Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunks between the Avaya Aura™ Conferencing Standard Edition 6.0 and Cisco Unified Communications Manager 7.0 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Avaya Aura™ Conferencing Standard Edition and Cisco Unified Communications Manager via direct SIP trunks.

1. Introduction

As shown in **Figure 1**, Avaya Aura™ Conferencing Standard Edition Server is a fully integrated audio and data conferencing solution. The Server is responsible for SIP signaling and multiplexing and streaming RTP to the conference participants. 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. It can provide both audio and web conferencing to Cisco Unified Communications Manager users. These Application Notes only describe configuration steps for audio conferencing. A SIP trunk is used to connect Avaya Aura™ Conferencing Standard Edition with Cisco Unified Communications Manager over the LAN. 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 **[3]**. 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]**.

For the sample configuration, the telephones are configured in the 500x extension range, while the conference access number (DNIS) on the Avaya Aura™ Conferencing Standard Edition is set to 7111. Cisco Unified Communications Manager runs on Cisco 2811 router, while Avaya Aura™ Conferencing Standard Edition runs on S8800 server. Avaya Aura™ Conferencing Standard Edition is managed by either Avaya Aura™ Conferencing Manager or Avaya Aura™ System Manager, if one already exists.

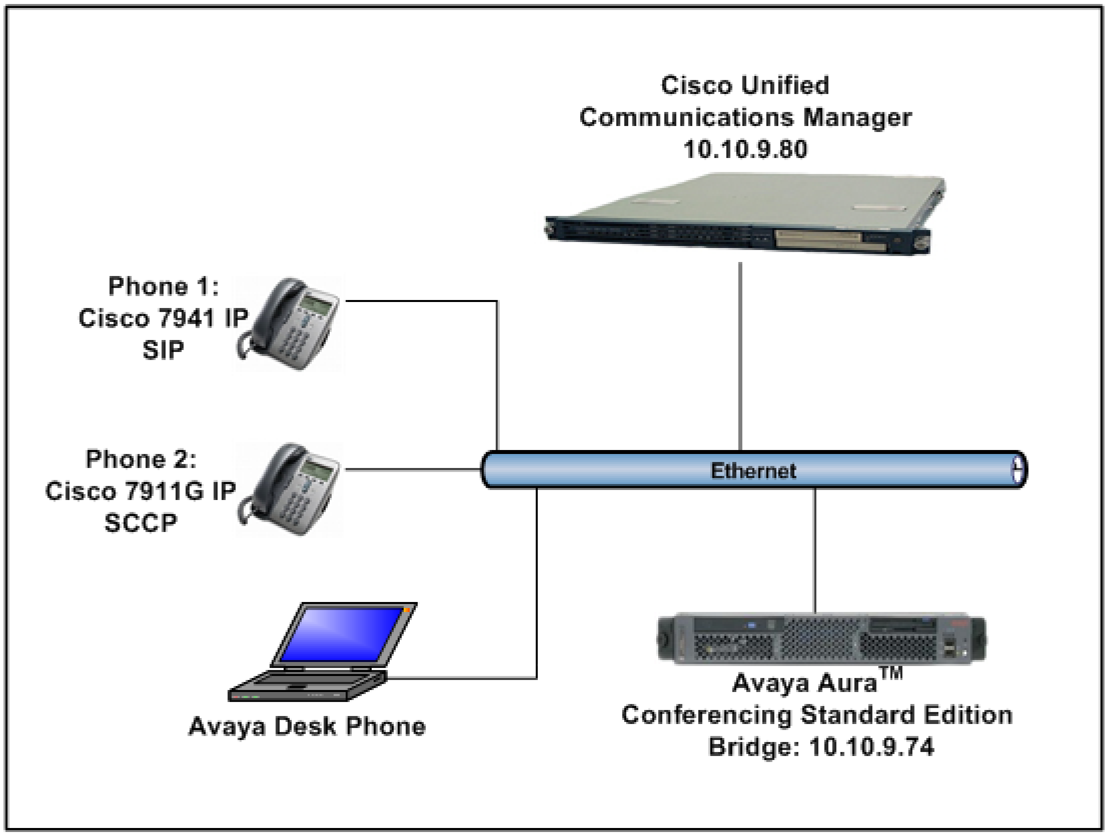

Cisco Unified
Communications Manager
10.10.9.80
Phone 1:
Cisco 7941 IP
SIP
Phone 2:
Cisco 7911G IP
SCCP
Ethernet
Avaya Desk Phone
Avaya AuraTM
Conferencing Standard Edition
Bridge: 10.10.9.74

Figure 1

2. Equipment and Software Validated

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

Equipment	Software
Avaya Aura^{TM} Standard Conferencing Server (S8800)	Avaya Aura^{TM} Standard Conferencing Server 6.0.0.0.262 + Release Patches
Cisco Unified Communications Manager	7.0.2.100000-18
Cisco 7911G SIP Telephone	SIP 11.8-4-3S
Cisco 7911G SCCP Telephone	SCCP 11.8-3-4SR1S

Table 1: Hardware and Software Versions

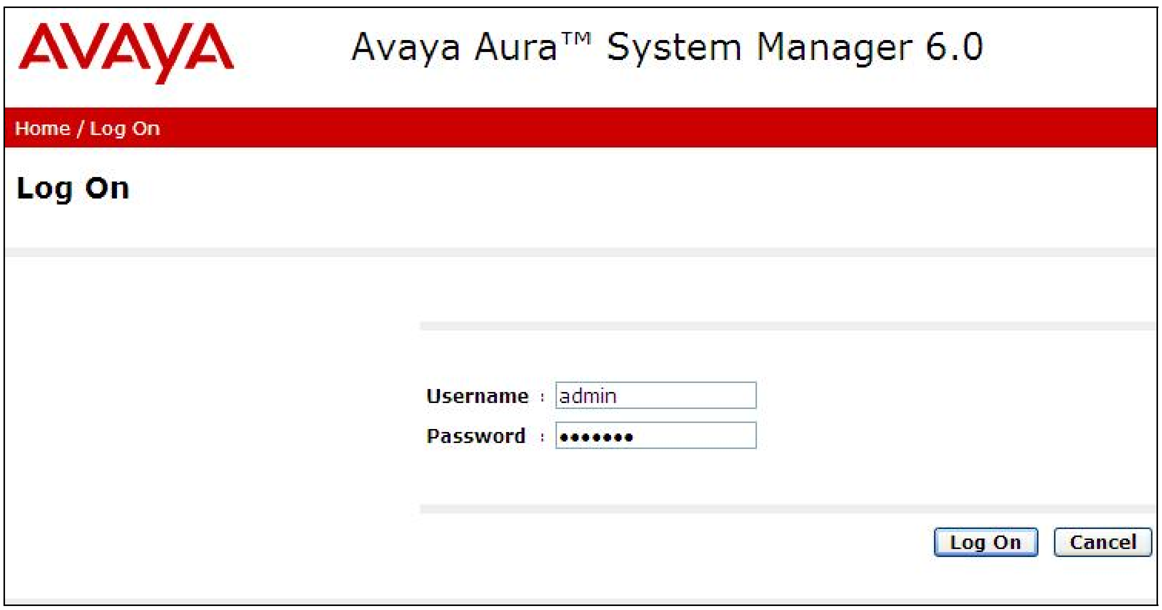
3. Configure Avaya Aura™ Conferencing Standard Edition

This section describes the procedure for configuring the Conferencing Standard Edition to interoperate with Cisco Unified Communications Manager via direct SIP trunks. The procedures include the following areas:

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

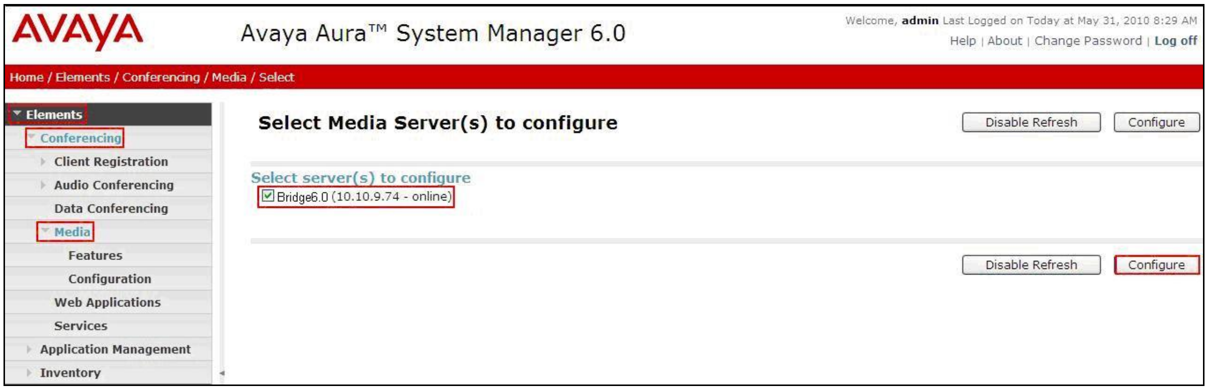
3.1. Log in to Avaya Aura™ System Manager

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.

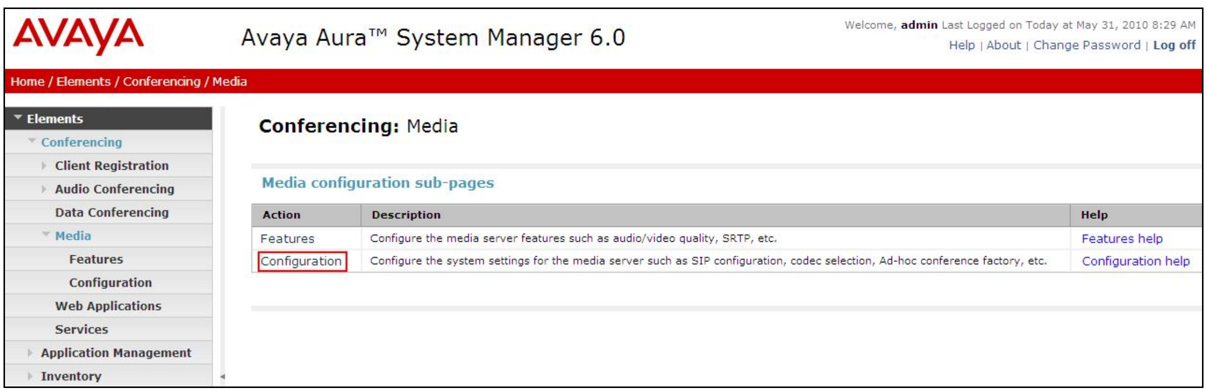


3.2. Configuring SIP Connectivity

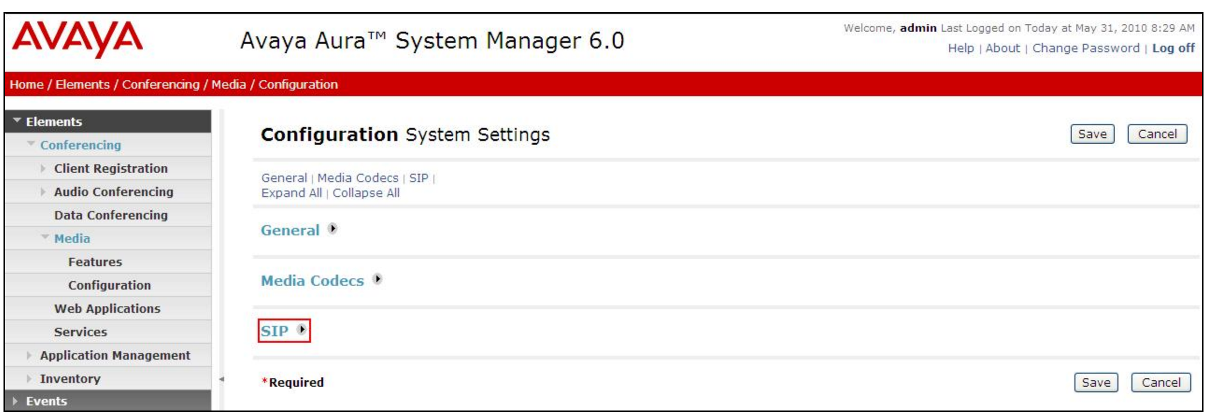
Configure settings that enable SIP connectivity between the conferencing bridge and other devices. Select **Elements → Conferencing → Media** on the left panel menu. From the right panel menu, select the media server to configure by selecting the tick box and select **Configure**.



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



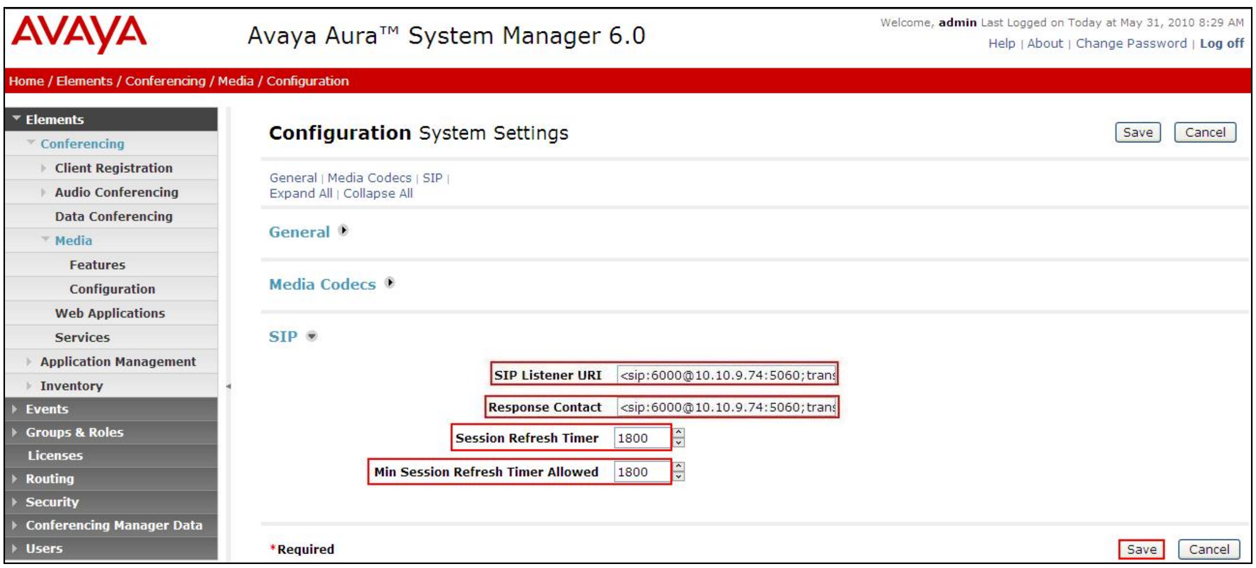
From the right panel menu, select **SIP**.



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**

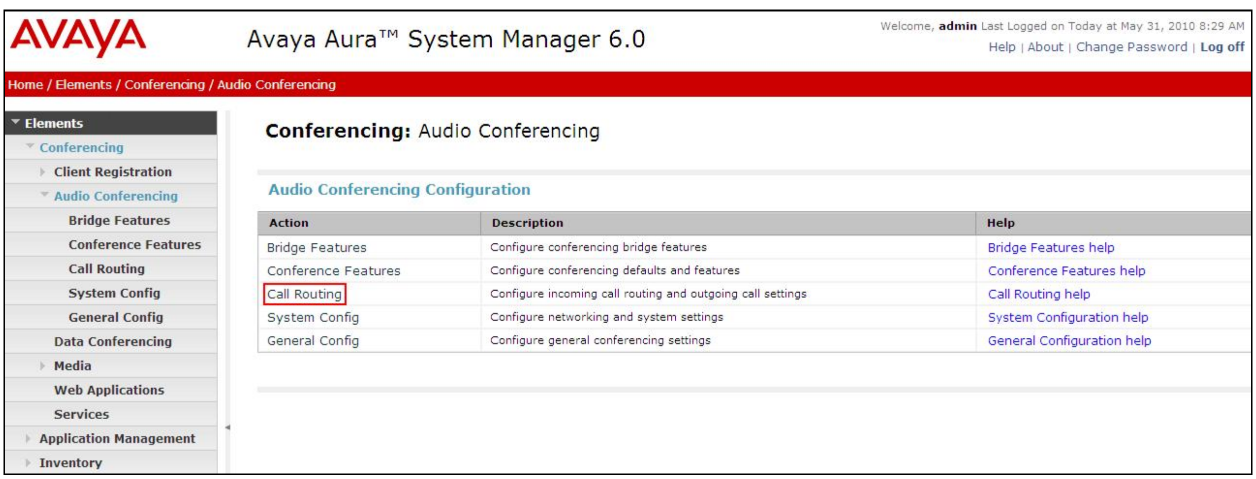
Click the **Save** button.



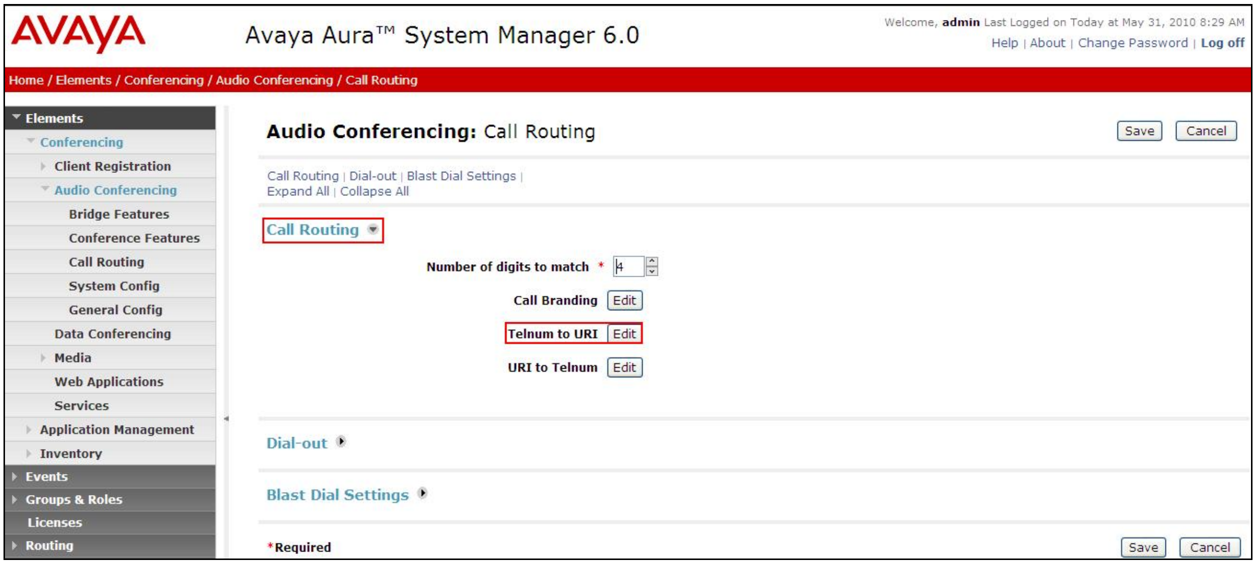
3.3. Configure Dialout

To enable Dial-Out from the Conferencing to the Cisco Unified Communications Manager, configure the **telnumToUri**, which is used to map the number dialed to a corresponding URI. Select **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**.

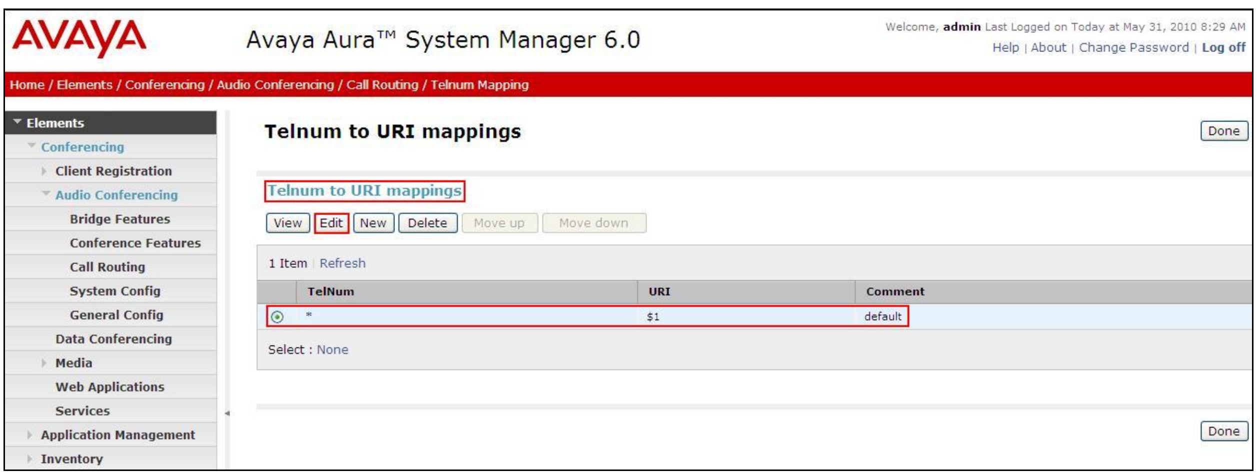
From the right panel menu, select **Call Routing**.



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



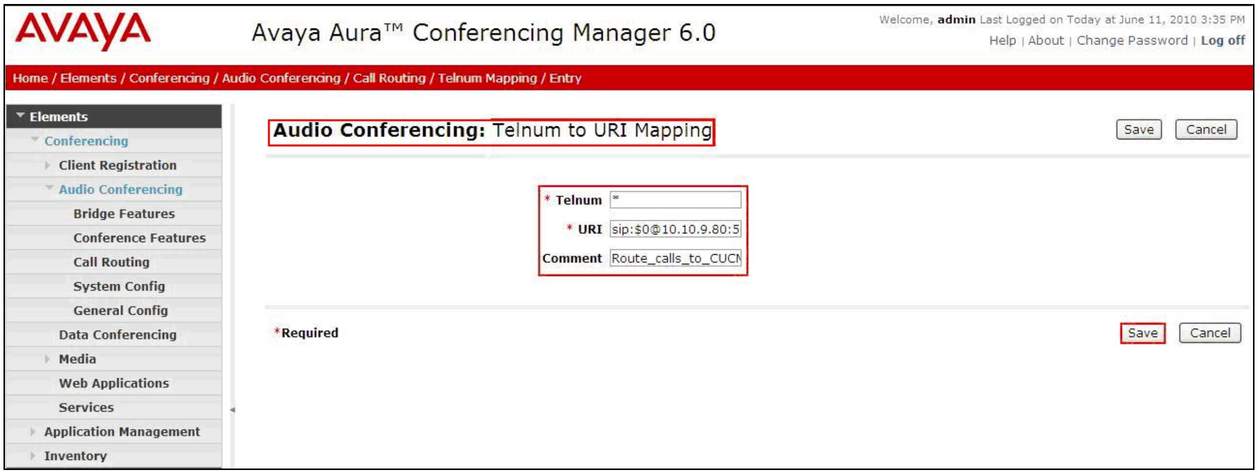
From the right panel menu select the default **Telnum to URI mappings** and select **Edit**.



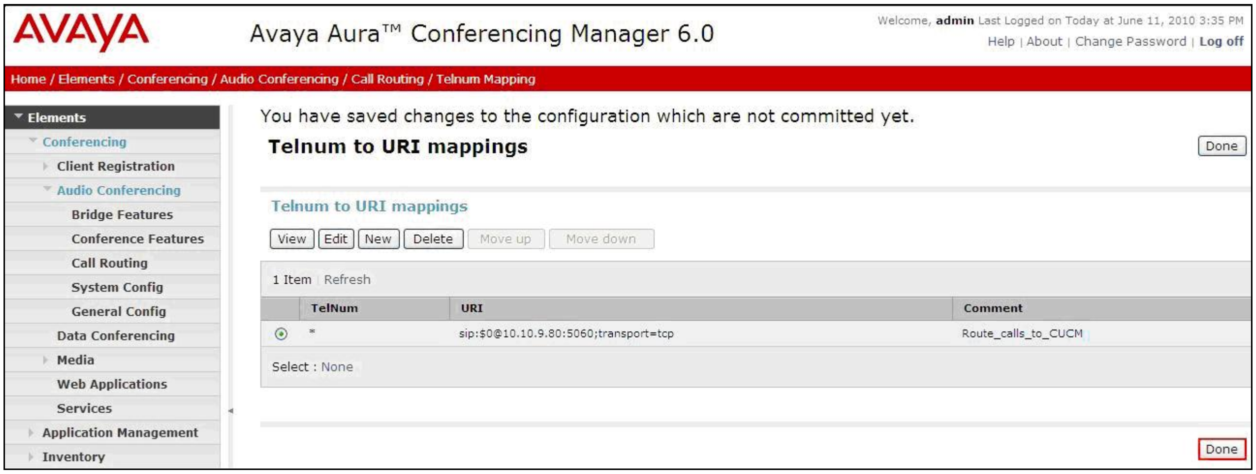
From the right panel menu complete the following options; under **Audio Conferencing: Telnum to URI Mapping**

- **Telnum** *
- **URI** **sip:$0@10.10.9.80:5060;transport=tcp**
  To route outbound calls from the Conferencing Bridge to the CUCM.
- **Comment** A descriptive comment

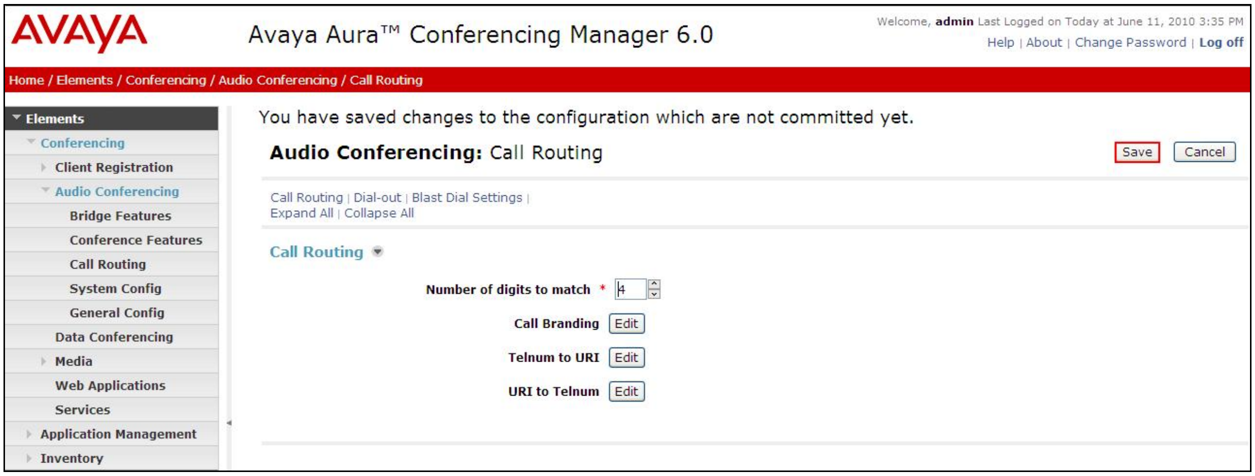
Click the **Save** button.



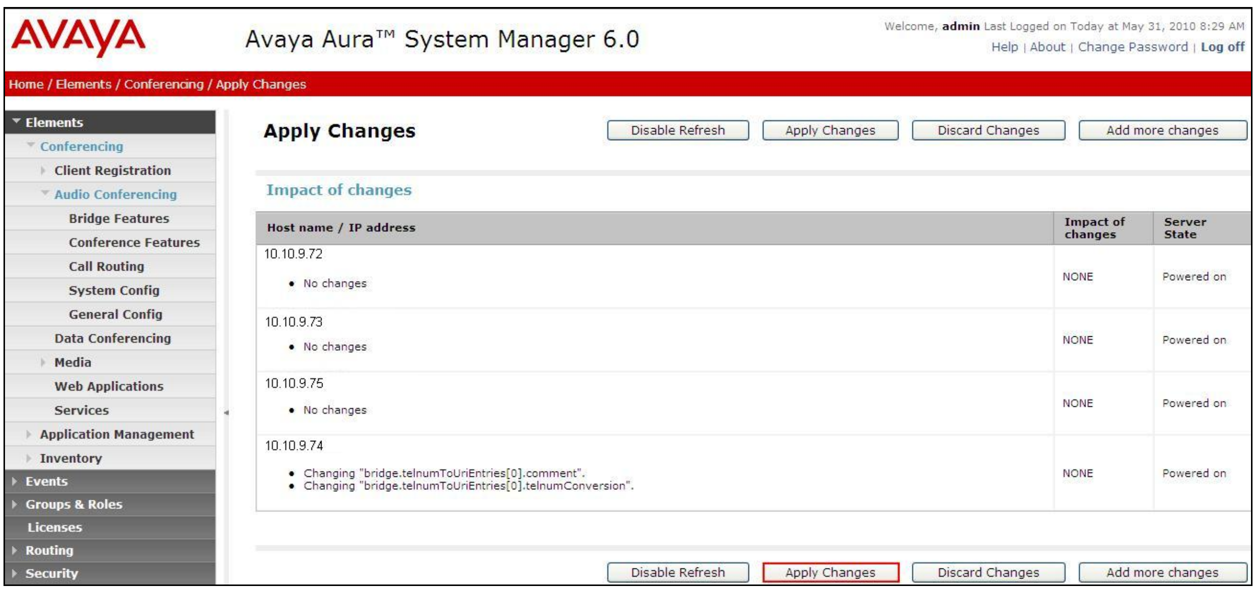
From the right panel menu select **Done**.



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



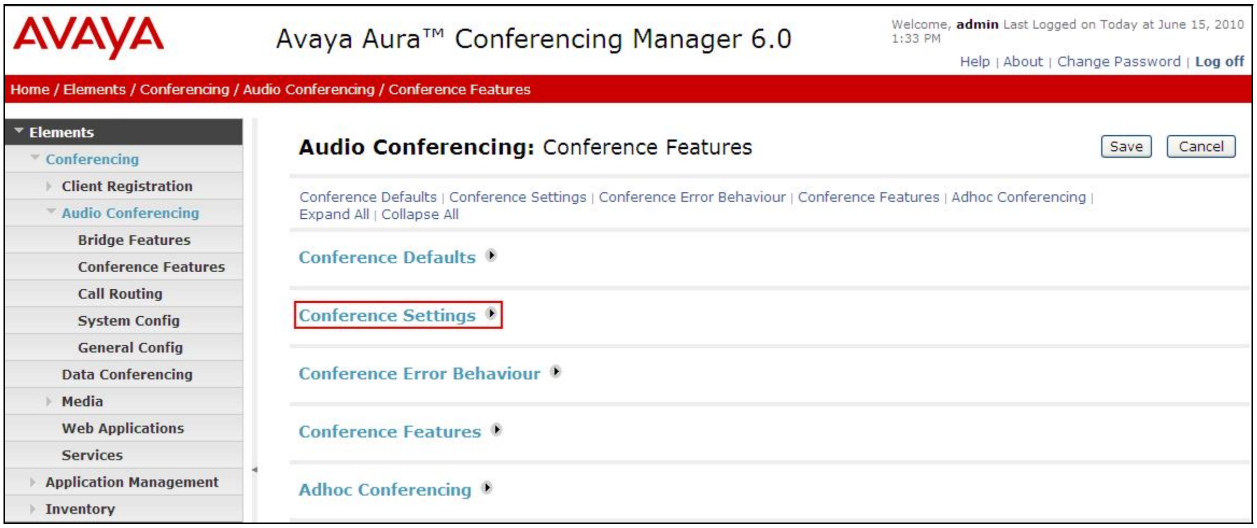
From the right panel menu select **Apply Changes**.



To enable Dial-Out from the Conferencing Bridge to the Cisco Unified Communications 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**.

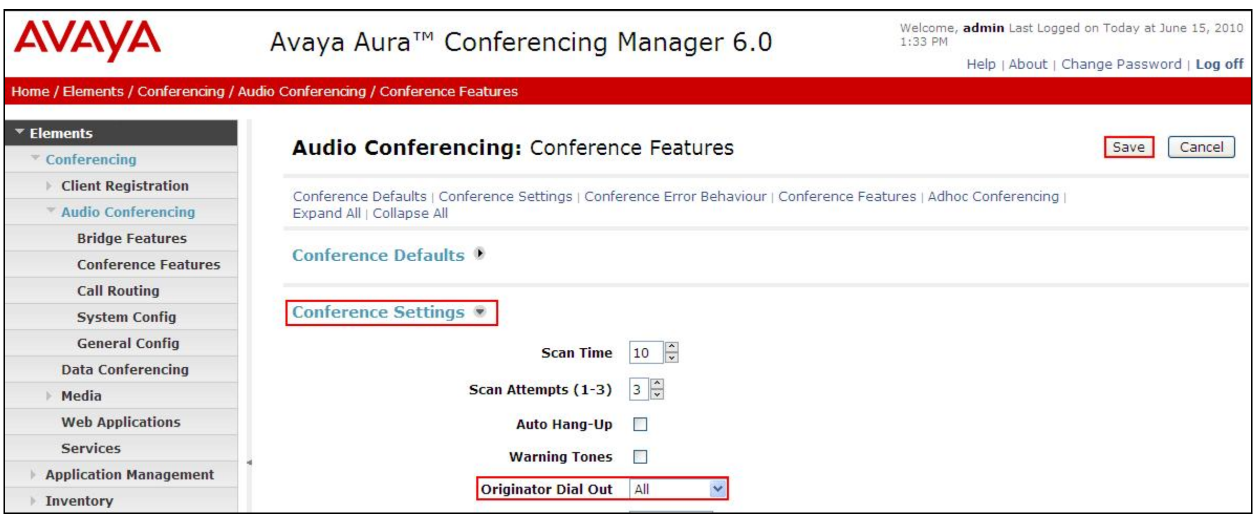
From the right panel menu, select **Conference Settings**.



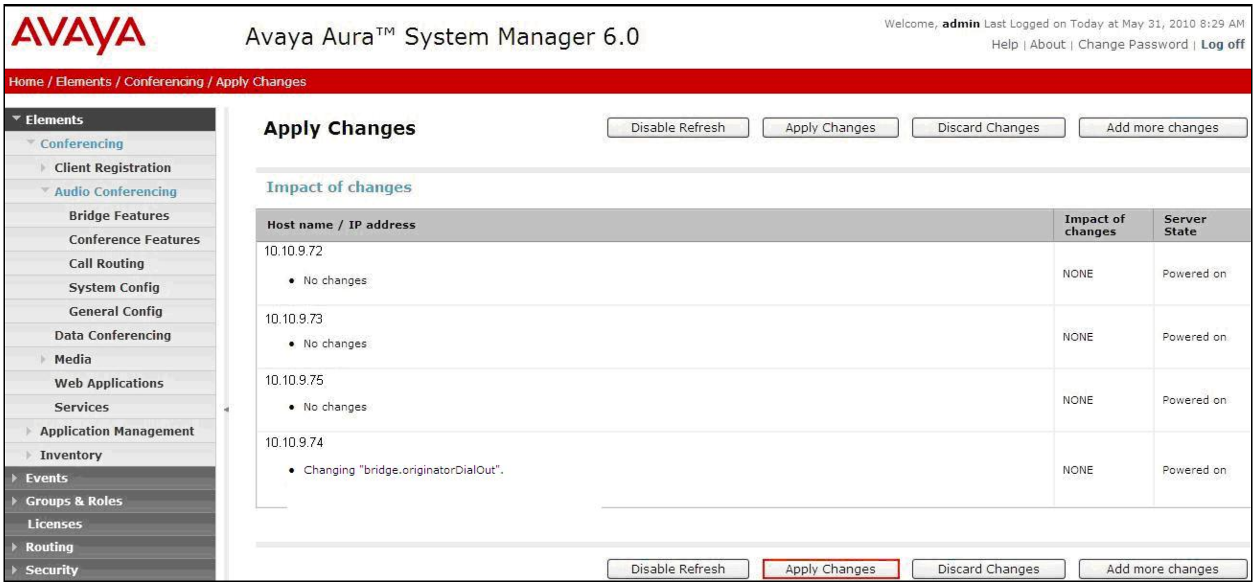
From the **Conference Settings** menu on the right panel, select the following parameter and leave the remaining parameters at their default values.

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

Click the **Save** button.



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

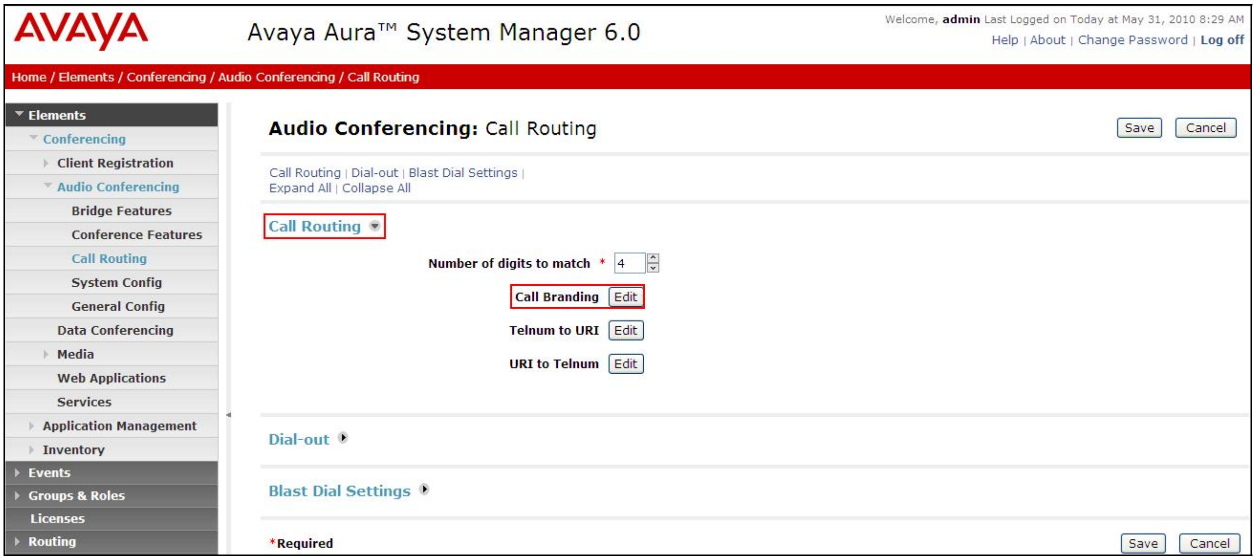


3.4. Map DNIS Entries

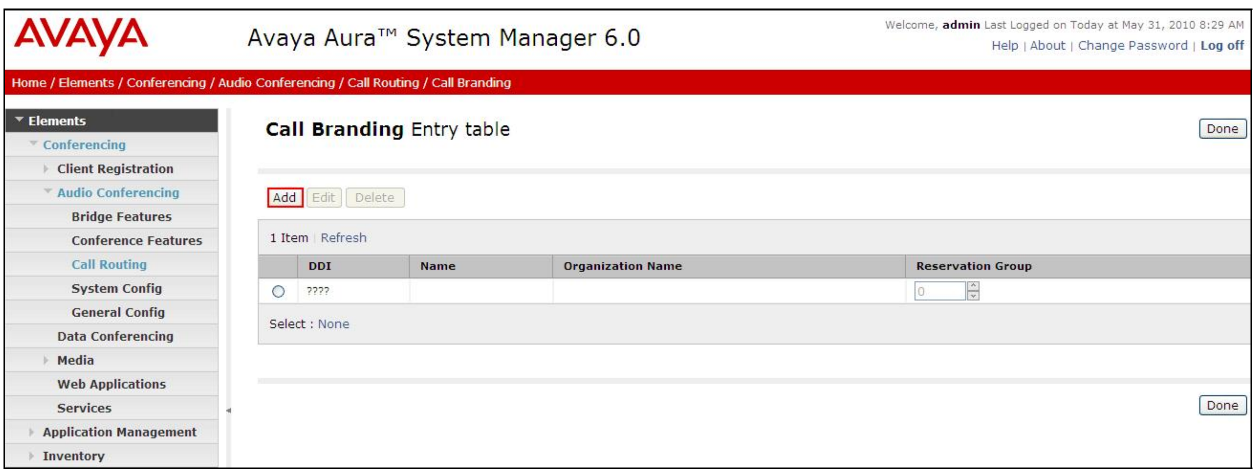
To map DNIS entries, run the Call Branding utility by selecting **Elements → Conferencing → Audio Conferencing** on the left panel menu. From the right panel menu select the conferencing server to configure by selecting the tick box and select **Configure**.

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

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



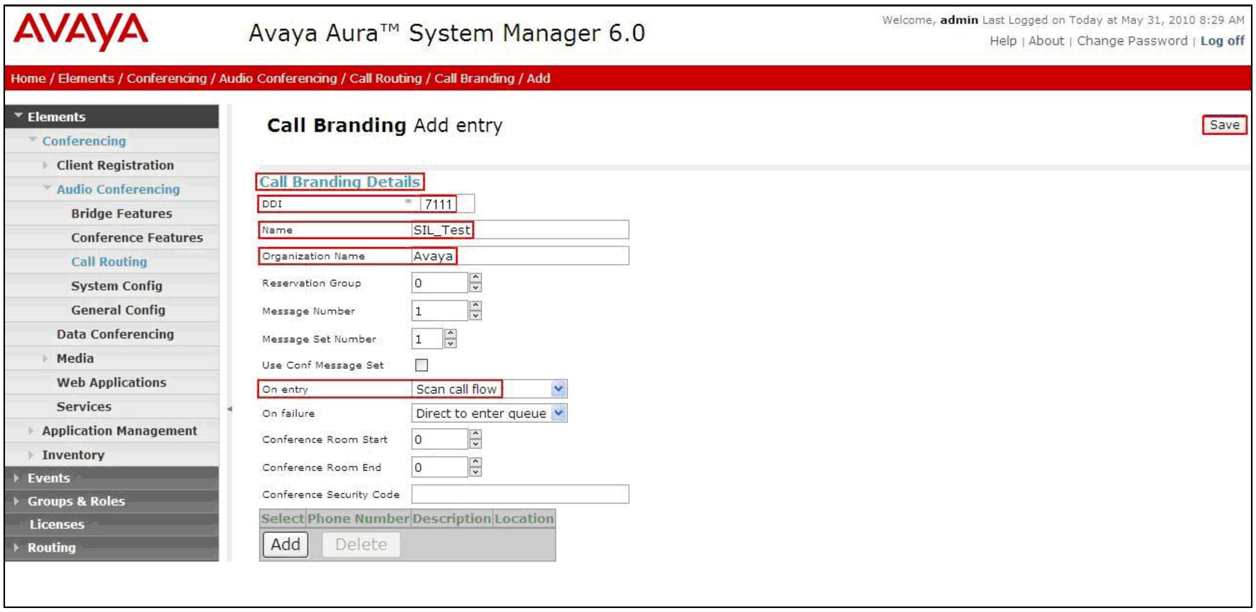
From the right panel menu select the **Add** button to create a new call branding entry.



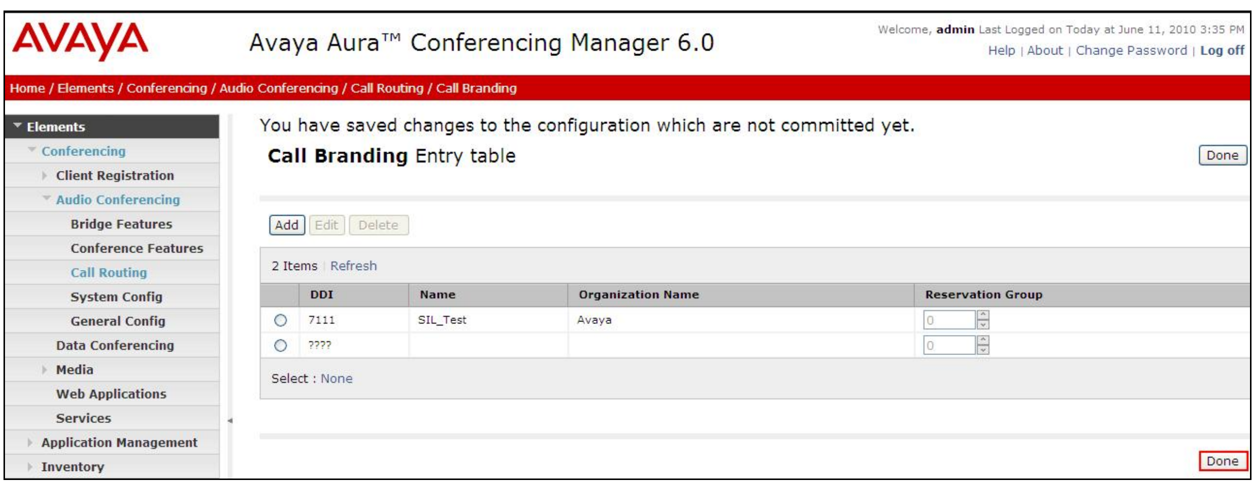
In this sample configuration for **Call Branding Details** complete the following options and use defaults for the remaining fields:
Under **Call Branding Details**

- **DDI** **7111**, a 4 digit number used to dial into conference.
- **Name** A descriptive name
- **Organisation Name** A descriptive name
- **On Entry** Select **Scan call flow** from the drop down menu.

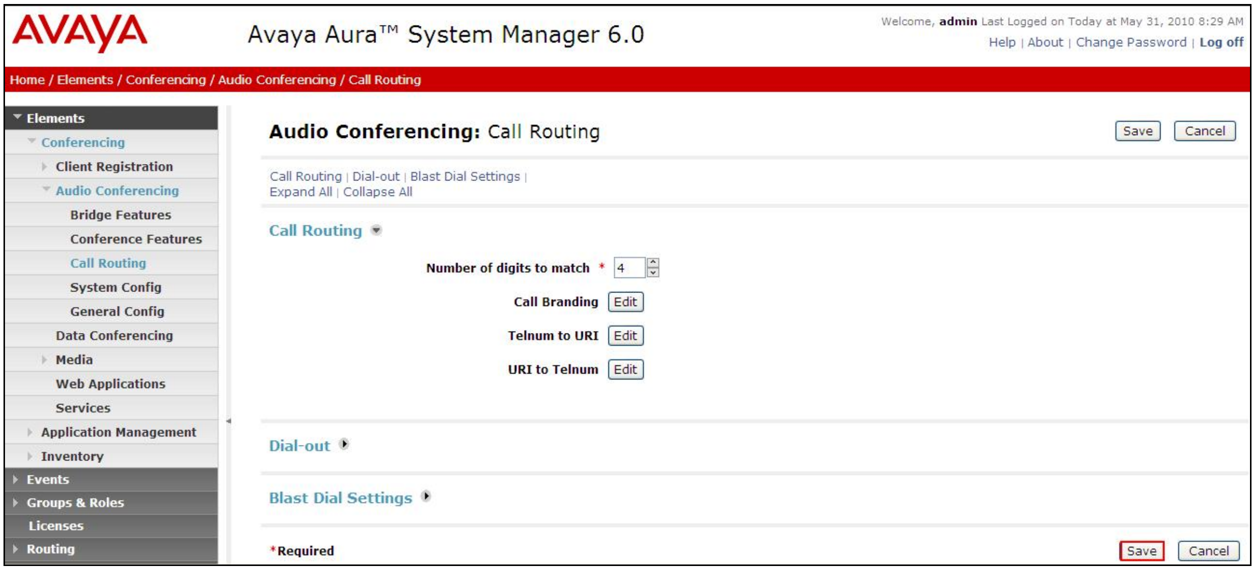
Click the **Save** button.



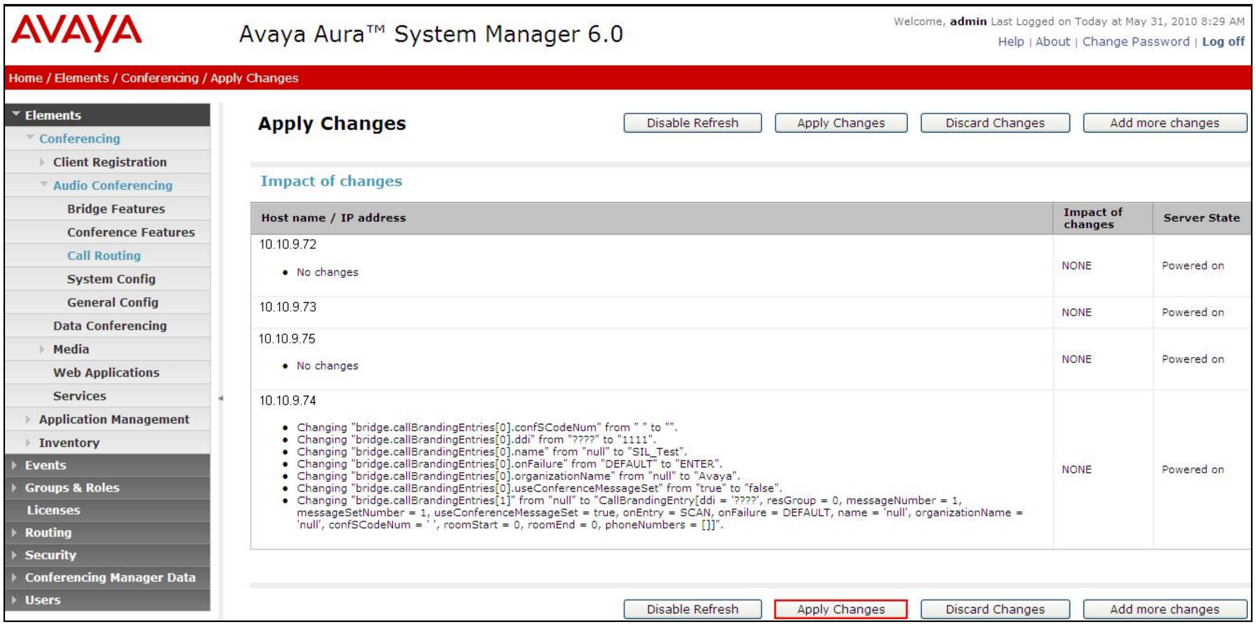
From the right panel menu select **Done**.



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



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



4.0. Configure Cisco Unified Communications Manager

This section provides the procedures for configuring Cisco Unified Communications Manager. These Application Notes assume that the basic configuration needed to support Cisco IP telephones has been completed. For further information on Cisco Unified Communications Manager, please consult **references [4]**, **[5]** and **[6]**. The procedures include configuration of the following items:

- Log in to Cisco Unified Communications Manager
- Administer SIP Trunk Security Profile
- Administer SIP Trunk
- Administer Route Pattern
- Administer Route Group
- Administer Phone

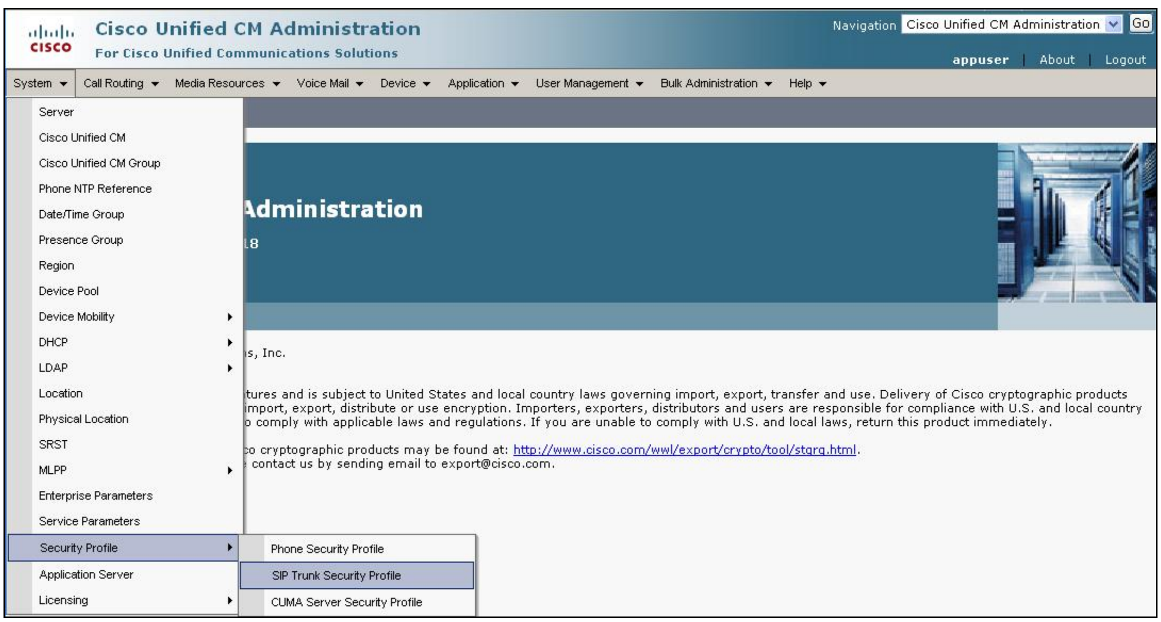
4.1. Log in to Cisco Unified Communications Manager

Open the Cisco Unified Communications Manager Administration web interface by using the URL "http://<ip-address>" in an Internet browser window, where "<ip-address>" is the IP address of the Cisco Unified Communications Manager. Click on **Cisco Unified Communications Manager Administration** at the bottom of the screen.

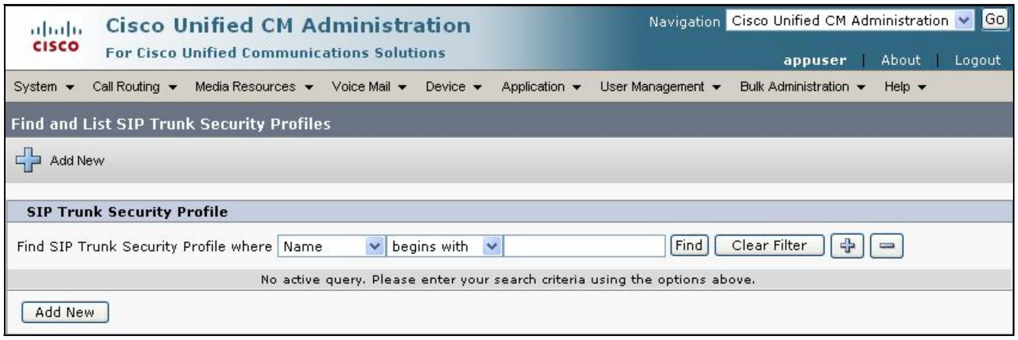
The **Cisco Unified CM Administration** screen is displayed. Select **Cisco Unified CM Administration** from the **Navigation** drop-down list, and log in with appropriate credentials.

4.2. Administer SIP Trunk Security Profile

Scroll to the top of the screen, and select **System → Security Profile → SIP Trunk Security Profile** as shown below.



The **SIP Trunk Security Profile** screen is displayed. Click **Add New** to add a new SIP Trunk Security Profile.

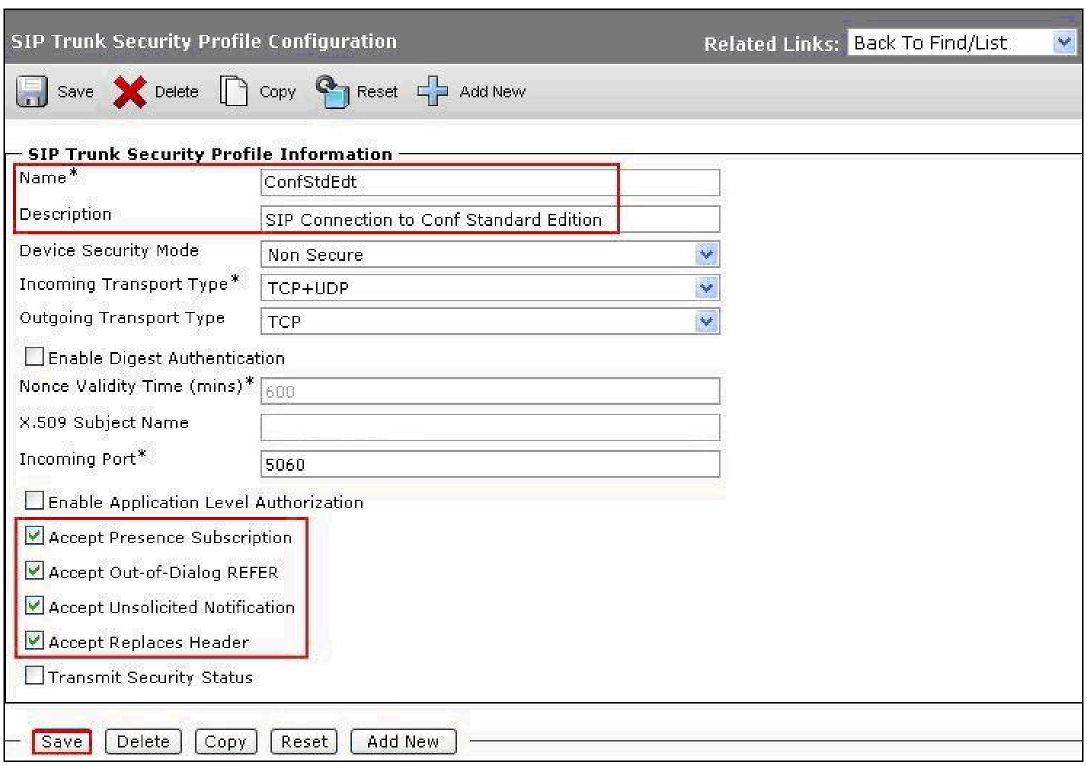


The **SIP Trunk Security Profile Information** configuration screen is displayed which was used in the sample network. Select the following parameters, leaving the remaining parameters at their default values. Click **Save** to commit the changes.

- **Name** A descriptive name
- **Description** An informative description

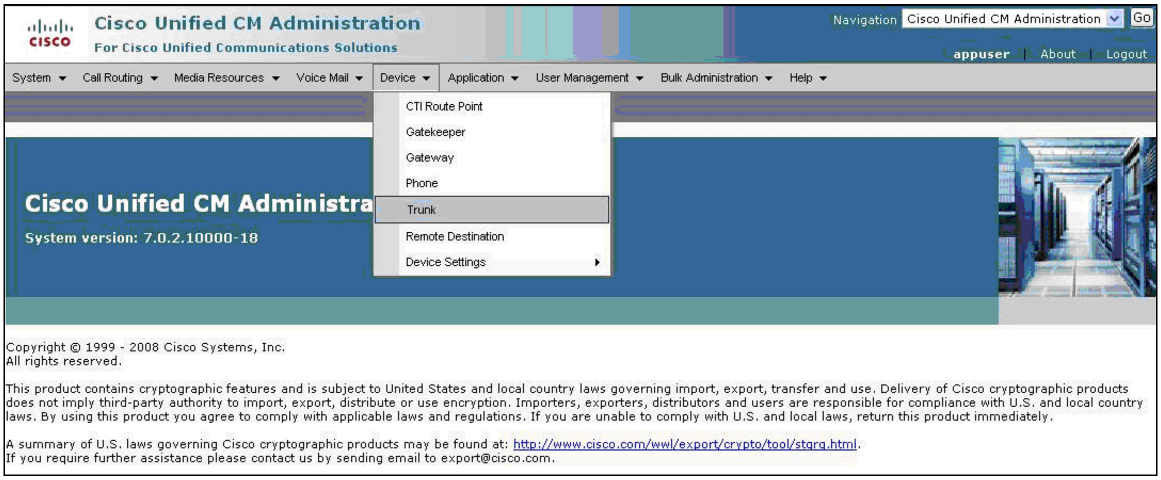
Ensure the following parameters are selected.

- **Accept Presence Subscription**
- **Accept Out-of-Dialog REFER**
- **Accept Unselected Notification**
- **Accept Replaces Header**



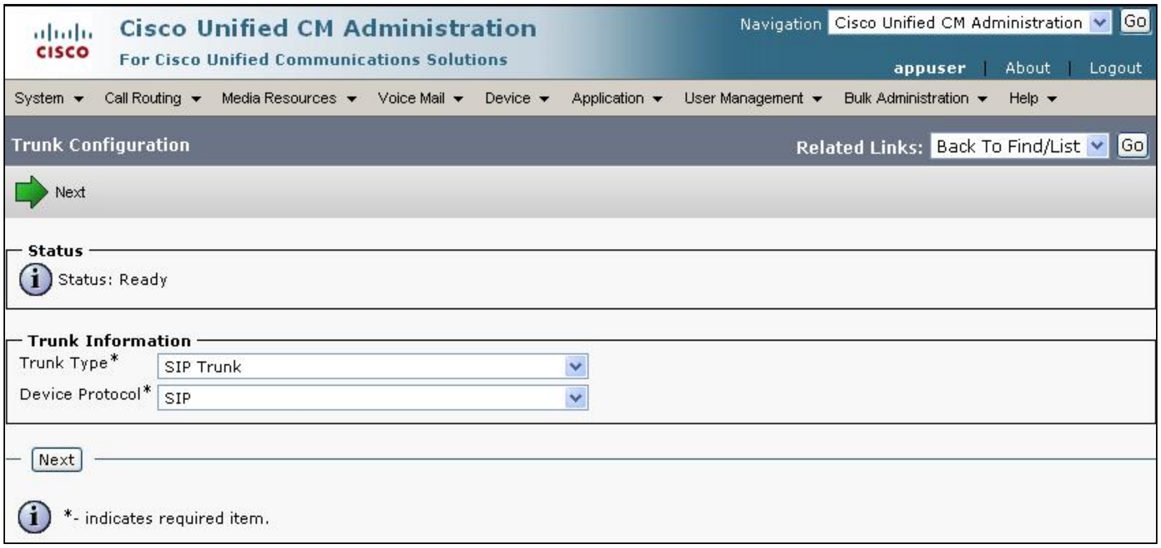
4.3. Administer SIP Trunk

Scroll to the top of the screen, and select **Device → Trunk** as shown below.



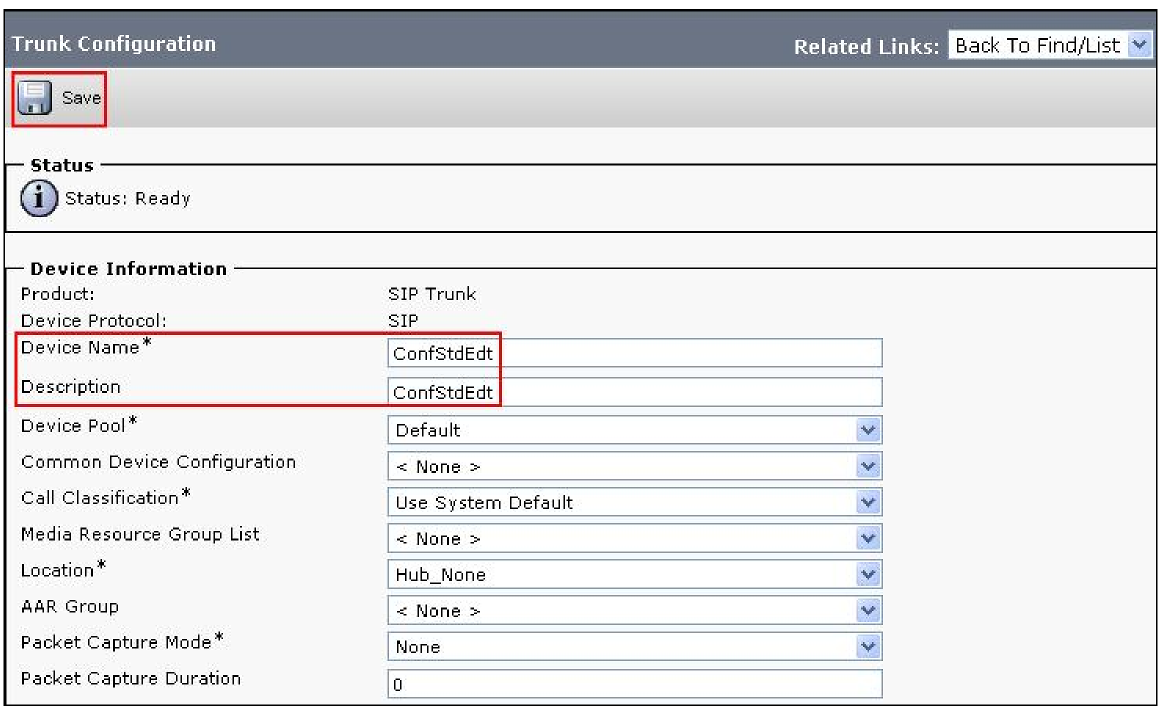
The **Find and List Trunks** screen is displayed. Click **Add New** to add a new SIP Trunk.

Select **SIP Trunk** as the **Trunk Type** and the **Device Protocol** field will automatically be changed to **SIP**. Click **Next** to continue.



The **SIP Trunk Configuration** screen is displayed. Select the following parameters, leaving the remaining parameters at their default values. Click **Save** to commit the changes.

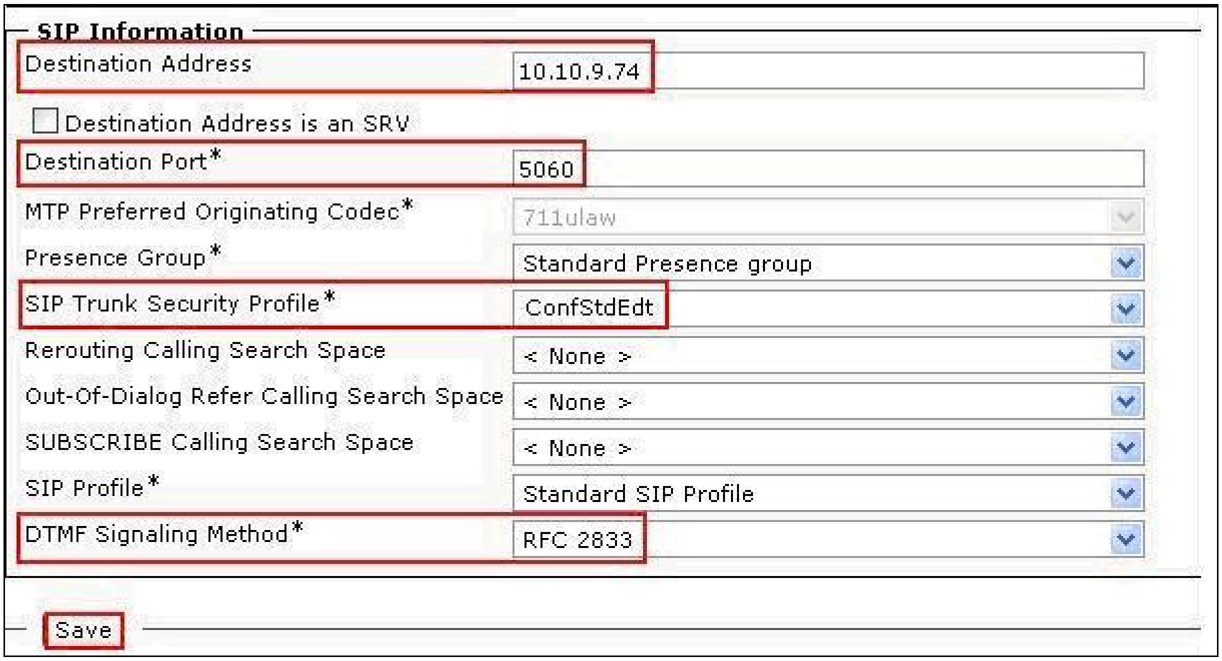
- **Device Name** A descriptive name
- **Description** An informative description for this trunk.



Navigate to the SIP Information section and select the following parameters, leaving the remaining parameters at their default values.

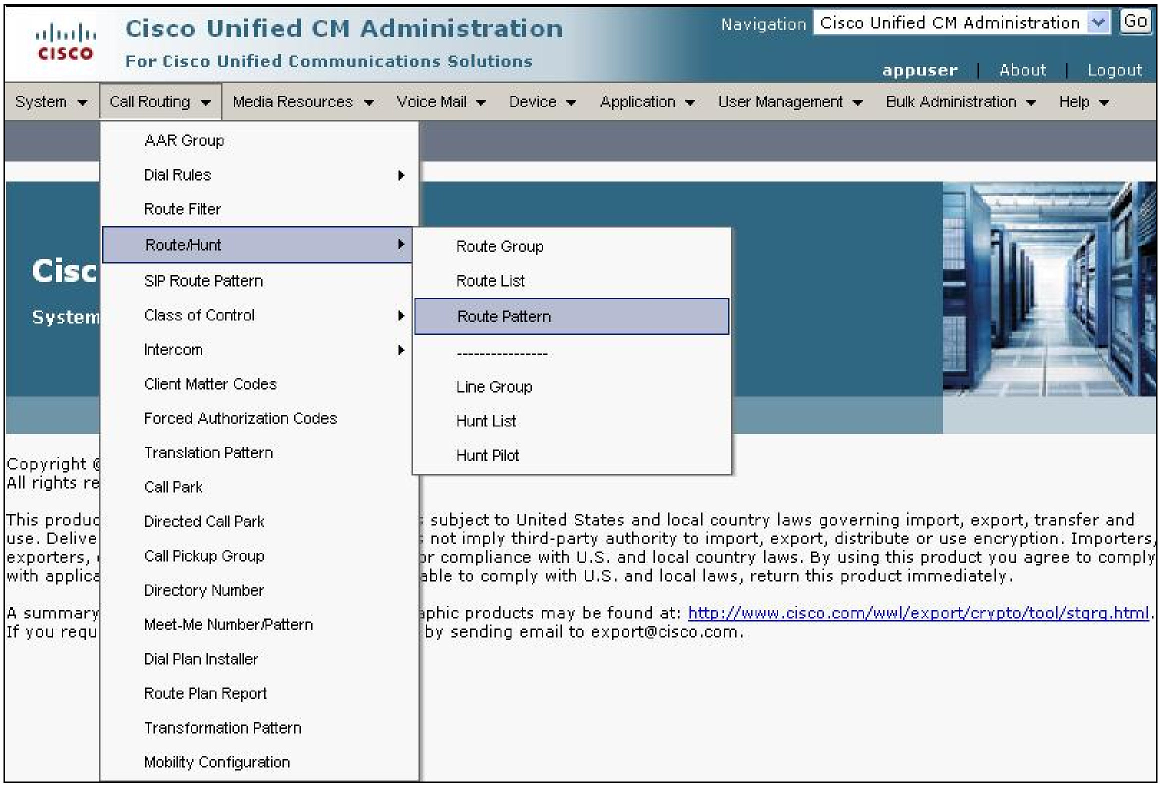
- **Destination Address** IP address of the Conferencing Standard Edition
- **Destination Port** Destination port number use for SIP Communications
- **SIP Trunk Security Profile** Profile configured in **Section 4.2**
- **DTMF Signaling Method** Select **RFC 2833**

Click **Save** to commit the changes.

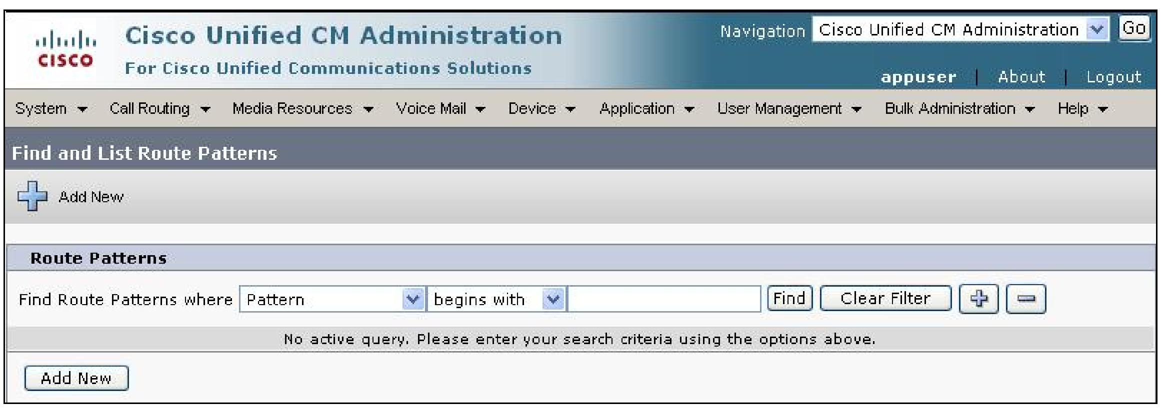


4.4. Administer Route Pattern

Scroll to the top of the screen, and select **Call Routing → Route/Hunt → Route Pattern** as shown below.



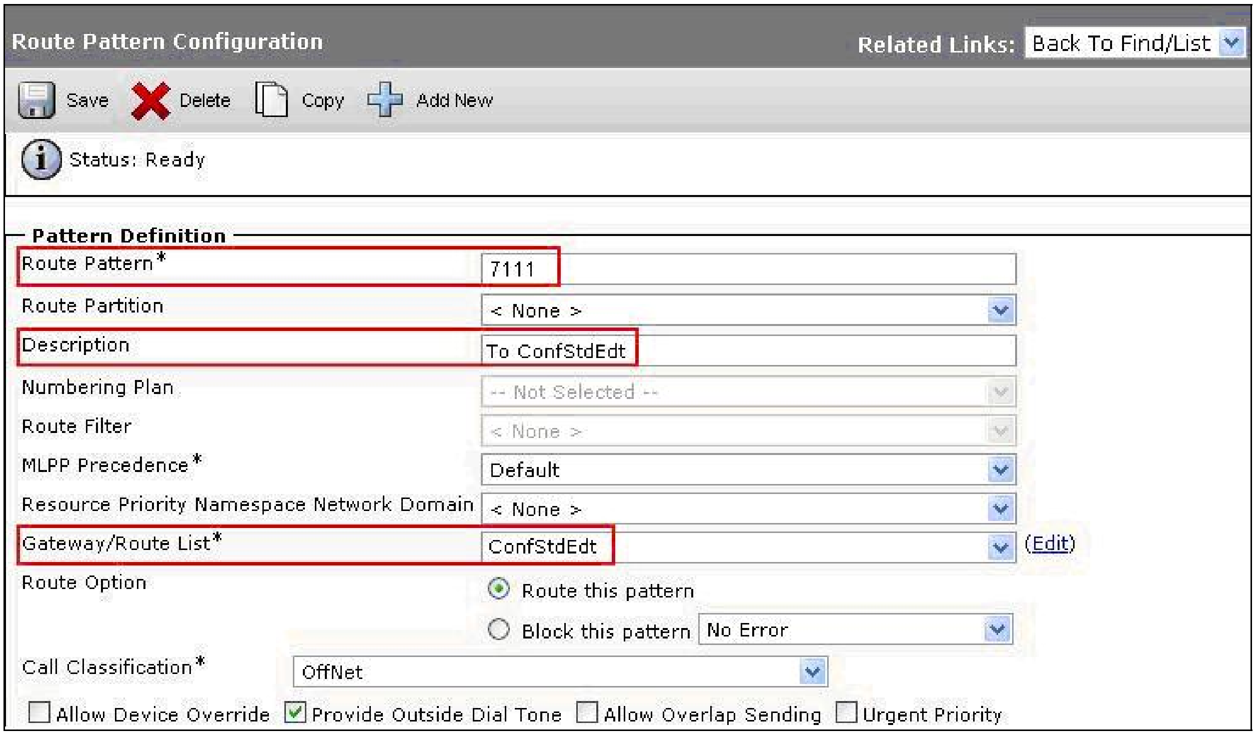
The **Find and List Route Patterns** screen is displayed. Click **Add New** to add a new Route Pattern.



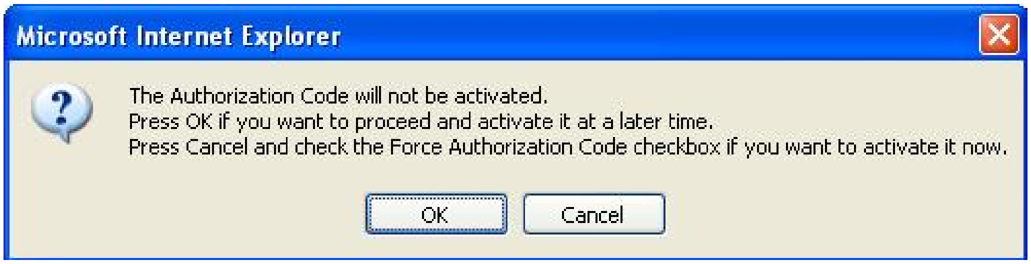
The following screen shows the route pattern used in the sample network. Select the following parameters, leaving the remaining parameters at their default values.

- **Route Pattern** **7111**, created in **Section 3.4**
- **Description** An informative description
- **Gateway/Route List** Select **ConfStdEdt**, created in **Section 4.3**, all calls to be routed through ConfStdEdt

Click **Save** to commit the changes (not shown).



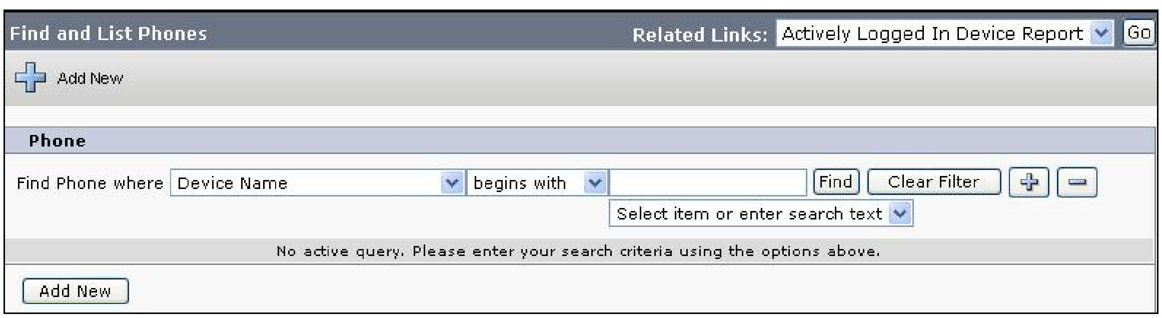
Click **OK** on the two subsequent pop up dialog boxes.



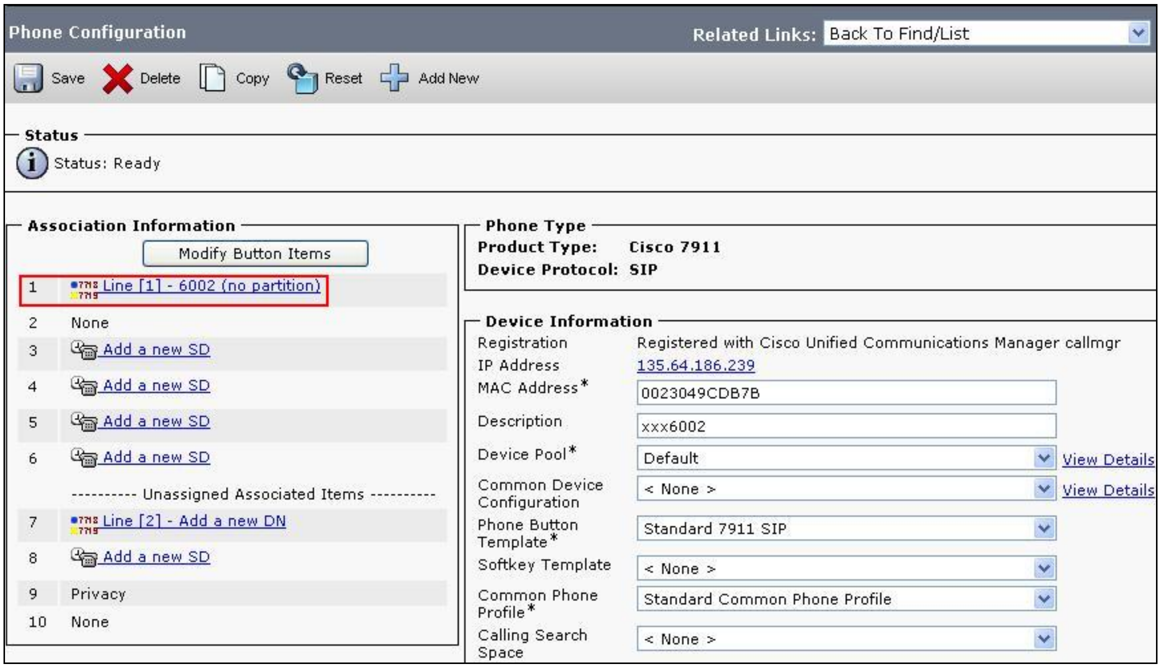
4.5. Administer Phones

Scroll to the top of the screen and select **Device → Phone** as shown below.

The **Find and List Phones** screen is displayed.

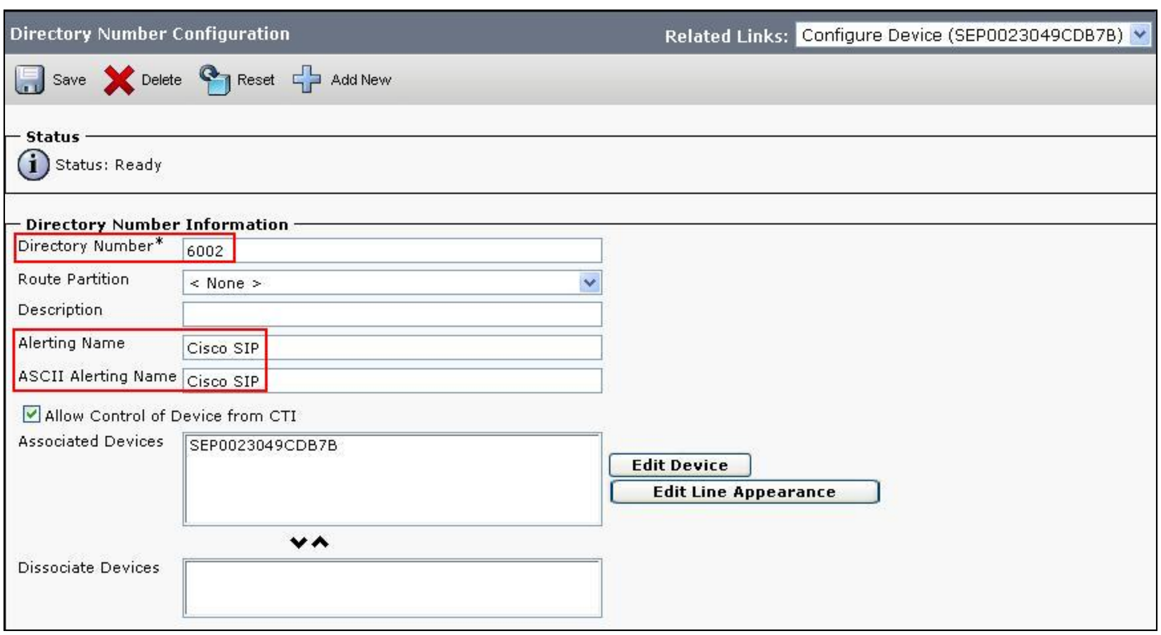


The following screen shows the display after a device has been selected. Click on the line for the device as highlighted in the screen below.



The following screen shows the display after the line has been selected. Select the following parameters, leaving the remaining parameters at their default values.

- **Directory Number** Select **6002**
- **Alerting Name** A descriptive Name
- **ASCII Alerting Name** A descriptive name



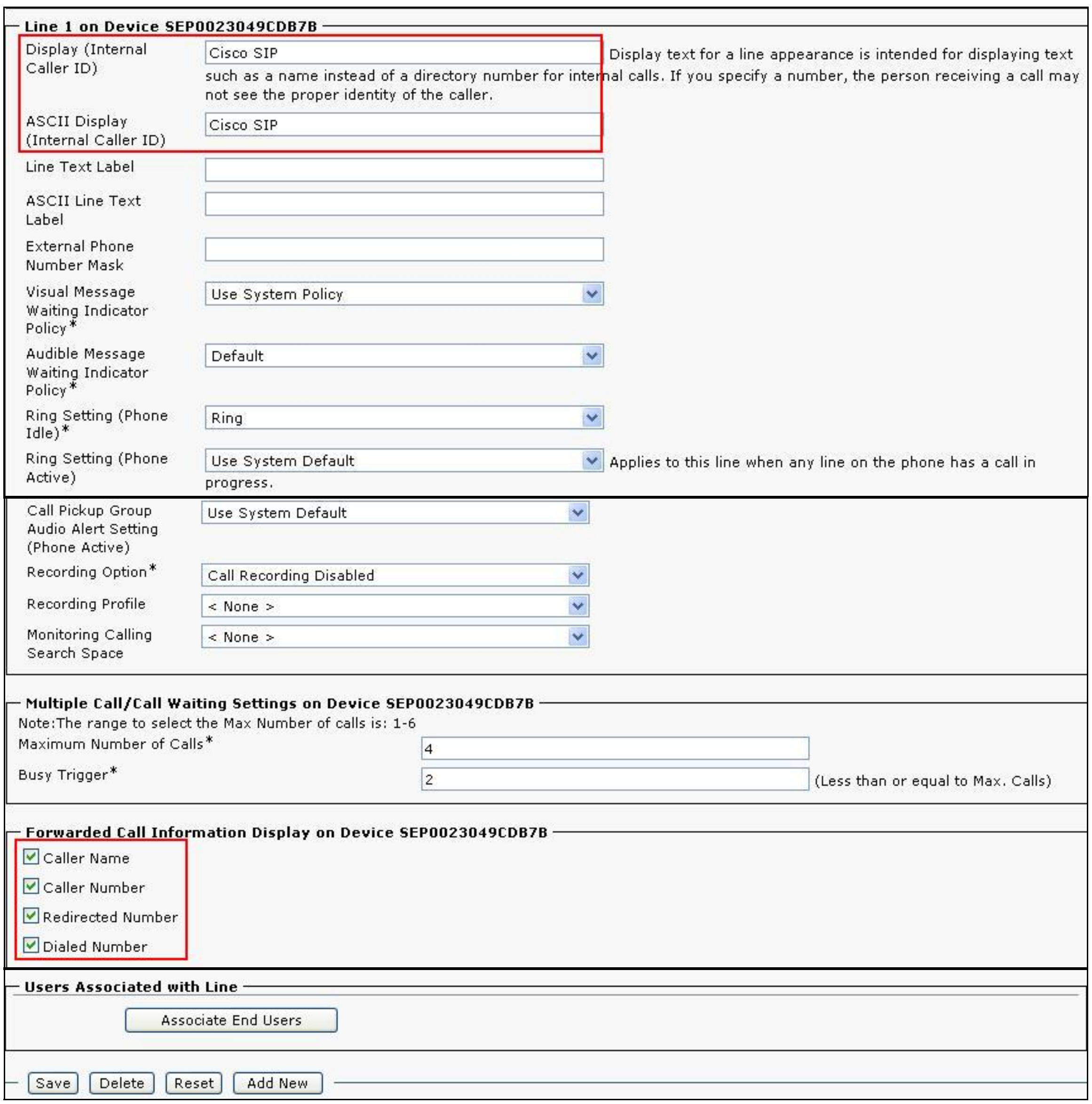
Navigate to **Line 1 on Device** section and select the following parameters, leaving the remaining parameters at their default values. This will be displayed on the called party phone on all outgoing calls.

- **Display (Internal Caller ID)** Descriptive details
- **ASCII Display** Descriptive details

Ensure the following parameters are selected.

- **Caller Name**
- **Caller Number**
- **Redirected Number**
- **Dialed Number**

Click **Save** to complete.



5. Verification Scenarios

Verify end to end audio between Conferencing Standard Edition and Cisco Unified Communications Manager, this is accomplished by:

- Placing a call from the 7941 IP Telephone (SIP) and the Cisco 7911G IP Telephone into conference ensuring 1 of the callers is a moderator.
- Verify both callers are in the same conference and there is two way talk-path between the callers.
- Initiate dial out by dialing *1 xxxx on the moderator phones touch pad, where xxxx is the extension for an end point. 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 talk-path between the callers.

6. Conclusion

As illustrated in these Application Notes, Avaya Aura™ Conferencing Standard Edition can interoperate successfully with Cisco Unified Communications Manager using SIP trunks.

7. 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.*

Cisco references are available at http://cisco.com

[4] *Cisco Unified Communications Manager Administration Guide for Cisco Unified Communications Manager Business Edition, Release 7.0(1), Part Number: OL-15405-01*

[5] *Cisco Unified Communications Manager Features and Services Guide for Cisco Unified Communications Manager Business Edition, Release 7.0(1), Part Number: OL-15409-01*

[6] *Cisco Unified Real-Time Monitoring Tool Administration Guide, Release 7.0(1), Part Number: OL-14994-01*

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