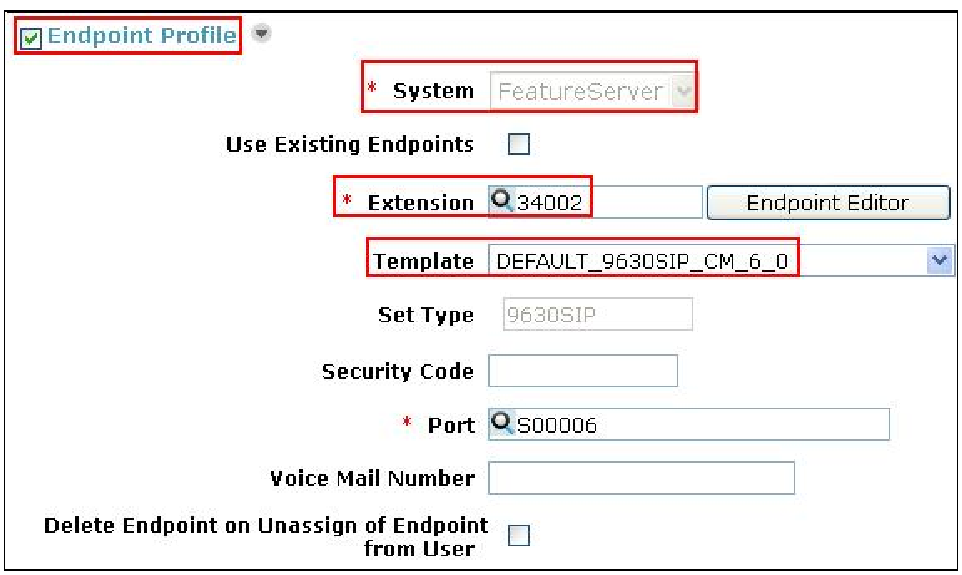
- **Primary Session Manager** Select **SessionManager**
- **Origination Application Sequence** Select **App Sequence**
- **Termination Application Sequence** Select **App Sequence**
- **Home Location** Select **Dublin Stack**



Click on **Endpoint Profile** to expand that section. Select the following parameters, leaving the remaining parameters at their default values.

- **System** Select the CM Entity created in **Section 4.11**
- **Extension** Enter a desired extension number
- **Template** Select a telephone type template

Click on **Commit** to save (not shown).



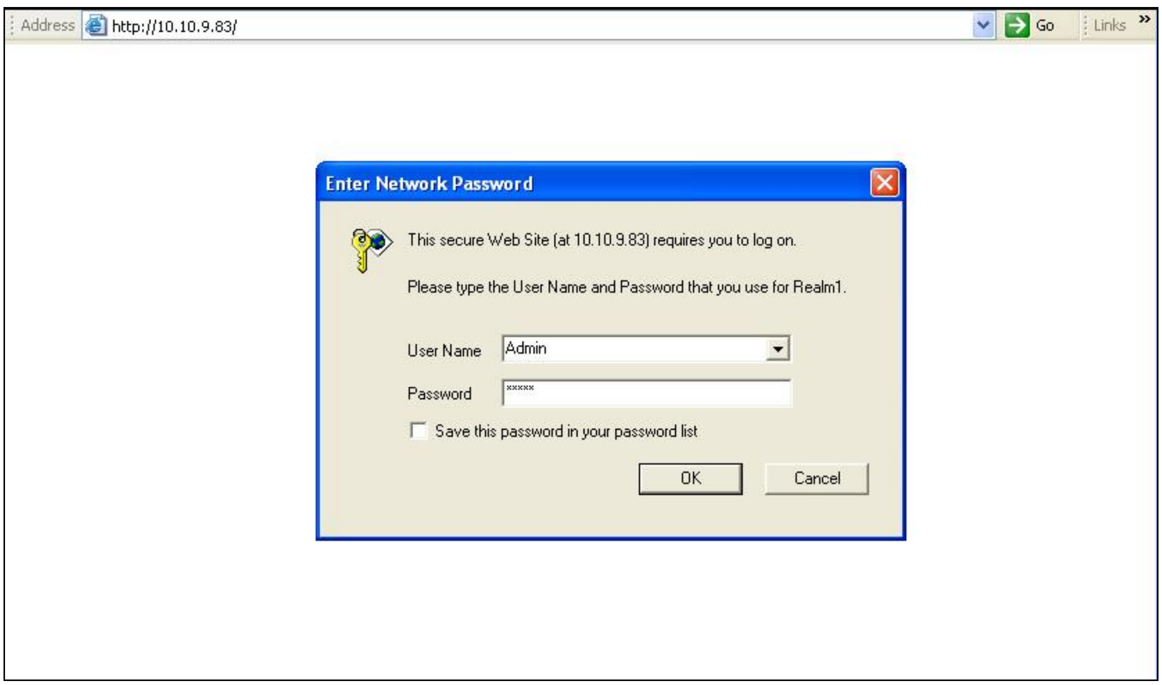
5. Configure AudioCodes Mediant 2000 Gateway

The following sections describe the configuration steps required to implement E1/PRI QSIG trunks on the Mediant 2000, using the web interface. It is assumed that basic hardware and software installation has been performed, details can be found in reference **[10]**. This section focuses on the following configuration areas:

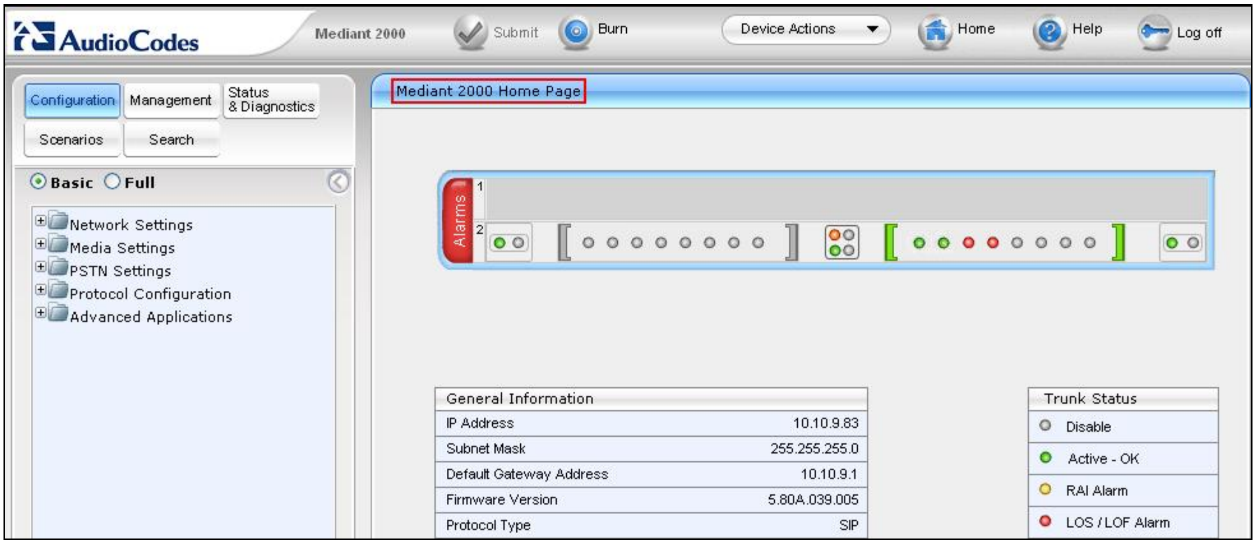
- Access Web Configuration Interface
- Administer TDM Bus Settings
- Administer PSTN Trunk Settings
- Administer SIP Protocol Parameters
- Administer Audio Codecs
- Administer DTMF Signaling
- Administer Proxy & Registration
- Administer Routing Tables
- Administer SIP General Parameters for TCP
- Save the Configuration

5.1. Access Web Configuration Interface

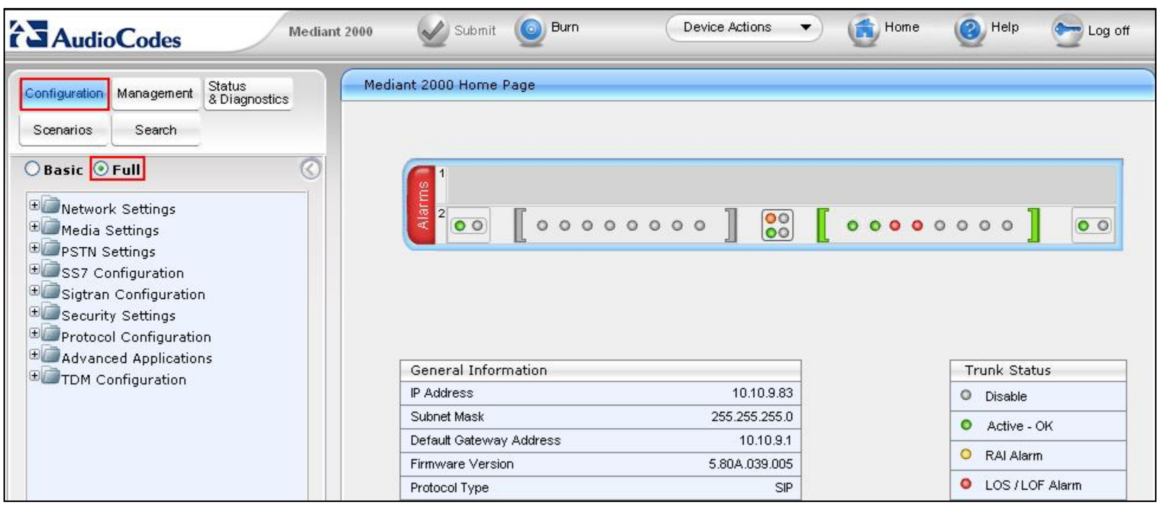
Access the Mediant 2000 GUI using a Web Browser and entering **http://<ip-address>**, where <ip-address> is the IP address of Mediant 2000. Log in using appropriate credentials and accept the subsequent Copyright Legal Notice.



The **Mediant 2000 Home Page** screen is displayed.



Select **Configuration** and set the mode to **Full** on the left panel menu. The menus on the left can be expanded as necessary to configure the appropriate features, as described in the following sections.

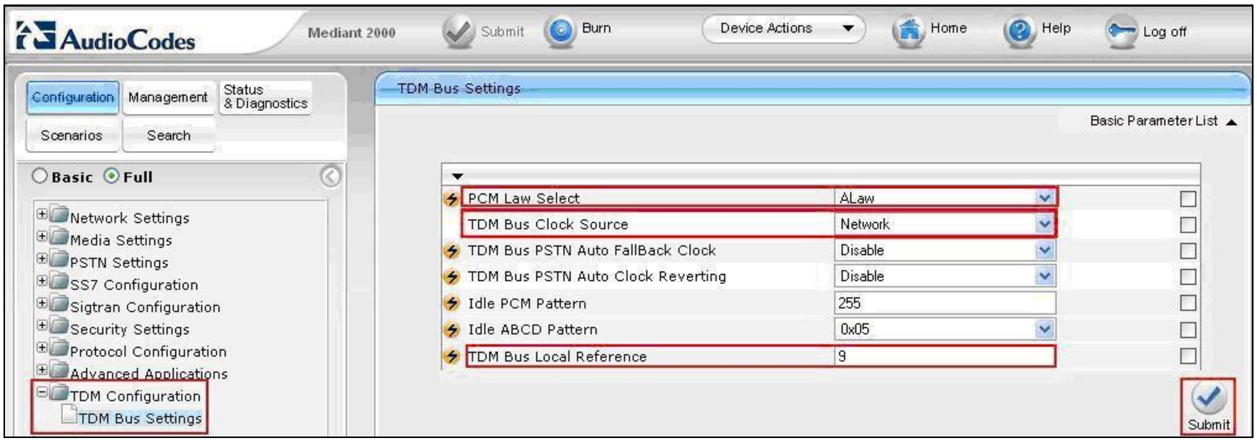


5.2. Administer TDM Bus Settings

Select **TDM Configuration → TDM Bus Settings** on the left panel menu. In the sample configuration the internal clock of the Mediant 2000 provides the clocking for the E1 PRI trunk. Select the following parameters, leaving the remaining parameters at their default values.

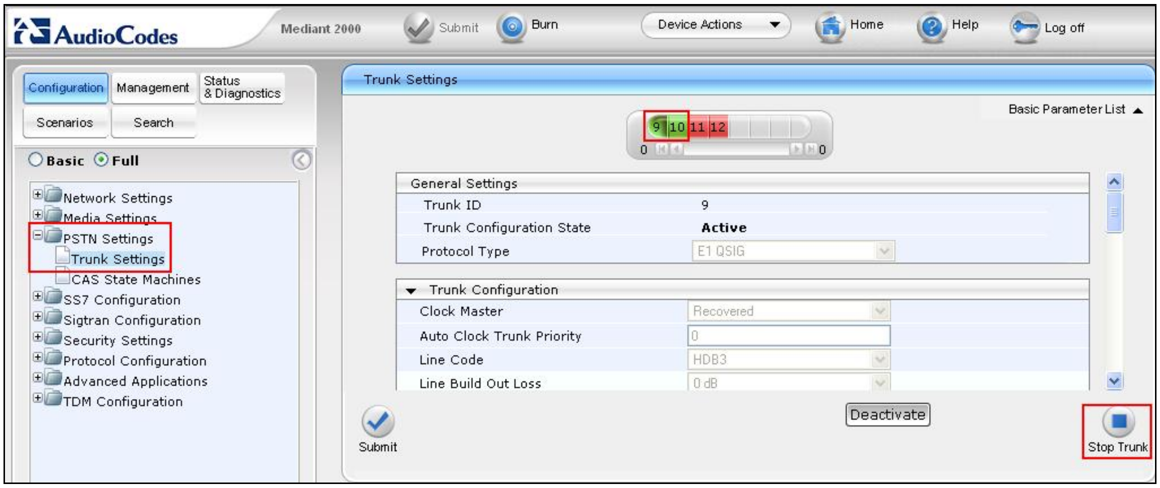
- **PCM Law Select** SelectA-Law.
- **TDM Bus Clock Source:** Select **Network**
- **TDM Bus Local Reference** Select **9**, first trunk that will take the clocking.

Click on **Submit** to save changes.



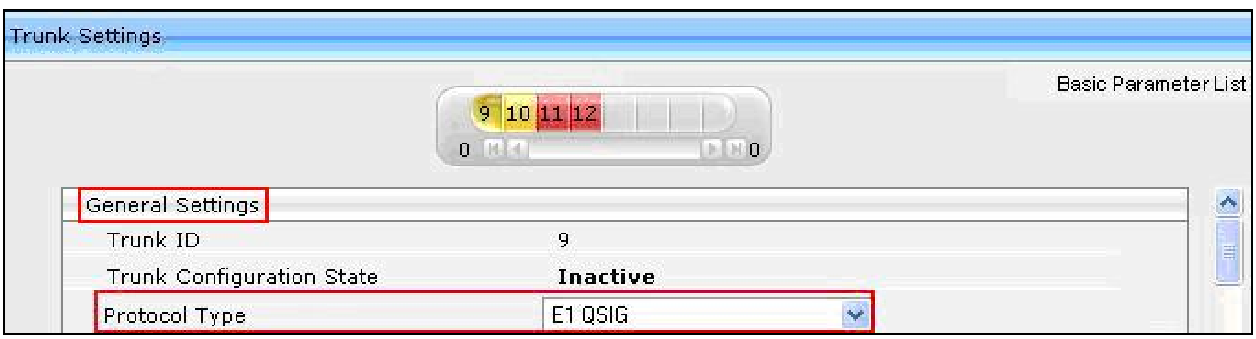
5.3. Administer PSTN Trunk Settings

Select **PSTN Settings → Trunk Settings** on the left panel menu. Click **Stop Trunk**, which will enable editing of the parameters.



Select the following parameters, leaving the remaining parameters at their default values.
Under **General Settings:**

- **Protocol Type:** Select **E1 QSIG**

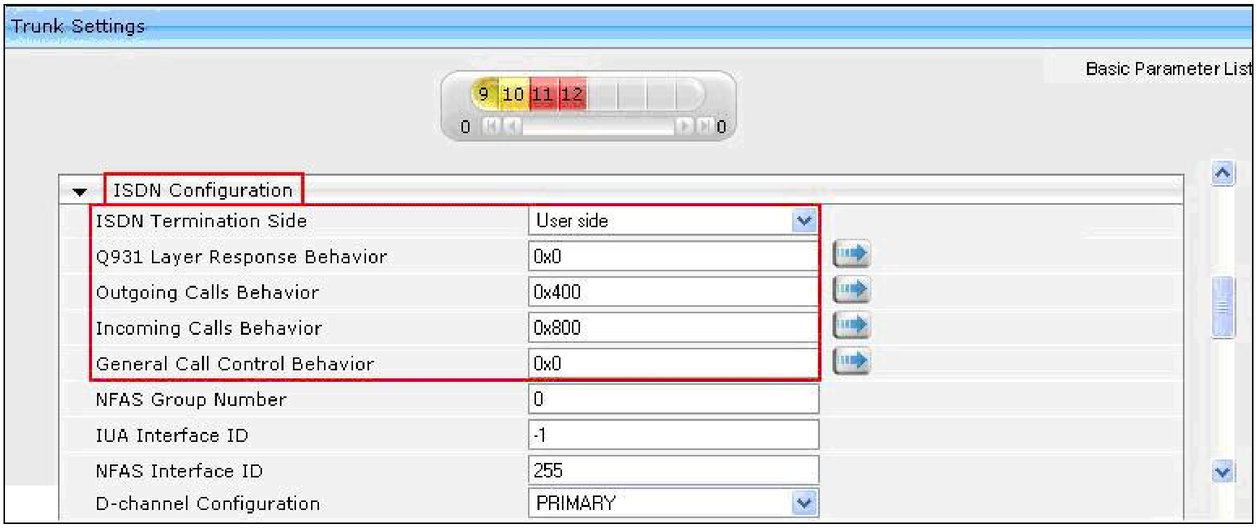


Under **Trunk Configuration**:

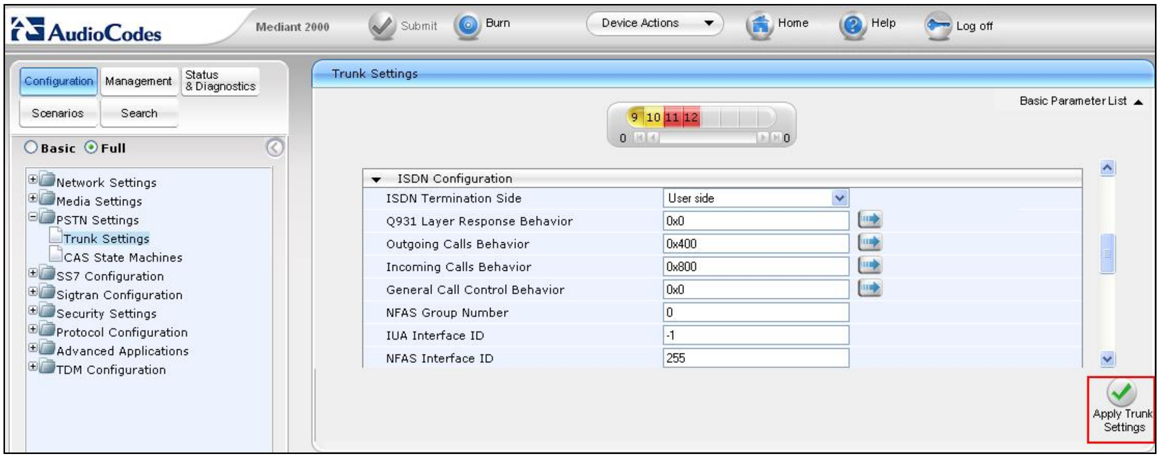
- **Clock Master:** Select **Recovered**
- **Line Code:** Select **HDB3**
- **Framing Method:** Select **E1 Framing MFF CRC4 EXT**

Under **ISDN Configuration:**

- **ISDN Termination Side:** Select **User side**
- **Q931 Layer Response Behavior**: Select **0x0**
- **Outgoing Calls Behavior:** Select **0x400**
- **Incoming Calls Behavior:** Select **0x800**
- **General Call Control Behavior** Select **0x0**



Click on **Apply Trunk Settings** to save all of the above changes and put the trunk into service. Successful trunk configuration will be indicated by the green status indications for the trunk board, as shown in **Section 5.1**.

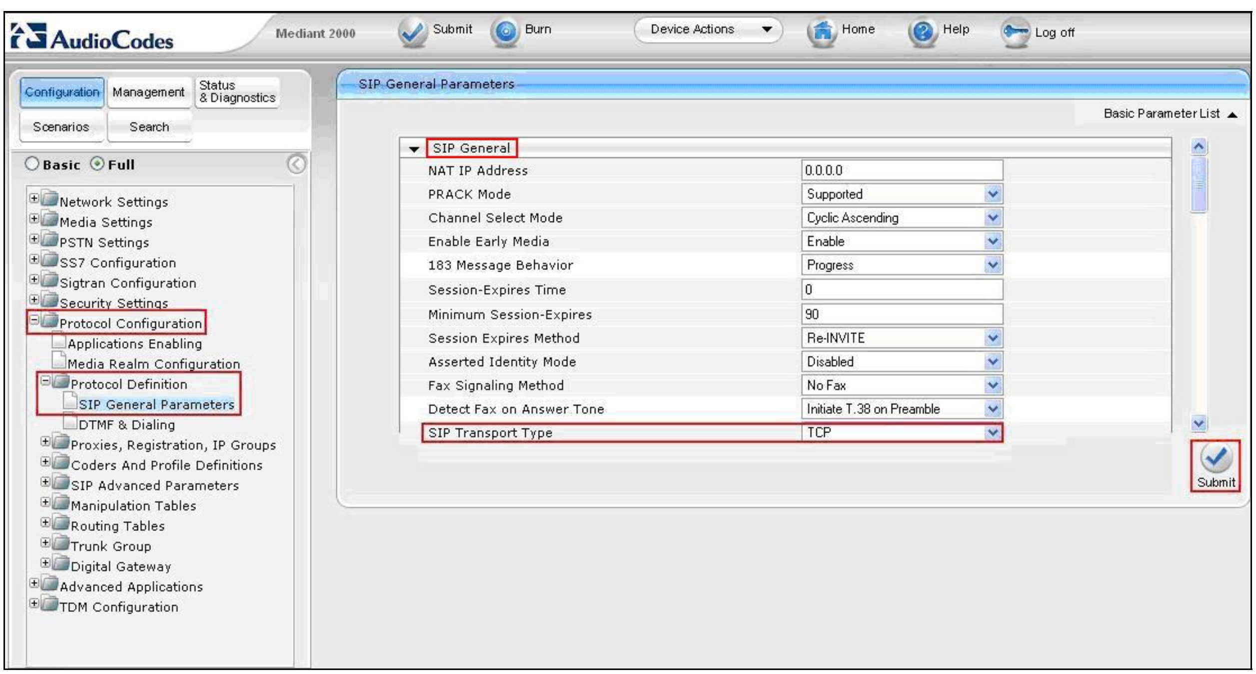


5.4. Administer SIP Protocol Parameters

To configure SIP parameters used when signaling with Conference Standard Edition, select **Protocol Configuration → Protocol Definition → SIP General Parameters** on the left panel menu. Select the following parameters, leaving the remaining parameters at their default values. Under **SIP General:**

- **SIP Transport Type:** Select **TCP**

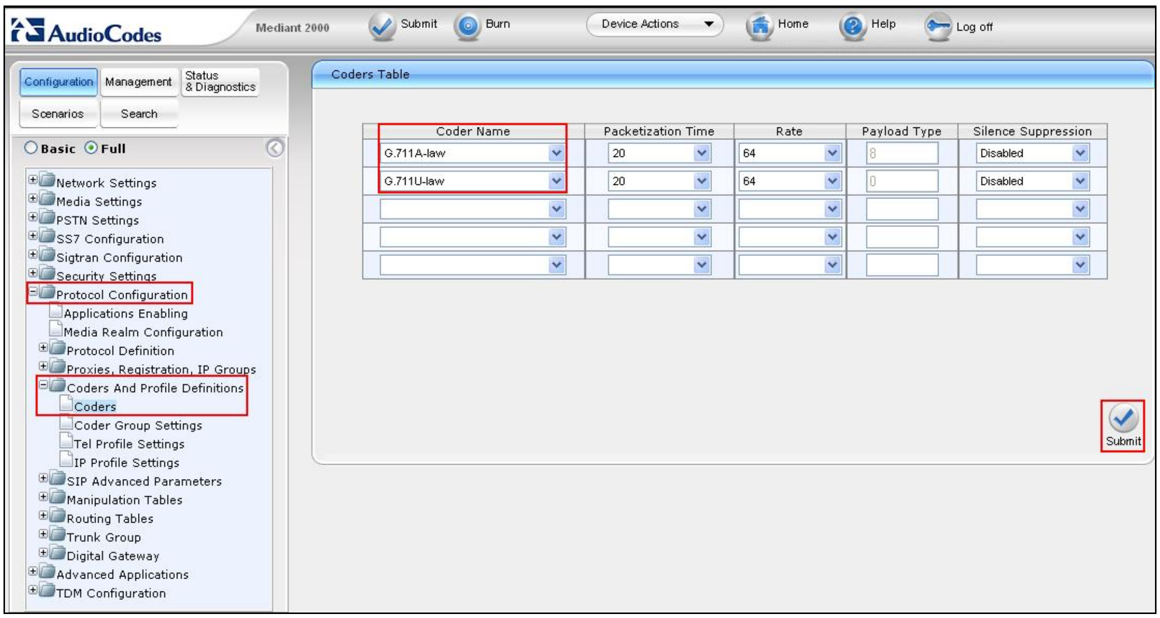
Click on **Submit** to save changes.



5.5. Administer Audio Codecs

Select **Protocol Configuration → Coders And Profile Definitions → Coders** on the left panel menu. Select the following parameters, leaving the remaining parameters at their default values. Configure **Coder Name** that is compatible with Conferencing Standard Edition. Conference Standard Edition only supports **G.711A-law** and **G.711U-law**.

Note: The first coder is the highest priority coder and is used by the Mediant 2000 whenever possible. If the far end SIP User Agent cannot use the coder assigned as the first coder, the gateway attempts to use the next coder and so forth. Click on **Submit** to save changes.

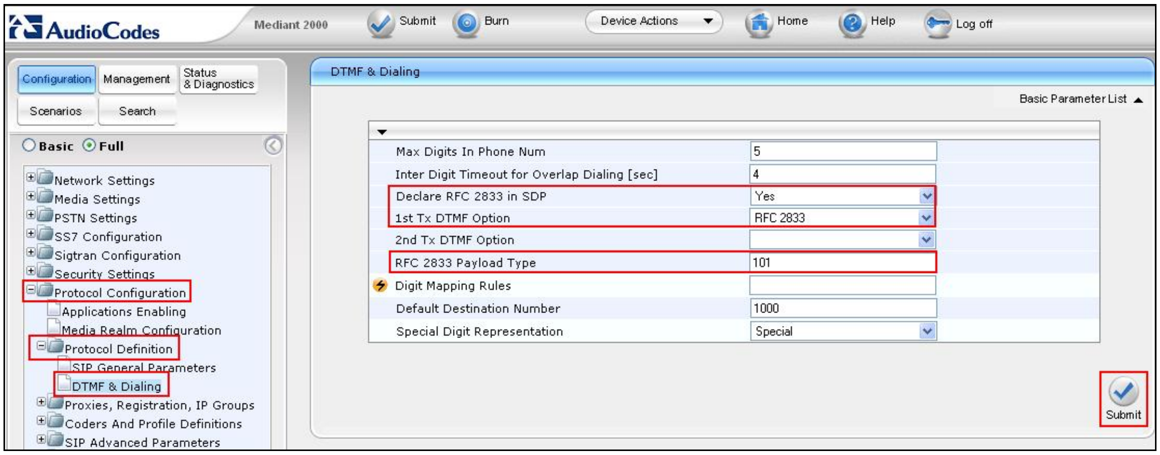


5.6. Administer DTMF Signaling

To configure Out Of Band, select **Protocol Configuration → Protocol Definition →DTMF & Dialing** on the left panel menu. Select the following parameters, leaving the remaining parameters at their default values.

- **Declare RFC 2833 in SDP:** Select **Yes**
- **1st Tx DTMF Option:** Select **RFC 2833**
- **RFC 2833 Payload Type:** Select **101**

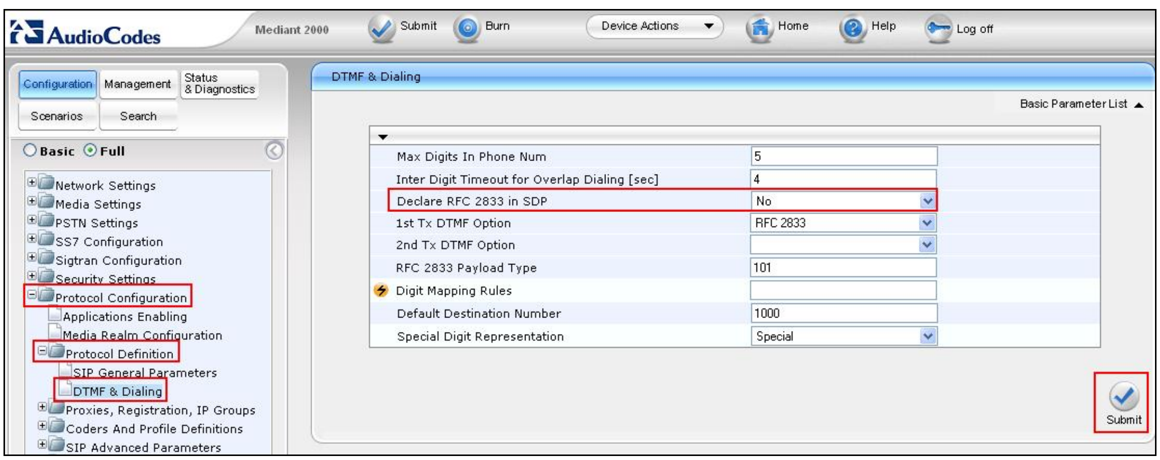
Click on **Submit** to save changes.



To configure In Band, select **Protocol Configuration → Protocol Definition → DTMF & Dialing** on the left panel menu**.** Select the following parameters, leaving the remaining parameters at their default values.

- **Declare RFC 2833 in SDP** Select **No**

Click on **Submit** to save changes.

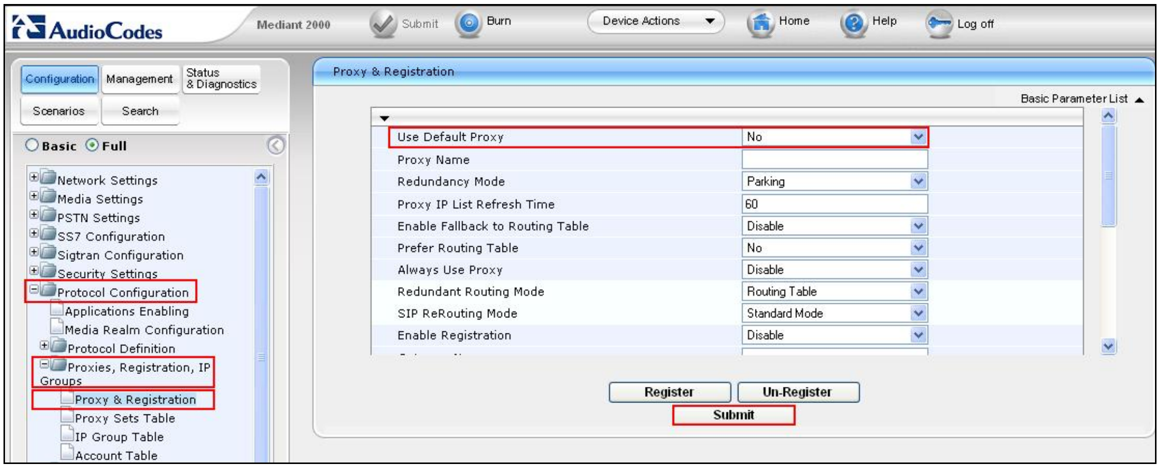


5.7. Administer Proxy & Registration

Select **Protocol Configuration → Proxies, Registration, IP Groups → Proxy & Registration**. Select the following parameters, leaving the remaining parameters at their default values.

- **Use Default Proxy** Select **No**

Click on **Submit** to save changes.

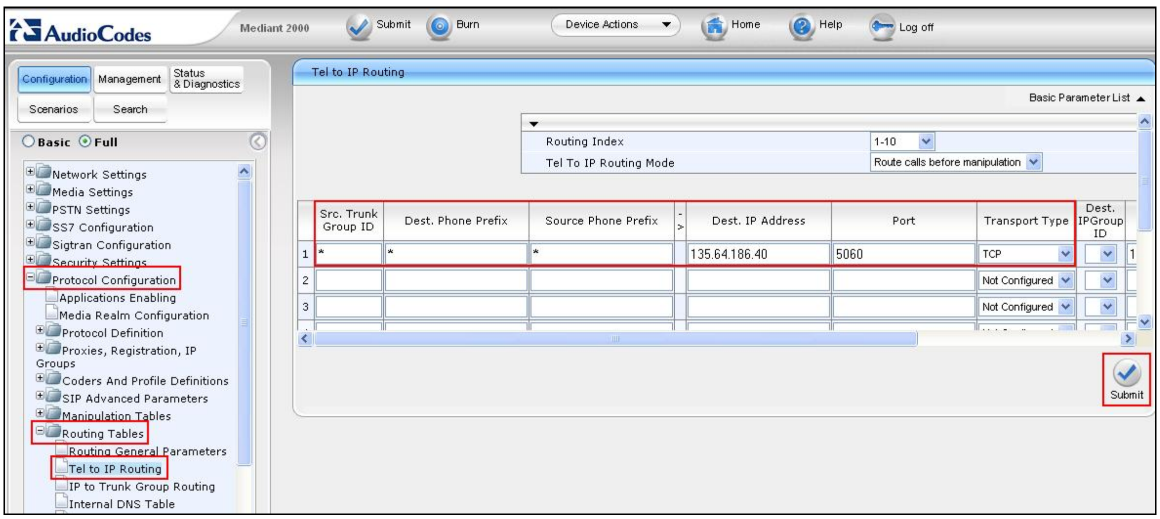


5.8. Administer Routing Tables

To configure the tables used for routing calls between the E1 and SIP interfaces, select **Protocol Configuration → Routing Tables → Tel to IP Routing** on the left panel menu. Since use of a SIP proxy was disabled in **Section 5.7**, the **Tel to IP Routing** needs to be configured. All calls from the PSTN are routed to the Software Asset Card of the Session Manager based on the dialed number. Select the following parameters, leaving the remaining parameters at their default values.

- **Src. Trunk Group ID** Select *****, wild card entry
- **Dest. Phone Prefix** Select *****, wild card entry
- **Source Phone Prefix** Select *****, wild card entry
- **Dest. IP Address** **135.64.186.40**, IP Address of Software Asset Card
- **Port** Select **5060**
- **Transport Type** Select **TCP**

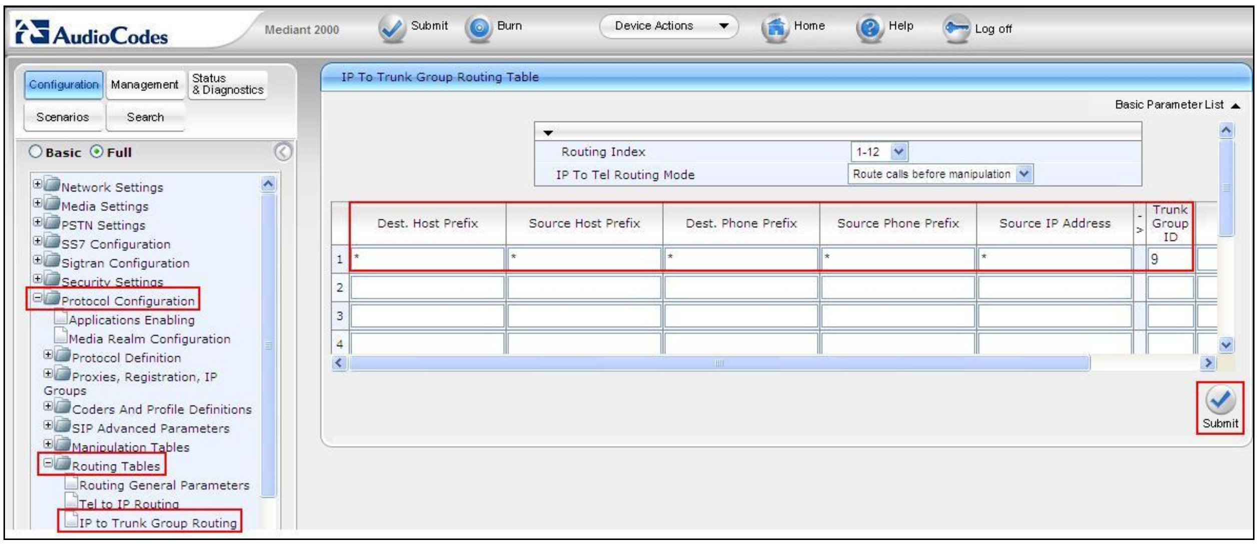
Click on **Submit** to save changes.



To configure routing from SIP to E1, select **Protocol Configuration → Routing Tables → IP to Trunk Group Routing** on the left panel menu. Select the following parameters, leaving the remaining parameters at their default values. These values specify that all SIP calls are to be routed to the E1 PRI interface.

- **Dest. Host Prefix:** Select *****, wild card entry *****
- **Source Host Prefix:** Select *****, wild card entry *****
- **Dest. Phone Prefix** Select *****, wild card entry *****
- **Source Phone Prefix:** Select *****, wild card entry *****
- **Source IP Address:** Select *****, wild card entry *****
- **Trunk Group ID** Select **9**, defined in **Section 5.3**

Click on **Submit** to save changes.

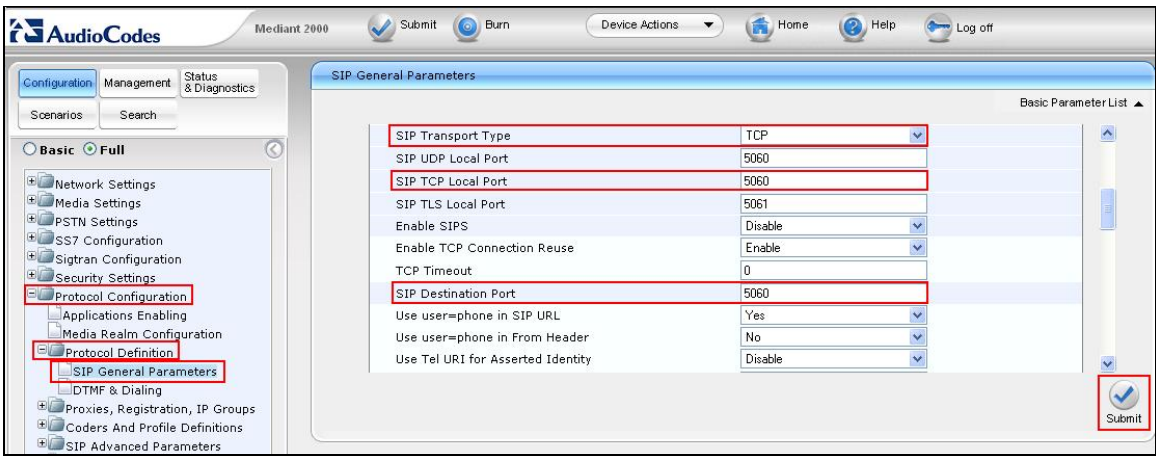


5.9. Administer SIP General Parameters for TCP

Select **Protocol Configuration → Protocol Definition → SIP General Parameters** on the left panel menu. Select the following parameters, leaving the remaining parameters at their default values.

- **SIP Transport Type** Select **TCP**
- **SIP TCP Local Port** Select **5060**
- **SIP Destination Port** Select **5060**

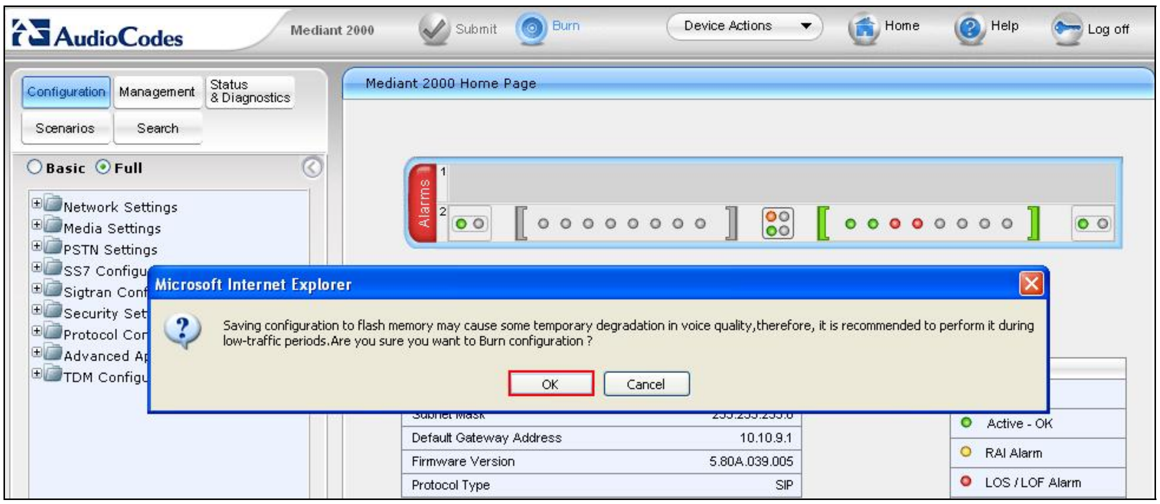
Click on **Submit** to save changes.



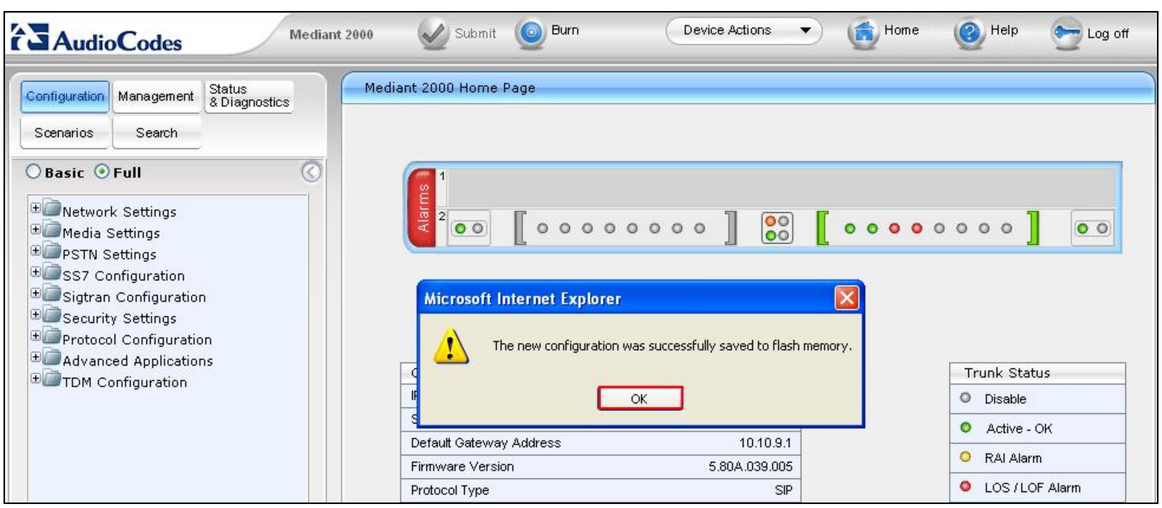
5.10. Save the Configuration

Click on **Burn** on the Mediant 2000 Toolbar.

Click **OK** to confirm the message below.



Click **OK** to confirm the message below.



6. Verification Steps

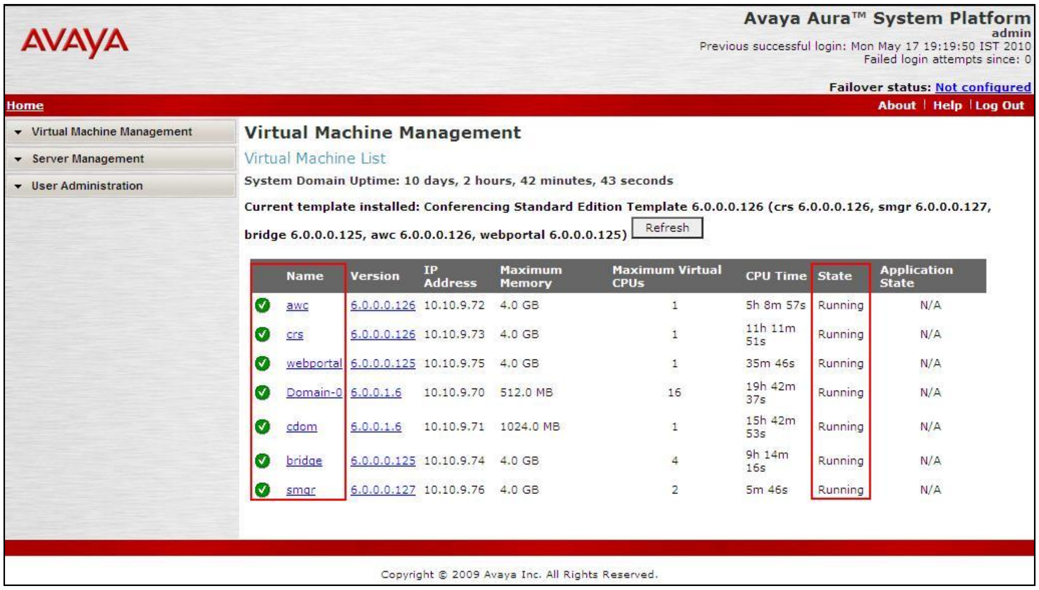
The following steps were used to verify the administrative steps presented in these Application Notes and are applicable for similar configurations in the field. The verification steps in this section validated the following:

- The Conferencing Standard Edition configuration
- Session Manager

6.1. Avaya Aura™ Conferencing Standard Edition

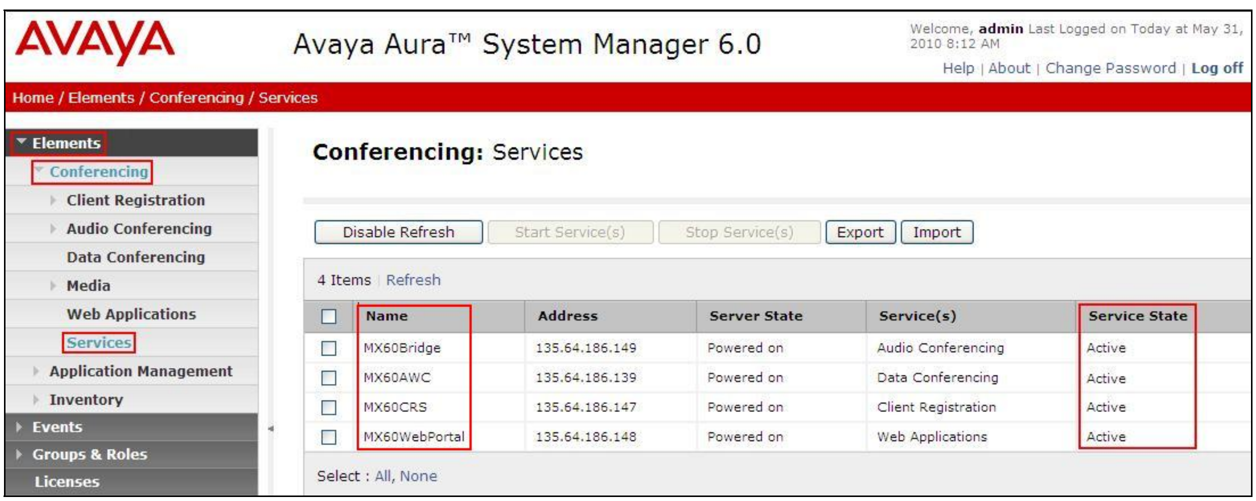
Verify all Virtual Machines are in a running state. Access the System Platform using a Web Browser and entering ***https://<ip-address>/webconsole***, where <ip-address> is the IP address of System Platform. Log in using appropriate credentials.

Verify all Virtual Machines are in a **Running State**.



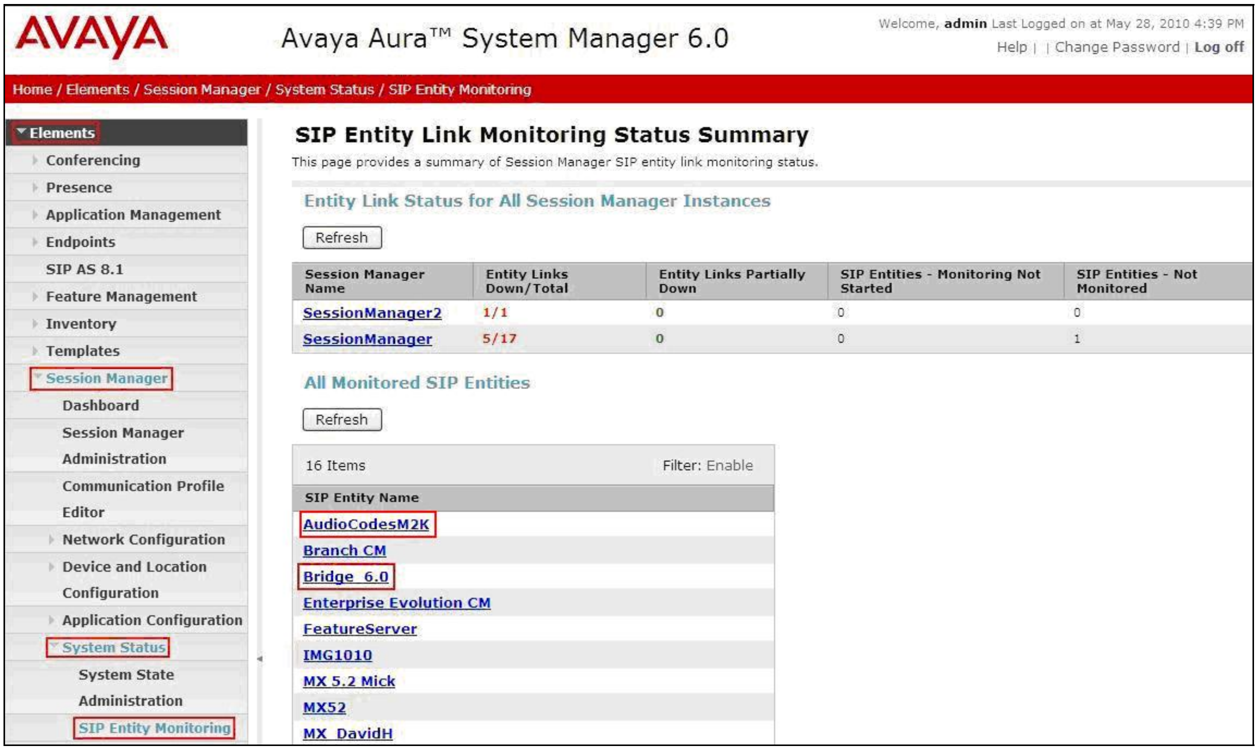
6.1.1. Conferencing Standard Edition Services

Using System Manager as shown below, check the **Service State** between the Conferencing bridge and other devices by configuring the SIP System Settings by selecting **Elements → Conferencing → Services** on the left panel menu. From the right panel menu ensure the **Conferencing Services** are in an **Active** Service State.

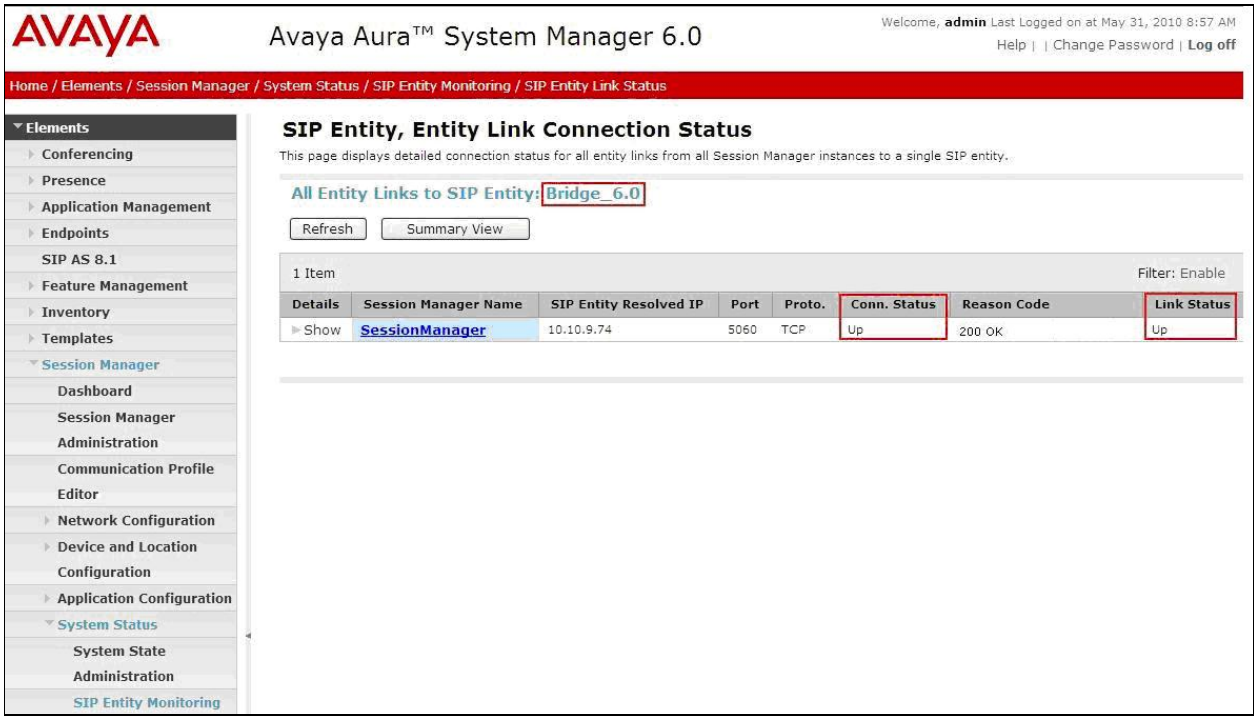


6.2. SIP Monitoring on Avaya Aura™ Session Manager

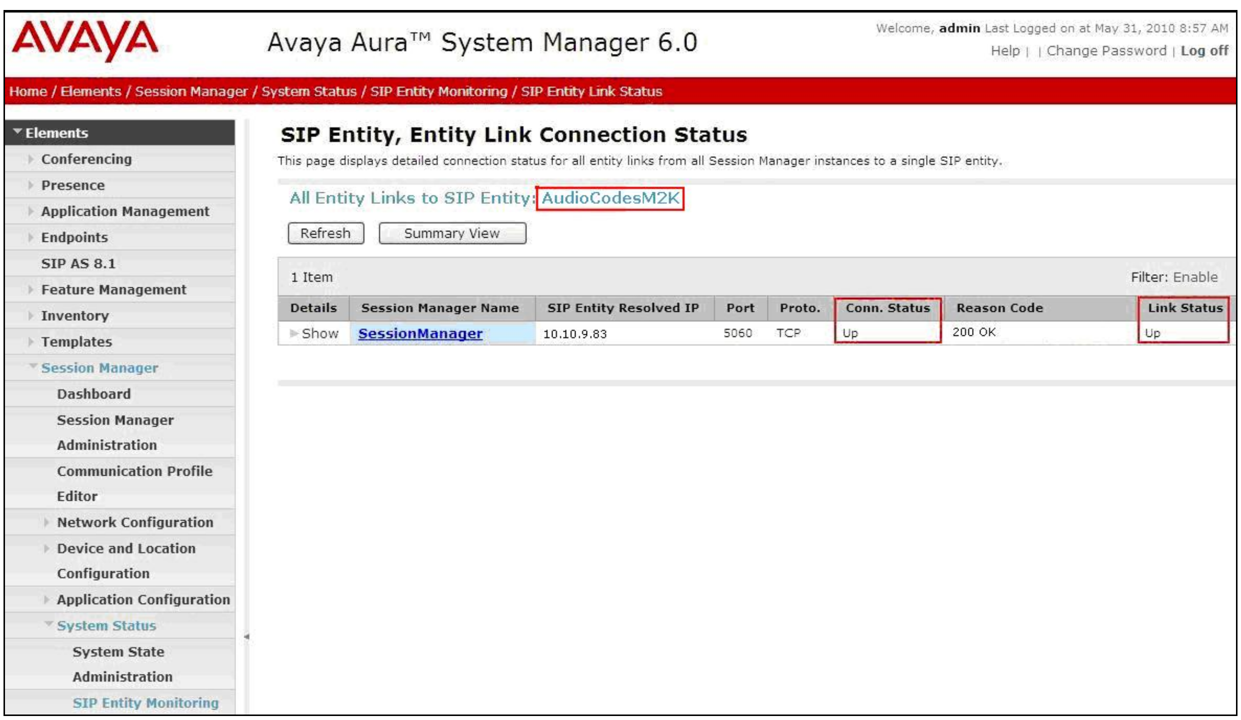
Verify that none of the links to the defined SIP entities are down, indicating that they are all reachable for call routing by selecting **Elements → Session Manager → System Status → SIP Entity Monitoring** on the left panel menu. From the right panel menu select the SIP elements created in **Section 4.4**



Click on the SIP Entity Name **Bridge 6.0**, shown in the previous screen, and verify that the **Conn. Status** and **Link Status** are **Up**.



Click on the SIP Entity Name **AudioCodesM2K**, and verify that the **Conn. Status** and **Link Status** are **Up**.



6.3. Verification Scenarios

Verify end to end signalling/media connectivity between the Mediant 2000 and Conferencing Standard Edition via the Session Manager, this is accomplished by:

- Placing a call from two endpoints into conference ensuring one of the callers is a moderator.
- Verify both callers are in the same conference and there is two way speech between the callers.
- Initiate dial out by dialing *1 xxxx on the moderator phones touch pad, where xxxx is the extension for an endpoint. Follow the instructions provided by the Conferencing bridge.
- After answering the call, on the moderator phone dial *2 to join the new participant into the conference.
- Verify both callers are in the same conference and there is two way speech between the callers.

7. Conclusion

As illustrated in these Application Notes, Avaya Aura™ Conferencing Standard Edition can interoperate successfully with Avaya Aura™ Session Manager and AudioCodes Mediant 2000 Gateway.

8. Additional References

This section references the product documentation relevant to these Application Notes.

Avaya Aura™ Conferencing Standard Edition 6.0

[1] *Implementing Avaya Aura™ Conferencing Standard Edition, Doc ID 04-603508, June 2010, available at http://support.avaya.com.*

[2] *Operating Avaya Aura™ Conferencing Standard Edition, Doc ID 04-603510, June 2010, available at http://support.avaya.com.*

[3] *Using Avaya Aura™ Conferencing Standard Edition, Doc ID 04-603509, June 2010, available at http://support.avaya.com.*

[4] *Avaya Aura™ Conferencing Standard Edition Release Notes, Doc ID 04-603528, June 2010, available at http://support.avaya.com*

Avaya Aura™ Session Manager 6.0

[5] *Avaya Aura™ Session Manager Overview, Doc ID 03-603323, available at http://support.avaya.com.*

[6] *Administering Avaya Aura™ Session Manager, Doc ID 03-603324 available at http://support.avaya.com.*

[7] *Installing and Upgrading Avaya Aura™ Session Manager 6.0, Doc ID 03-603324, available at http://support.avaya.com.*

[8] *Installing and Upgrading Avaya Aura™ System Manager 6.0, available at http://support.avaya.com.*

[9] *Maintaining and Troubleshooting Avaya Aura™ Session Manager 6.0, Doc ID 03-603321, available at http://support.avaya.com.*

AudioCodes Mediant 2000 Gateway

[10] Technical support and System Deployment Guides are available at *http://audiocodes.com*

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com