



## **Avaya Solution & Interoperability Test Lab**

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

## **Abstract**

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

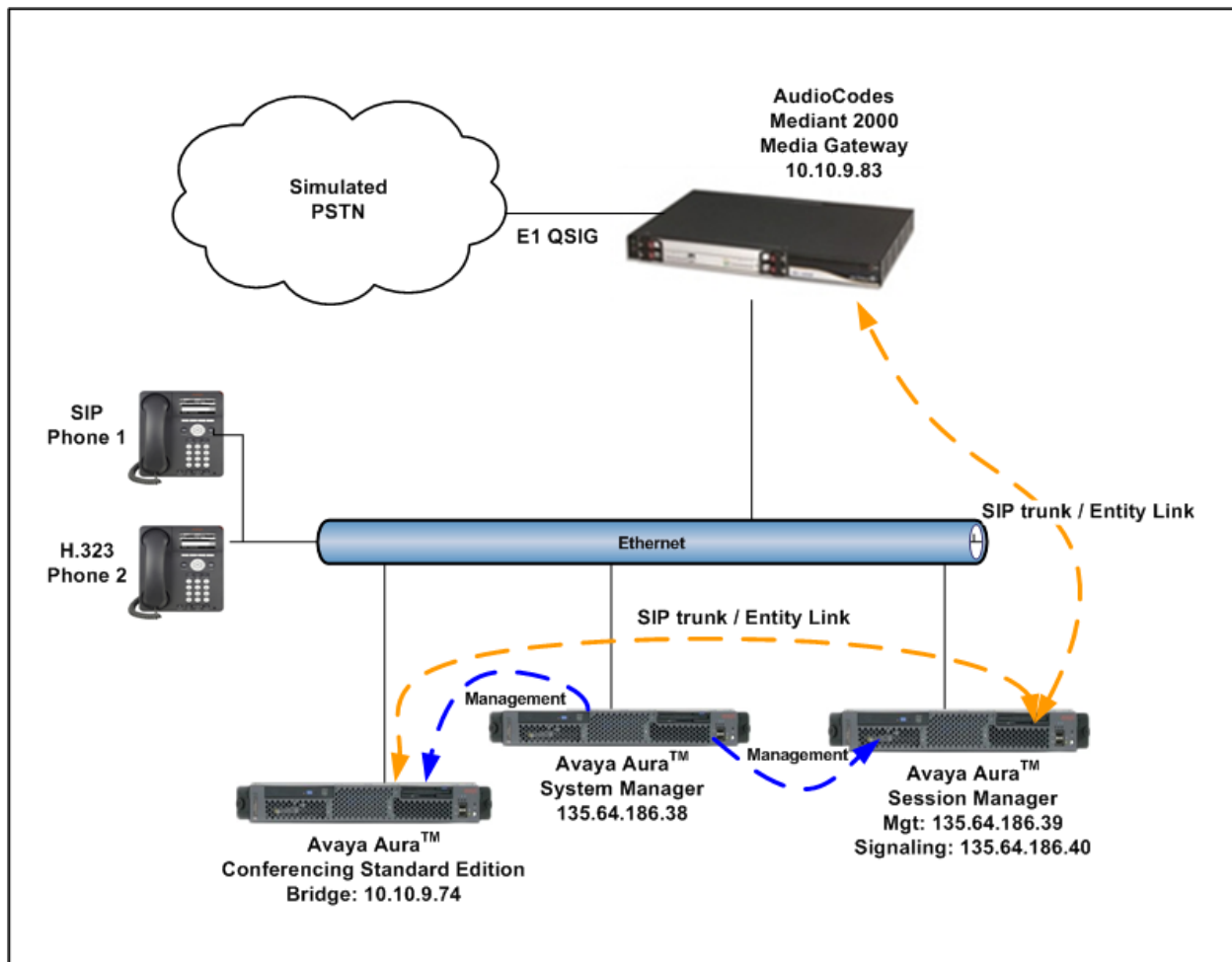
# 1. Introduction

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

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

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

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



**Figure 1 – Test Configuration used in these Application Notes**

## 2. Equipment and Software Validated

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

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

**Table 1: Hardware and Software Versions**

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

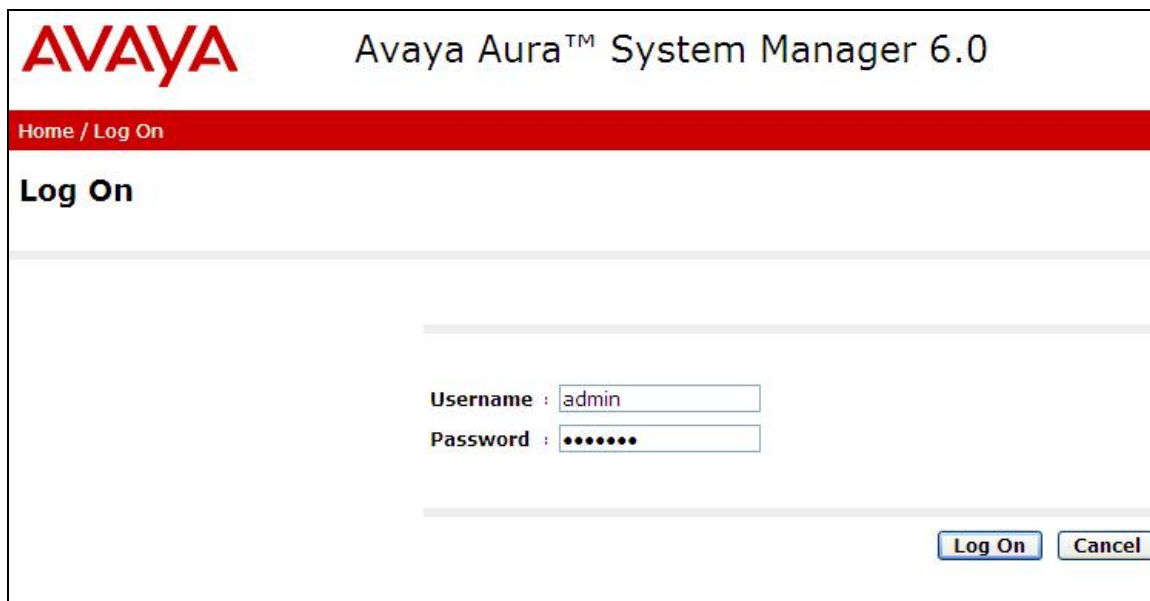
### 3. Configure Avaya Aura™ Conferencing Standard Edition

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

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

#### 3.1. Log in to Avaya Aura™ System Manager

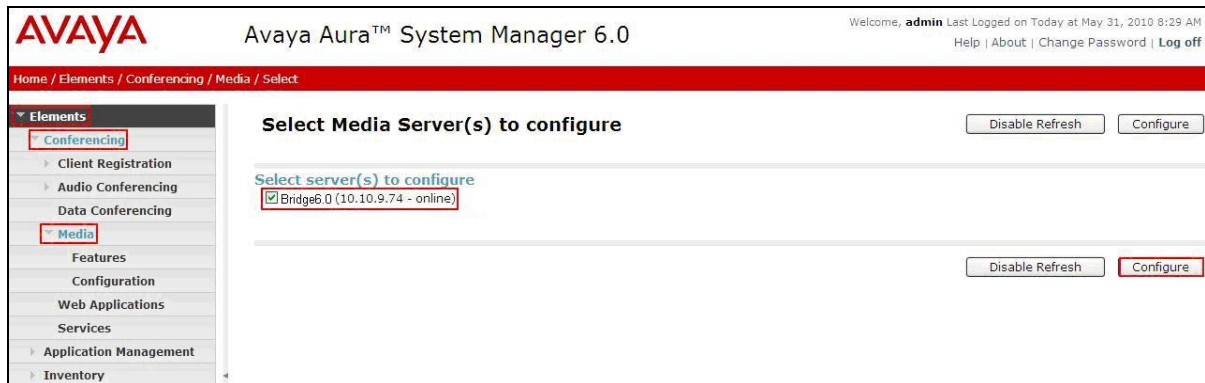
Access the System Manager using a Web Browser and entering *https://<ip-address>/smgr*, where <ip-address> is the IP address of System Manager. Log in using appropriate credentials and accept the subsequent Copyright Legal Notice.



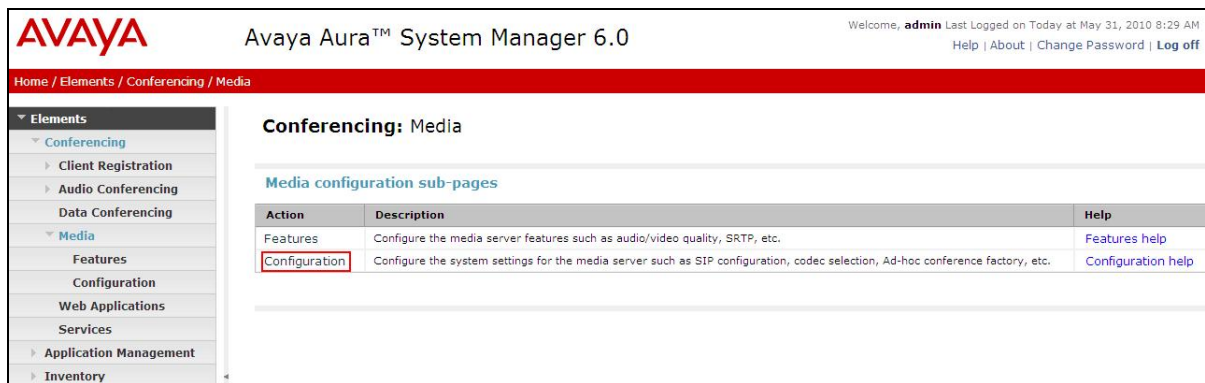
The screenshot shows the Avaya Aura™ System Manager 6.0 login interface. At the top, the Avaya logo is on the left and the title "Avaya Aura™ System Manager 6.0" is on the right. Below the title bar is a red navigation bar with the text "Home / Log On". The main content area is titled "Log On" and contains a login form. The form has two input fields: "Username" with the value "admin" and "Password" with masked characters (dots). Below the password field are two buttons: "Log On" and "Cancel".

## 3.2. Configuring SIP Connectivity

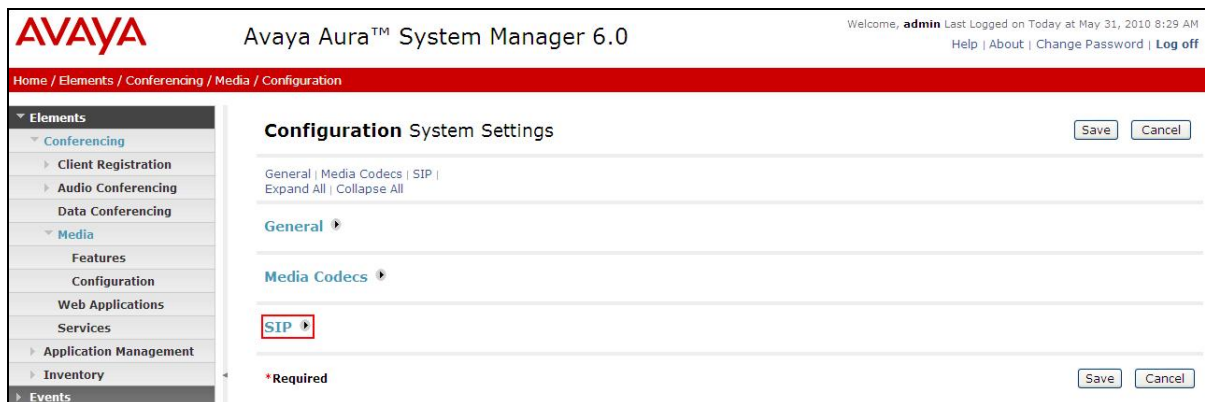
Configure settings that enable SIP connectivity between the Conferencing bridge and other devices by configuring the SIP System Settings by selecting **Elements** → **Conferencing** → **Media** on the left panel menu. From the right panel menu select the media server to configure by selecting the tick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



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



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



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

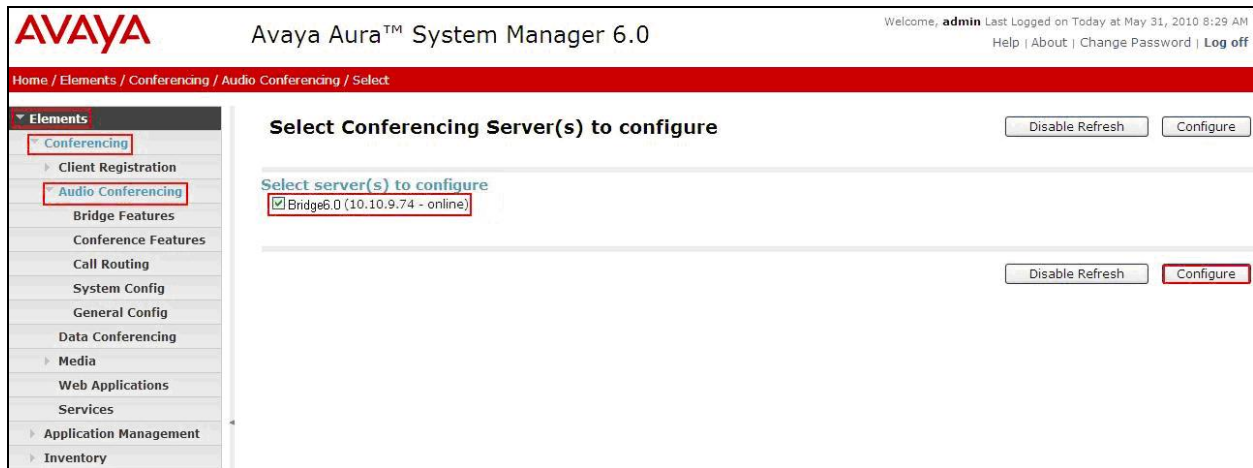
- **SIP Listener URI** <sip:6000@10.10.9.74:5060;transport=tcp>  
Depending on the SIP signalling protocol, TCP or UDP, configure the following line to populate the From Header Field in SIP INVITE messages:  
**Note:** The user field 6000, defined for this SIP URI must conform to RFC 3261. For consistency,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

Click the **Save** button.

The screenshot shows the Avaya Aura System Manager 6.0 interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura™ System Manager 6.0', and a welcome message for the 'admin' user. The breadcrumb trail is 'Home / Elements / Conferencing / Media / Configuration'. The left sidebar lists various system elements, with 'Media' expanded to show 'Features', 'Configuration', 'Web Applications', and 'Services'. The main content area is titled 'Configuration System Settings' and contains tabs for 'General', 'Media Codecs', and 'SIP'. The 'SIP' tab is active, showing four configuration fields: 'SIP Listener URI' (set to <sip:6000@10.10.9.74:5060;transport=tcp>), 'Response Contact' (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). The 'SIP Listener URI' and 'Response Contact' fields are highlighted with red boxes. At the bottom of the configuration area, there is a 'Required' label and 'Save' and 'Cancel' buttons.

### 3.3. Configure Dialout

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



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**

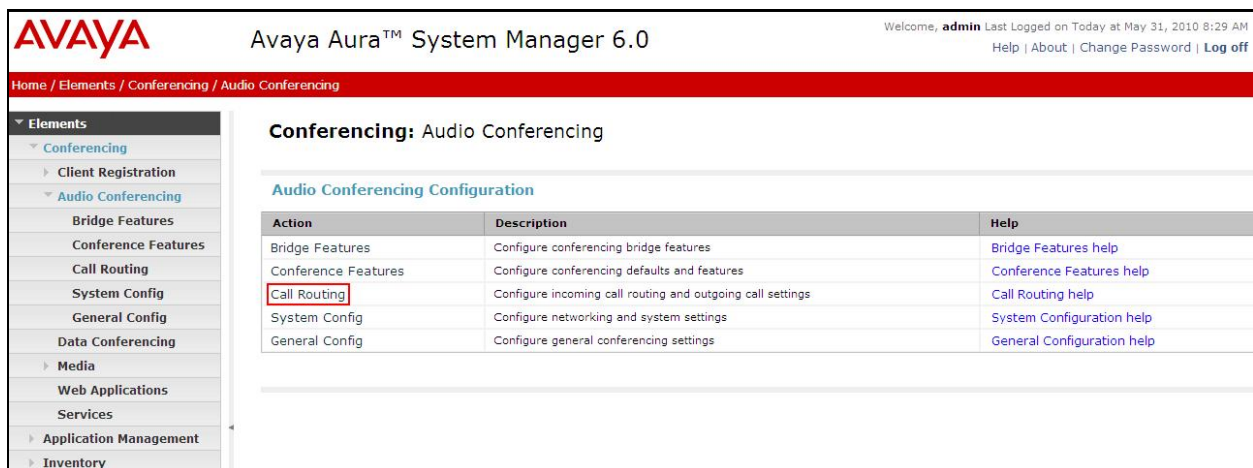
Disable Refresh Configure

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	<a href="#">Bridge Features help</a>
Conference Features	Configure conferencing defaults and features	<a href="#">Conference Features help</a>
<b>Call Routing</b>	Configure incoming call routing and outgoing call settings	<a href="#">Call Routing help</a>
System Config	Configure networking and system settings	<a href="#">System Configuration help</a>
General Config	Configure general conferencing settings	<a href="#">General Configuration help</a>



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

Click the **Save** button.

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 / Entry

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

\* Telnum \*

\* URI sip:\$0@135.64.186.40

Comment Route\_calls\_to\_Asset

\*Required [Save] [Cancel]

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

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

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@135.64.186.40:5060;transport=tcp	Route_calls_to_Asset_Card

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"><li>No changes</li></ul>	NONE	Powered on
10.10.9.73 <ul style="list-style-type: none"><li>No changes</li></ul>	NONE	Powered on
10.10.9.75 <ul style="list-style-type: none"><li>No changes</li></ul>	NONE	Powered on
10.10.9.74 <ul style="list-style-type: none"><li>Changing "bridge.telnumToUriEntries[0].comment".</li><li>Changing "bridge.telnumToUriEntries[0].telnumConversion".</li></ul>	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	<a href="#">Bridge Features help</a>
<b>Conference Features</b>	Configure conferencing defaults and features	<a href="#">Conference Features help</a>
Call Routing	Configure incoming call routing and outgoing call settings	<a href="#">Call Routing help</a>
System Config	Configure networking and system settings	<a href="#">System Configuration help</a>
General Config	Configure general conferencing settings	<a href="#">General Configuration help</a>

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**

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 RefreshApply ChangesDiscard ChangesAdd more changes

#### Impact of changes

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

Disable RefreshApply ChangesDiscard ChangesAdd 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	<a href="#">Bridge Features help</a>
Conference Features	Configure conferencing defaults and features	<a href="#">Conference Features help</a>
<b>Call Routing</b>	Configure incoming call routing and outgoing call settings	<a href="#">Call Routing help</a>
System Config	Configure networking and system settings	<a href="#">System Configuration help</a>
General Config	Configure general conferencing settings	<a href="#">General Configuration help</a>

Disable Refresh Configure

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

▼ 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

**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 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

▼ 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

**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
- **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		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]

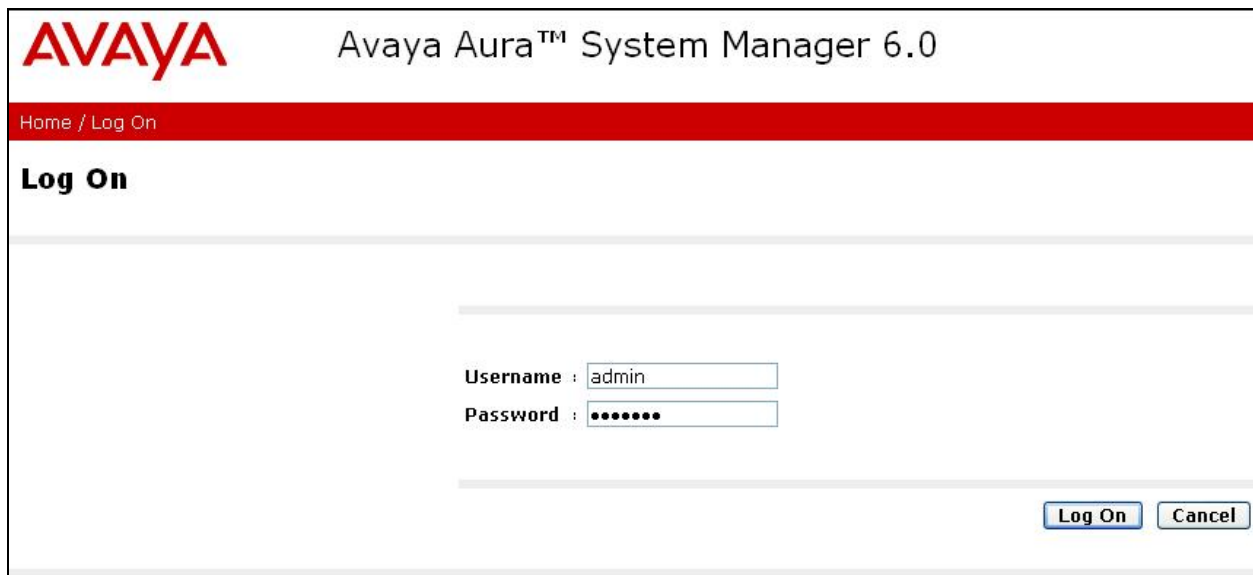
## 4. Configure Avaya Aura™ Session Manager

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

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

### 4.1. Log in to Avaya Aura™ System Manager

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



**AVAYA** Avaya Aura™ System Manager 6.0

Home / Log On

**Log On**

Username : admin

Password : .....

Log On Cancel

## 4.2. Administer Domains

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

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

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

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 breadcrumb trail indicates the current location: Home / Routing / Domains. The left-hand navigation pane lists various system components, with 'Routing' expanded to show 'Domains' as the active selection. 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 is shown with the name 'silstack.com' and type 'sip'. Below the table, there is a 'Select' dropdown menu set to 'All'.

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

### 4.3. Add Locations

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

Under **General**:

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

Under **Location Pattern**:

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

Click **Commit** to save changes.

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.*	

## 4.4. Add SIP Elements

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

Under **General**:

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

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

Click **Commit** to save changes.

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 welcome message for 'admin' with the last login time. A navigation breadcrumb trail 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 'SessionManager'), 'FQDN or IP Address' (set to '135.64.186.40'), 'Type' (set to 'Session Manager'), 'Notes' (empty), 'Location' (set to 'Dublin Stack'), 'Outbound Proxy' (empty), 'Time Zone' (set to 'Europe/Dublin'), and 'Credential name' (empty). At the bottom, there is a 'SIP Link Monitoring' section with a dropdown set to 'Use Session Manager Configuration'. 'Commit' and 'Cancel' buttons are located in the top right corner of the form area.

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

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

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

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

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

Under **General**:

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

Click **Commit** to save changes.

**AVAYA** Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at June 11, 2010 1:08 PM  
[Help](#) | [Change Password](#) | [Log off](#)

Home / Routing / SIP Elements / SIP Elements Details

**SIP Element Details**

**General**

\* **Name:** AudioCodesM2K

\* **FQDN or IP Address:** 10.10.9.83

**Type:** Gateway

**Notes:** AudioCodesMediant2000

**Adaptation:** [Dropdown]

**Location:** Dublin Stack

**Time Zone:** Europe/Dublin

Override Port & Transport with DNS SRV: ☐

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

**Credential name:** [Text Field]

**Commit** **Cancel**



The following screen shows addition of Conferencing Standard Edition (**Bridge\_6.0**). Select the following parameters, leaving the remaining parameters at their default values.

Under **General**:

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

Click **Commit** to save changes.

**AVAYA** 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 / SIP Elements / SIP Elements Details

**SIP Element Details** **Commit** Cancel

**General**

\* **Name:** Bridge\_6.0

\* **FQDN or IP Address:** 10.10.9.74

**Type:** SIP Trunk

**Notes:** Bridge Conferencing 6.0

**Adaptation:**

**Location:** Dublin Stack

**Time Zone:** Europe/Dublin

Override Port & Transport with DNS SRV: ☐

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

**Credential name:**

**Call Detail Recording:** both

**SIP Link Monitoring**

**SIP Link Monitoring:** Use Session Manager Configuration



## 4.5. Add Element Links

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

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

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

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at June 11, 2010 1:08 PM  
Help | Change Password | Log off

Home / Routing / Element Links

Element Links

Edit New Duplicate Delete More Actions Commit

28 Items Refresh Filter: Enable

<input type="checkbox"/>	Name	SIP Element 1	Protocol	Port	SIP Element 2	Port	Trusted
<input type="checkbox"/>	asm60-asm52	SessionManager	TCP	5060	asm 5.2	5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	AudioCodesM2K	SessionManager	TCP	5060	AudioCodesM2K	5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	Bridge_6.0	SessionManager	TCP	5060	Bridge_6.0	5060	<input checked="" type="checkbox"/>

## 4.6. Administer Time Ranges

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

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

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

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

## 4.7. Administer Routing Policies

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

Under **General**:

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

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

The screenshot displays the Avaya Aura System Manager 6.0 interface. The left sidebar shows the navigation menu with 'Routing' expanded and 'Policies' selected. The main content area is titled 'Routing Policy Details' and includes 'Commit' and 'Cancel' buttons. The 'General' tab is active, showing the 'Name' field set to 'Bridge 6.0', a 'Disabled' checkbox, and a 'Notes' field. The 'SIP Element as Destination' tab is also visible, showing a 'Select' button and a table with columns: Name, FQDN or IP Address, Type, and Notes. The table contains one entry: 'Bridge\_6.0', '10.10.9.74', 'SIP Trunk', and 'Bridge Conferencing 6.0'. The 'Time of Day' tab is active, showing 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. Below these buttons is a table with columns: Ranking, Name, Mon, Tue, Wed, Thu, Fri, Sat, Sun, Start Time, End Time, and Notes. The table contains one entry: '0', '24/7', with checkboxes for Mon through Sun all checked, '00:00' for Start Time, '23:59' for End Time, and 'Time Range 24/7' for Notes.

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

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	✓	✓	✓	✓	✓	✓	✓	00:00	23:59	Time Range 24/7

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

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

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

Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at June 11, 2010 1:08 PM  
Help | Change Password | Log off

Home / Routing / Policies / Policy Details

**Routing Policy Details** [Commit] [Cancel]

**General**

\* Name: AudioCodesM2K

Disabled: ☐

Notes:

**SIP Element as Destination**

Select

Name	FQDN or IP Address	Type	Notes
AudioCodesM2K	10.10.9.83	Gateway	AudioCodesMediant2000

**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

## 4.8. Administer Dial Patterns

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

Under **General**

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

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

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 the following fields: 'Pattern' (7111), 'Min' (4), 'Max' (4), 'SIP Domain' (-ALL-), and 'Notes'. Below the form is the 'Originating Locations and Routing Policies' section, which includes an 'Add' button and a table with 1 item. The table has columns for 'Originating Location Name', 'Originating Location Notes', 'Routing Policy Name', 'Rank', 'Routing Policy Disabled', 'Routing Policy Destination', and 'Routing Policy Notes'.

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

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

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	

## 4.9. Administer Avaya Aura™ Session Manager

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

Under **General**:

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

Under **Security Module**:

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

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

**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



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

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

### 4.10.1. Create an Application Entity

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

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

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

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

Click on **Commit** to save.

**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]



### 4.10.2. Create a Feature Server Application

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

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

Click on **Commit** to save.

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]

### 4.10.3. Create a Feature Server Application Sequence

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

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**

CommitCancel

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"/>	<div>▲▼✖</div>	FeatureServer	FeatureServer	<input checked="" type="checkbox"/>	

Select : All, None

Available Applications

1 Item RefreshFilter: Enable

	Name	SIP Entity	Description
<div>+</div>	FeatureServer	FeatureServer	

#### 4.10.4. Synchronize Avaya Aura™ Communication Manager Data

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

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: 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

**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

## 4.11. Add Users for SIP Phones

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

Under **General**:

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

**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 4.2**
- **Password**      Password for user to log into SMGR
- **Shared Communication Profile Password**  
    Password to be entered by the user when logging into the phone

**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**
- **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**

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

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

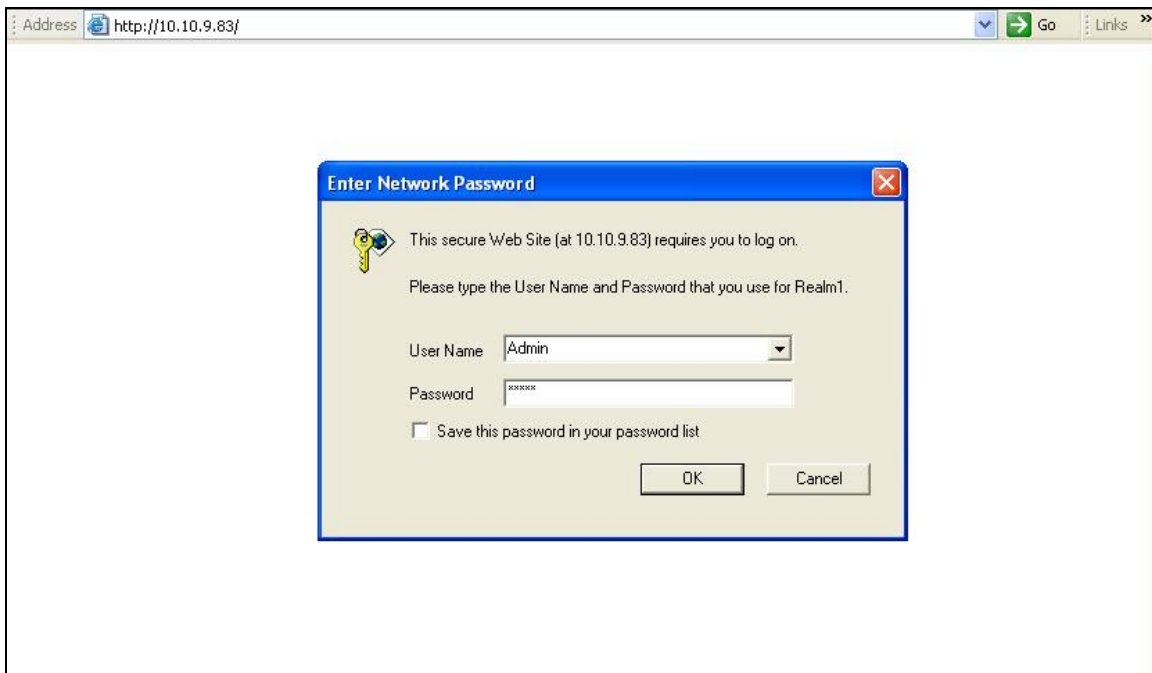
## 5. Configure AudioCodes Mediant 2000 Gateway

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

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

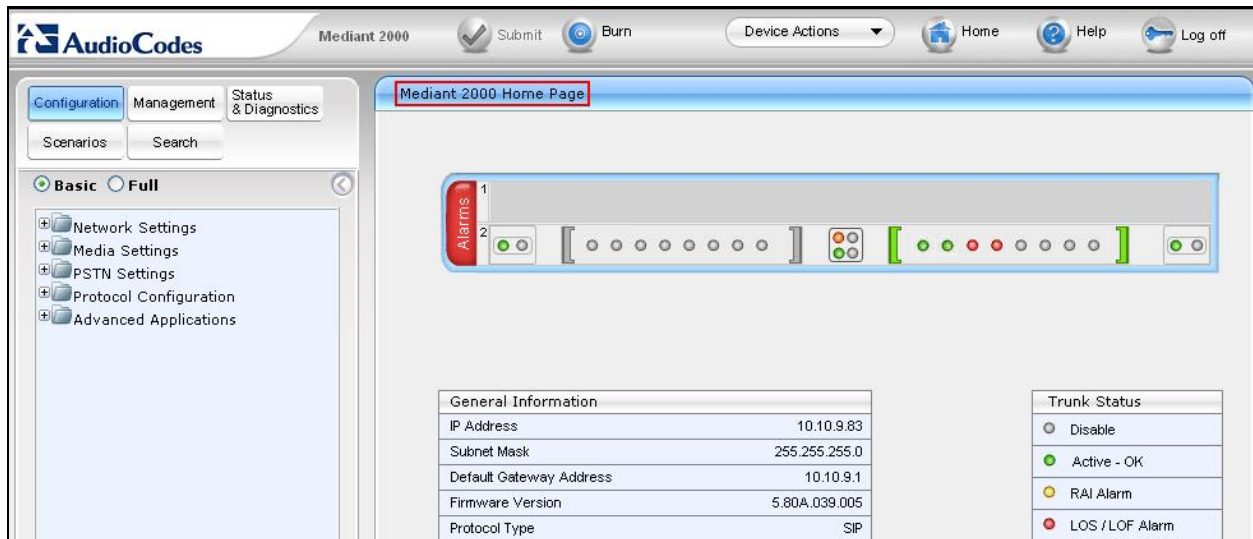
### 5.1. Access Web Configuration Interface

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

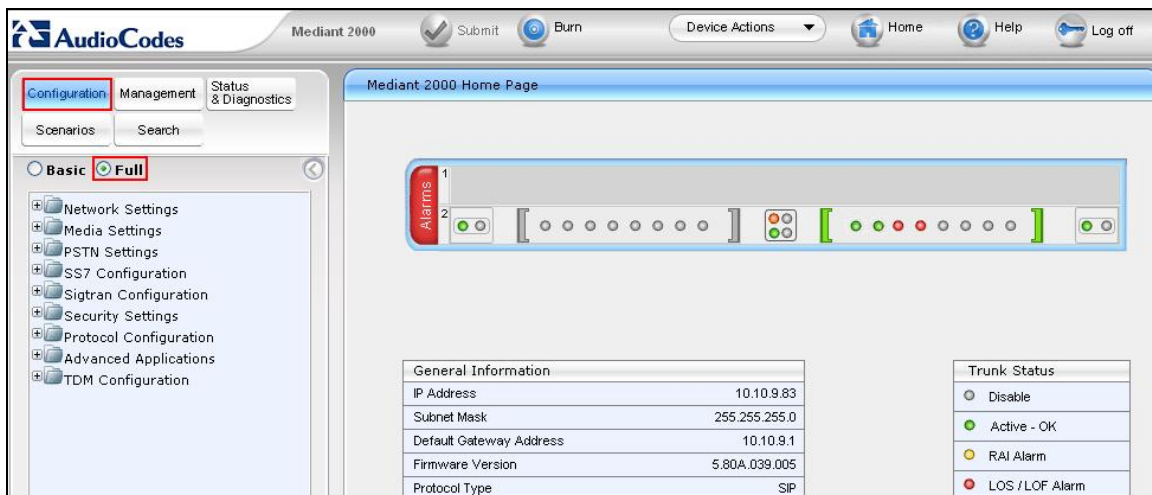




The **Mediant 2000 Home Page** screen is displayed.



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

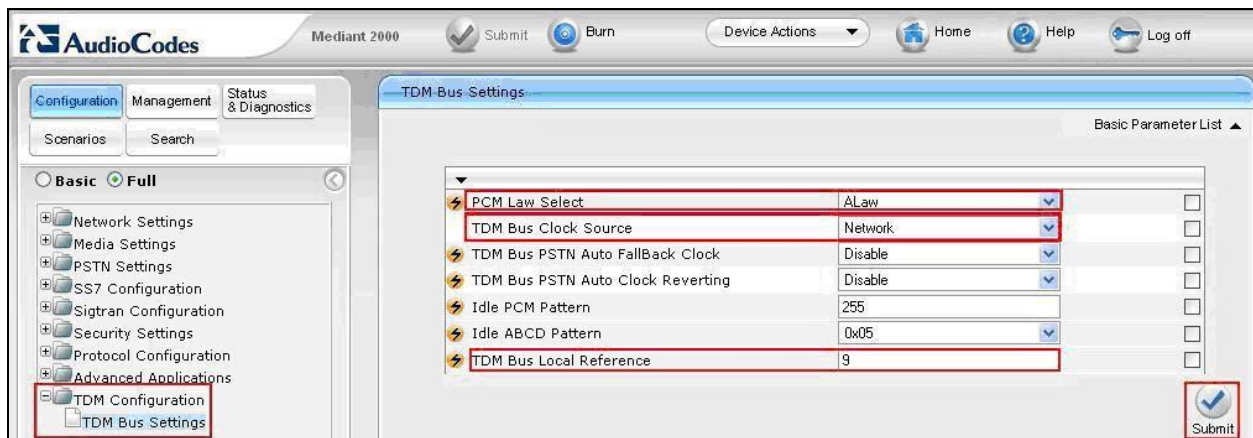


## 5.2. Administer TDM Bus Settings

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

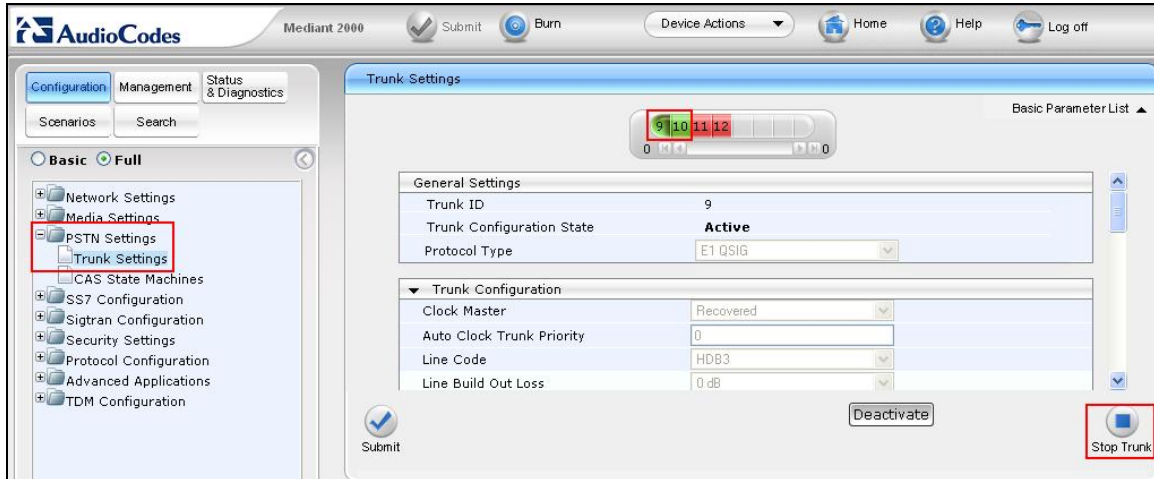
- **PCM Law Select** Select **A-Law**.
- **TDM Bus Clock Source:** Select **Network**
- **TDM Bus Local Reference** Select **9**, first trunk that will take the clocking.

Click on **Submit** to save changes.



### 5.3. Administer PSTN Trunk Settings

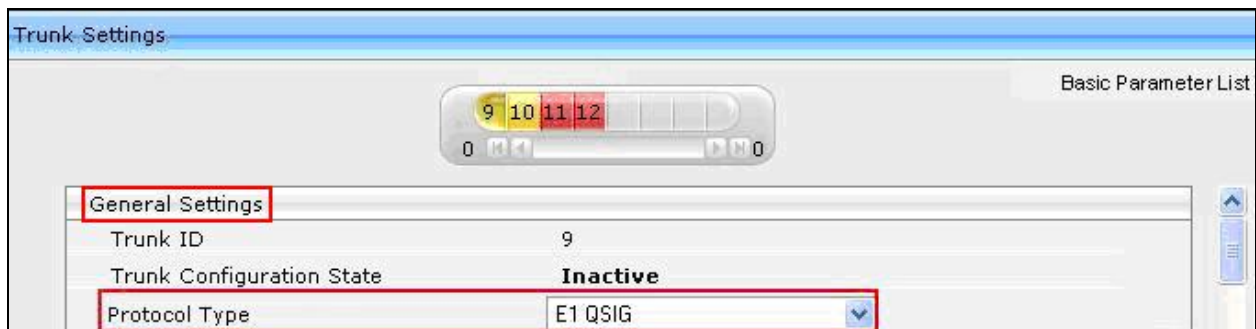
Select **PSTN Settings** → **Trunk Settings** on the left panel menu. Click **Stop Trunk**, which will enable editing of the parameters.



Select the following parameters, leaving the remaining parameters at their default values.

Under **General Settings**:

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



Under **Trunk Configuration**:

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

The screenshot shows the 'Trunk Settings' window with the 'Trunk Configuration' section expanded. The parameters are as follows:

Parameter	Value
Clock Master	Recovered
Auto Clock Trunk Priority	0
Line Code	HDB3
Line Build Out Loss	0 dB
Trace Level	No Trace
Line Build Out Overwrite	OFF
Framing Method	E1 FRAMING MFF CRC4 EXT

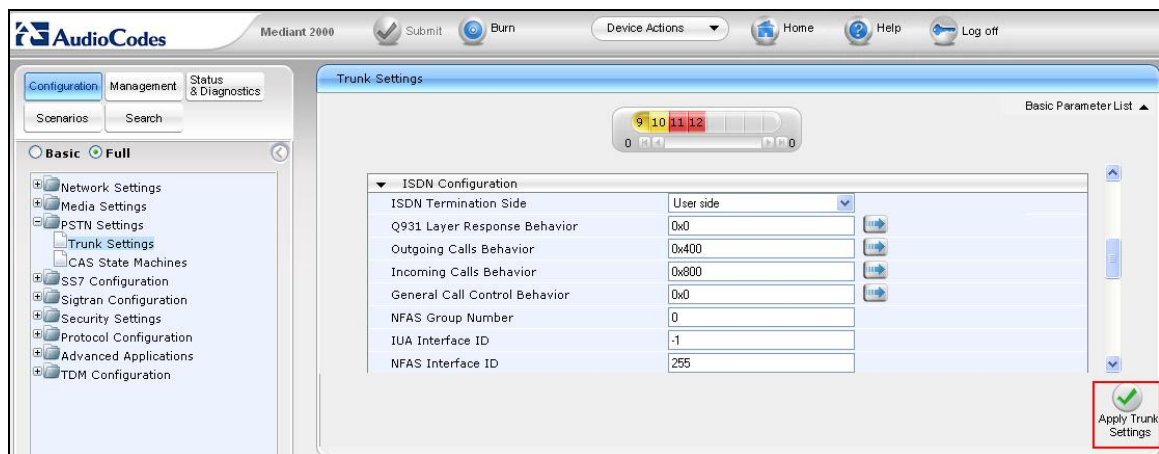
Under **ISDN Configuration**:

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

The screenshot shows the 'Trunk Settings' window with the 'ISDN Configuration' section expanded. The parameters are as follows:

Parameter	Value
ISDN Termination Side	User side
Q931 Layer Response Behavior	0x0
Outgoing Calls Behavior	0x400
Incoming Calls Behavior	0x800
General Call Control Behavior	0x0
NFAS Group Number	0
IUA Interface ID	-1
NFAS Interface ID	255
D-channel Configuration	PRIMARY

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

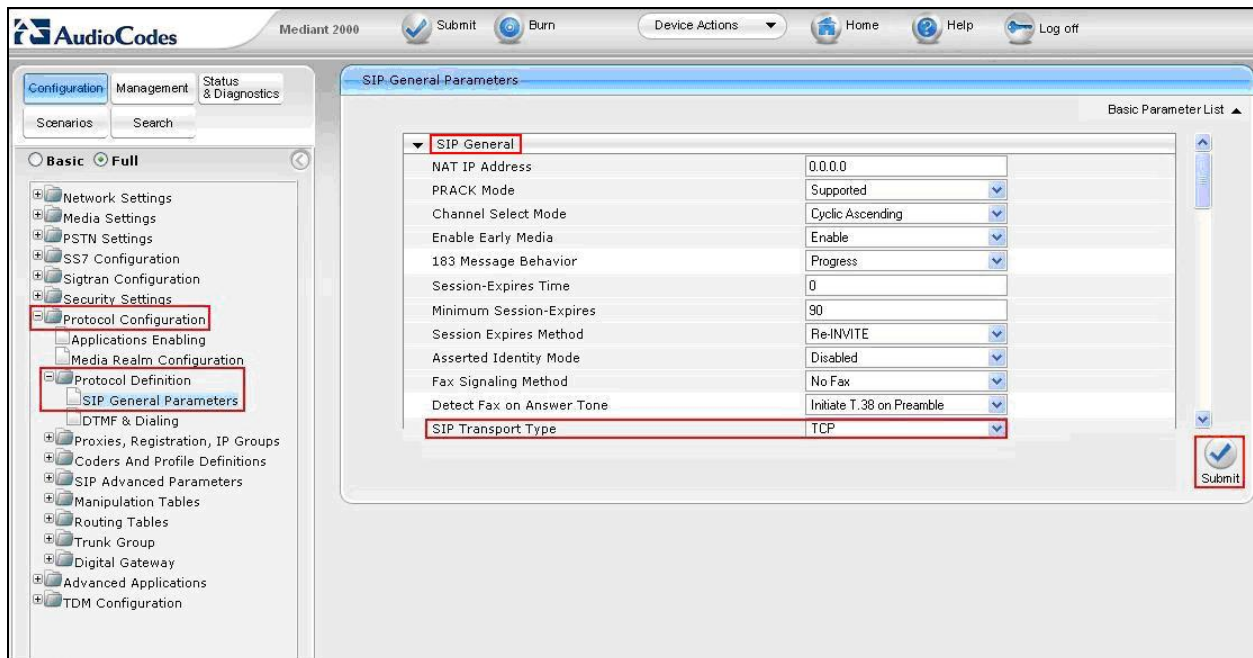


## 5.4. Administer SIP Protocol Parameters

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

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

Click on **Submit** to save changes.



## 5.5. Administer Audio Codecs

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

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

The screenshot shows the Mediant 2000 configuration interface. The left sidebar contains a tree view with the following structure:

- Configuration (selected)
- Management
- Status & Diagnostics
- Scenarios
- Search
- Basic (radio button)
- Full (radio button, selected)
- Network Settings
- Media Settings
- PSTN Settings
- SS7 Configuration
- Sigtran Configuration
- Security Settings
- Protocol Configuration (highlighted with a red box)
  - Applications Enabling
  - Media Realm Configuration
  - Protocol Definition
  - Proxies, Registration, IP Groups
  - Coders And Profile Definitions (highlighted with a red box)
    - Coders (highlighted with a red box)
    - Coder Group Settings
    - Tel Profile Settings
    - IP Profile Settings
  - SIP Advanced Parameters
  - Manipulation Tables
  - Routing Tables
  - Trunk Group
  - Digital Gateway
  - Advanced Applications
  - TDM Configuration

The main panel displays the 'Coders Table' with the following data:

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711A-law	20	64	8	Disabled
G.711U-law	20	64	0	Disabled

A 'Submit' button is located in the bottom right corner of the main panel.



## 5.6. Administer DTMF Signaling

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

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

Click on **Submit** to save changes.

Basic Parameter List	
Max Digits In Phone Num	5
Inter Digit Timeout for Overlap Dialing [sec]	4
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option	RFC 2833
2nd Tx DTMF Option	RFC 2833
RFC 2833 Payload Type	101
Digit Mapping Rules	
Default Destination Number	1000
Special Digit Representation	Special

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

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

Click on **Submit** to save changes.

Basic Parameter List	
Max Digits In Phone Num	5
Inter Digit Timeout for Overlap Dialing [sec]	4
Declare RFC 2833 in SDP	No
1st Tx DTMF Option	RFC 2833
2nd Tx DTMF Option	RFC 2833
RFC 2833 Payload Type	101
Digit Mapping Rules	
Default Destination Number	1000
Special Digit Representation	Special



## 5.7. Administer Proxy & Registration

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

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

Click on **Submit** to save changes.

The screenshot shows the AudioCodes Mediant 2000 configuration interface. The left sidebar contains a tree view with the following items: Configuration, Management, Status & Diagnostics, Scenarios, Search, Basic, and Full. Under the 'Full' section, the following items are listed: Network Settings, Media Settings, PSTN Settings, SS7 Configuration, Sigtran Configuration, Security Settings, Protocol Configuration, Applications Enabling, Media Realm Configuration, Protocol Definition, Proxies, Registration, IP Groups, Proxy & Registration, Proxy Sets Table, IP Group Table, and Account Table. The 'Proxy & Registration' item is selected. The main area displays the 'Proxy & Registration' configuration page. The 'Basic Parameter List' table is shown with the following parameters and values:

Parameter	Value
Use Default Proxy	No
Proxy Name	
Redundancy Mode	Parking
Proxy IP List Refresh Time	60
Enable Fallback to Routing Table	Disable
Prefer Routing Table	No
Always Use Proxy	Disable
Redundant Routing Mode	Routing Table
SIP ReRouting Mode	Standard Mode
Enable Registration	Disable

At the bottom of the page, there are three buttons: Register, Un-Register, and Submit. The Submit button is highlighted with a red box.

## 5.8. Administer Routing Tables

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

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

Click on **Submit** to save changes.

The screenshot shows the AudioCodes Mediant 2000 configuration interface. The left sidebar contains a navigation tree with the following items: Configuration, Management, Status & Diagnostics, Scenarios, Search, Basic, Full, Network Settings, Media Settings, PSTN Settings, SS7 Configuration, Sigtran Configuration, Security Settings, Protocol Configuration, Applications Enabling, Media Realm Configuration, Protocol Definition, Proxies, Registration, IP Groups, Coders And Profile Definitions, SIP Advanced Parameters, Manipulation Tables, Routing Tables, Routing General Parameters, Tel to IP Routing, IP to Trunk Group Routing, and Internal DNS Table. The 'Tel to IP Routing' item is selected. The main area displays the 'Tel to IP Routing' configuration page. At the top, there is a 'Basic Parameter List' section with a 'Routing Index' dropdown set to '1-10' and a 'Tel To IP Routing Mode' dropdown set to 'Route calls before manipulation'. Below this is a table with the following columns: Src. Trunk Group ID, Dest. Phone Prefix, Source Phone Prefix, Dest. IP Address, Port, Transport Type, and Dest. IP Group ID. The table has three rows. Row 1 is highlighted with a red border and contains the values: \*, \*, \*, 135.64.186.40, 5060, TCP, and 1. Row 2 contains empty fields. Row 3 contains empty fields. A 'Submit' button is located in the bottom right corner of the table area.

Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Port	Transport Type	Dest. IP Group ID
*	*	*	135.64.186.40	5060	TCP	1
					Not Configured	
					Not Configured	

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

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

Click on **Submit** to save changes.

The screenshot shows the AudioCodes Mediant 2000 configuration interface. The left sidebar contains a navigation tree with the following items: Configuration, Management, Status & Diagnostics, Scenarios, Search, Basic, Full, Network Settings, Media Settings, PSTN Settings, SS7 Configuration, Sigtran Configuration, Security Settings, Protocol Configuration, Applications Enabling, Media Realm Configuration, Protocol Definition, Proxies, Registration, IP Groups, Coders And Profile Definitions, SIP Advanced Parameters, Manipulation Tables, Routing Tables, Routing General Parameters, Tel to IP Routing, and IP to Trunk Group Routing. The 'IP to Trunk Group Routing' item is selected. The main area displays the 'IP To Trunk Group Routing Table' configuration page. The page includes a 'Routing Index' dropdown set to '1-12', a 'Routing Mode' dropdown set to 'IP To Tel Routing Mode', and a 'Route calls before manipulation' dropdown. Below these is a table with the following columns: Dest. Host Prefix, Source Host Prefix, Dest. Phone Prefix, Source Phone Prefix, Source IP Address, and Trunk Group ID. The table has 4 rows. The first row is populated with asterisks (\*) for the first five columns and '9' for the Trunk Group ID. The other three rows are empty. A 'Submit' button is located in the bottom right corner of the table area.

	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	Trunk Group ID
1	*	*	*	*	*	9
2						
3						
4						

## 5.9. Administer SIP General Parameters for TCP

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

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

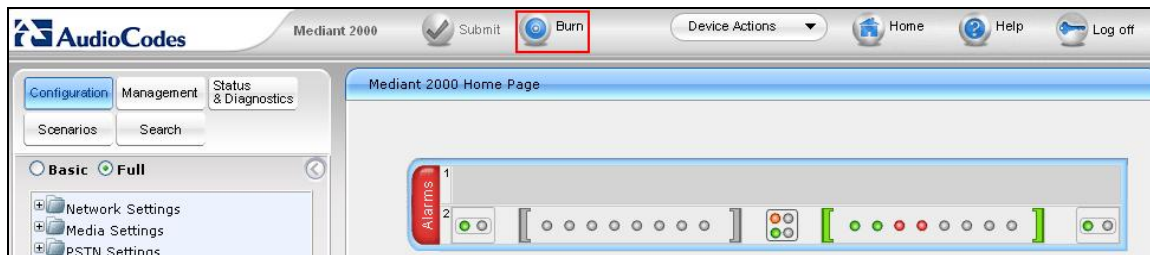
Click on **Submit** to save changes.

The screenshot shows the AudioCodes Mediant 2000 configuration interface. The left sidebar contains a tree view with the following items: Configuration, Management, Status & Diagnostics, Scenarios, Search, Basic, Full, Network Settings, Media Settings, PSTN Settings, SS7 Configuration, Sigtran Configuration, Security Settings, Protocol Configuration, Applications Enabling, Media Realm Configuration, Protocol Definition, SIP General Parameters, DTMF & Dialing, Proxies, Registration, IP Groups, Coders And Profile Definitions, and SIP Advanced Parameters. The 'SIP General Parameters' item is selected. The main panel displays the 'SIP General Parameters' configuration page. The parameters are listed in a table with their current values. Red boxes highlight the 'SIP Transport Type', 'SIP TCP Local Port', and 'SIP Destination Port' fields, which are set to 'TCP', '5060', and '5060' respectively. A 'Submit' button is visible at the bottom right.

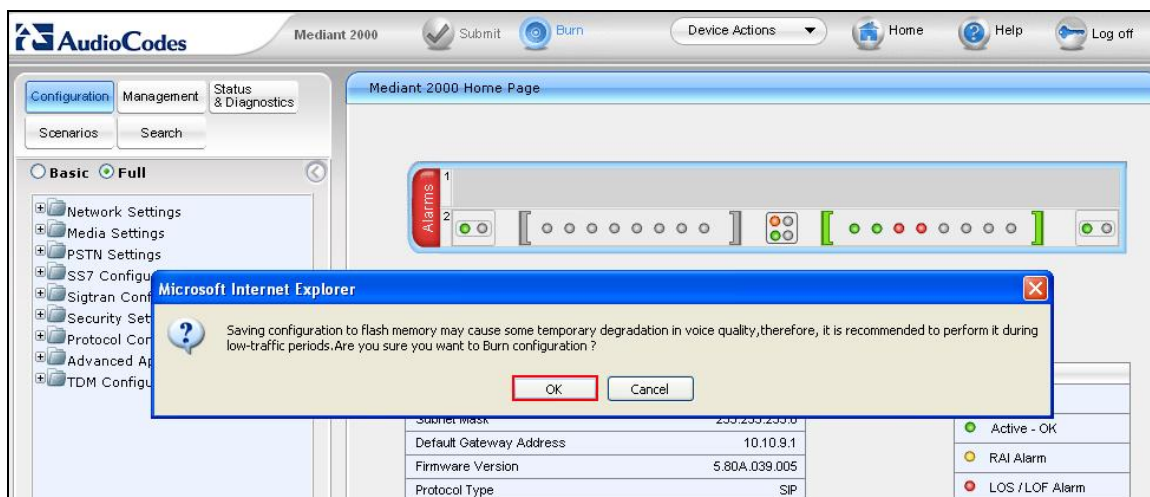
Parameter	Value
SIP Transport Type	TCP
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPs	Disable
Enable TCP Connection Reuse	Enable
TCP Timeout	0
SIP Destination Port	5060
Use user=phone in SIP URL	Yes
Use user=phone in From Header	No
Use Tel URI for Asserted Identity	Disable

## 5.10. Save the Configuration

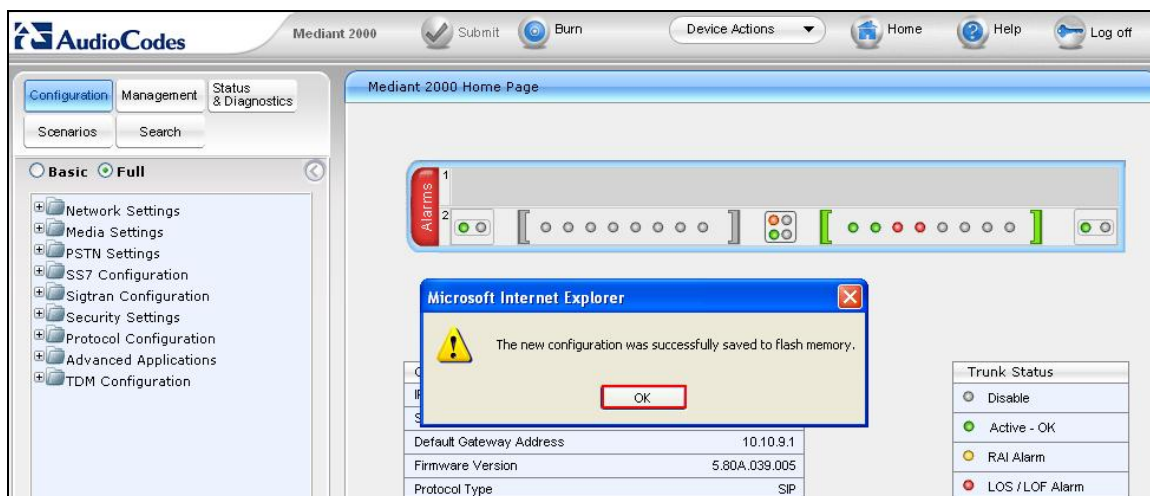
Click on **Burn** on the Mediant 2000 Toolbar.



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



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



## 6. Verification Steps

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

- The Conferencing Standard Edition configuration
- Session Manager

### 6.1. Avaya Aura™ Conferencing Standard Edition

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



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 "Login" at the top of the box.
- A "User Id" label followed by a text input field containing the value "admin".
- A "Password" label followed by a password input field filled with ten asterisks "\*\*\*\*\*".
- Two buttons at the bottom: "Reset" and "Log On".

At the bottom of the page, below a red horizontal line, is the copyright notice: "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

Home

- Virtual Machine Management
- Server Management
- User Administration

**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
cdom	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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### 6.1.1. Conferencing Standard Edition Services

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

**AVAYA** 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

- Elements
  - Conferencing
    - Client Registration
    - Audio Conferencing
    - Data Conferencing
    - Media
    - Web Applications
    - Services
    - Application Management
    - Inventory
  - Events
  - Groups & Roles
  - Licenses

**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



## 6.2. SIP Monitoring on Avaya Aura™ Session Manager

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

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 a user status message: 'Welcome, admin Last Logged on at May 28, 2010 4:39 PM'. Below this is a red breadcrumb trail: 'Home / Elements / Session Manager / System Status / SIP Entity Monitoring'.

The left sidebar contains a tree view of the system's configuration. The 'Elements' section is expanded, showing 'Session Manager' as the selected category. Within 'Session Manager', 'System Status' is selected, and 'SIP Entity Monitoring' is highlighted in the sub-menu.

The main content area is titled 'SIP Entity Link Monitoring Status Summary'. It includes a sub-header 'Entity Link Status for All Session Manager Instances' and a 'Refresh' button. Below this is a table with the following data:

Session Manager Name	Entity Links Down/Total	Entity Links Partially Down	SIP Entities - Monitoring Not Started	SIP Entities - Not Monitored
<a href="#">SessionManager2</a>	1/1	0	0	0
<a href="#">SessionManager</a>	5/17	0	0	1

Below the table, there is a section titled 'All Monitored SIP Entities' with another 'Refresh' button. It shows a list of 16 items with a 'Filter: Enable' option. The list contains the following SIP entity names:

- [AudioCodesM2K](#)
- [Branch\\_CM](#)
- [Bridge\\_6.0](#)
- [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 interface. The left sidebar contains a navigation menu with categories like Elements, Session Manager, System Status, and SIP Entity Monitoring. The main content area is titled "SIP Entity, Entity Link Connection Status" and displays a table of connection status for the selected SIP entity, Bridge 6.0. The table has columns for Session Manager Name, SIP Entity Resolved IP, Port, Proto., Conn. Status, Reason Code, and Link Status. The Conn. Status and Link Status are both "Up".

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	200 OK	Up

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

The screenshot shows the Avaya Aura System Manager 6.0 interface. The left sidebar contains a navigation menu with categories like Elements, Session Manager, System Status, and SIP Entity Monitoring. The main content area is titled "SIP Entity, Entity Link Connection Status" and displays a table of connection status for the selected SIP entity, AudioCodesM2K. The table has columns for Session Manager Name, SIP Entity Resolved IP, Port, Proto., Conn. Status, Reason Code, and Link Status. The Conn. Status and Link Status are both "Up".

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	SessionManager	10.10.9.83	5060	TCP	Up	200 OK	Up

### 6.3. Verification Scenarios

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

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

## 7. Conclusion

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

## 8. Additional References

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

### Avaya Aura™ Conferencing Standard Edition 6.0

- [1] *Implementing Avaya Aura™ Conferencing Standard Edition, Doc ID 04-603508, June 2010, available at <http://support.avaya.com>.*
- [2] *Operating Avaya Aura™ Conferencing Standard Edition, Doc ID 04-603510, June 2010, available at <http://support.avaya.com>.*
- [3] *Using Avaya Aura™ Conferencing Standard Edition, Doc ID 04-603509, June 2010, available at <http://support.avaya.com>.*
- [4] *Avaya Aura™ Conferencing Standard Edition Release Notes, Doc ID 04-603528, June 2010, available at <http://support.avaya.com>*

### Avaya Aura™ Session Manager 6.0

- [5] *Avaya Aura™ Session Manager Overview, Doc ID 03-603323, available at <http://support.avaya.com>.*
- [6] *Administering Avaya Aura™ Session Manager, Doc ID 03-603324 available at <http://support.avaya.com>.*
- [7] *Installing and Upgrading Avaya Aura™ Session Manager 6.0, Doc ID 03-603324, available at <http://support.avaya.com>.*
- [8] *Installing and Upgrading Avaya Aura™ System Manager 6.0, available at <http://support.avaya.com>.*
- [9] *Maintaining and Troubleshooting Avaya Aura™ Session Manager 6.0, Doc ID 03-603321, available at <http://support.avaya.com>.*

### AudioCodes Mediant 2000 Gateway

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

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