



**Application Notes for Configuring Avaya Aura<sup>®</sup>  
Communication Manager R6.0.1, Avaya Aura<sup>®</sup> Session  
Manager R6.1 and Avaya Session Border Controller  
Advanced for Enterprise to Support BT  
Wholesale/HIPCOM SIP Trunk Service – Issue 1.0**

**Abstract**

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between BT Wholesale (BTW)/HIPCOM's SIP Trunk Service and an Avaya SIP enabled enterprise solution. The Avaya solution consists of Avaya Aura<sup>®</sup> Session Manager, Avaya Aura<sup>®</sup> Communication Manager and Avaya Session Border Controller Advanced for Enterprise. BT is a member of the DevConnect Service Provider program.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

# 1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between BT Wholesale/HIPCOM SIP Trunk Service and an Avaya SIP enabled enterprise solution. The Avaya solution consists of Avaya Aura<sup>®</sup> Session Manager, Avaya Aura<sup>®</sup> Communication Manager configured as an Evolution Server and Avaya Session Border Controller Advanced for Enterprise. Customers using this Avaya SIP-enabled enterprise solution with the BT Wholesale/HIPCOM SIP Trunk Service are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. This approach generally results in lower cost for the enterprise.

## 2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of Session Manager and Communication Manager. The enterprise site was configured to use the SIP Trunk Service provided by BTW/HIPCOM.

### 2.1. Interoperability Compliance Testing

The interoperability test included the following:

- SIP Registration was enabled on the Avaya Session Border Controller for Enterprise and registration was tested.
- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by BTW/HIPCOM. Incoming PSTN calls were made to H.323, SIP, Digital and Analogue telephones at the enterprise.
- Outgoing calls from the enterprise site were completed via BTW/HIPCOM to PSTN destinations. Outgoing calls from the enterprise to the PSTN were made from H.323, SIP, Digital and Analogue telephones.
- Calls using G.729, G.711A and G.711Mu codec's.
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using the T.38 mode.
- DTMF transmission using RFC 2833 with successful Vector navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction.
- Call coverage and call forwarding for endpoints at the enterprise site.

## 2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the BTW/HIPCOM SIP Trunk Service with the following observations:

- All tests were completed using H.323, SIP, Digital and Analogue phone types. The Avaya one-X® Communicator was used to test Soft client functionality.
- No inbound toll free numbers were tested, however routing of inbound DID numbers and the relevant number translation was successfully tested.
- Routing to emergency numbers (such as 999) was not tested.
- Direct IP-to-IP media (also known as “shuffling”) with SIP and H.323 telephones was turned off during this test.
- T.38fax must be negotiated using a SIP domain name specified by HIPCOM that is resolveable by the enterprise when the re-invite comes from HIPCOM.

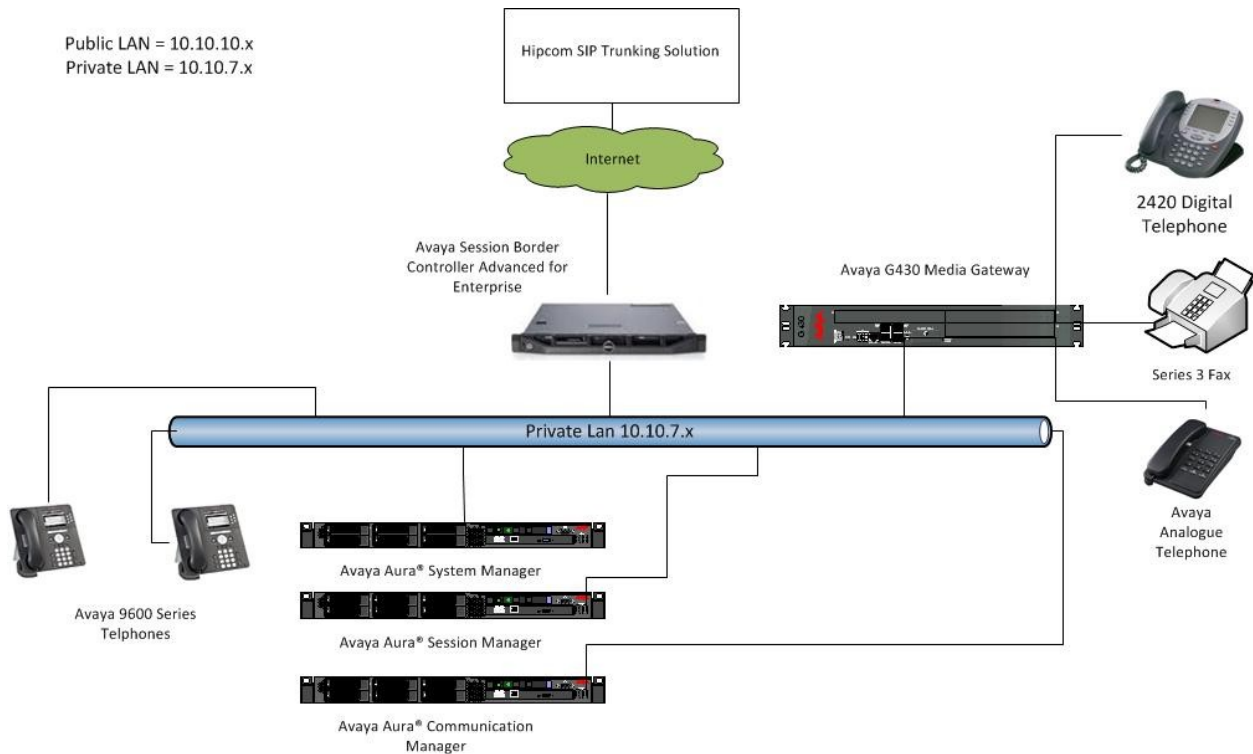
## 2.3. Support

For technical support on BTW/HIPCOM products please refer to the following websites:

<http://www.hipcom.co.uk/support> or <http://ipvoicesupport.btwholesale.com>

### 3. Reference Configuration

**Figure 1** illustrates the test configuration. The test configuration shows an enterprise site connected to the BTW/HIPCOM SIP Trunk Service. Located at the enterprise site is a Session Manager and Communication Manager. Endpoints are Avaya 9600 series IP telephones, Avaya 2400 series Digital Telephone, an Analogue Telephone and Fax Machine. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.



**Figure 1: BTW/HIPCOM SIP Solution Topology**

## 4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software
Avaya S8800 Server	Avaya Aura® Communication Manager R6.0.1 (R016x.00.1.510.1-19009)
Avaya G430 Media Gateway MM711 Analogue MM712 Digital	HW31 FW093 HW07 FW009
Avaya S8800 Server	Avaya Aura® Session Manager R6.1 (6.1.4.0.614005)
Avaya S8800 Server	Avaya Aura® System Manager R6.1 (6.1.0.0.7345-6.1.5.115) Update revision No: 6.1.8.1.1551
Dell R310 running Avaya Session Border Controller Advanced for Enterprise.	Avaya Session Border Controller Advanced for Enterprise R4.0.5.Q02
Avaya 9620 Phone (H.323)	3.11
Avaya 9620 Phone (SIP)	2.6.4.0
Avaya 2420 Digital Phone	N/A
Analog Phone	N/A
BTW/HIPCOM SIP Trunk Service	Acme Packet 4500 Net-Net SBC ver SCX6.1.0 Broadsoft - ver 14 Service Pack 9 Configuration version - HIPCOM v8.1

## 5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager for SIP Trunking. SIP trunks are established between Communication Manager and Session Manager. These SIP trunks will carry SIP Signaling associated with BTW/HIPCOM SIP Trunk Service. For incoming calls, the Session Manager receives SIP messages from BTW/HIPCOM and directs the incoming SIP messages to Communication Manager. Once the message arrives at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects a SIP trunk, the SIP signaling is routed to the Session Manager. The Session Manager directs the outbound SIP messages to the BTW/HIPCOM network. Communication Manager configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. The general installation of the Avaya S8800 Server and Avaya G430 Media Gateway is presumed to have been previously completed and is not discussed here.

### 5.1. Confirm System 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. Use the **display system-parameters customer-options** command and on **Page 2**, verify that the **Maximum Administered SIP Trunks** supported by the system is sufficient for the combination of trunks to the BTW/HIPCOM network, and any other SIP trunks used.

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	3
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	0
Maximum Video Capable IP Softphones:	18000	0
<b>Maximum Administered SIP Trunks:</b>	<b>24000</b>	<b>30</b>

On **Page 4**, verify that **IP Trunks** field 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? y
    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? n                                           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? n
                                IP Trunks? y

IP Attendant Consoles? y
(NOTE: You must logoff & login to effect the permission changes.)
```

## 5.2. Administer IP Node Names

The node names defined here will be used in other configuration screens to define a SIP signaling group between Communication Manager and Session Manager. In the **IP Node Names** form, assign the node **Name** and **IP Address** for the Session Manager. In this case, **sm100** and **10.10.7.61** are the **Name** and **IP Address** for the Session Manager. Also note the **procr** name as this is the interface that Communication Manager will use as the SIP signaling interface to Session Manager.

```
display node-names ip
                                IP NODE NAMES

Name          IP Address
procr         10.10.7.52
sm100         10.10.7.61
default       0.0.0.0
```

### 5.3. Administer IP Network Region

Use the **change ip-network-region 1** command to set the following values:

- The **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **lab.ic.static.hipcom.co.uk**.
- By default, **IP-IP Direct Audio** (both **Intra-** and **Inter-Region**) is set to **yes** to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set **1** will be used.

```
change ip-network-region 1                                     Page 1 of 20
                                                           IP NETWORK REGION
Region: 1
Location: 1           Authoritative Domain: lab.ic.static.hipcom.co.uk
Name: Default NR
MEDIA PARAMETERS                                           Intra-region IP-IP Direct Audio: yes
Codec Set: 1                                               Inter-region IP-IP Direct Audio: yes
UDP Port Min: 35000                                       IP Audio Hairpinning? n
UDP Port Max: 50001
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
```

### 5.4. Administer IP Codec Set

Use the **change ip-codec-set** command for the codec set specified in the **IP Network Region** form. Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test, the codec's supported by BTW/HIPCOM were configured, namely **G.711A**, **G.711MU** and **G.729**.

```
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.711A   n                     2          20
2: G.729    n                     2          20
3: G.711MU  n                     2          20
```



BTW/HIPCOM SIP Trunk Service supports the T.38 fax protocol. Configure the T.38 fax protocol by setting the **Fax Mode** to **t.38-standard** on **Page 2** of the codec set form as shown below.

change ip-codec-set 1		Page 2 of 2
IP Codec Set		
Allow Direct-IP Multimedia? n		
<b>FAX</b>	<b>Mode</b>	Redundancy
	<b>t.38-standard</b>	<b>0</b>
Modem	off	0
TDD/TTY	US	3
Clear-channel	n	0

## 5.5. Administer SIP Signaling Groups

This signaling group (and trunk group) will be used for inbound and outbound PSTN calls to BTW/HIPCOM SIP Trunk Service and will be configured using TCP (Transmission Control Protocol) and the default tcp port of **5060**. Configure the **Signaling Group** using the **add signaling-group n** command as follows:

- Set the **Group Type** field to **sip**
- The **Transport Method** field is set to **tcp**
- Set the **Near-end Node Name** to the processor interface (node name **procr**). This value is taken from the **IP Node Names** form shown in **Section 5.2**
- Set the **Far-end Node Name** to the node name defined for the Session Manager (node name **smp100**), also shown in **Section 5.2**
- Ensure that the recommended TCP port value of **5060** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields
- In the **Far-end Network Region** field, enter the IP Network Region configured in **Section 5.3**. This field logically establishes the **far-end** for calls using this signaling group as network region 1
- Set the **Far-end-Domain** to BTW/HIPCOM domain name, in this case **lab.ic.static.hipcom.co.uk**
- The **Direct IP-IP Audio Connections** field is set to **y**
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833

The default values for the other fields may be used.

```
add signaling-group 1
                                SIGNALING GROUP

Group Number: 1                Group Type: sip
                                Transport Method: tcp

IMS Enabled? n

Near-end Node Name: procr      Far-end Node Name: sm100
Near-end Listen Port: 5060     Far-end Listen Port: 5060
                                Far-end Network Region: 1
Far-end Domain: lab.ic.static.hipcom.co.uk

Incoming Dialog Loopbacks: eliminate
                                Bypass If IP Threshold Exceeded? n
                                RFC 3389 Comfort Noise? n
DTMF over IP: rtp-payload      Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3
                                IP Audio Hairpinning? n
                                Enable Layer 3 Test? n
                                Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? y
                                Alternate Route Timer(sec): 6
```

## 5.6. Administer SIP Trunk Group

A trunk group is associated with the signaling group described in **Section 5.5**. Configure the trunk group using the **add trunk-group x** command, where **x** is an available trunk group. On **Page 1** of this form:

- Set the **Group Type** field to **sip**
- Choose a descriptive **Group Name**
- Specify a trunk access code (**TAC**) consistent with the dial plan, i.e. **135**
- The **Direction** is set to **two-way** to allow incoming and outgoing calls
- Set the **Service Type** field to **tie**
- Specify the signaling group associated with this trunk group in the **Signaling Group** field as previously configured in **Section 5.5**
- Specify the **Number of Members** supported by this SIP trunk group

add trunk-group 1		Page 1 of 21	
TRUNK GROUP			
Group Number: 1	Group Type: sip	CDR Reports: y	
Group Name: smpub	COR: 1	TN: 1	TAC: 135
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: tie	Auth Code? n		
Signaling Group: 1			
Number of Members: 30			

On **Page 2** of the trunk-group form the **Preferred Minimum Session Refresh Interval (sec)** field should be set to a value mutually agreed with BTW/HIPCOM to prevent unnecessary SIP messages during call setup.

add trunk-group 1		Page 2 of 21	
Group Type: sip			
TRUNK PARAMETERS			
Unicode Name: auto			
		Redirect On OPTIM Failure: 8000	
SCCAN? n	Digital Loss Group: 18		
Preferred Minimum Session Refresh Interval(sec): 1800			

On **Page 3**, set the **Numbering Format** field to **private**. This allows the number to be sent to BTW/HIPCOM without the + used in the E164 numbering format.

add trunk-group 1		Page 3 of 21
TRUNK FEATURES		
ACA Assignment? n	Measured: none	Maintenance Tests? y
<b>Numbering Format: private</b>		
	UI Treatment: service-provider	
	Replace Restricted Numbers? n	
	Replace Unavailable Numbers? n	
<b>Modify Tandem Calling Number:</b>		

On **Page 4**, set the **Mark Users as Phone** to **y**, this field inserts a parameter to SIP requests indicating to any receiving SIP entity that the user part of the request URI should be treated as a telephone number. Set **Send Transferring Party Information** to **y**, to allow trunk to trunk transfers. Set **Telephone Event Payload Type** to **120**.

add trunk-group 1		Page 4 of 21
PROTOCOL VARIATIONS		
<b>Mark Users as Phone? y</b>		
Prepend '+' to Calling Number? n		
<b>Send Transferring Party Information? y</b>		
Network Call Redirection? n		
Send Diversion Header? n		
Support Request History? y		
<b>Telephone Event Payload Type: 120</b>		

## 5.7. Administer Calling Party Number Information

### 5.7.1. Set Private Unknown Numbering

Use the **change private-unknown-numbering** command to configure Communication Manager to send the calling party number. In the sample configuration, all stations with a **4-digit** extension beginning with **1** will send the calling party number **44203xxxxxx** to BTW/HIPCOM SIP Trunk Service. This calling party number will be sent in the SIP From, Contact and PAI headers, and displayed on display-equipped PSTN telephones. Public DID numbers have been masked for security purposes.

change private-unknown-numbering 0					Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT					
Ext	Ext	Trk	CPN	Total	
Len	Code	Grp(s)	Prefix	CPN	
				Len	
<b>4</b>	<b>1</b>	<b>1</b>	<b>44203xxxxxx</b>	<b>12</b>	Total Administered: 1
					Maximum Entries: 240

## 5.8. Administer Route Selection for Outbound Calls

In these Application Notes, the Automatic Route Selection (ARS) feature will be used to route outbound calls via the SIP trunk to BTW/HIPCOM SIP Trunk Service. In the sample configuration, the single digit **9** is used as the ARS access code. Avaya telephone users will dial **9** to reach an outside line. Use the **change feature-access-codes** command to configure or observe **9** as the **Auto Route Selection (ARS) - Access Code 1**.

<b>change feature-access-codes</b>		Page 1 of 9
FEATURE ACCESS CODE (FAC)		
Abbreviated Dialing List1 Access Code:		
Abbreviated Dialing List2 Access Code:		
Abbreviated Dialing List3 Access Code:		
Abbreviated Dial - Prgm Group List Access Code:		
Announcement Access Code: *37		
Answer Back Access Code: *12		
Attendant Access Code:		
Auto Alternate Routing (AAR) Access Code: 7		
<b>Auto Route Selection (ARS) - Access Code 1: 9</b>		Access Code 2: *99
Automatic Callback Activation:		Deactivation:
Call Forwarding Activation Busy/DA: *87	All: *88	Deactivation: #88
Call Forwarding Enhanced Status:	Act:	Deactivation:

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. A small sample of dial patterns are illustrated here. Further administration of ARS is beyond the scope of these Application Notes. The example entries shown will match outgoing calls to numbers beginning **0** or **00**. Calls are sent to **Route Pattern 1**, which contains the previously configured SIP Trunk Group.

<b>change ars analysis 02</b>		Page 1 of 2
ARS DIGIT ANALYSIS TABLE		
Location: all		Percent Full: 1
Dialed String	Total Min Max	Route Pattern
0	11 11	1
00	13 13	1
	Call Type	Node Num ANI Req'd
	pubu	n
	pubu	n

Use the **change route-pattern** command to add the SIP trunk group to the route pattern that ARS selects. In this configuration, route pattern **1** is used to route calls to trunk group 1.

change route-pattern 1													Page 1 of 3						
Pattern Number: 1 Pattern Name: tosm100																			
SCCAN? n Secure SIP? n																			
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted						DCS/	IXC					
No			Mrk	Lmt	List	Del	Digits						QSIG						
							Dgts						Intw						
1:	1	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. <b>Numbering</b> LAR																			
0	1	2	M	4	W	Request							Dgts	<b>Format</b>					
													Subaddress						
1:	y	y	y	y	y	n	n	rest					none						
2:	y	y	y	y	y	n	n	rest					none						

## 5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DID calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from BTW/HIPCOM can be manipulated as necessary to route calls to the desired extension. In the examples used in the compliance testing, the incoming DID numbers provided by BTW/HIPCOM correlate to the internal extensions assigned within Communication Manager. The entries displayed below translates incoming DID numbers 44203xxxxxx to a 4 digit extension by deleting all of the incoming digits and inserting an extension. Public DID numbers have been masked for security purposes.

change inc-call-handling-trmt trunk-group 1					Page	1 of	3
INCOMING CALL HANDLING TREATMENT							
Service/	Number	Number	Del Insert				
Feature	Len	Digits					
public-ntwrk	12	44203xxxxxxx	all	1306			
public-ntwrk	12	44203xxxxxxx	all	1307			

Save Communication Manager changes by enter **save translation** to make them permanent.

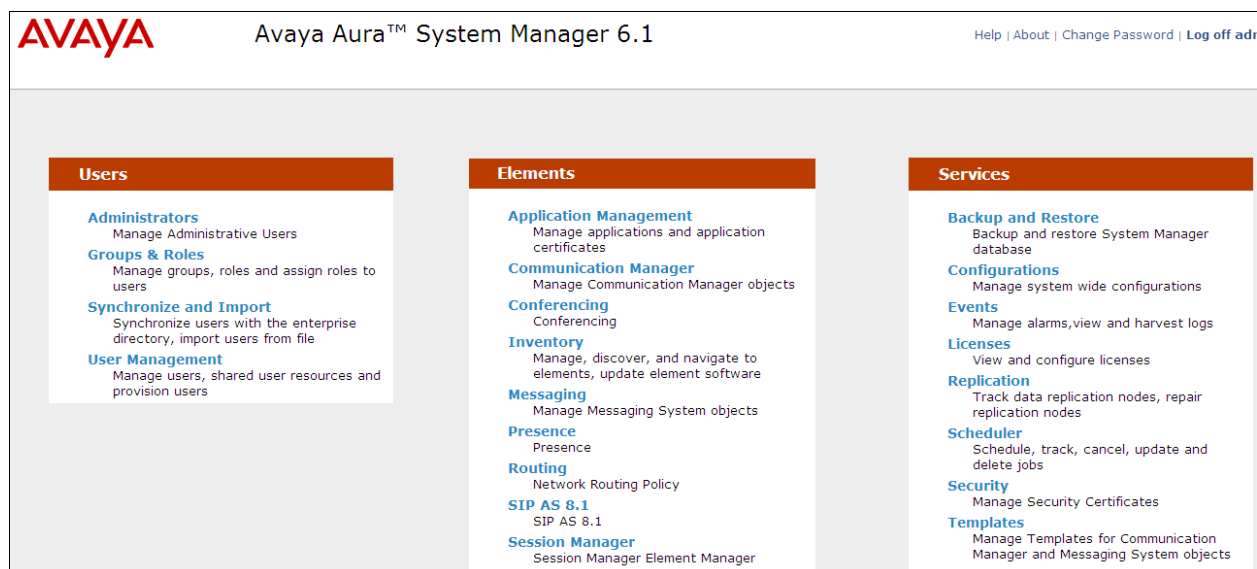
## 6. Configuring Avaya Aura® Session Manager

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 Routing Policies
- Administer Dial Patterns
- Administer Avaya Aura® Communication Manager as Managed Element
- Administer Application for Avaya Aura® Communication Manager
- Administer Application Sequence for Avaya Aura® Communication Manager
- Administer SIP Extensions

### 6.1. Log in to Avaya Aura® System Manager

Access the System Manager using a Web Browser by entering **http://<FQDN>/SMGR**, where **<FQDN>** is the fully qualified domain name of System Manager. Log in using appropriate credentials (not shown) and the Home tab will be presented with menu options shown below.



## 6.2. Administer SIP Domain

To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu (not shown) and in the resulting tab select **SIP Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name (e.g., **lab.ic.static.hipcom.co.uk**) and optionally a description for the domain in the **Notes** field. Click **Commit** to save changes (not shown).

Home / Elements / Routing / Domains - Domain Management

Domain Management

Edit New Duplicate Delete More Actions

1 Item Refresh Filter:

<input type="checkbox"/>	Name	Type	Default	Notes
<input type="checkbox"/>	lab.ic.static.hipcom.co.uk	sip	<input type="checkbox"/>	

Select : All, None

## 6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for the purposes of bandwidth management. One location is added to the sample configuration for the enterprise SIP entities. On the **Routing** tab select **Locations** from the left hand menu. Under **General**, in the **Name** field enter an informative name for the location. Scroll to the bottom of the page and under **Location Pattern**, click **Add**, then enter an **IP Address Pattern** in the resulting new row, \* is used to specify any number of allowed characters at the end of the string. Below is the location configuration used for the simulated enterprise.

Home / Elements / Routing / Locations - Location Details

Location Details

Commit

Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth.  
See Session Manager -> Session Manager Administration -> Global Setting

General

\* Name: SPLab7

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec

Total Bandwidth:

Per-Call Bandwidth Parameters

\* Default Audio Bandwidth: 80 Kbit/sec

Location Pattern

Add Remove

3 Items Refresh Filter: E

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.10.9.*	
<input type="checkbox"/>	* 10.10.8.*	
<input type="checkbox"/>	* 10.10.7.*	



## 6.4. Administer SIP Entities

A SIP Entity must be added for each SIP-based telephony system supported by a SIP connection to the Session Manager. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown). The following will need to be entered for each SIP Entity.

Under **General**:

- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signaling interface on the connecting system
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager SIP entity and **Gateway** for the SBC SIP entity
- In the **Location** field select the appropriate location from the drop down menu
- In the **Time Zone** field enter the time zone for the SIP Entity

In this configuration there are three SIP Entities.

- Session Manager SIP Entity
- Communication Manager SIP Entity
- Session Border Controller SIP Entity

### 6.4.1. Avaya Aura® Session Manager SIP Entity

The following screens show the SIP entity for Session Manager. The **FQDN or IP Address** field is set to the IP address of the Session Manager SIP signaling interface.

The screenshot displays the 'SIP Entity Details' configuration page for a Session Manager SIP Entity. The left sidebar shows a navigation menu with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and includes a 'Commit' button in the top right. The 'General' tab is active, showing the following fields:

- Name:** ASM61
- FQDN or IP Address:** 10.10.7.61
- Type:** Session Manager (dropdown menu)
- Notes:** (empty text field)
- Location:** SPLab7 (dropdown menu)
- Outbound Proxy:** (empty dropdown menu)
- Time Zone:** Etc/GMT (dropdown menu)
- Credential name:** (empty text field)
- SIP Link Monitoring:** Use Session Manager Configuration (dropdown menu)

Below the 'General' tab, there are sections for 'SIP Link Monitoring' and 'Entity Links', which are currently collapsed.

The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

- In the **Port** field enter the port number on which the system listens for SIP requests
- In the **Protocol** field enter the transport protocol to be used for SIP requests
- In the **Default Domain** field, from the drop down menu select **lab.ic.static.hipcom.co.uk** as the default domain

**Port**

3 Items Refresh Filter: Enable

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	lab.ic.static.hipcom.co.uk	
<input type="checkbox"/>	5060	UDP	lab.ic.static.hipcom.co.uk	
<input type="checkbox"/>	5061	TLS	lab.ic.static.hipcom.co.uk	

Select : All, None

### 6.4.2. Avaya Aura® Communication Manager SIP Entity

The following screens show the SIP entity for Communication Manager. The **FQDN or IP Address** field is set to the IP address of the Interface that will be providing SIP signaling. The entity **Type** is set to **CM**.

Home / Elements / Routing / SIP Entities - SIP Entity Details

**SIP Entity Details**

**General**

\* Name: CMEVO

\* FQDN or IP Address: 10.10.7.52

Type: CM

Notes:

Adaptation:

Location: SPLab7

Time Zone: Etc/GMT

Override Port & Transport with DNS SRV: ☐

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

Credential name:

Call Detail Recording: none

**SIP Link Monitoring**

SIP Link Monitoring: Use Session Manager Configuration

### 6.4.3. Avaya Session Border Controller Advanced for Enterprise SIP Entity

The following screen shows the SIP entity for the Avaya Session Border Controller Advanced for Enterprise. The **FQDN or IP Address** field is set to the IP address of the public interface administered in **Section 7** of this document. Choose the **Adaptation; remove 00** from the drop down list.

The screenshot displays the 'SIP Entity Details' configuration page for the Avaya Session Border Controller Advanced for Enterprise. The page is titled 'Home / Elements / Routing / SIP Entities - SIP Entity Details' and includes a 'Commit' button in the top right corner. A left-hand navigation menu lists various configuration options: Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is divided into two sections: 'General' and 'SIP Link Monitoring'. The 'General' section contains several fields: 'Name' (ASBCAE), 'FQDN or IP Address' (10.10.7.220), 'Type' (Gateway), 'Notes' (empty), 'Adaptation' (remove 00), 'Location' (SPLab7), 'Time Zone' (Etc/GMT), 'Override Port & Transport with DNS SRV' (unchecked), 'SIP Timer B/F (in seconds)' (4), 'Credential name' (empty), and 'Call Detail Recording' (none). The 'SIP Link Monitoring' section contains a single dropdown menu for 'SIP Link Monitoring' set to 'Use Session Manager Configuration'.

Routing

Domains Routing

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / SIP Entities - SIP Entity Details

SIP Entity Details

Commit

General

\* Name: ASBCAE

\* FQDN or IP Address: 10.10.7.220

Type: Gateway

Notes:

Adaptation: remove 00

Location: SPLab7

Time Zone: Etc/GMT

Override Port & Transport with DNS SRV: ☐

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

Credential name:

Call Detail Recording: none

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

## 6.5. Administer Entity Links

A SIP trunk between a Session Manager and another system is described by an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button . Fill in the following fields in the new row that is displayed.

- In the **Name** field enter an informative name
- In the **SIP Entity 1** field select **SessionManager**
- In the **Port** field enter the port number to which the other system sends its SIP requests
- In the **SIP Entity 2** field enter the other SIP Entity for this link, created in **Section 6.4**
- In the **Port** field enter the port number to which the other system expects to receive SIP requests
- Select the **Trusted** tick box to make the other system trusted
- In the **Protocol** field enter the transport protocol to be used to send SIP requests

Click **Commit** (not shown) to save changes. The following screen shows the Entity Links used in this configuration.

SIP Entities	1 Item   Refresh <span style="float: right;">Filter: t</span>							
Entity Links	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
Time Ranges	* toCM	* ASM61	TCP	* 5060	* CMEVO	* 5060	Trusted	
Routing Policies								
Dial Patterns								
Regular Expressions								
Defaults	* Input Required							Commit

SIP Entities	1 Item   Refresh <span style="float: right;">Filter: E</span>							
Entity Links	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
Time Ranges	* toASBCAE	* ASM61	TCP	* 5060	* ASBCAE	* 5060	Trusted	
Routing Policies								
Dial Patterns								
Regular Expressions								
Defaults	* Input Required							Commit

## 6.6. Administer Routing Policies

Routing policies must be created to direct how calls will be routed to a system. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown).

Under **General**:

- Enter an informative name in the **Name** field
- Under **SIP Entity as Destination**, click **Select**, and then select the appropriate SIP entity to which this routing policy applies

The following screen shows the routing policy for Communication Manager:

Routing Policy Details			
General			
* Name: TO CMEVO			
Disabled: <input type="checkbox"/>			
Notes:			
SIP Entity as Destination			
Select			
Name	FQDN or IP Address	Type	Notes
CMEVO	10.10.7.52	CM	

The following screen shows the routing policy for Avaya Session Border Controller Advanced for Enterprise:

Routing Policy Details			
General			
* Name: to_ASBCAE			
Disabled: <input type="checkbox"/>			
Notes:			
SIP Entity as Destination			
Select			
Name	FQDN or IP Address	Type	Notes
ASBCAE	10.10.7.220	Gateway	

Commit

## 6.7. Administer Dial Patterns

A dial pattern must be defined to direct calls to the appropriate telephony system. To configure a dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

Under **General**:

- In the **Pattern** field enter a dialed number or prefix to be matched
- In the **Min** field enter the minimum length of the dialed number
- In the **Max** field enter the maximum length of the dialed number
- In the **SIP Domain** field select the domain configured in **Section 6.2**

Under **Originating Locations and Routing Policies**. Click **Add**, in the resulting screen (not shown) under **Originating Location** select **Locations** created in **Section 6.3** and under **Routing Policies** select one of the routing policies defined in **Section 6.6**. Click **Select** button to save (not shown). The following screen shows an example dial pattern configured for BTW/HIPCOM SIP Trunk Service.

Home / Elements / Routing / Dial Patterns - Dial Pattern Details

Dial Pattern Details Commit

**General**

\* Pattern: 0

\* Min: 11

\* Max: 11

Emergency Call: ☐

SIP Domain: -ALL-

Notes:

**Originating Locations and Routing Policies**

Add Remove

1 Item Refresh Filter: E

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Notes
<input type="checkbox"/>	SPlab7		to_ASBCAE	0	<input type="checkbox"/>	ASBCAE	

The following screen shows an example dial pattern configured for the Communication Manager.

**Dial Pattern Details** Commit

**General**

\* Pattern: 44203551

\* Min: 10

\* Max: 12

Emergency Call: ☐

SIP Domain: -ALL-

Notes:

**Originating Locations and Routing Policies**

Add Remove

1 Item Refresh Filter: E

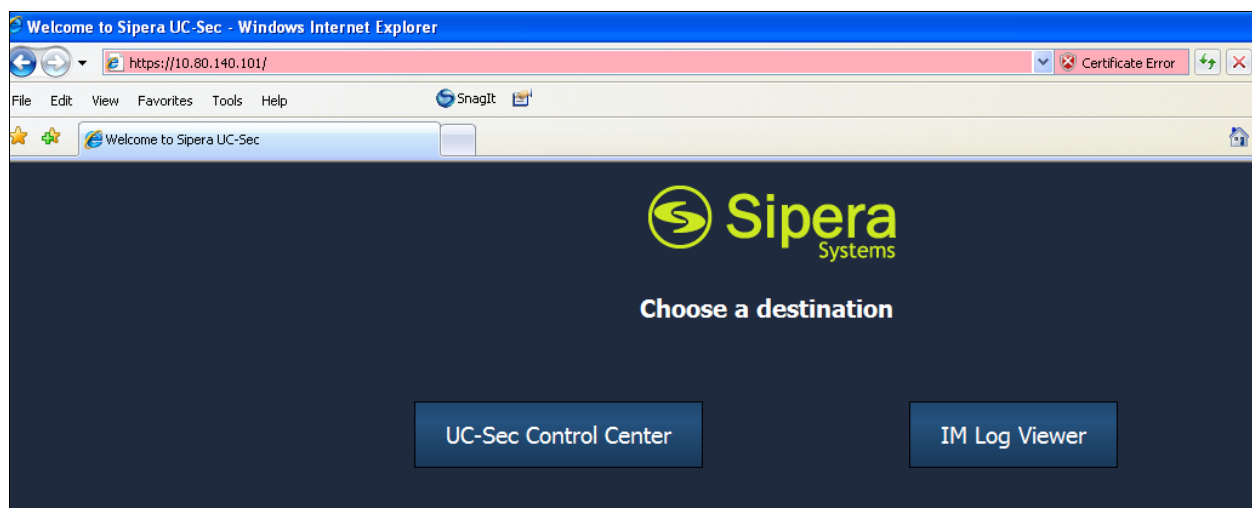
<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Notes
<input type="checkbox"/>	-ALL-	Any Locations	TO CMEVO	0	<input type="checkbox"/>	CMEVO	

## 7. Avaya Session Border Controller Advanced for Enterprise Configuration

This section provides the procedures for configuring Session Border Controller Advanced for Enterprise.

### 7.1. Accessing UC-Sec Control Centre

Access the web interface by typing <https://x.x.x.x> (where x.x.x.x is the management IP of the ASBCAE)



Select **UC-Sec Control Center** and enter the login ID and password.





## 7.2. Global Profiles

Global Profiles allows for configuration of parameters across all UC-Sec appliances.

### 7.2.1. Server Interworking Avaya Side

Server Internetworking allows you to configure and manage various SIP call server-specific capabilities such as call hold and T38.

- Select **Global Profiles** → **Server Interworking**
- Select **Add Profile**
- On the **General** tab:
  - Enter profile name: **SM7-HCOM**
  - Check **Hold Support: RFC3264**
  - Check **T38 Support Yes**
  - All other options on the **General** tab can be left at default
  - Click **Next**
- At the **Privacy** tab
  - Click **Next**
- At the **Internetworking Profile** tab
  - Click **Next**.
- On the **Advanced** tab
  - Click **Next**
  - Click **Finish** (not shown)

The screen below is a result of the details configured above.

The screenshot displays the UC-Sec Control Center interface. On the left is a navigation tree with categories like Administration, System Management, Global Profiles, and Troubleshooting. The 'Global Profiles' section is expanded, showing 'Server Interworking' as the selected option. The main panel is titled 'Global Profiles > Server Interworking: SM7-HCOM'. It features a list of 'Interworking Profiles' on the left, including 'cs2100', 'avaya-ru', 'OCS-Edge-Server', 'cisco-ccm', 'cups', 'Sipera-Halo', 'OCS-FrontEnd-Server', 'KPN-SM9', 'SM9-KPN', 'HCOM-SM7', and 'SM7-HCOM' (which is highlighted). To the right of this list are buttons for 'Add Profile', 'Rename Profile', and 'Clone Profile'. The main configuration area has tabs for 'General', 'Timers', 'URI Manipulation', 'Header Manipulation', and 'Advanced'. The 'General' tab is active, showing a table of parameters and their values. Below this, there are sections for 'Privacy' and 'DTMF' settings. An 'Edit' button is located at the bottom right of the configuration area.

General	
Hold Support	RFC3264
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
T.38 Support	Yes
URI Scheme	SIP
Via Header Format	RFC3261

Privacy	
Privacy Enabled	No
User Name	
P-Asserted-Identity	No
P-Preferred-Identity	No
Privacy Header	

DTMF	
DTMF Support	None

### 7.2.2. Server Interworking – HIPCOT side

Server Interworking allows you to configure and manage various SIP call server-specific capabilities such as call hold and T38.

- Select **Global Profiles** from the menu on the left-hand side
- Select the **Server Interworking**
- Select **Add Profile**
- On the **General** tab:
  - Enter profile name: **HCOM-SM7**
  - **Check T38 Support: Yes**
  - All other options on the **General** tab can be left at default
  - Click **Next**
- At the **Privacy** tab
  - Click **Next**
- At the **Interworking Profile** tab
  - Click **Next**.
- On the **Advanced** tab
  - Click **Next**
  - Click **Finish**

The screen below is a result of the details configured above.

The screenshot displays the UC-Sec Control Center interface. On the left is a navigation tree with categories like Administration, System Management, Global Profiles, and Server Interworking. The 'Server Interworking' option is selected. The main area shows the configuration for the 'HCOM-SM7' profile. It includes tabs for General, Timers, URI Manipulation, Header Manipulation, and Advanced. The 'General' tab is active, showing a table of settings for various SIP call server-specific capabilities.

General	
Hold Support	RFC2543
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
T.38 Support	Yes
URI Scheme	SIP
Via Header Format	RFC3261

Privacy	
Privacy Enabled	No
User Name	
P-Asserted-Identity	No
P-Preferred-Identity	No
Privacy Header	

DTMF	
DTMF Support	None

Buttons: Add Profile, Rename Profile, Clone Profile, Edit

### 7.2.3. Routing – Avaya side

The Routing Profile allows you to manage parameters related to routing SIP signaling messages.

- Select **Global Profiles → Routing**
- Select **Add Profile**
- Enter Profile Name: **Call\_Server\_SM7**
- Click **Next**
- **Next Hop Server 1: 10.10.7.61** (Session Manager IP address)
- Select **Routing Priority Based on Next Hop Server**
  - Select **Use Next Hop for In-Dialog Messages**
  - **Outgoing Transport: TCP**
  - Click **Finish** (not shown)

The screen below is a result of the details configured above.

The screenshot shows the UC-Sec Control Center interface. On the left is a navigation tree with 'Routing' selected. The main panel is titled 'Global Profiles > Routing: Call\_Server\_SM7'. It features a 'Routing Profiles' list on the left with 'Call\_Server\_SM7' highlighted. The main area shows the 'Routing Profile' configuration for 'Call\_Server\_SM7'. At the top, there are buttons for 'Add Profile', 'Rename Profile', 'Clone Profile', and 'Delete Profile'. Below these is a yellow bar with the text 'Click here to add a description.' and an 'Add Routing Rule' button. A table displays the routing rule configuration:

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport
1	*	10.10.7.61	--	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	TCP

### 7.2.4. Routing – HIPCOM side

The Routing Profile allows you to manage parameters related to routing SIP signaling messages.

- Select **Global Profiles → Routing**
- Select **Add Profile**
- Enter Profile Name: **Trunk\_Server\_HCOM**
- Click **Next**
- **Next Hop Server 1: 10.10.10.20** (IP Address provided by HIPCOM)
  - Select **Routing Priority Based on Next Hop Server**
  - Select **Use Next Hop for In-Dialog Messages**
  - **Outgoing Transport: UDP**
  - Click **Finish** (not shown)

The screen below is a result of the details configured above.

The screenshot shows the UC-Sec Control Center interface. On the left is a navigation tree with 'Routing' selected. The main panel is titled 'Global Profiles > Routing: Trunk\_Server\_HCOM'. It features a 'Routing Profiles' list on the left with 'Trunk\_Server\_HCOM' highlighted. The main area shows the 'Routing Profile' configuration for 'Trunk\_Server\_HCOM'. At the top, there are buttons for 'Add Profile', 'Rename Profile', 'Clone Profile', and 'Delete Profile'. Below these is a yellow bar with the text 'Click here to add a description.' and an 'Add Routing Rule' button. A table displays the routing rule configuration:

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport
1	*	10.10.10.20	--	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	UDP

## 7.2.5. Server Configuration – Avaya SM

The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow you to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, IP Server type, heartbeat signaling parameters and some advanced options.

- Select **Global Profiles** → **Server Configuration**
- Select **Add Profile**
- Enter profile name: **Call\_Server\_SM7**
- On the **Add Server Configuration Profile** Tab:
  - Select Server Type: **Call Server**
  - **IP Address: 10.10.7.61** (Session Manager IP Address)
  - **Supported Transports:** Check **UDP** and **TCP**
  - **TCP Port: 5060**
  - **UDP Port: 5060**
  - Click **Next**
- At the **Authentication** tab
  - Click **Next**
- At the **Heartbeat** tab
  - Click **Next**
- On the **Advanced** Tab
  - Select **HCOM-SM7** for Interworking Profile
  - Click **Next**
- Click **Finish**

The screen below is a result of the details configured above.

The screenshot shows the UC-Sec Control Center interface. On the left is a navigation tree with 'Server Configuration' selected. The main panel is titled 'Global Profiles > Server Configuration: Call\_Server\_SM7'. It features a 'Profile' list on the left with 'Call\_Server\_SM7' highlighted. The right side has tabs for 'General', 'Authentication', 'Heartbeat', and 'Advanced'. The 'General' tab is active, displaying the following configuration:

General	
Server Type	Call Server
IP Addresses / FQDNs	10.10.7.61
Supported Transports	TCP, UDP, TLS
TCP Port	5060
UDP Port	5060
TLS Port	5061

An 'Edit' button is located at the bottom right of the configuration table.

The screenshot shows the same UC-Sec Control Center interface, but with the 'Advanced' tab selected. The configuration details are as follows:

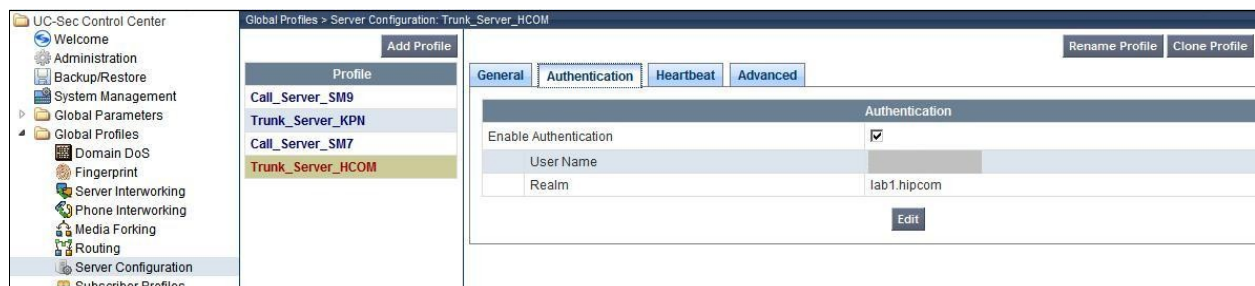
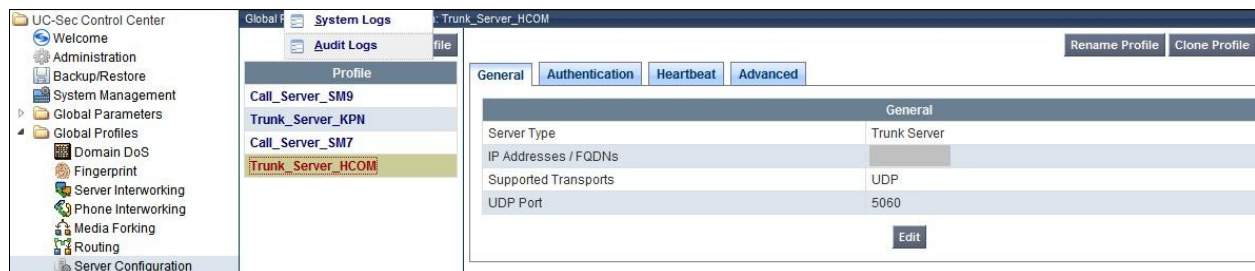
Advanced	
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	HCOM-SM7
TLS Client Profile	None
Signaling Manipulation Script	None
TCP Connection Type	SUBID
UDP Connection Type	SUBID
TLS Connection Type	SUBID

An 'Edit' button is located at the bottom right of the configuration table.

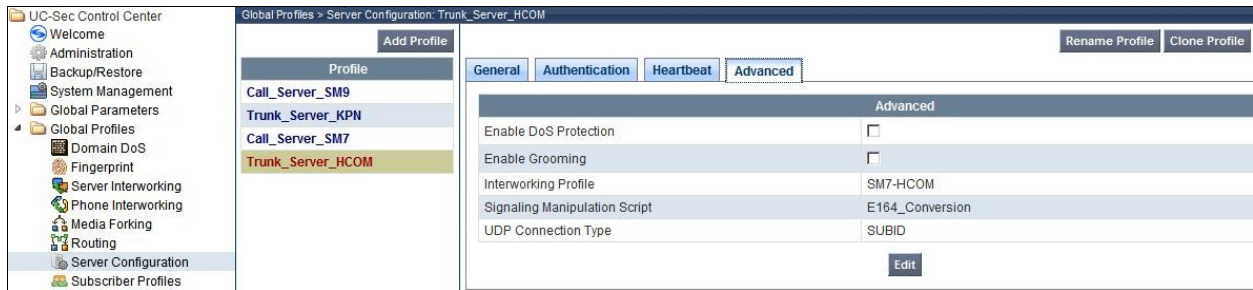
## 7.2.6. Server Configuration– HIPCOT side

The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow you to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, server type, heartbeat signaling parameters and some advanced options.

- Select **Global Profiles** → **Server Configuration**
- Select **Add Profile**
- Enter profile name: **Trunk\_Server\_HCOM**
- On the **Add Server Configuration Profile** Tab:
  - Select Server Type: **Trunk Server**
  - **IP Address: 10.10.10.20** (HIPCOT Trunk Server )
  - **Supported Transports: Check UDP**
  - **UDP Port: 5060**
  - Click **Next**
- At the **Authentication** tab
  - Select **Enable Authentication**
  - Enter the **User Name** and **Realm** provided by HIPCOT
  - At the **Heartbeat** tab
  - Click **Next**.
- On the **Advanced** Tab
  - Select **SM7-HCOM** for interworking profile
  - Click **Next**
- Click **Finish**



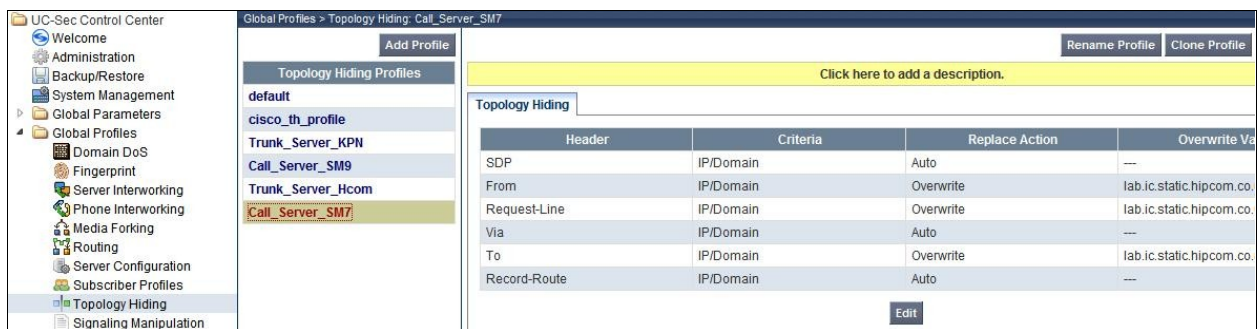




## 7.2.7. Topology Hiding – Avaya side

The **Topology Hiding** screen allows you to manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks.

- Select **Global Profiles → Topology Hiding**
- Click **default** profile and select **Clone Profile**
- **Enter Profile Name: Call\_Server\_SM7**
- Click on **Edit**
- For the Header **To**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**
  - In the **Overwrite Value** column: **lab.ic.static.hipcom.co.uk**
- For the Header **From**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**
  - In the **Overwrite Value** column: **lab.ic.static.hipcom.co.uk**
- For the Header **Request Line**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**
- In the **Overwrite Value** column: **lab.ic.static.hipcom.co.uk**
- Click **Finish** (not shown)



### 7.2.8. Topology Hiding – HIPCOM side

The **Topology Hiding** screen allows you to manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks. Select **Global Profiles** from the menu on the left-hand side

- Select the **Global Profiles → Topology Hiding**
- Click **default** profile and select **Clone Profile**
- Enter Profile Name: **Trunk\_Server\_Hcom**
- For the Header **To**,
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**
  - In the **Overwrite Value** column: **lab.ic.static.hipcom.co.uk**
- For the Header **From**,
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**
  - In the **Overwrite Value** column: **lab.ic.static.hipcom.co.uk**
- For the Header **Request Line**,
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**
- In the **Overwrite Value** column: **lab.ic.static.hipcom.co.uk**
- Click **Finish**

The screenshot shows the UC-Sec Control Center interface. On the left is a navigation tree with 'Global Profiles' expanded and 'Topology Hiding' selected. The main panel is titled 'Global Profiles > Topology Hiding: Trunk\_Server\_Hcom'. It features a list of 'Topology Hiding Profiles' on the left, including 'default', 'cisco\_th\_profile', 'Trunk\_Server\_KPN', 'Call\_Server\_SM9', 'Trunk\_Server\_Hcom' (highlighted), and 'Call\_Server\_SM7'. On the right, there's a form for the 'Trunk\_Server\_Hcom' profile. It has buttons for 'Add Profile', 'Rename Profile', and 'Clone Profile'. Below these is a yellow bar with the text 'Click here to add a description.' The main table is titled 'Topology Hiding' and has columns: 'Header', 'Criteria', 'Replace Action', and 'Overwrite Value'. The table contains the following data:

Header	Criteria	Replace Action	Overwrite Value
SDP	IP/Domain	Auto	---
From	IP/Domain	Overwrite	lab.ic.static.hipcom.co.uk
Request-Line	IP/Domain	Overwrite	lab.ic.static.hipcom.co.uk
Via	IP/Domain	Auto	---
To	IP/Domain	Overwrite	lab.ic.static.hipcom.co.uk
Record-Route	IP/Domain	Auto	---

An 'Edit' button is located at the bottom right of the table.

## 7.3. Device Specific Settings

The Device Specific Settings feature for SIP allows you to view aggregate system information, and manage various device-specific parameters which determine how a particular device will function when deployed in the network. Specifically, you have the ability to define and administer various device-specific protection features such as Message Sequence Analysis (MSA) functionality, end-point and session call flows and Network Management.

- Select **Device Specific Settings → Network Management**
- Enter in the **IP Address** and **Gateway Address** for both the Inside and the Outside interfaces.
- Select the physical interface used in the Interface column

UC-Sec Control Center

Device Specific Settings > Network Management: GSSCPTRNK

UC-Sec Devices

GSSCPTRNK

Network Configuration | Interface Configuration

Modifications or deletions of an IP address or its associated data require an application restart before taking effect. Application restarts can be issued from System Management.

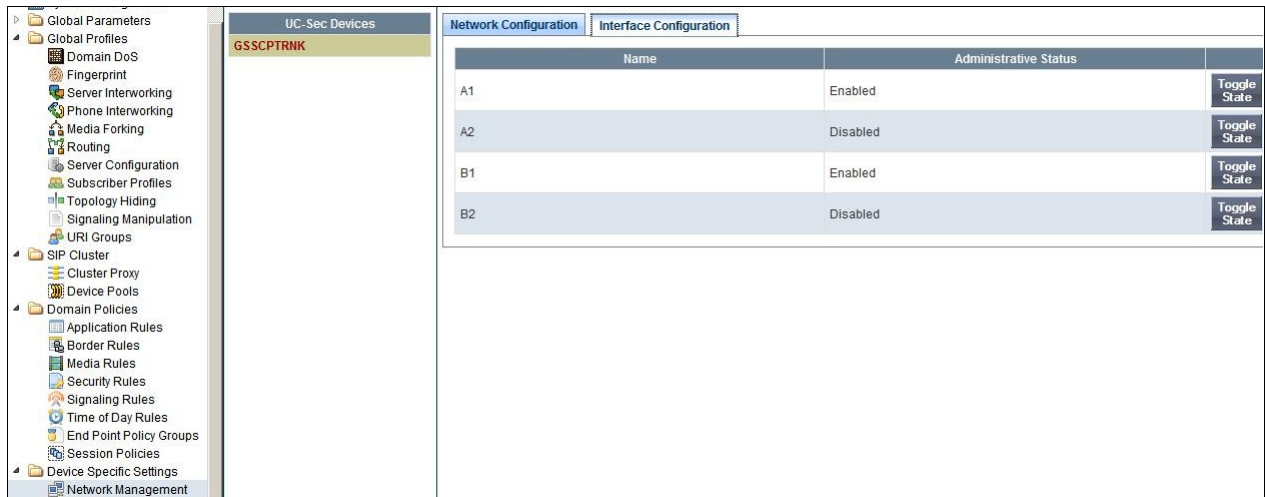
A1 Netmask: 255.255.255.0    A2 Netmask:    B1 Netmask: 255.255.255.128    B2 Netmask:   

Add IP    Save Changes    Clear Changes

IP Address	Public IP	Gateway	Interface
10.10.7.220		10.10.7.1	A1
10.10.10.10		10.10.10.10	B1



Select the **Interface Configuration** tab and enable the state of the physical interfaces being used.

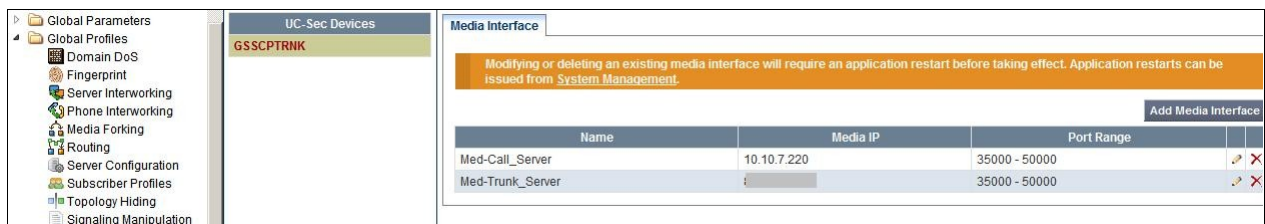


### 7.3.1. Media Interfaces

This section is used to configure the interface and port range used to transport media.

Select **Device Specific Settings** → **Media Interface**

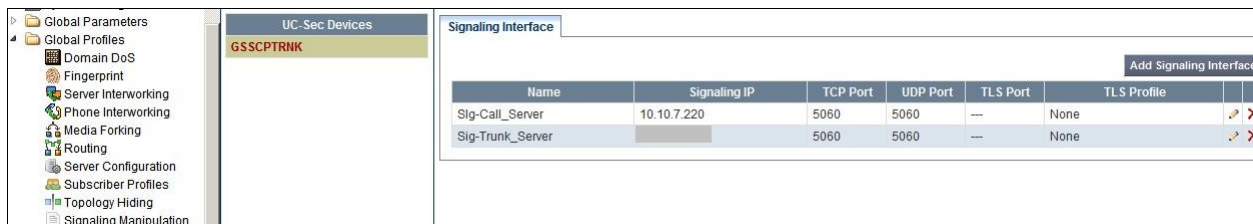
- Select **Add Media Interface**
  - **Name:** Med-Call\_Server
  - **Media IP:** 10.10.7.220 (Internal Address toward the Callserver)
  - **Port Range:** 35000-50000
  - Click **Finish** (not shown)
- Select **Add Media Interface**
  - **Name:** Med-Trunk\_Server
  - **Media IP:** 10.10.10.10 (External Internet Address toward HIPCOM trunk)
  - **Port Range:** 35000-50000
  - Click **Finish** (not shown)



### 7.3.2. Signaling Interface

This section is used to configure the interface and port range used to transport media.

- Select **Device Specific Settings** → **Signaling Interface**
- Select **Add Signaling Interface**
  - **Name:** Sig-Call\_Server
  - **Media IP:** 10.10.7.220 (Internal Address toward the Callserver)
  - **TCP Port:** 5060
  - **UDP Port:** 5060
  - Click **Finish**
- Select **Add Media Interface**
  - **Name:** Sig-Trunk\_Server
  - **Media IP:** 10.10.10.10(External Internet Address toward HIPCOM trunk)
  - **TCP Port:** 5060
  - **UDP Port:** 5060
  - Click **Finish**



Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile		
Sig-Call_Server	10.10.7.220	5060	5060	---	None		
Sig-Trunk_Server		5060	5060	---	None		

### 7.3.3. End Point Flows – Avaya\_SM

This section is used to determine the method and interfaces used to transport sip media and signaling to the Callserver.

- Select **Device Specific Settings** → **Endpoint Flows**
- Select the **Server Flows** tab
- Select **Add Flow**
- **Name:** Avaya\_SM
  - **Server Configuration:** SM7-HCOM
  - **URI Group:** \*
  - **Transport:** \*
  - **Remote Subnet:** \*
  - **Received Interface:** Sig-Trunk\_Server
  - **Signaling Interface:** Sig-Call\_Server
  - **Media Interface:** Med-Call\_Server
  - **End Point Policy Group:** default-low
  - **Routing Profile:** Trunk\_Server\_HCOM
  - **Topology Hiding Profile:** Call\_Server\_SM7
  - **File Transfer Profile:** None
  - Click **Finish** (not shown)

### 7.3.4. End Point Flows – SIP Trunk

This section is used to determine the method and interfaces used to transport sip media and signaling to Hipcom.

- Select **Device Specific Settings → Endpoint Flows**
- Select the **Server Flows** tab
- Select **Add Flow**
- **Name: SIP Trunk**
- **Server Configuration: HCOM-SM7**
- **URI Group: \***
- **Transport: \***
- **Remote Subnet: \***
- **Received Interface: Sig-Call\_Server**
- **Signaling Interface: Sig-Trunk\_Server**
- **Media Interface: Med-Trunk-Server**
- **End Point Policy Group: default-low**
- **Routing Profile: Call\_Server\_SM7**
- **Topology Hiding Profile: Trunk\_Server\_Hcom**
- **File Transfer Profile: None**
- Click **Finish** (not shown)

The screenshot displays the 'Server Flows' configuration window. On the left is a navigation tree with categories like Global Parameters, Global Profiles, SIP Cluster, and Device Specific Settings. The main area shows two tables of server configurations.

**Server Configuration: Call\_Server\_SM7**

Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile		
1	SM7-HCOM	*	*	*	Sig-Trunk_Server	Sig-Call_Server	Med-Call_Server	default-low	Trunk_Server_HCOM	Call_Server_SM7	None		

**Server Configuration: Trunk\_Server\_HCOM**

Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile		
1	HCOM-SM7	*	*	*	Sig-Call_Server	Sig-Trunk_Server	Med-Trunk_Server	default-low	Call_Server_SM7	Trunk_Server_Hcom	None		

## 8. BT Wholesale/HIPCOM Configuration

The configuration required by BTW/HIPCOM to allow the tests to be carried out is not covered in this document and any further information required should be obtained through the local BTW/HIPCOM representative.

## 9. Verification Steps

This section provides steps that may be performed to verify that the solution is configured correctly.

1. From System Manager Home tab click on Session Manager and navigate to **Session Manager → System Status → SIP Entity Monitoring**. Select the relevant SIP Entity from the list and observe if the **Conn Status** and **Link Status** are showing as **up**.

This is the SIP Entity link to the Communication Manager:

Communication Profile	All Entity Links to SIP Entity: CMEVO						
Editor	Summary View						
Network Configuration	1 Item Refresh Filter: Enable						
Device and Location Configuration							
Application Configuration							
System Status							
SIP Entity Monitoring							
	Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code
	Show	ASM61	10.10.7.52	5060	TCP	Up	200 OK
							Link Status
							Up

This is the SIP Entity link to the Avaya Session Border Controller Advanced for Enterprise:

Communication Profile	All Entity Links to SIP Entity: ASBCAE						
Editor	Summary View						
Network Configuration	1 Item Refresh Filter: En						
Device and Location Configuration							
Application Configuration							
System Status							
SIP Entity Monitoring							
	Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code
	Show	ASM61	10.10.7.220	5060	TCP	Up	200 OK
							Link Status
							Up

From the Communication Manager SAT interface run the command **status trunk x** where **x** is a previously configured SIP trunk. Observe if all channels on the trunk group display **In service/ idle**.

```
status trunk 1
```

TRUNK GROUP STATUS			
Member	Port	Service State	Mtce Connected Ports Busy
0001/001	T00001	in-service/idle	no
0001/002	T00007	in-service/idle	no
0001/003	T00008	in-service/idle	no
0001/004	T00009	in-service/idle	no
0001/005	T00010	in-service/idle	no

2. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
3. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
4. Verify that the user on the PSTN can end an active call by hanging up.
5. Verify that an endpoint at the enterprise site can end an active call by hanging up.

## 10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager, Avaya Aura® Session Manager and the Avaya Session Border Controller Advanced for Enterprise to BTW/HIPCOM SIP Trunk Service. The testing was successfully performed with BTW/HIPCOM, refer to **Section 2.2** for more details.

## 11. References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Installing and Configuring Avaya Aura® System Platform*, Release 6.03, February 2011.
- [2] *Administering Avaya Aura® System Platform*, Release 6.03, February 2011.
- [3] *Administering Avaya Aura® Communication Manager*, August 2010, Document Number 03-300509.
- [4] *Avaya Aura® Communication Manager Feature Description and Implementation*, May 2009, Document Number 555-245-205.
- [5] *Installing and Upgrading Avaya Aura® System Manager Release 6.1*, November 2010.
- [6] *Installing and Configuring Avaya Aura® Session Manager*, January 2011, Document Number 03-603473
- [7] *Administering Avaya Aura® Session Manager*, March 2011, Document Number 03-603324.
- [8] RFC 3261 *SIP: Session Initiation Protocol*, <http://www.ietf.org/>

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