



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.

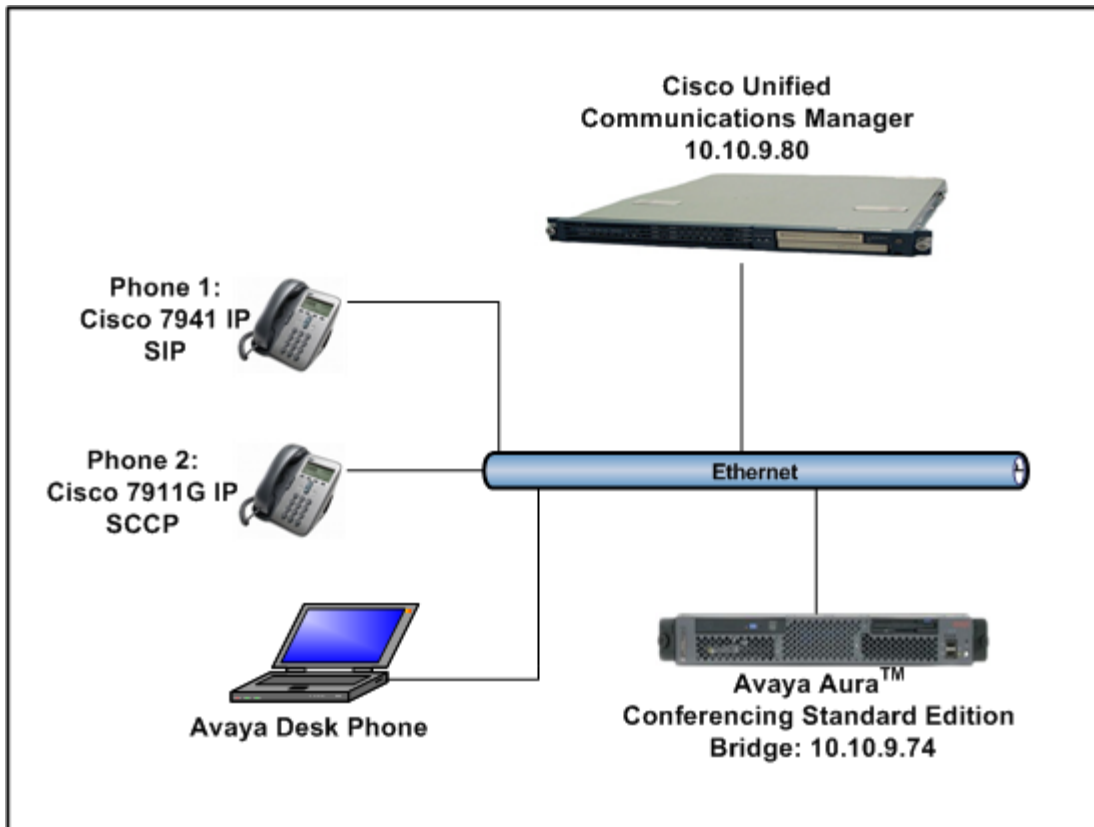


Figure 1

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™ Standard Conferencing Server (S8800)	Avaya Aura™ 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

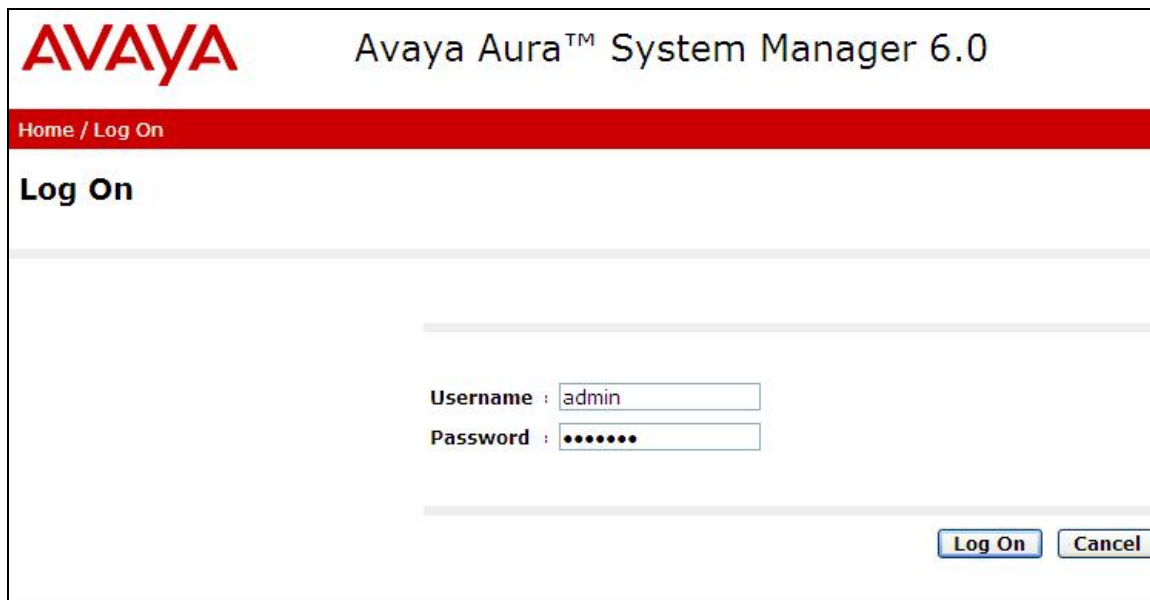
3. Configure Avaya Aura™ Conferencing Standard Edition

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.



AVAYA Avaya Aura™ System Manager 6.0

Home / Log On

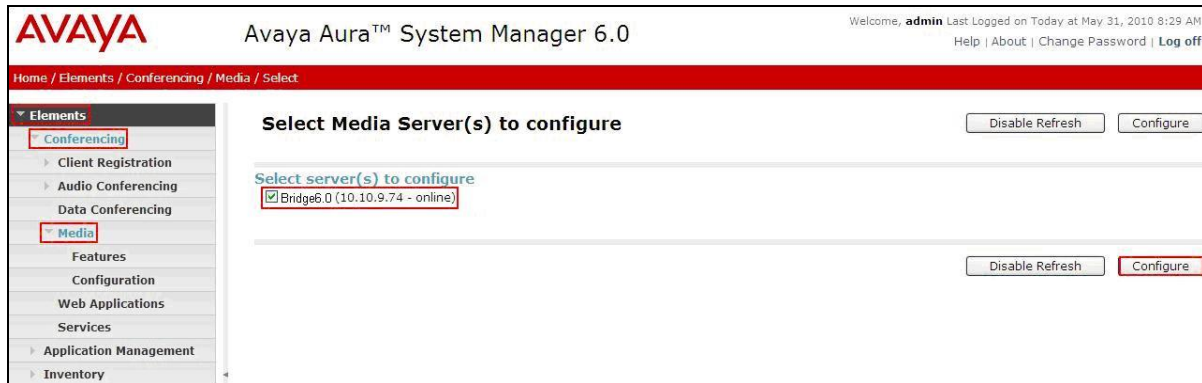
Log On

Username :

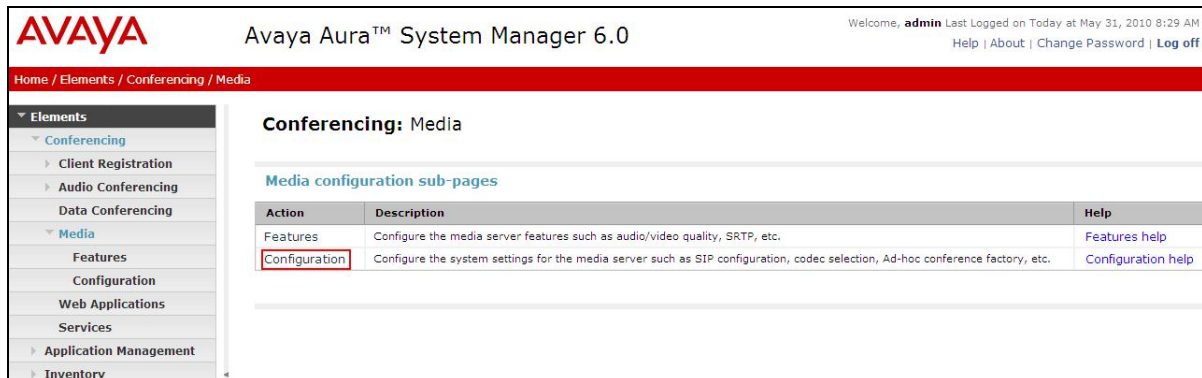
Password :

3.2. Configuring SIP Connectivity

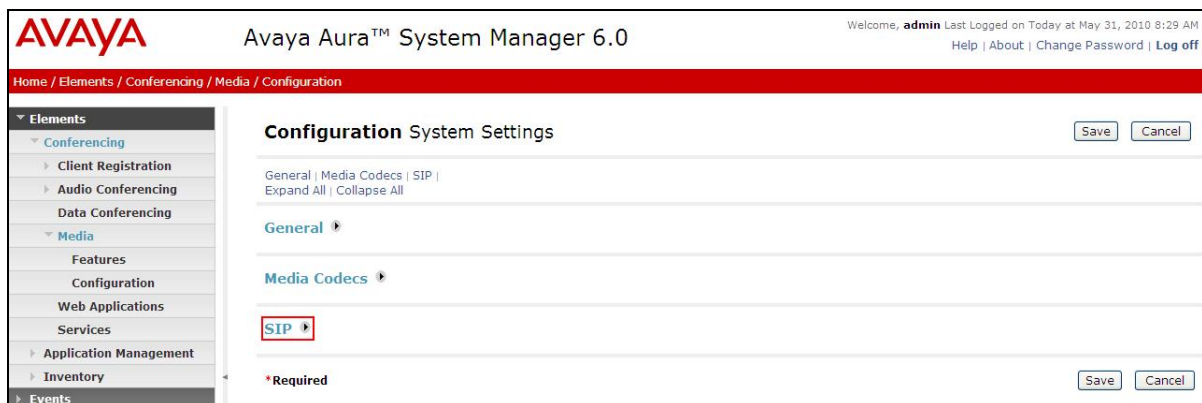
Configure settings that enable SIP connectivity between the conferencing bridge and other devices. Select **Elements** → **Conferencing** → **Media** on the left panel menu. From the right panel menu, select the media server to configure by selecting the tick box and select **Configure**.



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.

AVAYA Avaya Aura™ System Manager 6.0 Welcome, **admin** Last Logged on Today at May 31, 2010 8:29 AM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Media / Configuration

Configuration System Settings [Save] [Cancel]

General | Media Codecs | SIP | Expand All | Collapse All

General

Media Codecs

SIP

SIP Listener URI <sip:6000@10.10.9.74:5060;transport=tcp>

Response Contact <sip:6000@10.10.9.74:5060;transport=tcp>

Session Refresh Timer 1800

Min Session Refresh Timer Allowed 1800

* Required [Save] [Cancel]

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**.

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on Today at May 31, 2010 8:29 AM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Audio Conferencing / Select

Elements

- Conferencing
 - Client Registration
 - Audio Conferencing**
 - Bridge Features
 - Conference Features
 - Call Routing
 - System Config
 - General Config
 - Data Conferencing
- Media
 - Web Applications
 - Services
- Application Management
- Inventory

Select Conferencing Server(s) to configure

Disable Refresh Configure

Select server(s) to configure

☒ Bridge6.0 (10.10.9.74 - online)

Disable Refresh **Configure**

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

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on Today at May 31, 2010 8:29 AM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Audio Conferencing

Elements

- Conferencing
 - Client Registration
 - Audio Conferencing**
 - Bridge Features
 - Conference Features
 - Call Routing
 - System Config
 - General Config
 - Data Conferencing
- Media
 - Web Applications
 - Services
- Application Management
- Inventory

Conferencing: Audio Conferencing

Audio Conferencing Configuration

Action	Description	Help
Bridge Features	Configure conferencing bridge features	Bridge Features help
Conference Features	Configure conferencing defaults and features	Conference Features help
Call Routing	Configure incoming call routing and outgoing call settings	Call Routing help
System Config	Configure networking and system settings	System Configuration help
General Config	Configure general conferencing settings	General Configuration help

Disable Refresh **Configure**

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

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on Today at May 31, 2010 8:29 AM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Audio Conferencing / Call Routing

Audio Conferencing: Call Routing [Save] [Cancel]

Call Routing | Dial-out | Blast Dial Settings |
Expand All | Collapse All

Call Routing ▼

Number of digits to match * 4

Call Branding [Edit]

Telnum to URI [Edit]

URI to Telnum [Edit]

Dial-out ▶

Blast Dial Settings ▶

*Required [Save] [Cancel]

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

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on Today at May 31, 2010 8:29 AM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Audio Conferencing / Call Routing / Telnum Mapping

Telnum to URI mappings [Done]

Telnum to URI mappings

[View] [Edit] [New] [Delete] [Move up] [Move down]

1 Item Refresh

TelNum	URI	Comment
*	\$1	default

Select : None

[Done]

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.

Avaya Aura™ Conferencing Manager 6.0

Welcome, **admin** Last Logged on Today at June 11, 2010 3:35 PM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Audio Conferencing / Call Routing / Telnum Mapping / Entry

Audio Conferencing: Telnum to URI Mapping [Save] [Cancel]

* Telnum *

* URI sip:\$0@10.10.9.80:5

Comment Route_calls_to_CUCM

* Required [Save] [Cancel]

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

Avaya Aura™ Conferencing Manager 6.0

Welcome, **admin** Last Logged on Today at June 11, 2010 3:35 PM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Audio Conferencing / Call Routing / Telnum Mapping

You have saved changes to the configuration which are not committed yet.

Telnum to URI mappings [Done]

Telnum to URI mappings

[View] [Edit] [New] [Delete] [Move up] [Move down]

1 Item Refresh

TelNum	URI	Comment
*	sip:\$0@10.10.9.80:5060;transport=tcp	Route_calls_to_CUCM

Select : None

[Done]

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

AVAYA Avaya Aura™ Conferencing Manager 6.0

Welcome, **admin** Last Logged on Today at June 11, 2010 3:35 PM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Audio Conferencing / Call Routing

▼ Elements

- ▼ Conferencing
 - Client Registration
 - ▼ Audio Conferencing
 - Bridge Features
 - Conference Features
 - Call Routing
 - System Config
 - General Config
 - Data Conferencing
 - Media
 - Web Applications
 - Services
 - Application Management
 - Inventory

You have saved changes to the configuration which are not committed yet.

Audio Conferencing: Call Routing Save Cancel

Call Routing | Dial-out | Blast Dial Settings |
Expand All | Collapse All

Call Routing ▼

Number of digits to match *

Call Branding Edit

Telnum to URI Edit

URI to Telnum Edit

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

AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on Today at May 31, 2010 8:29 AM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Apply Changes

▼ Elements

- ▼ Conferencing
 - Client Registration
 - ▼ Audio Conferencing
 - Bridge Features
 - Conference Features
 - Call Routing
 - System Config
 - General Config
 - Data Conferencing
 - Media
 - Web Applications
 - Services
 - Application Management
 - Inventory
- ▼ Events
- ▼ Groups & Roles
- Licenses
- Routing
- Security

Apply Changes Disable Refresh Apply Changes Discard Changes Add more changes

Impact of changes

Host name / IP address	Impact of changes	Server State
10.10.9.72 <ul style="list-style-type: none">No changes	NONE	Powered on
10.10.9.73 <ul style="list-style-type: none">No changes	NONE	Powered on
10.10.9.75 <ul style="list-style-type: none">No changes	NONE	Powered on
10.10.9.74 <ul style="list-style-type: none">Changing "bridge.telnumToUriEntries[0].comment".Changing "bridge.telnumToUriEntries[0].telnumConversion".	NONE	Powered on

Disable Refresh Apply Changes Discard Changes Add more 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**.

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on Today at May 31, 2010 8:29 AM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Elements / Conferencing / Audio Conferencing / Select

Elements

- Conferencing
 - Client Registration
 - Audio Conferencing
 - Bridge Features
 - Conference Features
 - Call Routing
 - System Config
 - General Config
 - Data Conferencing
 - Media
 - Web Applications
 - Services
 - Application Management
 - Inventory

Select Conferencing Server(s) to configure

[Disable Refresh](#) [Configure](#)

Select server(s) to configure

☒ Bridge6.0 (10.10.9.74 - online)

[Disable Refresh](#) [Configure](#)

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

Avaya Aura™ Conferencing Manager 6.0

Welcome, **admin** Last Logged on Today at June 15, 2010 1:33 PM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Elements / Conferencing / Audio Conferencing

Elements

- Conferencing
 - Client Registration
 - Audio Conferencing
 - Bridge Features
 - Conference Features
 - Call Routing
 - System Config
 - General Config
 - Data Conferencing
 - Media
 - Web Applications
 - Services
 - Application Management
 - Inventory

Conferencing: Audio Conferencing

Audio Conferencing Configuration

Action	Description	Help
Bridge Features	Configure conferencing bridge features	Bridge Features help
Conference Features	Configure conferencing defaults and features	Conference Features help
Call Routing	Configure incoming call routing and outgoing call settings	Call Routing help
System Config	Configure networking and system settings	System Configuration help
General Config	Configure general conferencing settings	General Configuration help

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

AVAYA Avaya Aura™ Conferencing Manager 6.0

Welcome, **admin** Last Logged on Today at June 15, 2010 1:33 PM

Help | About | Change Password | **Log off**

Home / Elements / Conferencing / Audio Conferencing / Conference Features

▼ Elements

- ▼ Conferencing
 - Client Registration
 - ▼ Audio Conferencing
 - Bridge Features
 - Conference Features
 - Call Routing
 - System Config
 - General Config
 - Data Conferencing
 - Media
 - Web Applications
 - Services
 - Application Management
 - Inventory

Audio Conferencing: Conference Features [Save] [Cancel]

Conference Defaults | Conference Settings | Conference Error Behaviour | Conference Features | Adhoc Conferencing | Expand All | Collapse All

Conference Defaults ▸

Conference Settings ▾

Conference Error Behaviour ▸

Conference Features ▸

Adhoc Conferencing ▸

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.

AVAYA Avaya Aura™ Conferencing Manager 6.0

Welcome, **admin** Last Logged on Today at June 15, 2010 1:33 PM

Help | About | Change Password | **Log off**

Home / Elements / Conferencing / Audio Conferencing / Conference Features

▼ Elements

- ▼ Conferencing
 - Client Registration
 - ▼ Audio Conferencing
 - Bridge Features
 - Conference Features
 - Call Routing
 - System Config
 - General Config
 - Data Conferencing
 - Media
 - Web Applications
 - Services
 - Application Management
 - Inventory

Audio Conferencing: Conference Features [Save] [Cancel]

Conference Defaults | Conference Settings | Conference Error Behaviour | Conference Features | Adhoc Conferencing | Expand All | Collapse All

Conference Defaults ▸

Conference Settings ▾

Scan Time 10

Scan Attempts (1-3) 3

Auto Hang-Up ☐

Warning Tones ☐

Originator Dial Out All

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

AVAYA Avaya Aura™ System Manager 6.0 Welcome, **admin** Last Logged on Today at May 31, 2010 8:29 AM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

[Home](#) / [Elements](#) / [Conferencing](#) / [Apply Changes](#)

Apply Changes Disable Refresh Apply Changes Discard Changes Add more changes

Impact of changes

Host name / IP address	Impact of changes	Server State
10.10.9.72 • No changes	NONE	Powered on
10.10.9.73 • No changes	NONE	Powered on
10.10.9.75 • No changes	NONE	Powered on
10.10.9.74 • Changing "bridge.originatorDialOut".	NONE	Powered on

Disable Refresh Apply Changes Discard Changes Add more 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**.

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on Today at May 31, 2010 8:29 AM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Audio Conferencing / Select

Elements

- Conferencing
 - Client Registration
 - Audio Conferencing**
 - Bridge Features
 - Conference Features
 - Call Routing
 - System Config
 - General Config
 - Data Conferencing
- Media
 - Web Applications
 - Services
- Application Management
- Inventory

Select Conferencing Server(s) to configure

Disable Refresh Configure

Select server(s) to configure

☒ Bridge6.0 (10.10.9.74 - online)

Disable Refresh **Configure**

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

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on Today at May 31, 2010 8:29 AM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Audio Conferencing

Elements

- Conferencing
 - Client Registration
 - Audio Conferencing**
 - Bridge Features
 - Conference Features
 - Call Routing
 - System Config
 - General Config
 - Data Conferencing
- Media
 - Web Applications
 - Services
- Application Management
- Inventory

Conferencing: Audio Conferencing

Audio Conferencing Configuration

Action	Description	Help
Bridge Features	Configure conferencing bridge features	Bridge Features help
Conference Features	Configure conferencing defaults and features	Conference Features help
Call Routing	Configure incoming call routing and outgoing call settings	Call Routing help
System Config	Configure networking and system settings	System Configuration help
General Config	Configure general conferencing settings	General Configuration help

Disable Refresh Configure

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

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on Today at May 31, 2010 8:29 AM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Elements / Conferencing / Audio Conferencing / Call Routing

Audio Conferencing: Call Routing Save Cancel

Call Routing | Dial-out | Blast Dial Settings |
Expand All | Collapse All

Call Routing ▾

Number of digits to match * 4 ▾

Call Branding Edit

Telnum to URI Edit

URI to Telnum Edit

Dial-out ▾

Blast Dial Settings ▾

*Required Save Cancel

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

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on Today at May 31, 2010 8:29 AM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Elements / Conferencing / Audio Conferencing / Call Routing / Call Branding

Call Branding Entry table Done

Add Edit Delete

1 Item Refresh

DDI	Name	Organization Name	Reservation Group
○	????		0 ▾

Select : None

Done

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.

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on Today at May 31, 2010 8:29 AM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Elements / Conferencing / Audio Conferencing / Call Routing / Call Branding / Add

Call Branding Add entry Save

Call Branding Details

DDI * 7111

Name SIL_Test

Organization Name Avaya

Reservation Group 0

Message Number 1

Message Set Number 1

Use Conf Message Set ☐

On entry Scan call flow

On failure Direct to enter queue

Conference Room Start 0

Conference Room End 0

Conference Security Code

Select Phone Number Description Location

Add Delete

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

Avaya Aura™ Conferencing Manager 6.0

Welcome, **admin** Last Logged on Today at June 11, 2010 3:35 PM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Elements / Conferencing / Audio Conferencing / Call Routing / Call Branding

You have saved changes to the configuration which are not committed yet.

Call Branding Entry table Done

Add Edit Delete

2 Items Refresh

DDI	Name	Organization Name	Reservation Group
7111	SIL_Test	Avaya	0
???			0

Select : None

Done

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

Avaya Aura™ System Manager 6.0

Welcome, admin Last Logged on Today at May 31, 2010 8:29 AM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Audio Conferencing / Call Routing

Audio Conferencing: Call Routing [Save] [Cancel]

Call Routing | Dial-out | Blast Dial Settings |
Expand All | Collapse All

Call Routing ▾

Number of digits to match * 4 ▾

Call Branding [Edit]

Telnum to URI [Edit]

URI to Telnum [Edit]

Dial-out ▾

Blast Dial Settings ▾

*Required [Save] [Cancel]

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

Avaya Aura™ System Manager 6.0

Welcome, admin Last Logged on Today at May 31, 2010 8:29 AM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Apply Changes

Apply Changes [Disable Refresh] [Apply Changes] [Discard Changes] [Add more changes]

Impact of changes

Host name / IP address	Impact of changes	Server State
10.10.9.72	No changes	Powered on
10.10.9.73	No changes	Powered on
10.10.9.75	No changes	Powered on
10.10.9.74	Changing "bridge.callBrandingEntries[0].confSCodeNum" from "" to "". Changing "bridge.callBrandingEntries[0].ddi" from "?????" to "1111". Changing "bridge.callBrandingEntries[0].name" from "null" to "SIL_Test". Changing "bridge.callBrandingEntries[0].onFailure" from "DEFAULT" to "ENTER". Changing "bridge.callBrandingEntries[0].organizationName" from "null" to "Avaya". Changing "bridge.callBrandingEntries[0].useConferenceMessageSet" from "true" to "false". Changing "bridge.callBrandingEntries[1]" from "null" to "CallBrandingEntry[ddi = '????', resGroup = 0, messageNumber = 1, messageSetNumber = 1, useConferenceMessageSet = true, onEntry = SCAN, onFailure = DEFAULT, name = 'null', organizationName = 'null', confSCodeNum = '1', roomStart = 0, roomEnd = 0, phoneNumbers = []]".	Powered on

[Disable Refresh] [Apply Changes] [Discard Changes] [Add more 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.

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This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at: <http://www.cisco.com/www/export/crypto/tool/starg.html>.
If you require further assistance please contact us by sending email to export@cisco.com.

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.

System

- Server
 - Cisco Unified CM
 - Cisco Unified CM Group
 - Phone NTP Reference
 - Date/Time Group
 - Presence Group
 - Region
 - Device Pool
 - Device Mobility
 - DHCP
 - LDAP
 - Location
 - Physical Location
 - SRST
 - MLPP
 - Enterprise Parameters
 - Service Parameters
- Security Profile
 - Phone Security Profile
 - SIP Trunk Security Profile
 - CUMA Server Security Profile
- Application Server
- Licensing

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.

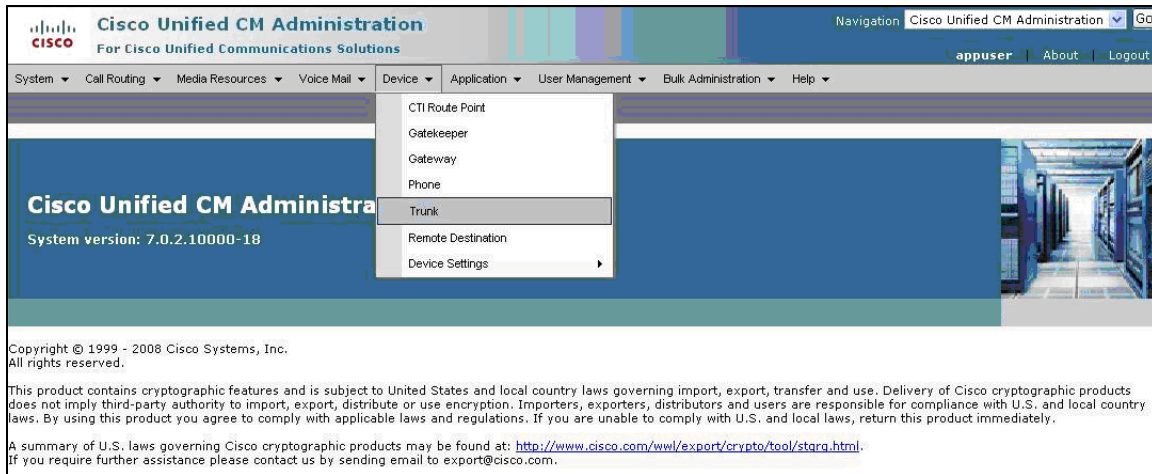
- **Name** A descriptive name
- **Description** An informative description

Ensure the following parameters are selected.

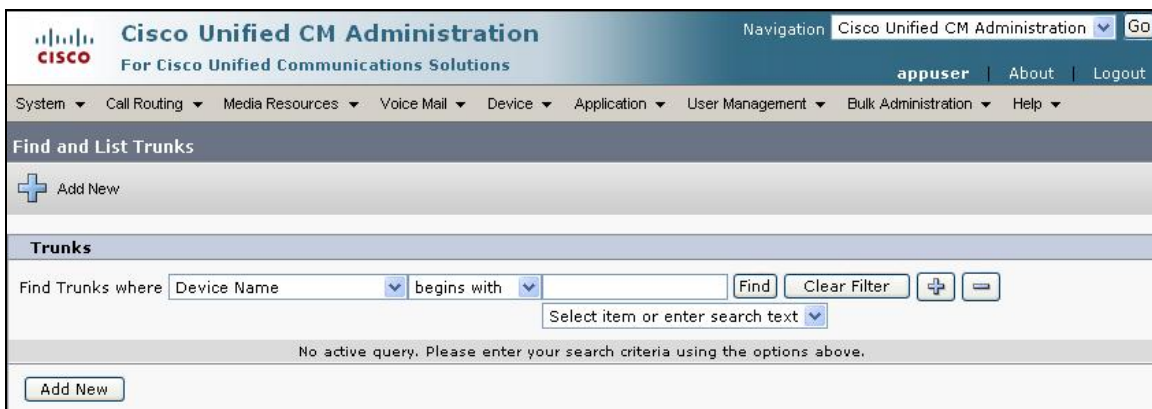
- **Accept Presence Subscription**
- **Accept Out-of-Dialog REFER**
- **Accept Unselected Notification**
- **Accept Replaces Header**

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.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

appuser | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration Related Links: Back To Find/List Go

Next

Status
Status: Ready

Trunk Information
Trunk Type* SIP Trunk
Device Protocol* SIP

Next

* - indicates required item.

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.

Trunk Configuration Related Links: Back To Find/List

Save

Status
Status: Ready

Device Information
Product: SIP Trunk
Device Protocol: SIP
Device Name* ConfStdEdt
Description ConfStdEdt
Device Pool* Default
Common Device Configuration < None >
Call Classification* Use System Default
Media Resource Group List < None >
Location* Hub_None
AAR Group < None >
Packet Capture Mode* None
Packet Capture Duration 0

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**

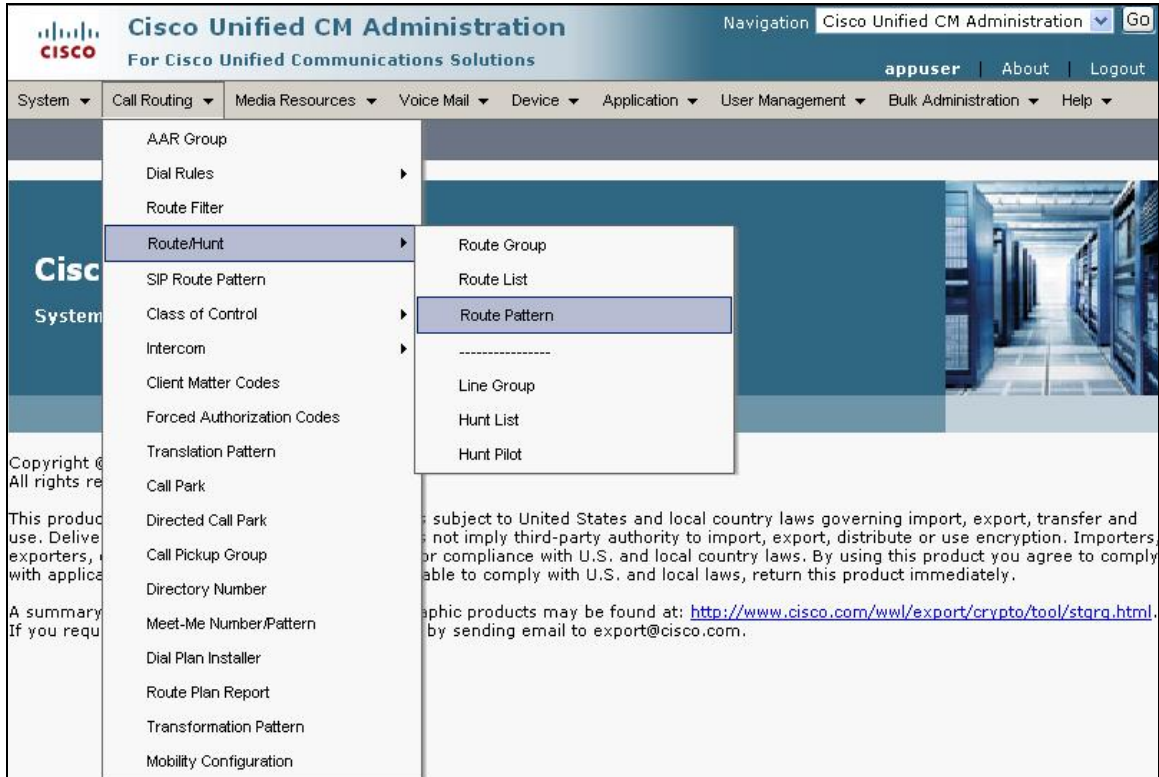
Click **Save** to commit the changes.

The screenshot shows a web-based configuration interface for SIP Information. The title bar reads "SIP Information". Below it, several fields are visible, with five of them highlighted by red rectangular boxes: "Destination Address" (containing "10.10.9.74"), "Destination Port*" (containing "5060"), "SIP Trunk Security Profile*" (containing "ConfStdEdt"), "DTMF Signaling Method*" (containing "RFC 2833"), and a "Save" button at the bottom left. Other fields include "Destination Address is an SRV" (unchecked checkbox), "MTP Preferred Originating Codec*" (dropdown with "711ulaw"), "Presence Group*" (dropdown with "Standard Presence group"), "Rerouting Calling Search Space" (dropdown with "< None >"), "Out-Of-Dialog Refer Calling Search Space" (dropdown with "< None >"), "SUBSCRIBE Calling Search Space" (dropdown with "< None >"), and "SIP Profile*" (dropdown with "Standard SIP Profile").

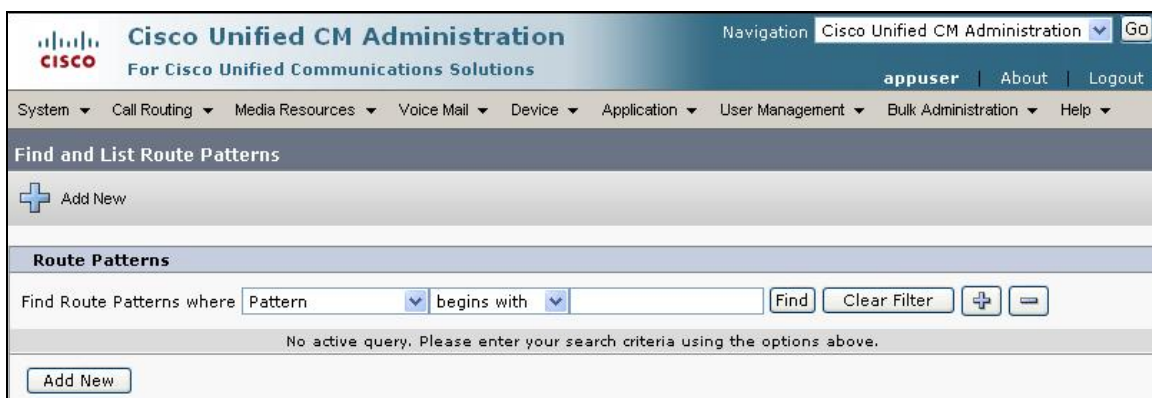
SIP Information	
Destination Address	10.10.9.74
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	5060
MTP Preferred Originating Codec*	711ulaw
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	ConfStdEdt
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	RFC 2833
Save	

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.

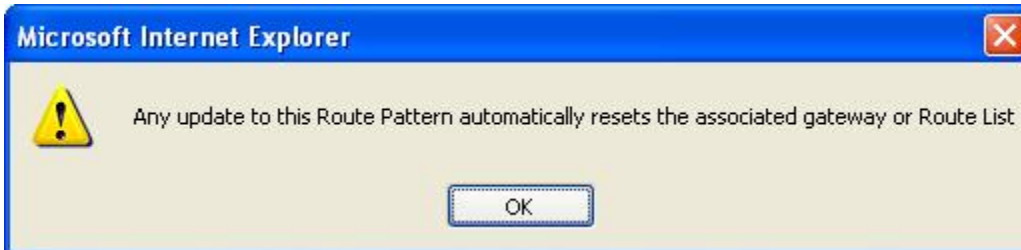
- **Route Pattern** 7111, created in **Section 3.4**
- **Description** An informative description
- **Gateway/Route List** Select **ConfStdEdt**, created in **Section 4.3**, all calls to be routed through ConfStdEdt

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

The screenshot shows the 'Route Pattern Configuration' window. At the top, there's a title bar with 'Route Pattern Configuration' and 'Related Links: Back To Find/List'. Below the title bar is a toolbar with icons for Save, Delete, Copy, and Add New. A status bar indicates 'Status: Ready'. The main section is titled 'Pattern Definition' and contains several fields: 'Route Pattern*' (7111), 'Route Partition' (< None >), 'Description' (To ConfStdEdt), 'Numbering Plan' (-- Not Selected --), 'Route Filter' (< None >), 'MLPP Precedence*' (Default), 'Resource Priority Namespace Network Domain' (< None >), 'Gateway/Route List*' (ConfStdEdt), and 'Route Option' (Route this pattern). There are also checkboxes for 'Allow Device Override', 'Provide Outside Dial Tone', 'Allow Overlap Sending', and 'Urgent Priority'. The 'Provide Outside Dial Tone' checkbox is checked.

Pattern Definition	
Route Pattern*	7111
Route Partition	< None >
Description	To ConfStdEdt
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
Resource Priority Namespace Network Domain	< None >
Gateway/Route List*	ConfStdEdt (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
Call Classification*	OffNet
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	

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

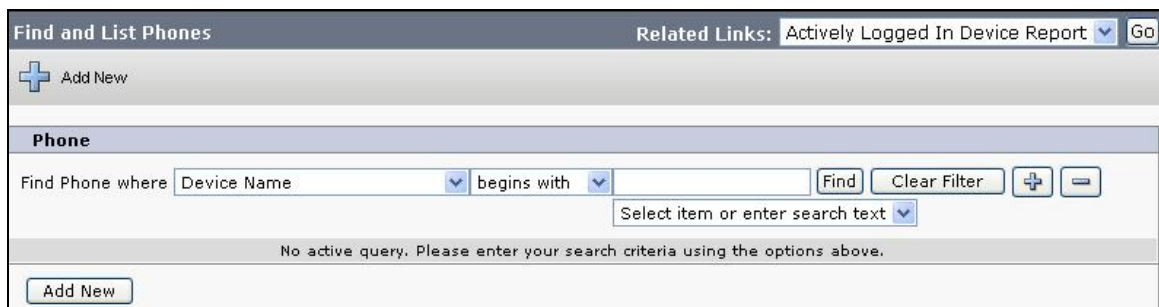


4.5. Administer Phones

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.

Phone Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Reset Add New

Status
Status: Ready

Association Information
Modify Button Items

1	Line [1] - 6002 (no partition)
2	None
3	Add a new SD
4	Add a new SD
5	Add a new SD
6	Add a new SD
----- Unassigned Associated Items -----	
7	Line [2] - Add a new DN
8	Add a new SD
9	Privacy
10	None

Phone Type
Product Type: Cisco 7911
Device Protocol: SIP

Device Information

Registration	Registered with Cisco Unified Communications Manager callmgr	
IP Address	135.64.186.239	
MAC Address*	0023049CDB7B	
Description	xxx6002	
Device Pool*	Default	View Details
Common Device Configuration	< None >	View Details
Phone Button Template*	Standard 7911 SIP	
Softkey Template	< None >	
Common Phone Profile*	Standard Common Phone Profile	
Calling Search Space	< None >	

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

Directory Number Configuration Related Links: [Configure Device \(SEP0023049CDB7B\)](#)

Save Delete Reset Add New

Status
Status: Ready

Directory Number Information

Directory Number*	6002
Route Partition	< None >
Description	
Alerting Name	Cisco SIP
ASCII Alerting Name	Cisco SIP
<input checked="" type="checkbox"/> Allow Control of Device from CTI	
Associated Devices	SEP0023049CDB7B
Edit Device Edit Line Appearance	
Dissociate Devices	

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.

Line 1 on Device SEP0023049CDB7B		
Display (Internal Caller ID)	Cisco SIP	Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)	Cisco SIP	
Line Text Label		
ASCII Line Text Label		
External Phone Number Mask		
Visual Message Waiting Indicator Policy*	Use System Policy	
Audible Message Waiting Indicator Policy*	Default	
Ring Setting (Phone Idle)*	Ring	
Ring Setting (Phone Active)	Use System Default	Applies to this line when any line on the phone has a call in progress.
Call Pickup Group	Use System Default	
Audio Alert Setting (Phone Active)		
Recording Option*	Call Recording Disabled	
Recording Profile	< None >	
Monitoring Calling Search Space	< None >	
Multiple Call/Call Waiting Settings on Device SEP0023049CDB7B Note: The range to select the Max Number of calls is: 1-6 Maximum Number of Calls* <input type="text" value="4"/> Busy Trigger* <input type="text" value="2"/> (Less than or equal to Max. Calls)		
Forwarded Call Information Display on Device SEP0023049CDB7B <input checked="" type="checkbox"/> Caller Name <input checked="" type="checkbox"/> Caller Number <input checked="" type="checkbox"/> Redirected Number <input checked="" type="checkbox"/> Dialed Number		
Users Associated with Line <input type="button" value="Associate End Users"/>		
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/>		

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*

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