



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura™ Communication Manager 6.0 and Avaya Aura™ Session Manager 6.0 to Allow Interoperability between Avaya Aura™ Conferencing Standard Edition 6.0 using SIP Trunks – Issue 1.0

Abstract

These Application Notes present a sample configuration for a network that uses Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager to connect Avaya Aura™ Conferencing Standard Edition using SIP trunks.

Testing was conducted at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes present a sample configuration for a network that uses Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager to connect Avaya Aura™ Conferencing Standard Edition using SIP trunks. SIP trunks connect Avaya Aura™ Communication Manager and Avaya Aura™ Conferencing Standard Edition to Avaya Aura™ Session Manager. 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. Installation and configuration details can be found in references [7], [8], [9], [10] and [11].

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 (CRS) Front End. Installation details can be found in references [2] and [3].

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

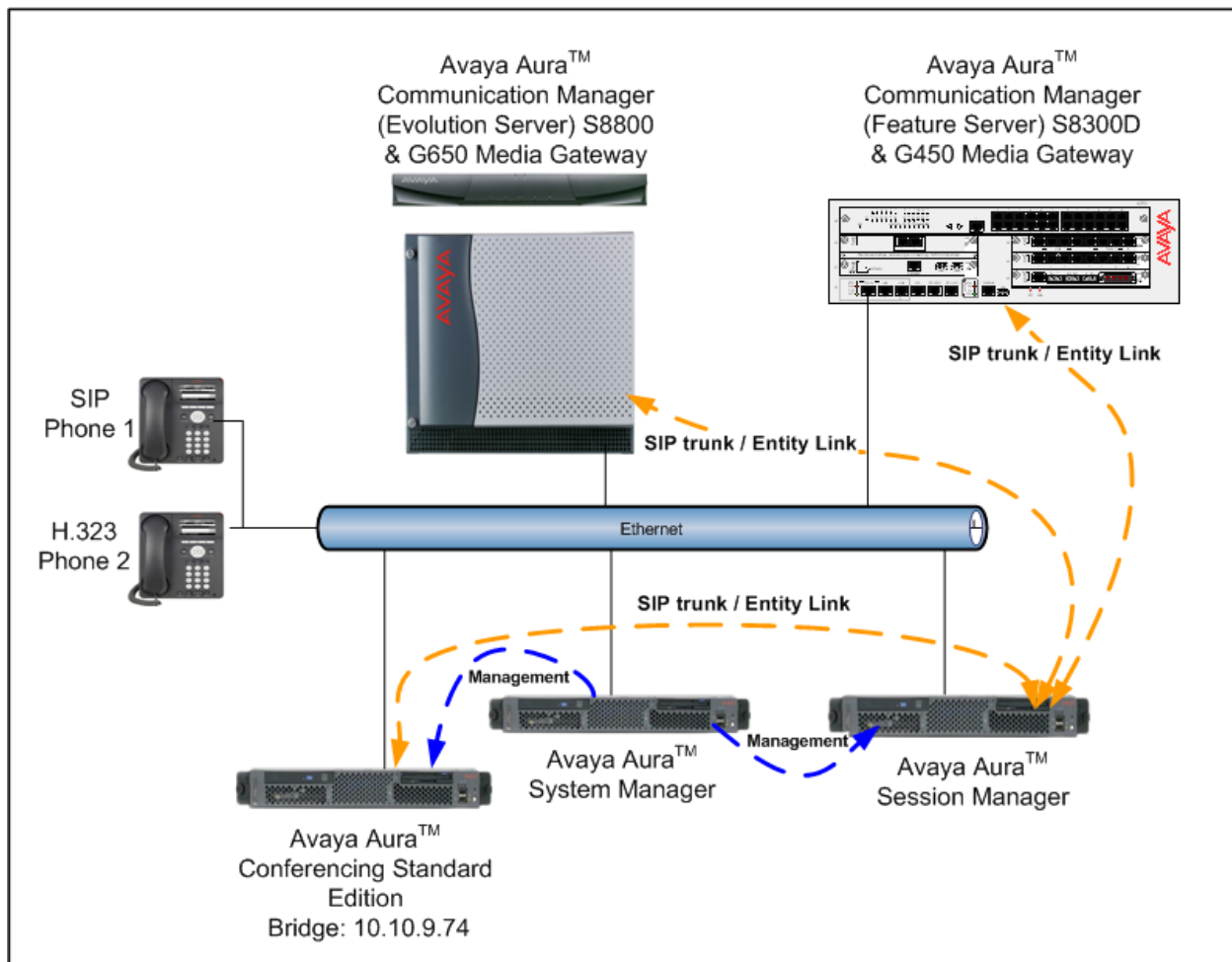


Figure 1 – Test Configuration used in these Application Notes

2. Equipment and Software Validated

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

Equipment	Software
S8800 Server	Avaya Aura™ Communication Manager 6.0 SP 0 (Load 345, Patch 18246)
Avaya G650 Media Gateway - CLAN - TN799DP - MedPro - TN2602AP	HW01 FW034 HW02 FW049
Avaya S8300D Server	Avaya Aura™ Communication Manager 6.0 SP 0 (Load 345, Patch 18246)
Avaya G450 Media Gateway	Firmware 30.11.3
IBM 3550	Avaya Aura™ Session Manager 6.0, Load 600020
IBM 3550	Avaya Aura™ System Manager 6.0, Load 600020
Avaya Aura™ Conferencing Standard Edition server, IBM 3550	Avaya Aura™ Conferencing Standard Edition Server 6.0.0.0.262 + Release Patches
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.

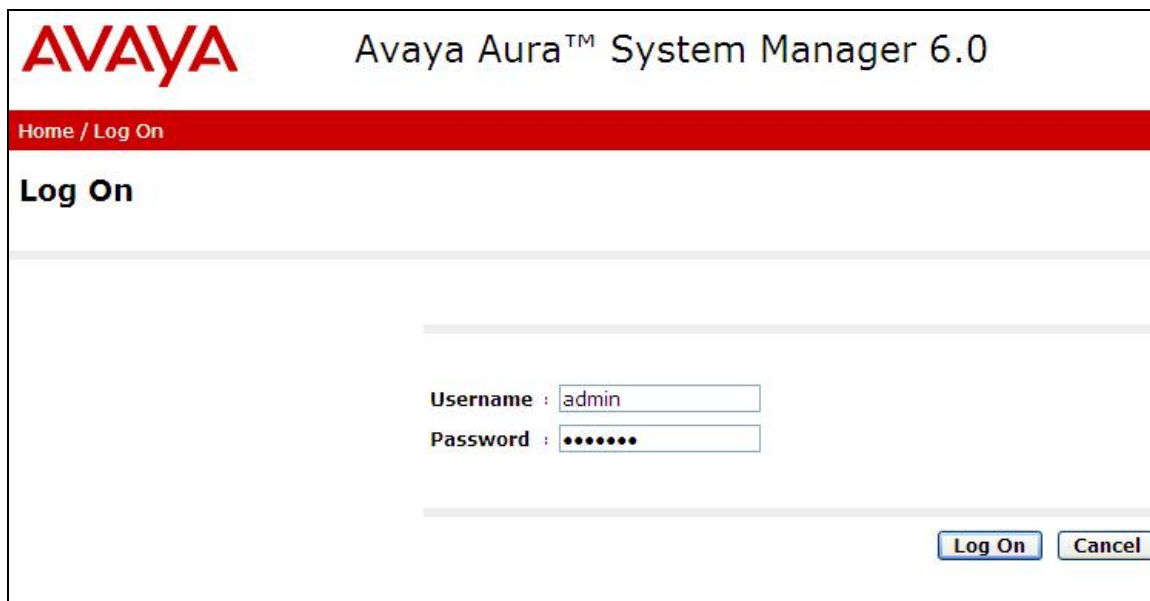
3. Configure Avaya Aura™ Conferencing Standard Edition

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

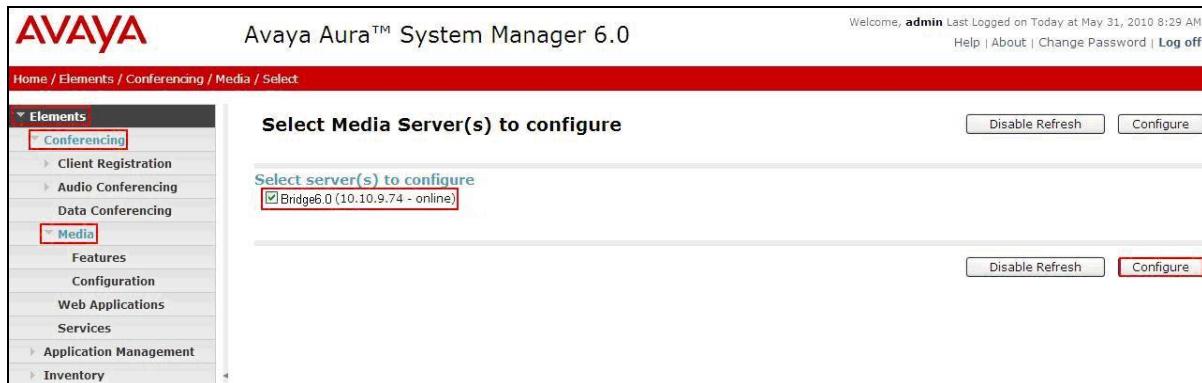
Access the System Manager using a Web Browser and entering *https://<ip-address>/smgr*, where <ip-address> is the IP address of System Manager. Log in using appropriate credentials and accept the subsequent Copyright Legal Notice.



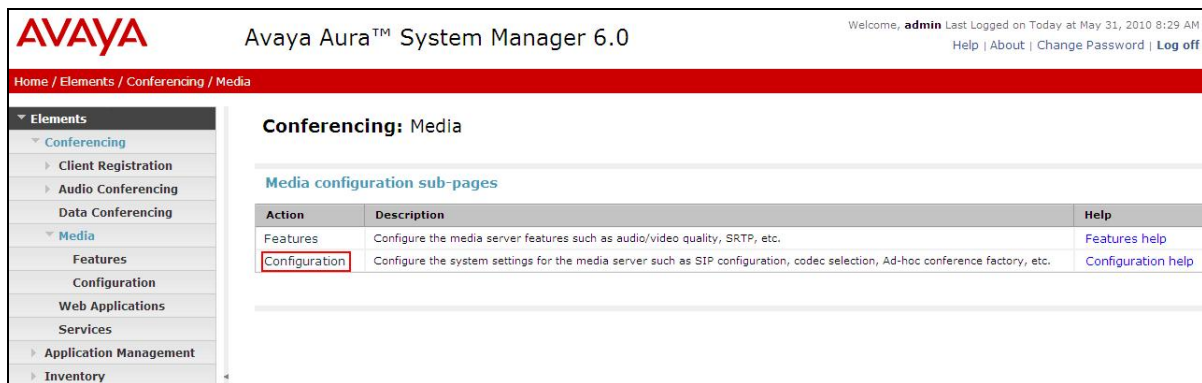
The screenshot shows the Avaya Aura™ System Manager 6.0 login interface. At the top left is the AVAYA logo in red. To its right is the text "Avaya Aura™ System Manager 6.0". Below this is a red navigation bar with the text "Home / Log On" in white. Underneath the navigation bar is a section titled "Log On" in bold black text. The main area contains a login form with two fields: "Username : admin" and "Password : " followed by a series of dots. At the bottom right of the form are two buttons: "Log On" and "Cancel".

3.2. Configuring SIP Connectivity

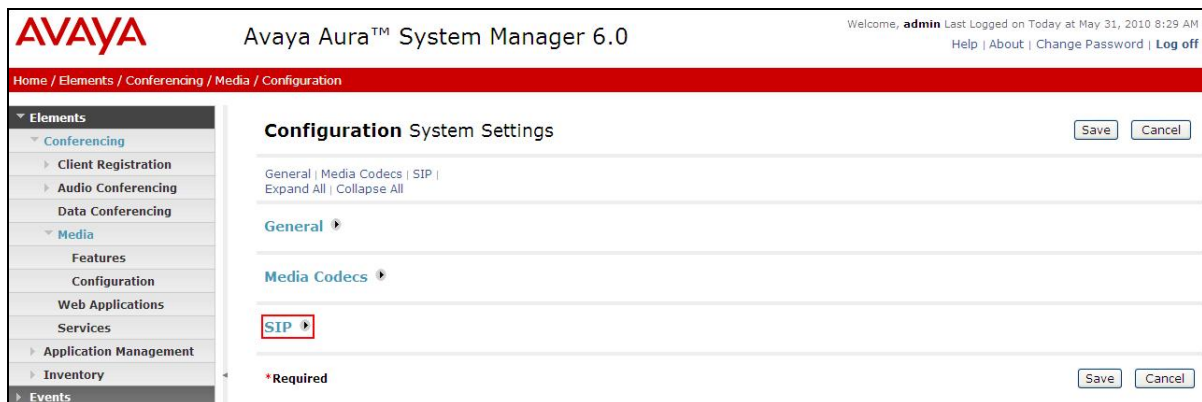
Configure settings that enable SIP connectivity between the Conferencing bridge and other devices by configuring the SIP System Settings. Select **Elements** → **Conferencing** → **Media** on the left panel menu. From the right panel menu select the media server to configure by selecting the tick box 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 Conferencing Standard Edition:
- **Session Refresh Timer** 1800
- **Min Session Refresh Timer Allowed** 1800

Click the **Save** button.

The screenshot displays the Avaya Aura™ System Manager 6.0 interface. The top header shows the Avaya logo, the system name, and a welcome message for the 'admin' user. The breadcrumb trail indicates the current location: Home / Elements / Conferencing / Media / Configuration. The left sidebar lists various system elements, with 'Media' expanded to show 'Configuration'. The main content area is titled 'Configuration System Settings' and includes tabs for General, Media Codecs, and SIP. The SIP tab is active, showing four configuration fields: 'SIP Listener URI' and 'Response Contact' (both set to <sip:6000@10.10.9.74:5060;transport=tcp>), 'Session Refresh Timer' (set to 1800), and 'Min Session Refresh Timer Allowed' (set to 1800). A 'Save' button is visible at the bottom right of the configuration area.

3.3. Configure Dialout

To enable Dial-Out from the Conferencing 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**.

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

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

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 / 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 *

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 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@135.64.186.40:5060;transport=tcp**
To route outbound calls from the Conferencing bridge to the Session Manager.
- **Comment** A descriptive comment

Click the **Save** button.

From the right panel menu select **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 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**.

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
<input checked="" type="checkbox"/> 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 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** 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.

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 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** 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** select the following parameters, leaving the remaining parameters at their default values.

Under **Call Branding Details**

- **DDI** 7111
- **Name** A descriptive name
- **Organization 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. Configure Avaya Aura™ Communication Manager as Evolution Server

This section describes the steps for configuring the Communication Manager as an Evolution Server. All configurations in the section are administered using the System Access Terminal (SAT). These Application Notes assume that the basic Communication Manager configuration has already been administered. The procedures include the following areas:

- Confirm Necessary Features
- Administer IP Node Names
- Administer IP Network Region
- Administer IP Codec Set
- Administer SIP Signaling Group
- Administer SIP Trunk Group
- Administer Private Numbering
- Administer Route patterns
- Administer Uniform Dialplan
- Administer AAR
- Save Translations

Throughout this section the administration of Communication Manager is performed using a System Access Terminal (SAT). The following commands are entered on the system with the appropriate administrative permissions. Some administration screens have been abbreviated for clarity. These instructions assume that the Communication Manager has been installed, configured, licensed and provided with a functional dial plan. Refer to the appropriate documentation as described in reference [5] for more details.

4.1. Confirm Necessary Features

The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity. Log into the Communication Manager SAT interface and use the **display system-parameters customer-options** command to determine these values. On **Page 2** verify that the available **Maximum Administered SIP Trunks** is equal to or greater than the desired number of simultaneous SIP trunk connections.

display system-parameters customer-options		Page	2 of 11
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
Maximum Administered H.323 Trunks:		12000	0
Maximum Concurrently Registered IP Stations:		18000	7
Maximum Administered Remote Office Trunks:		12000	0
Maximum Concurrently Registered Remote Office Stations:		18000	0
Maximum Concurrently Registered IP eCons:		414	0
Max Concur Registered Unauthenticated H.323 Stations:		100	0
Maximum Video Capable Stations:		18000	1
Maximum Video Capable IP Softphones:		18000	1
Maximum Administered SIP Trunks:		24000	30
Maximum Administered Ad-hoc Video Conferencing Ports:		24000	0
Maximum Number of DS1 Boards with Echo Cancellation:		522	0
Maximum TN2501 VAL Boards:		128	1
Maximum Media Gateway VAL Sources:		250	0
Maximum TN2602 Boards with 80 VoIP Channels:		128	0
Maximum TN2602 Boards with 320 VoIP Channels:		128	2
Maximum Number of Expanded Meet-me Conference Ports:		300	0

On **Page 4** verify the field **IP Trunks** is set to **y**.

display system-parameters customer-options		Page	4 of 11
OPTIONAL FEATURES			
Emergency Access to Attendant? y	IP Stations? y		
Enable 'dadmin' Login? y			
Enhanced Conferencing? y	ISDN Feature Plus? n		
Enhanced EC500? y	ISDN/SIP Network Call Redirection? y		
Enterprise Survivable Server? n	ISDN-BRI Trunks? y		
Enterprise Wide Licensing? n	ISDN-PRI? y		
ESS Administration? y	Local Survivable Processor? n		
Extended Cvg/Fwd Admin? y	Malicious Call Trace? y		
External Device Alarm Admin? y	Media Encryption Over IP? n		
Five Port Networks Max Per MCC? n	Mode Code for Centralized Voice Mail? n		
Flexible Billing? n			
Forced Entry of Account Codes? y	Multifrequency Signaling? y		
Global Call Classification? y	Multimedia Call Handling (Basic)? y		
Hospitality (Basic)? y	Multimedia Call Handling (Enhanced)? y		
Hospitality (G3V3 Enhancements)? y	Multimedia IP SIP Trunking? y		
IP Trunks? y			
IP Attendant Consoles? y			

4.2. Administer IP Node Names

Use the **change node-names ip** command to add entries for the Communication Manager and Session Manager that will be used for connectivity. In the sample network, **clan1a3** and **135.64.186.6** are entered as **Name** and **IP Address** for the CLAN card in Communication Manager running on the Avaya S8800 Server. In addition, **SessionM1** and **135.64.186.40** are entered for Session Manager.

change node-names ip		IP NODE NAMES
Name	IP Address	
clan1a3	135.64.186.6	
GatewaySub3	135.64.186.65	
mprola2	135.64.186.8	
mprolb2	135.64.186.9	
SessionM1	135.64.186.40	
procr	135.64.186.70	

4.3. Administer IP Network Region

Use the **change ip-network-region n** command, where **n** is the network region number to configure the network region being used. In the sample network ip-network-region **1** is used. For the **Authoritative Domain** field, enter the SIP domain name configured for this enterprise and a descriptive **Name** for this ip-network-region. Set **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** to **yes** to allow for direct media between endpoints. Set the **Codec Set** to **1** to use ip-codec-set 1. See the following section.

change ip-network-region 1		Page 1 of 20
IP NETWORK REGION		
Region: 1		
Location: 1	Authoritative Domain: silstack.com	
Name: Dublin		
MEDIA PARAMETERS	Intra-region IP-IP Direct Audio: yes	
Codec Set: 1	Inter-region IP-IP Direct Audio: yes	
UDP Port Min: 2048	IP Audio Hairpinning? n	
UDP Port Max: 3329		
DIFFSERV/TOS PARAMETERS		
Call Control PHB Value: 46		
Audio PHB Value: 46		
Video PHB Value: 26		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 6		
Audio 802.1p Priority: 6		
Video 802.1p Priority: 5	AUDIO RESOURCE RESERVATION PARAMETERS	
H.323 IP ENDPOINTS	RSVP Enabled? n	
H.323 Link Bounce Recovery? y		
Idle Traffic Interval (sec): 20		
Keep-Alive Interval (sec): 5		
Keep-Alive Count: 5		

4.4. Administer IP Codec Sets

Use the **change ip-codec-set n** command, where **n** is the codec set specified in the **IP Network Region** form. Enter the codecs eligible to be used; the codecs defined here must be supported by the far end device.

change ip-codec-set 1				Page	1 of	2
IP Codec Set						
Codec Set: 1						
Audio	Silence	Frames	Packet			
Codec	Suppression	Per Pkt	Size(ms)			
1: G.711MU	n	2	20			
2: G.711A	n	2	20			
3:						
4:						
5:						

4.5. Administer SIP Signaling Group

Use the **add signaling-group n** command, where **n** is the number of the SIP signaling-group to create. Set the **Group Type** field to be **sip** and **Transport Method** to **tcp**. Use the values defined in **Section 4.2** and **4.3** for **Near-end Node Name**, **Far-End Node-Name** and **Far-End Network Region**. The **Far-end Domain** is left blank so that the signaling group accepts any authoritative domain.

add signaling-group 120		Page	1 of	1
SIGNALING GROUP				
Group Number: 120		Group Type: sip		
IMS Enabled? n		Transport Method: tcp		
Q-SIP? n		SIP Enabled LSP? n		
IP Video? n		Enforce SIPS URI for SRTP? y		
Peer Detection Enabled? y		Peer Server: SM		
Near-end Node Name: clan1a3		Far-end Node Name: SessionM1		
Near-end Listen Port: 5060		Far-end Listen Port: 5060		
		Far-end Network Region: 1		
Far-end Domain:				
Incoming Dialog Loopbacks: eliminate		Bypass If IP Threshold Exceeded? n		
DTMF over IP: rtp-payload		RFC 3389 Comfort Noise? n		
Session Establishment Timer(min): 3		Direct IP-IP Audio Connections? y		
Enable Layer 3 Test? n		IP Audio Hairpinning? n		
H.323 Station Outgoing Direct Media? n		Initial IP-IP Direct Media? n		
		Alternate Route Timer(sec): 6		

4.6. Administer SIP Trunk Group

Use the **add trunk-group n** command where **n** is the new trunk group number being added to the system. The following screens show the settings used for trunk group 120. Set the **Group Type** field to be **sip**.

add trunk-group 120		Page 1 of 21	
TRUNK GROUP			
Group Number: 120	Group Type: sip	CDR Reports: y	
Group Name: TO ASM	COR: 1	TN: 1	TAC: 120
Direction: two-way	Outgoing Display? y	Night Service:	
Dial Access? n			
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Signaling Group: 120	
		Number of Members: 20	

Navigate to **Page 3** and enter **private** for **Numbering Format**.

add trunk-group 120		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: private		UI Treatment: service-provider	
		Replace Restricted Numbers? n	
		Replace Unavailable Numbers? N	
Modify Tandem Calling Number: no			
Show ANSWERED BY on Display? y			

4.7. Administer Private Numbering

Use the **change private-numbering** command to define the calling party number to be sent out through the SIP trunk. All calls originating from a **5**-digit extension beginning with **2** will result in a 5-digit calling number. The calling party number will be in the SIP “From” header.

change private-numbering 0					Page 1 of 2
NUMBERING - PRIVATE FORMAT					
Ext	Ext	Trk	Private	Total	
Len	Code	Grp(s)	Prefix	Len	
5	2	120		5	Total Administered: 1
					Maximum Entries: 540

4.8. Administer Route Patterns

Configure a route pattern to correspond to the newly added SIP trunk group. Use the **change route-pattern n** command, where **n** is the route pattern number specified in **Section 4.10**. Configure this route pattern to route calls to trunk group number **120** configured in **Section 4.6**. Assign the lowest **FRL** (facility restriction level) to allow all callers to use this route pattern.

change route-pattern 120													Page 1 of 3	
Pattern Number: 120													Pattern Name: To ASM	
SCCAN? n													Secure SIP? n	
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted						DCS/	IXC
No			Mrk	Lmt	List	Del	Digits						QSIG	
													Intw	
1:	120	0											n	user
2:												n	user	
3:												n	user	
4:												n	user	
5:												n	user	
6:												n	user	
		BCC VALUE		TSC	CA-TSC		ITC BCIE Service/Feature PARM				No.	Numbering	LAR	
		0	1	2	M	4	W	Request						
													Dgts Format	
													Subaddress	
1:	y	y	y	y	y	n	n	rest				next		
2:	y	y	y	y	y	n	n	rest				none		
3:	y	y	y	y	y	n	n	rest				none		
4:	y	y	y	y	y	n	n	rest				none		
5:	y	y	y	y	y	n	n	rest				none		
6:	y	y	y	y	y	n	n	rest				none		

4.9. Administer Uniform Dialplan

Use the **change uniform-dialplan n** command to administer the uniform dialplan. In this configuration extension **7111** is configured as **aar** to send calls via the aar analysis table.

change uniform-dialplan 7						Page	1 of	2
UNIFORM DIAL PLAN TABLE						Percent Full: 0		
Matching			Insert					
Pattern	Len	Del	Digits	Net	Conv	Num		
7111	4	0		aar	n			

4.10. Administer AAR

Use the **change aar analysis n** command to specify which route pattern to use based upon the number dialed. In this example, **Route Pattern 120** is used for extension **7111**.

change aar analysis 7						Page	1 of	2
AAR DIGIT ANALYSIS TABLE						Percent Full: 1		
Location: all								
Dialed	Total		Route	Call	Node	ANI		
String	Min	Max	Pattern	Type	Num	Reqd		
7111	4	4	120	aar		n		

4.11. Save Translations

Configuration of Communication Manager is complete. Use the **save translation** command to save these changes.

5. Configure Avaya Aura™ Communication Manager as Feature Server

This section describes the steps for configuring the Communication Manager as a Feature Server. All configurations in the section are administered using the System Access Terminal (SAT). These Application Notes assume that the basic Communication Manager configuration has already been administered. The procedures include the following areas:

- Administer IP Node Names
- Administer IP Network Region
- Administer IP Codec Set
- Administer SIP Signaling Group
- Administer SIP Trunk Group
- Administer Private Numbering
- Administer Route patterns
- Administer Uniform Dialplan
- Administer AAR
- Save Translations

Throughout this section the administration of Communication Manager is performed using a System Access Terminal (SAT). The following commands are entered on the system with the appropriate administrative permissions. Some administration screens have been abbreviated for clarity. These instructions assume that the Communication Manager has been installed, configured, licensed and provided with a functional dial plan. Refer to the appropriate documentation as described in reference [5] and [6] for more details.

5.1. Administer IP Node Names

Use the **change node-names ip** command to add entries for the Session Manager that will be used for connectivity. In addition, **SM** and **135.64.186.40** are entered for Session Manager. Also note the IP Address of the **procr** and **135.64.186.55** as this will be used to configure the SIP signaling group.

change node-names ip		IP NODE NAMES
Name	IP Address	
SILStackAES	135.64.186.28	
SM	135.64.186.40	
default	0.0.0.0	
fs-acm	135.64.186.55	
procr	135.64.186.55	

5.2. Administer IP Network Region

Use the **change ip-network-region n** command, where **n** is the network region number to configure the network region being used. In the sample network ip-network-region 1 is used. For the **Authoritative Domain** field, enter the SIP domain name configured for this enterprise and a descriptive **Name** for this ip-network-region. Set **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** to **yes** to allow for direct media between endpoints. Set the **Codec Set** to **1** to use ip-codec-set 1. See the following section.

change ip-network-region 1		Page 1 of 20
IP NETWORK REGION		
Region: 1		
Location: 1 Authoritative Domain: silstack.com		
Name: To ASM		
MEDIA PARAMETERS		
Codec Set: 1 Intra-region IP-IP Direct Audio: yes		
Inter-region IP-IP Direct Audio: yes		
UDP Port Min: 2048 IP Audio Hairpinning? n		
UDP Port Max: 3329		
DIFFSERV/TOS PARAMETERS		
Call Control PHB Value: 46		
Audio PHB Value: 46		
Video PHB Value: 26		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 6		
Audio 802.1p Priority: 6		
Video 802.1p Priority: 5		
AUDIO RESOURCE RESERVATION PARAMETERS		
H.323 IP ENDPOINTS		
RSVP Enabled? n		
H.323 Link Bounce Recovery? y		
Idle Traffic Interval (sec): 20		
Keep-Alive Interval (sec): 5		
Keep-Alive Count: 5		

5.3. Administer IP codec sets

Use the **change ip-codec-set n** command, where **n** is the codec set specified in the IP Network Region form. Enter the codecs eligible to be used. The codecs defined here must be supported by the far end device.

change ip-codec-set 1		Page 1 of 2	
IP Codec Set			
Codec Set: 1			
Audio	Silence	Frames	Packet
Codec	Suppression	Per Pkt	Size(ms)
1: G.711MU	n	2	20
2: G.711A	n	2	20
3:			
4:			
5:			

5.4. Administer SIP Signaling Group

Use the **add signaling-group n** command, where **n** is the number of the SIP signaling-group to create. Set the **Group Type** field to be **sip** and **Transport Method** to **tcp**. Set the **IMS Enabled** field to **y**. Use the values defined in **Section 5.2** and **5.3** for **Near-end Node Name**, **Far-End Node-Name** and **Far-End Network Region**. The **Far-end Domain** is left blank so that the signaling group accepts any authoritative domain.

add signaling-group 150		Page 1 of 1
SIGNALING GROUP		
Group Number: 150	Group Type: sip	
IMS Enabled? y	Transport Method: tcp	
Q-SIP? n	SIP Enabled LSP? n	
IP Video? n		
Peer Detection Enabled? y	Peer Server: SM	
Near-end Node Name: procr	Far-end Node Name: SM	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 1	
Far-end Domain:		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n	
	Alternate Route Timer(sec): 6	

5.5. Administer SIP Trunk Group

Use the **add trunk-group n** command where **n** is the new trunk group number being added to the system. The following screens show the settings used for trunk group 150. Set the **Group Type** field to be **sip**.

add trunk-group 150		Page 1 of 21	
TRUNK GROUP			
Group Number: 150	Group Type: sip	CDR Reports: y	
Group Name: To ASM SM100	COR: 1	TN: 1	TAC: 150
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Signaling Group: 150	
		Number of Members: 10	

On **Page 3** of the trunk-group form set the **Numbering Format** field to **private**.

add trunk-group 150		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: private		UUI Treatment: service-provider	
		Replace Restricted Numbers? n	
		Replace Unavailable Numbers? n	
Modify Tandem Calling Number: no			
Show ANSWERED BY on Display? y			
DSN Term? n			

5.6. Administer Private Number

Use the **change private-numbering** command to define the calling party number to be sent out through the SIP trunk. All calls originating from a **5**-digit extension beginning with **4** will result in a 5-digit calling number. The calling party number will be in the SIP “From” header.

change private-numbering 0					Page 1 of 2
NUMBERING - PRIVATE FORMAT					
Ext	Ext	Trk	Private	Total	
Len	Code	Grp(s)	Prefix	Len	
5	4	150		5	Total Administered: 1
					Maximum Entries: 540

5.7. Administer Route Patterns

Configure a route pattern to correspond to the newly added SIP trunk group. Use the **change route-pattern n** command, where **n** is the route pattern number specified in **Section 5.8**. Configure this route pattern to route calls to trunk group number **150** configured in **Section 5.5**. Assign the lowest **FRL** (facility restriction level) to allow all callers to use this route pattern.

change route-pattern 150										Page 1 of 3	
Pattern Number: 150										Pattern Name: FeatureCM	
SCCAN? n										Secure SIP? n	
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/	IXC		
No			Mrk	Lmt	List	Del	Digits	QSIG			
								Intw			
1:	150	0							n	user	
2:										n	user
3:										n	user
4:										n	user
5:										n	user
6:										n	user
BCC VALUE		TSC	CA-TSC	ITC BCIE		Service/Feature		PARM	No.	Numbering	LAR
0	1	2	M	4	W	Request		Dgts		Format	
										Subaddress	
1:	y	y	y	y	y	n	n	rest	next		
2:	y	y	y	y	y	n	n	rest	none		
3:	y	y	y	y	y	n	n	rest	none		
4:	y	y	y	y	y	n	n	rest	none		
5:	y	y	y	y	y	n	n	rest	none		
6:	y	y	y	y	y	n	n	rest	none		

5.8. Administer Uniform Dialplan

Use the **change uniform-dialplan n** command to administer the uniform dialplan. In this configuration extension **7111** is configured as **aar** to send calls via the aar analysis table.

change uniform-dialplan 7						Page	1 of	2
UNIFORM DIAL PLAN TABLE						Percent Full: 0		
Matching	Len Del		Insert	Node				
Pattern			Digits	Net	Conv	Num		
7111	4	0		aar	n			

5.9. Administer AAR

Use the **change aar analysis n** command to specify which route pattern to use based upon the number dialed. In this example, **Route Pattern 150** is used for extension **7111**.

change aar analysis 7						Page	1 of	2
AAR DIGIT ANALYSIS TABLE						Percent Full: 1		
Location: all								
Dialed	Total		Route	Call	Node	ANI		
String	Min	Max	Pattern	Type	Num	Reqd		
7111	4	4	150	aar		n		

5.10. Save Translations

Configuration of Communication Manager is complete. Use the **save translation** command to save these changes.

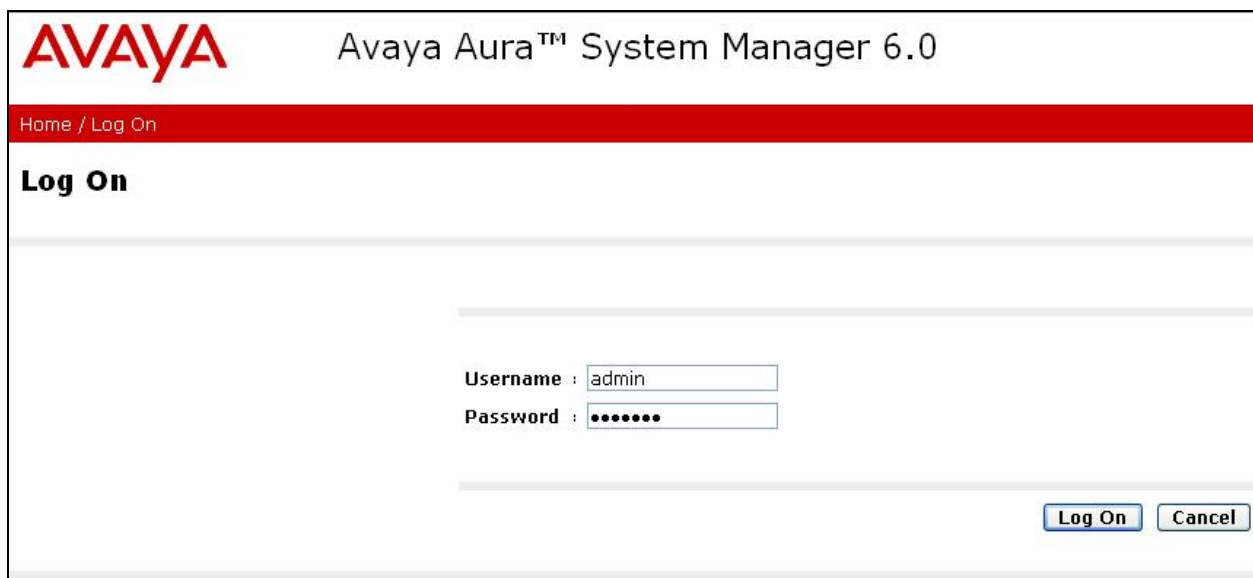
6. Configure Avaya Aura™ Session Manager

This section provides the procedures for configuring Session Manager. The 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

6.1. Log in to Avaya Aura™ System Manager

Access the Avaya Aura™ System Manager using a Web Browser and entering **http://<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 : admin

Password : ••••••

Log On Cancel

6.2. 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.

The screenshot displays the Avaya Aura™ System Manager 6.0 web interface. The top header includes the Avaya logo, the product name and version, and a welcome message for the 'admin' user. A red navigation bar contains the breadcrumb 'Home / Routing / Domains'. On the left, a sidebar menu lists various system components, with 'Routing' and its sub-item 'Domains' highlighted. The main content area, titled 'Domain Management', features action buttons (Edit, New, Duplicate, Delete, More Actions) and a table listing domain entries. A single entry for 'silstack.com' is shown with type 'sip'. Below the table is a 'Select' dropdown menu.

	Name	Type	Default	Notes
<input type="checkbox"/>	silstack.com	sip	<input type="checkbox"/>	

6.3. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management. Location is added to the configuration for both Communication Manager 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
- **Notes:** Descriptive text (optional)

Under **Location Pattern**:

- **IP Address Pattern:** A pattern used to logically identify the location
- **Notes:** Descriptive text (optional)

Click **Commit** to save changes.

Avaya Aura™ System Manager 6.0

Welcome, admin Last Logged on at June 1, 2010 12:21 PM

Help | Change Password | Log off

Home / Routing / Locations / Location Details

Location Details

General

* Name: Dublin Stack

Notes:

Managed Bandwidth:

* Average Bandwidth per Call: 80 Kbit/sec

Location Pattern

Add Remove

2 Items Refresh Filter: Enable

	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.10.9.*	
<input type="checkbox"/>	* 135.64.186.*	

6.4. 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 Entity is added for the Session Manager, the C-LAN board in the Avaya G650 Media Gateway, the procr in the Avaya G450 Media Gateway and the Conferencing bridge. To add a SIP Element, select **Routing** → **SIP Element** 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 ASM or the signaling interface on the telephony system.
- **Type:** Select between **Session Manager** for Session Manager, **CM** for Communication Manager and **SIP Trunk** for Conferencing server
- **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 Session Manager’s SIP routing interface.

AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at April 28, 2010 2:06 PM
Help | Change Password | Log off

Home / Routing / SIP Elements / SIP Elements Details

SIP Element Details [Commit] [Cancel]

General

* Name: SessionManager

* FQDN or IP Address: 135.64.186.40

Type: Session Manager

Notes:

Location: Dublin Stack

Outbound Proxy:

Time Zone: Europe/Dublin

Credential name:

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

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 used to listen to SIP requests.
- **Default Domain** The domain used for the enterprise (e.g., **silstack.com**).

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

Port

Add Remove

4 Items | Refresh Filter: Enable

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input checked="" type="checkbox"/>	5060	TCP	silstack.com	

The following screen shows addition of Evolution Server. 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 C-LAN board.
- **Type:** Select **CM**
- **Location:** Select one of the locations defined previously.

Click **Commit** to save changes.

AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at April 28, 2010 2:06 PM
Help | Change Password | Log off

Home / Routing / SIP Elements / SIP Elements Details

SIP Element Details [Commit] [Cancel]

General

* Name: Enterprise Evolution CM

* FQDN or IP Address: 135.64.186.6

Type: CM

Notes:

Adaptation:

Location: Dublin Stack

Time Zone: Europe/Paris

Override Port & Transport with DNS SRV: ☐

* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: none

The following screen shows addition of Feature Server. 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 procr
- **Type:** Select **CM**
- **Location:** Select one of the locations defined previously.

Click **Commit** to save changes.

The screenshot displays the Avaya Aura™ System Manager 6.0 web interface. The top header includes the Avaya logo, the product name and version, and a user status bar indicating 'Welcome, admin' and 'Last Logged on at April 28, 2010 2:06 PM'. A navigation bar shows the path 'Home / Routing / SIP Elements / SIP Elements Details'. On the left, a sidebar menu lists various system components, with 'Routing' expanded and 'SIP Elements' selected. The main content area is titled 'SIP Element Details' and features a 'General' tab. The form contains several fields: 'Name' (set to 'FeatureServer'), 'FQDN or IP Address' (set to '135.64.186.55'), 'Type' (set to 'CM'), 'Notes' (empty), 'Adaptation' (dropdown), 'Location' (dropdown), 'Time Zone' (set to 'Europe/Dublin'), 'Override Port & Transport with DNS SRV' (checkbox), '* SIP Timer B/F (in seconds)' (set to '4'), 'Credential name' (empty), and 'Call Detail Recording' (set to 'none'). 'Commit' and 'Cancel' buttons are located at the top right of the form area.

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**
- **Location:** Select one of the locations defined previously.

Click **Commit** to save changes.

The screenshot displays the Avaya Aura System Manager 6.0 web interface. The top navigation bar includes the Avaya logo, the product name 'Avaya Aura™ System Manager 6.0', and user information: 'Welcome, admin Last Logged on at June 1, 2010 12:21 PM'. A secondary bar contains links for 'Help', 'Change Password', and 'Log off'. The main content area is titled 'SIP Element Details' and features a 'General' tab. The 'General' tab contains several configuration fields: 'Name' (set to 'Bridge_6.0'), 'FQDN or IP Address' (set to '10.10.9.74'), 'Type' (set to 'SIP Trunk'), 'Notes' (set to 'Bridge Conferencing 6.0'), 'Adaptation' (a dropdown menu), 'Location' (set to 'Dublin Stack'), and 'Time Zone' (set to 'Europe/Dublin'). There are also checkboxes for 'Override Port & Transport with DNS SRV' and 'SIP Timer B/F (in seconds)' (set to 4), a 'Credential name' field, and a 'Call Detail Recording' dropdown (set to 'both'). At the bottom, there is a 'SIP Link Monitoring' section with a dropdown set to 'Use Session Manager Configuration'. A left sidebar provides navigation options, with 'SIP Elements' highlighted. A 'Commit' button is located in the top right corner of the main content area.

6.5. Add Element Links

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 6.4**
- **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.

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at April 28, 2010 2:05 PM

Help | Change Password | Log off

Home / Routing / Element Links

Element Links

Edit New Duplicate Delete More Actions Commit

11 Items Refresh Filter: Enable

<input type="checkbox"/>	Name	SIP Element 1	Protocol	Port	SIP Element 2	Port	Trusted
<input type="checkbox"/>	AudioCodesM2K	SessionManager	TCP	5060	AudioCodesM2K	5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	Evolution CM	SessionManager	TCP	5060	Enterprise Evolution CM	5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	IMG1010	SessionManager	TCP	5060	IMG1010	5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	MX52	SessionManager	TCP	5060	MX52	5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	Bridge_6.0	SessionManager	TCP	5060	MX6.0	5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	MX6.0	SessionManager	TCP	5060	MXStack6.0	5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	PresenceElementLink	SessionManager	TLS	5061	PresenceServer	5061	<input checked="" type="checkbox"/>
<input type="checkbox"/>	SessionManager_FeatureServer_5065_TLS	SessionManager	TLS	5065	FeatureServer	5065	<input checked="" type="checkbox"/>
<input type="checkbox"/>	SM_OCS	SessionManager	TCP	5060	OCS	5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	To Branch	SessionManager	TCP	5060	Branch CM	5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	To Voice Mail	SessionManager	TCP	5060	VoiceMail	5060	<input checked="" type="checkbox"/>

6.6. 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.

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at April 28, 2010 2:06 PM

Help | | Change Password | Log off

Home / Routing / Time Ranges

Time Ranges

Edit New Duplicate Delete More Actions Commit

2 Items | Refresh Filter: Enable

<input type="checkbox"/>	Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes
<input type="checkbox"/>	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7
<input type="checkbox"/>	Always	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	

Select : All, None

6.7. 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**)

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. Click **Commit** to save changes. The following screen shows the **Routing Policy Details** for the Conferencing server.

The screenshot displays the Avaya Aura System Manager 6.0 interface. The left sidebar shows a navigation menu with 'Routing' expanded and 'Policies' selected. The main content area is titled 'Routing Policy Details' and includes a 'Commit' button. The 'General' section shows the policy name 'Bridge 6.0' and a 'Disabled' checkbox. The 'SIP Element as Destination' section shows a 'Select' button and a table with one entry: 'Bridge_6.0' with FQDN '10.10.9.74' and Type 'SIP Trunk'. The 'Time of Day' section shows an 'Add' button and a table with one entry: '24/7' with a time range of '00:00' to '23:59'.

Avaya Aura™ System Manager 6.0

Welcome, admin Last Logged on at June 1, 2010 12:21 PM

Help | Change Password | Log off

Home / Routing / Policies / Policy Details

Routing Policy Details

Commit Cancel

General

* Name: Bridge 6.0

Disabled: ☐

Notes:

SIP Element as Destination

Select

Name	FQDN or IP Address	Type	Notes
Bridge_6.0	10.10.9.74	SIP Trunk	Bridge Conferencing 6.0

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

6.8. 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 the Conferencing server. 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**.

The screenshot shows the Avaya Aura System Manager 6.0 interface. The left sidebar contains a navigation menu with 'Routing' expanded and 'Dial Patterns' selected. The main area displays the 'Dial Pattern Details' form. The 'General' tab is active, showing fields for Pattern (7111), Min (4), Max (4), SIP Domain (-ALL-), and Notes. The 'Originating Locations and Routing Policies' section shows an 'Add' button and a table with one item.

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes



Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at June 1, 2010 12:21 PM

[Home](#) / [Routing](#) / [Dial Patterns](#) / [Dial Pattern Details](#) / [Locations and Policy List](#)

Elements

Events

Groups & Roles

Licenses

Routing

Domains

Locations

Adaptations

SIP Elements

Element Links

Time Ranges

Policies

Dial Patterns

Regular Expressions

Defaults

Security

System Manager Data

Users

Help

Originating Location and Routing Policy List

Select

Cancel

Originating Location

2 Items | Refresh

Filter: Enable

<input type="checkbox"/>	Name	Notes
<input checked="" type="checkbox"/>	-ALL-	Any Locations
<input type="checkbox"/>	Dublin Stack	

Select : All, None

Routing Policies

13 Items | Refresh

Filter: Enable

<input type="checkbox"/>	Name	Disabled	Destination	Notes
<input type="checkbox"/>	AudioCodesM2K	<input type="checkbox"/>	AudioCodesM2K	
<input type="checkbox"/>	Branch CM	<input type="checkbox"/>	Branch CM	
<input checked="" type="checkbox"/>	Bridge 6.0	<input type="checkbox"/>	Bridge_6.0	

6.9. 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 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 Session Manager's SIP routing interface
- **Network Mask** Enter the network mask corresponding to the IP address of Session Manager's SIP routing interface.
- **Default Gateway:** Enter the IP address for the Session Manager's SIP routing interface.

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

AVAYA Avaya Aura™ System Manager 6.0 Welcome, admin Last Logged on at April 28, 2010 6:06 PM
Help | About | Change Password | Log off

Home / Elements / Session Manager / Session Manager Administration / Edit Session Manager

Elements

- Conferencing
- Presence
- Application Management
- Endpoints
- SIP AS 8.1
- Feature Management
- Inventory
- Templates
- Session Manager**
 - Dashboard
 - Session Manager**
 - Administration**
 - Communication Profile Editor
 - Network Configuration
 - Device and Location Configuration
 - Application Configuration
 - System Status

Add Session Manager Commit Cancel

General | Security Module | NIC Bonding | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server |
Expand All | Collapse All

General

SIP Entity Name SessionManager

Description Enterprise ASM 1

*Management Access Point Host Name/IP 135.64.186.39

*Direct Routing to Endpoints Enable

Security Module

SIP Entity IP Address 135.64.186.40

*Network Mask 255.255.255.224

*Default Gateway 135.64.186.33

*Call Control PHB 46

*QOS Priority 6

6.10. 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.

6.10.1. 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.

AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at April 29, 2010 9:07 AM
Help | About | Change Password | Log off

Home / Elements / Application Management / Applications / Applications Details

Elements

- Conferencing
- Presence
- Application Management
- Endpoints
- SIP AS 8.1
- Feature Management
- Inventory**
- Manage Elements
- Discovered Inventory
- Discovery Management
- Synchronization
- Templates
- Session Manager

New CM Instance [Commit] [Cancel]

Application | Port | Access Point | SNMP Attributes | Attributes |
Expand All | Collapse All

Application

* Name FeatureServer

* Type CM

Description

* Node 135.64.186.55

* Version ☒ None ☐ V1 ☐ V3

Attributes

* Login init

Password *****

Confirm Password *****

Is SSH Connection ☒

* Port 5022

Alternate IP Address

RSA SSH Fingerprint (Primary IP)

RSA SSH Fingerprint (Alternate IP)

Is ASG Enabled ☐

ASG Key

Confirm ASG Key

Location

* Required [Commit] [Cancel]

6.10.2. 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 6.4**
- **CM System for SIP Entity** Select the CM Application Entity defined in **Section 6.10.1**

Click on **Commit** to save.

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at June 2, 2010 11:25 AM
Help | About | Change Password | Log off

Home / Elements / Session Manager / Application Configuration / Application Editor

Application Editor [Commit] [Cancel]

Application Editor

Name: FeatureServer

* SIP Entity: FeatureServer

* CM System for SIP Entity: FeatureServer [Refresh] [View/Add CM Systems](#)

Description:

Application Attributes (optional)

Name	Value
Application Handle	
URI Parameters	

* Required [Commit] [Cancel]

6.10.3. 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. In this case the Sequence name is **App Sequence**. 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.

Home / Elements / Session Manager / Application Configuration / Application Sequence Editor

▼ Elements

► Conferencing

► Presence

► Application Management

► Endpoints

SIP AS 8.1

► Feature Management

► Inventory

► Templates

▼ Session Manager

Dashboard

Session Manager

Administration

Communication Profile

Editor

► Network Configuration

► Device and Location

Configuration

▼ Application Configuration

Applications

Application Sequences

Implicit Users

Application Sequence Editor

Commit Cancel

Sequence Name

Name App Sequence

Description

Applications in this Sequence

Move First Move Last Remove

1 Item

<input type="checkbox"/>	Sequence Order (first to last)	Name	SIP Entity	Mandatory	Description
<input type="checkbox"/>		FeatureServer	FeatureServer	<input checked="" type="checkbox"/>	

Select : All, None

Available Applications

1 Item Refresh Filter: Enable

	Name	SIP Entity	Description
+	FeatureServer	FeatureServer	

6.10.4. 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.

The screenshot displays the Avaya Aura System Manager 6.0 web interface. The left-hand navigation pane shows a tree structure with 'Elements' expanded, and 'Inventory' → 'Synchronization' → 'Communication System' selected. The main content area is titled 'Synchronize CM Data and Configure Options'. It includes a sub-header 'Synchronize CM Data/Launch Element Cut Through' and a table with 2 items. The table has columns for Element Name, FQDN/IP Address, Last Sync Time, Last Translation Time, Sync Type, Sync Status, Location, and Softphone. The 'FeatureServer' entry is selected with a checkmark. Below the table, there are radio button options: 'Initialize data for selected devices' (selected), 'Incremental Sync data for selected devices', and 'Save Translations for selected devices'. At the bottom, there are buttons for 'Now', 'Schedule', 'Cancel', and 'Launch Element Cut Through'.

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at June 1, 2010 7:54 PM
Help | About | Change Password | Log off

Home / Elements / Inventory / Synchronization / Communication System

Elements

- Conferencing
- Presence
- Application Management
- Endpoints
- SIP AS 8.1
- Feature Management
- Inventory**
 - Manage Elements
 - Discovered Inventory
 - Discovery Management
 - Synchronization**
 - Communication System**
 - Messaging System
 - Templates
 - Session Manager
- Events
- Groups & Roles
- Licenses
- Routing
- Security

Synchronize CM Data and Configure Options

Synchronize CM Data/Launch Element Cut Through | Configuration Options | Expand All | Collapse All

Synchronize CM Data/Launch Element Cut Through

2 Items | Refresh Filter: Enable

<input type="checkbox"/>	Element Name	FQDN/IP Address	Last Sync Time	Last Translation Time	Sync Type	Sync Status	Location	Softphone
<input type="checkbox"/>	CMES60	135.64.186.70	June 2, 2010 10:00:36 AM +01:00	10:00 pm TUE JUN 1, 2010	Incremental	Completed		R01
<input checked="" type="checkbox"/>	FeatureServer	135.64.186.55	June 2, 2010 10:00:27 AM +01:00	10:00 pm TUE JUN 1, 2010	Incremental	Completed		R01

Select : All, None

☒ Initialize data for selected devices
☐ Incremental Sync data for selected devices
☐ Save Translations for selected devices

Now Schedule Cancel Launch Element Cut Through

6.11. Add Users for SIP Phones

Users must be added via Session Manager and the details will be updated on the 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

AVAYA Avaya Aura™ System Manager 6.0 Welcome, **admin** Last Logged on at April 29, 2010 9:07 AM
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Users / Manage Users / User Edit

New User Profile Commit Cancel

General | Identity | Communication Profile | Roles | Override Permissions | Group Membership | Default Contact List | Private Contacts |
Expand All | Collapse All

General

* Last Name: Test

* First Name: System

Middle Name:

Description:

☐ Administrator
☐ Communication User
☐ Agent

User Type: ☐ Supervisor
☐ Resident Expert
☐ Service Technician
☐ Lobby Phone

Status: Offline

Update Time: April 28, 2010 4:04:3

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 6.2**
- **Password** Password for user to log into SMGR
- **Shared Communication Profile Password**
 Password to be entered by the user when logging into the phone

Identity ▾

* Login Name: 34002@silstack.com

* Authentication Type: Basic ▾

SMGR Login Password:

* Password: ••••••

* Confirm Password: ••••••

Shared Communication Profile Password: ••••••••

Confirm Password: ••••••••

Localized Display Name:

Endpoint Display Name:

Honorific:

Language Preference: ▾

Time Zone: ▾

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** from the drop down menu.
- **Fully Qualified Address** Enter the extension-number@domain

Click on **Add**.

Communication Profile

New Delete Done Cancel

Name

Primary

Select : None

* Name: Primary

Default : ☒

Communication Address

New Edit Delete

Type	Handle	Domain
No Records found		

Type: Avaya SIP

* Fully Qualified Address: 34002@silstack.com

Add Cancel

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**

☒ Session Manager Profile

* Primary Session Manager: SessionManager

Secondary Session Manager: (None)

Origination Application Sequence: App Sequence

Termination Application Sequence: App Sequence

Survivability Server: (None)

* Home Location: Dublin Stack

Primary	Secondary	Maximum
6	0	6

Primary	Secondary	Maximum

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

- **System** Select the Application Entity created in **Section 6.10.1**
- **Extension** Enter a desired extension number
- **Template** Select a telephone type template

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

☒ Endpoint Profile

* System: FeatureServer

Use Existing Endpoints: ☐

* Extension: 34002

Template: DEFAULT_9630SIP_CM_6_0

Set Type: 9630SIP

Security Code:

* Port: S00006

Voice Mail Number:

Delete Endpoint on Unassign of Endpoint from User: ☐

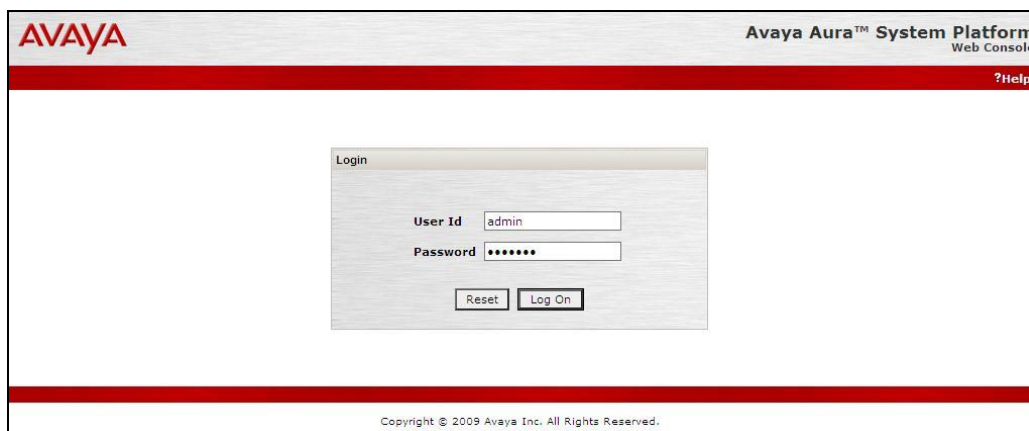
7. Verification Steps

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

7.1. 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.



The screenshot displays the Avaya Aura™ System Platform Web Console login interface. At the top left is the AVAYA logo, and at the top right is the text "Avaya Aura™ System Platform Web Console" with a "? Help" link. A central login box contains the following elements:

- A title bar labeled "Login".
- A "User Id" field with the text "admin" entered.
- A "Password" field with seven asterisks (*****).
- Two buttons at the bottom: "Reset" and "Log On".

At the bottom of the page, a copyright notice reads: "Copyright © 2009 Avaya Inc. All Rights Reserved."

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

Avaya Aura™ System Platform
admin
Previous successful login: Mon May 17 19:19:50 IST 2010
Failed login attempts since: 0
Failover status: **Not configured**
About | Help | Log Out

Virtual Machine Management
Virtual Machine List
System Domain Uptime: 10 days, 2 hours, 42 minutes, 43 seconds
Current template installed: Conferencing Standard Edition Template 6.0.0.0.126 (crs 6.0.0.0.126, smgr 6.0.0.0.127, bridge 6.0.0.0.125, awc 6.0.0.0.126, webportal 6.0.0.0.125) Refresh

Name	Version	IP Address	Maximum Memory	Maximum Virtual CPUs	CPU Time	State	Application State
awc	6.0.0.0.126	10.10.9.72	4.0 GB	1	5h 8m 57s	Running	N/A
crs	6.0.0.0.126	10.10.9.73	4.0 GB	1	11h 11m 51s	Running	N/A
webportal	6.0.0.0.125	10.10.9.75	4.0 GB	1	35m 46s	Running	N/A
Domain-0	6.0.0.1.6	10.10.9.70	512.0 MB	16	19h 42m 37s	Running	N/A
cdm	6.0.0.1.6	10.10.9.71	1024.0 MB	1	15h 42m 53s	Running	N/A
bridge	6.0.0.0.125	10.10.9.74	4.0 GB	4	9h 14m 16s	Running	N/A
smgr	6.0.0.0.127	10.10.9.76	4.0 GB	2	5m 46s	Running	N/A

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7.2. Conferencing Standard Edition Services

Check **Service State** between the Conferencing bridge and other devices by configuring the SIP System Settings. Select **Elements** → **Conferencing** → **Services** on the left panel menu. From the right panel menu ensure the **Conferencing Services** are in an **Active Service State**.

Avaya Aura™ System Manager 6.0
Welcome, admin Last Logged on Today at May 31, 2010 8:12 AM
Help | About | Change Password | Log off

Home / Elements / Conferencing / Services

Conferencing: Services

Disable Refresh Start Service(s) Stop Service(s) Export Import

4 Items Refresh

Name	Address	Server State	Service(s)	Service State
MX60Bridge	135.64.186.149	Powered on	Audio Conferencing	Active
MX60AWC	135.64.186.139	Powered on	Data Conferencing	Active
MX60CRS	135.64.186.147	Powered on	Client Registration	Active
MX60WebPortal	135.64.186.148	Powered on	Web Applications	Active

Select : All, None

7.3. 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 one of the SIP Entities created in **Section 6.4**

The screenshot displays the Avaya Aura System Manager 6.0 web interface. The top navigation bar includes the Avaya logo, the system name 'Avaya Aura™ System Manager 6.0', and a welcome message for user 'admin' with a 'Log off' link. A red breadcrumb trail shows the path: Home / Elements / Session Manager / System Status / SIP Entity Monitoring.

The left sidebar contains a tree view of the system's configuration. The 'Session Manager' folder is expanded, and the 'SIP Entity Monitoring' option is selected and highlighted with a red box.

The main content area is titled 'SIP Entity Link Monitoring Status Summary'. It includes a 'Refresh' button and a table summarizing the status of SIP entity links for all Session Manager instances.

Session Manager Name	Entity Links Down/Total	Entity Links Partially Down	SIP Entities - Monitoring Not Started	SIP Entities - Not Monitored
SessionManager2	1/1	0	0	0
SessionManager	5/17	0	0	1

Below the summary table, there is a section titled 'All Monitored SIP Entities' with another 'Refresh' button. It shows a list of 16 monitored entities, with a 'Filter: Enable' option. The entities listed are:

- [AudioCodesM2K](#)
- [Branch CM](#)
- [Bridge 6.0](#) (highlighted with a red box)
- [Enterprise Evolution CM](#)
- [FeatureServer](#)
- [IMG1010](#)
- [MX 5.2 Mick](#)
- [MX52](#)
- [MX_DavidH](#)

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**.

The screenshot shows the Avaya Aura System Manager 6.0 web interface. The top navigation bar includes the Avaya logo, the product name, and user information (Welcome, admin, Last Logged on at May 31, 2010 8:57 AM). The breadcrumb trail is: Home / Elements / Session Manager / System Status / SIP Entity Monitoring / SIP Entity Link Status.

The left sidebar contains a tree view of system elements, with 'SIP Entity Monitoring' selected. The main content area is titled 'SIP Entity, Entity Link Connection Status' and includes a sub-header 'All Entity Links to SIP Entity: Bridge_6.0'. Below this are 'Refresh' and 'Summary View' buttons.

A table displays the connection status for one item. The table has columns: Details, Session Manager Name, SIP Entity Resolved IP, Port, Proto., Conn. Status, Reason Code, and Link Status. The data row shows 'SessionManager' as the Session Manager Name, '10.10.9.74' as the SIP Entity Resolved IP, '5060' as the Port, 'TCP' as the Protocol, 'Up' as the Connection Status, '480 Temporarily Unavailable' as the Reason Code, and 'Up' as the Link Status.

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
► Show	SessionManager	10.10.9.74	5060	TCP	Up	480 Temporarily Unavailable	Up

7.4. Verification Scenarios

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

- Placing a call from 2 end points into conference ensuring 1 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 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 speech between the callers.

8. Conclusion

As illustrated in these Application Notes, Avaya Aura™ Conferencing Standard Edition can interoperate successfully with Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager using SIP trunks.

9. 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™ Communication Manager 6.0

- [5] *Administering Avaya Aura™ Communication Manager, Doc ID 03-300509, June 2010, available at <http://support.avaya.com>.*
- [6] *Administering Avaya Aura™ Communication Manager as a Feature Server, Doc # 03-603479, Issue 1.2, Release 5.2 January 2010, available at <http://support.avaya.com>.*

Avaya Aura™ Session Manager 6.0

- [7] *Avaya Aura™ Session Manager Overview, Doc ID 03-603323, available at <http://support.avaya.com>.*
- [8] *Administering Avaya Aura™ Session Manager, Doc ID 03-603324 available at <http://support.avaya.com>.*
- [9] *Installing and Upgrading Avaya Aura™ Session Manager 6.0, Doc ID 03-603324, available at <http://support.avaya.com>.*
- [10] *Installing and Upgrading Avaya Aura™ System Manager 6.0, available at <http://support.avaya.com>.*
- [11] *Maintaining and Troubleshooting Avaya Aura™ Session Manager 6.0, available at <http://support.avaya.com>.*

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