



# **Application Notes for Configuring Avaya Aura® Communication Manager R6.3 as an Evolution Server, Avaya Aura® Session Manager R6.3 and Avaya Session Border Controller for Enterprise R6.3 to support Swisscom Enterprise SIP Service - Issue 1.0**

## **Abstract**

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between Swisscom Enterprise SIP service and an Avaya SIP enabled enterprise solution.

The Avaya solution consists of Avaya Session Border Controller for Enterprise, Avaya Aura® Session Manager and Avaya Aura® Communication Manager as an Evolution Server. Swisscom is a member of the DevConnect Service Provider program.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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.

## Contents

1.	Introduction.....	4
2.	General Test Approach and Test Results.....	4
2.1.	Interoperability Compliance Testing.....	4
2.2.	Test Results .....	5
2.3.	Support .....	5
3.	Reference Configuration .....	6
4.	Equipment and Software Validated .....	7
5.	Configure Avaya Aura® Communication Manager .....	7
5.1.	Confirm System Features .....	8
5.2.	Administer IP Node Names.....	9
5.3.	Administer IP Network Region.....	10
5.4.	Administer IP Codec Set .....	11
5.5.	Administer SIP Signalling Groups .....	12
5.6.	Administer SIP Trunk Group .....	13
5.7.	Administer Calling Party Number Information .....	15
5.8.	Administer Route Selection for Outbound Calls.....	15
5.9.	Administer Incoming Digit Translation .....	17
5.10.	EC500 Configuration.....	17
6.	Configuring Avaya Aura® Session Manager .....	18
6.1.	Log in to Avaya Aura® System Manager.....	18
6.2.	Administer SIP Domain .....	19
6.3.	Administer Locations .....	20
6.4.	Administer Adaptations.....	21
6.5.	Administer SIP Entities .....	23
6.5.1.	Avaya Aura® Session Manager SIP Entity .....	24
6.5.2.	Avaya Aura® Communication Manager SIP Entity .....	25
6.5.3.	Avaya Session Border Controller for Enterprise SIP Entity.....	26
6.6.	Administer Entity Links .....	28
6.7.	Administer Routing Policies .....	29
6.8.	Administer Dial Patterns .....	30
7.	Configure Avaya Session Border Controller for Enterprise .....	32
7.1.	Accessing Avaya Session Border Controller for Enterprise .....	32
7.2.	Global Profiles.....	34
7.2.1.	Server Interworking Avaya.....	34
7.2.2.	Server Interworking – Swisscom.....	36
7.2.3.	Server Configuration– Avaya .....	38
7.2.4.	Server Configuration – Swisscom .....	39
7.2.5.	Routing.....	41
7.2.6.	Topology Hiding.....	44
7.3.	Define Network Information.....	46
7.4.	Define Interfaces .....	47
7.4.1.	Signalling Interfaces .....	47

7.4.2. Media Interfaces.....	48
7.5. Server Flows.....	49
8. Swisscom Enterprise SIP Service Configuration.....	52
9. Verification Steps.....	52
10. Conclusion .....	55
11. Additional References.....	55

# 1. Introduction

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between Swisscom's Enterprise SIP service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Session Border Controller for Enterprise (Avaya SBCE), Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Customers using this Avaya SIP-enabled enterprise solution with Swisscom Enterprise SIP 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 customer.

## 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 Communication Manager, Session Manager and Avaya SBCE. The enterprise site was configured to use the SIP trunking service provided by Swisscom. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

### 2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability the following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types including H.323, SIP, Digital and Analogue telephones at the enterprise.
- All inbound PSTN calls were routed to the enterprise across the SIP trunk from the Service Provider.
- Outgoing PSTN calls from various phone types including H.323, SIP, Digital, and Analogue telephones at the enterprise.
- All outbound PSTN calls were routed from the enterprise across the SIP trunk to the Service Provider.
- Calls using G.711A and G.711MU codecs.
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using T.38 and G.711A fallback.
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls.
- Inbound and outbound PSTN calls to/from Avaya One-X® Communicator Softphone clients.

- Various call types including: local, long distance, international, toll free (outbound) and directory assistance.
- Caller ID presentation and Caller ID restriction.
- User features such as hold and resume, call mute, transfer, and conference.
- Off-net call forwarding and mobile twinning.
- Transmission and response of SIP OPTIONS messages sent by Swisscom's SIP trunk requiring Avaya response and sent by Avaya requiring Swisscom's response.  
**Note:** Swisscom requested the transmission of OPTIONs from Avaya every 15 seconds. Please refer to **Section 6.5.3** for configuration settings.

## 2.2. Test Results

Interoperability testing of the test configuration was completed with successful results for Swisscom's Enterprise SIP service with the following observations:

- G.729 codec is not supported by Swisscom and therefore was not tested.
- No inbound toll free numbers were tested, however routing of inbound DDI numbers and the relevant number translation was successfully tested.
- Access to Emergency Services was not tested as no test call had been booked by the Service Provider with the Emergency Services Operator. However both three and four digit numbering format replicating Emergency Service's numbering formats was tested successfully.

## 2.3. Support

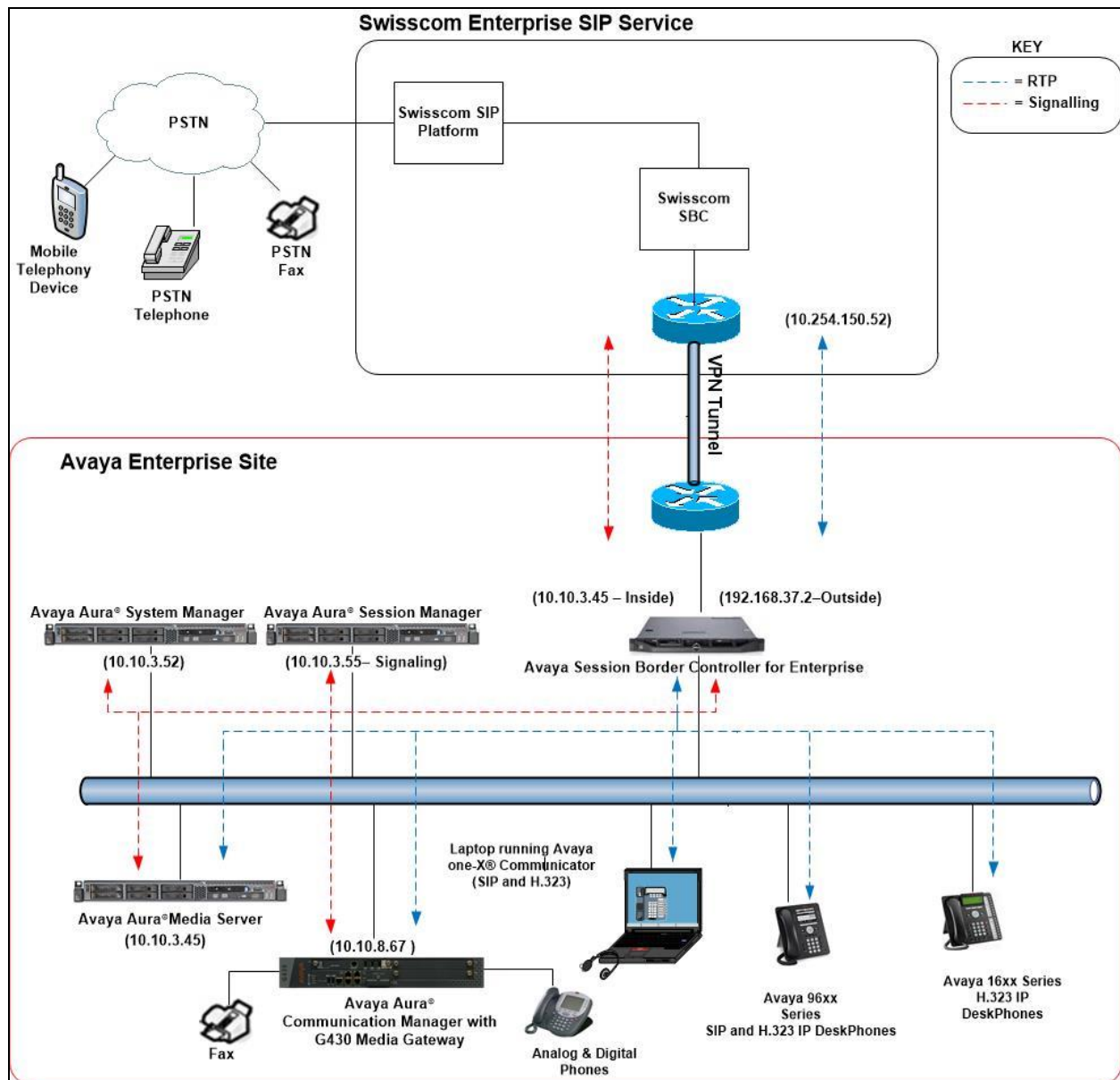
For technical support on the Avaya products described in these Application Notes visit <http://support.avaya.com>.

For technical support on Swisscom products please contact the Swisscom support team:

- Email: [ent.Incident-Voice@swisscom.com](mailto:ent.Incident-Voice@swisscom.com)

### 3. Reference Configuration

The following equipment in **Figure 1** illustrates the test configuration. The test configuration shows an Enterprise site connected to Swisscom Enterprise SIP service. Located at the Enterprise site is an Avaya Session Border Controller for Enterprise, Session Manager and Communication Manager. Endpoints are Avaya 96x0 series and Avaya 96x1 series IP telephones (with SIP and H.323 firmware), Avaya 16xx series IP telephones (with H.323 firmware), Avaya analogue telephones and an analogue fax machine. Also included in the test configuration was an Avaya one-X® Communicator soft phone running on a laptop PC.



**Figure 1: Test Setup Swisscom Enterprise SIP Service to Avaya Enterprise**

## 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
<b>Avaya</b>	
Dell PowerEdge R620 running System Manager on VM Version 8	6.3.10 Build No. - 6.3.0.8.5682-6.3.8.4580 Software Update Revision No: 6.3.10.7.2880 Service Pack 10
Dell PowerEdge R620 running Session Manager on VM Version 8	6.3.10.0.631008
Avaya S8300D Server running Avaya Aura® Communication Manager	6.3.16 - 03.0.124.0-23675 PLAT-rhel5.3-3018
Avaya G430 Media Gateway	6.3.15 (g430_sw_36_22.0)
Avaya Session Border Controller for Enterprise	6.3.3-08-6224
Avaya 1600 IP DeskPhone (H.323)	1.3.9
Avaya 9670 IP DeskPhone (H.323)	6.3
Avaya 96x0 IP DeskPhone (H.323)	6.3
Avaya 9611 IP DeskPhone (SIP)	6.3
Avaya 9608 IP DeskPhone (SIP)	6.3
Avaya 9621 IP DeskPhone (SIP)	6.3
Avaya 9608 IP DeskPhone (SIP)	6.3
Avaya one-X® Communicator (H.323 & SIP) on Lenovo T510 Laptop PC	6.2.11.03-FP11
Analogue Handset	N/A
Analogue Fax	N/A
<b>Swisscom</b>	
eSBC	Cisco 897va 15.5(3)
SBC	ACME Net-Net 6300 Firmware 7.3.0m1
SESM	Genband MCP_19_0.1

## 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 signalling associated with the Swisscom SIP trunk. For incoming calls, Session Manager receives SIP messages from the Avaya SBCE 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 signalling is routed to Session Manager. Session Manager directs the outbound SIP messages to the Avaya SBCE at the enterprise site that then sends the SIP messages to the Swisscom

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 Servers 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 Swisscom SIP network, and any other SIP trunks used.

<b>display system-parameters customer-options</b>		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:	41000	0
Maximum Video Capable IP Softphones:	18000	0
<b>Maximum Administered SIP Trunks:</b>	<b>4000</b>	<b>10</b>
Maximum Administered Ad-hoc Video Conferencing Ports:	24000	0
Maximum Number of DS1 Boards with Echo Cancellation:	522	0



On **Page 5**, verify that **IP Trunks** field is set to **y**.

display system-parameters customer-options		Page 5 of 11
OPTIONAL FEATURES		
Emergency Access to Attendant? y	IP Stations? y	
Enable 'dadmin' Login? y		
Enhanced Conferencing? y	ISDN Feature Plus? n	
Enhanced EC500? y	ISDN/SIP Network Call Redirection? y	
Enterprise Survivable Server? n	ISDN-BRI Trunks? y	
Enterprise Wide Licensing? n	ISDN-PRI? y	
ESS Administration? y	Local Survivable Processor? n	
Extended Cvg/Fwd Admin? y	Malicious Call Trace? y	
External Device Alarm Admin? y	Media Encryption Over IP? y	
Five Port Networks Max Per MCC? n	Mode Code for Centralized Voice Mail? n	
Flexible Billing? n		
Forced Entry of Account Codes? y	Multifrequency Signaling? y	
Global Call Classification? y	Multimedia Call Handling (Basic)? y	
Hospitality (Basic)? y	Multimedia Call Handling (Enhanced)? y	
Hospitality (G3V3 Enhancements)? y	Multimedia IP SIP Trunking? y	
IP Trunks? y		
IP Attendant Consoles? y		

## 5.2. Administer IP Node Names

The node names defined here will be used in other configuration screens to define a SIP signalling group between Communication Manager and Session Manager. In the **IP Node Names** form, assign the node **Name** and **IP Address** for Session Manager. In this case, **SM** and **10.10.3.55** are the **Name** and **IP Address** for the Session Manager SIP interface. Also note the **procr** name as this is the processor interface that Communication Manager will use as the SIP signalling interface to Session Manager.

display node-names ip		IP NODE NAMES
Name	IP Address	
ASM	10.10.3.55	
default	0.0.0.0	
procr	10.10.8.67	
procr6	::	

### 5.3. Administer IP Network Region

Use the **change ip-network-region x** command where x is the desired network-region 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 **avaya.com**.
- By default, **IP-IP Direct Audio** (both **Intra-** and **Inter-Region**) is enabled (**yes**) to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources. When a PSTN call is shuffled, the media stream is established directly between the enterprise end-point and the internal media interface of the Avaya SBCE.
- 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** is used.
- The rest of the fields can be left at default values.

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

## 5.4. Administer IP Codec Set

Open the **IP Codec Set** form for the codec set specified in the IP Network Region form in **Section 5.3**. Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test the codec supported by Swisscom was configured, namely **G.711A** and **G.711MU**.

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

Swisscom SIP trunk supports T.38-G.711-fallback for transmission of fax. Navigate to **Page 2** and define fax properties as follows:

- Set the **FAX - Mode** to **t.38-G711-fallback**.
- Leave **ECM** at default value of **y**.

change ip-codec-set 1				Page 2 of 2
IP CODEC SET				
Allow Direct-IP Multimedia? n				
	Mode	Redundancy	ECM: y	Packet Size (ms)
<b>FAX</b>	<b>t.38-G711-fallback</b>	<b>0</b>		
Modem	off	0		
TDD/TTY	US	3		
H.323 Clear-channel	n	0		
SIP 64K Data	n	0		20

## 5.5. Administer SIP Signalling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the Swisscom SIP trunk network. During test, this was configured to use TCP and port 5060 to facilitate tracing and fault analysis. It is recommended however, to use TLS (Transport Layer Security) and the default TLS port of 5061 for security. Configure the **Signaling Group** using the **add signaling-group x** command as follows:

- Set **Group Type** to **sip**.
- Set **Transport Method** to **tcp**.
- Set **Peer Detection Enabled** to **y** allowing Communication Manager to automatically detect if the peer server is a Session Manager.
- Set **Near-end Node Name** to the processor interface (node name **procr** as defined in the **IP Node Names** form shown in **Section 5.2**).
- Set **Far-end Node Name** to Session Manager (node name **SM** as defined in the **IP Node Names** form shown in **Section 5.2**).
- Set **Near-end Listen Port** and **Far-end Listen Port** to **5060** (Commonly used TCP port value).
- Set **Far-end Network Region** to the IP Network Region configured in **Section 5.3**. (logically establishes the far-end for calls using this signalling group as network region 1)
- Leave **Far-end Domain** blank (allows Communication Manager to accept calls from any SIP domain on the associated trunk).
- Leave **DTMF over IP** at default value of **rtp-payload** (Enables **RFC2833** for DTMF transmission from Communication Manager).
- Set **Direct IP-IP Audio Connections** to **y**.
- Set **Initial IP-IP Direct Media** to **y** as Direct Media was enabled for the compliance testing.
- Set **H.323 Station Outgoing Direct Media** to **n**.

The default values for the other fields may be used.

add signaling-group 1		Page 1 of 2
SIGNALING GROUP		
Group Number: 1	Group Type: sip	
IMS Enabled? n	Transport Method: tcp	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: SM	
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y		
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n		
Near-end Node Name: procr	Far-end Node Name: SM	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 1	
Far-end Domain:		
	Bypass If IP Threshold Exceeded? n	
Incoming Dialog Loopbacks: eliminate	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? y	Initial IP-IP Direct Media? y	
H.323 Station Outgoing Direct Media? n	Alternate Route Timer(sec): 6	

## 5.6. Administer SIP Trunk Group

A trunk group is associated with the signalling 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.
- The **Direction** is set to **two-way** to allow incoming and outgoing calls.
- Set the **Service Type** field to **public-ntwrk**.
- Specify the signalling group associated with this trunk group in the **Signaling Group** field as previously configured in **Section 5.5**.
- Specify the **Number of Members** administered for 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: OUTSIDE CALL	COR: 1	TN: 1	TAC: 101
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: public-ntwrk	Auth Code? n		
	Member Assignment Method: auto		
	Signaling Group: 1		
	Number of Members: 10		

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 Swisscom to prevent unnecessary SIP messages during call setup. During the compliance testing, **Preferred Minimum Session Refresh Interval (sec)** was set to **180**.

add trunk-group 1		Page 2 of 21	
Group Type: sip			
TRUNK PARAMETERS			
Unicode Name: auto			
Redirect On OPTIM Failure: 5000			
SCCAN? n	Digital Loss Group: 18		
Preferred Minimum Session Refresh Interval(sec): 180			
Disconnect Supervision - In? y Out? y			
XOIP Treatment: auto		Delay Call Setup When Accessed Via IGAR? n	
Caller ID for Service Link Call to H.323 1xC: station-extension			

On **Page 3**, set the **Numbering Format** field to **public**. This allows delivery of CLI in E164 format.

add trunk-group 1		Page 3 of 21
TRUNK FEATURES		
ACA Assignment? n	Measured: none	Maintenance Tests? y
Numbering Format: public		
UUI Treatment: service-provider		
Replace Restricted Numbers? n		
Replace Unavailable Numbers? n		

On **Page 4** of this form:

- Set **Mark Users as Phone** to **y**.
- Set **Send Transferring Party Information** to **n**.
- Set **Network Call Direction** to **n**.
- Set **Send Diversion Header** to **y**.
- Set **Support Request History** to **n**.
- Set the **Telephone Event Payload Type** to **101** to match the value preferred by Swisscom.
- Set **Always Use re-INVITE for Display Updates** to **y**.
- Set the **Identity for Calling Party Display** to **P-Asserted-Identity**.

PROTOCOL VARIATIONS	
Mark Users as Phone? y	
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n	
Send Transferring Party Information? n	
Network Call Redirection? n	
Build Refer-To URI of REFER From Contact For NCR? n	
Send Diversion Header? y	
Support Request History? n	
Telephone Event Payload Type: 101	
Convert 180 to 183 for Early Media? n	
Always Use re-INVITE for Display Updates? y	
Identity for Calling Party Display: P-Asserted-Identity	
Block Sending Calling Party Location in INVITE? n	
Accept Redirect to Blank User Destination? n	
Enable Q-SIP? n	
Interworking of ISDN Clearing with In-Band Tones: keep-channel-active	

## 5.7. Administer Calling Party Number Information

Use the **change public-unknown-numbering x** command to configure Communication Manager to send the calling party number in the format required E.164. This calling party number is sent in the SIP From, Contact and PAI headers, and displayed on display-equipped PSTN telephones.

change public-unknown-numbering					Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT					
Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len	
4	6102	1	4143xxxxx80	11	Total Administered: 6
4	6100	1	4143xxxxx81	11	Maximum Entries: 240
4	6003	1	4143xxxxx82	11	Note: If an entry applies to a SIP connection to Avaya Aura(R) Session Manager, the resulting number must be a complete E.164 number.
4	6104	1	4143xxxxx83	11	
					Communication Manager automatically inserts a '+' digit in this case.

## 5.8. Administer Route Selection for Outbound Calls

In the test environment, the Automatic Route Selection (ARS) feature was used to route outbound calls via the SIP trunk to Swisscom's Enterprise SIP service. The single digit **9** was used as the ARS access code providing a facility for telephone users to dial 9 to reach an outside line. Use the **change feature-access-codes** command to configure a digit as the **Auto Route Selection (ARS) - Access Code 1**.

change feature-access-codes		Page 1 of 10
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: *69		
Answer Back Access Code:		
Attendant Access Code:		
Auto Alternate Routing (AAR) Access Code: 7		
<b>Auto Route Selection (ARS) - Access Code 1: 9</b>		Access Code 2:

Use the **change ars analysis** command to configure the routing of dialled digits following the first digit 9. A small sample of dial patterns are shown here as an example. Further administration of ARS is beyond the scope of this document. The example entries shown will match outgoing calls to numbers beginning **0**. Note that exact maximum number lengths should be used where possible to reduce post-dial delay. Calls are sent to **Route Pattern 1**.

change ars analysis 0							Page 1 of 2
ARS DIGIT ANALYSIS TABLE							
Location: all							Percent Full: 0
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd	
0	11	14	1	pubu		n	
00	13	15	1	pubu		n	
0035391	13	13	1	pubu		n	
030	10	10	1	pubu		n	
0800	8	10	1	pubu		n	
0900	8	8	1	pubu		n	

Use the **change route-pattern x** command, where **x** is an available route pattern, 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**. **Numbering Format** is applied to CLI and is used to set TDM signalling parameters such as type of number and numbering plan indicator. This doesn't have the same significance in SIP calls and during testing it was set to **unk-unk**.

change route-pattern 1													Page	1 of	3						
Pattern Number: 1													Pattern Name:								
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.	Numbering	LAR
0	1	2	M	4	W	Request							Dgts	Format							
													Subaddress								
1:	y	y	y	y	y	n	n	rest					unk-unk	none							
2:	y	y	y	y	y	n	n	rest						none							
3:	y	y	y	y	y	n	n	rest						none							
4:	y	y	y	y	y	n	n	rest						none							
5:	y	y	y	y	y	n	n	rest						none							
6:	y	y	y	y	y	n	n	rest						none							



## 5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DDI calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from Swisscom can be manipulated as necessary to route calls to the desired extension. In the examples used in the compliance testing, the incoming DDI numbers provided by Swisscom correlate to the internal extensions assigned within Communication Manager. The entries displayed below translate incoming DDI numbers **+4143xxxxx80**, **+4143xxxxx81**, **+4143xxxxx82**, and **+4143xxxxx83** to a 4 digit extension by deleting all of the incoming digits and inserting an extension. Public DDI numbers have been masked for security purposes.

change inc-call-handling-trmt trunk-group 1					Page 1 of 3		
INCOMING CALL HANDLING TREATMENT							
Service/ Feature	Number Len	Number Digits	Del	Insert			
public-ntwrk	12	+4143xxxxx80	all	6102			
public-ntwrk	12	+4143xxxxx81	all	6100			
public-ntwrk	12	+4143xxxxx82	all	6030			
public-ntwrk	12	+4143xxxxx83	all	6104			

## 5.10. EC500 Configuration

When EC500 is enabled on the Communication Manager station, a call to that station will generate a new outbound call from Communication Manager to the configured EC500 destination, typically a mobile phone. The following screen shows an example EC500 configuration for the user with station extension 6102. Use the command **change off-pbx-telephone station-mapping x** where **x** is the Communication Manager station.

- The **Station Extension** field will automatically populate with station extension.
- For **Application** enter **EC500**.
- For the **Phone Number** enter the phone that will also be called (e.g.**0035389434xxxx**).
- Set the **Trunk Selection** to **1** so that Trunk Group 1 will be used for routing.
- Set the **Config Set** to **1**.

change off-pbx-telephone station-mapping 6102						Page	1 of	3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION								
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode	
6102	EC500	-		0035389434xxxx	1	1		
-								

**Note:** The phone number shown is for a mobile phone used for testing at Avaya Labs and is in international format. To use facilities for calls coming in from EC500 mobile phones, the number received in Communication Manager must exactly match the number specified in the above table.

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

## 6. Configuring Avaya Aura® Session Manager

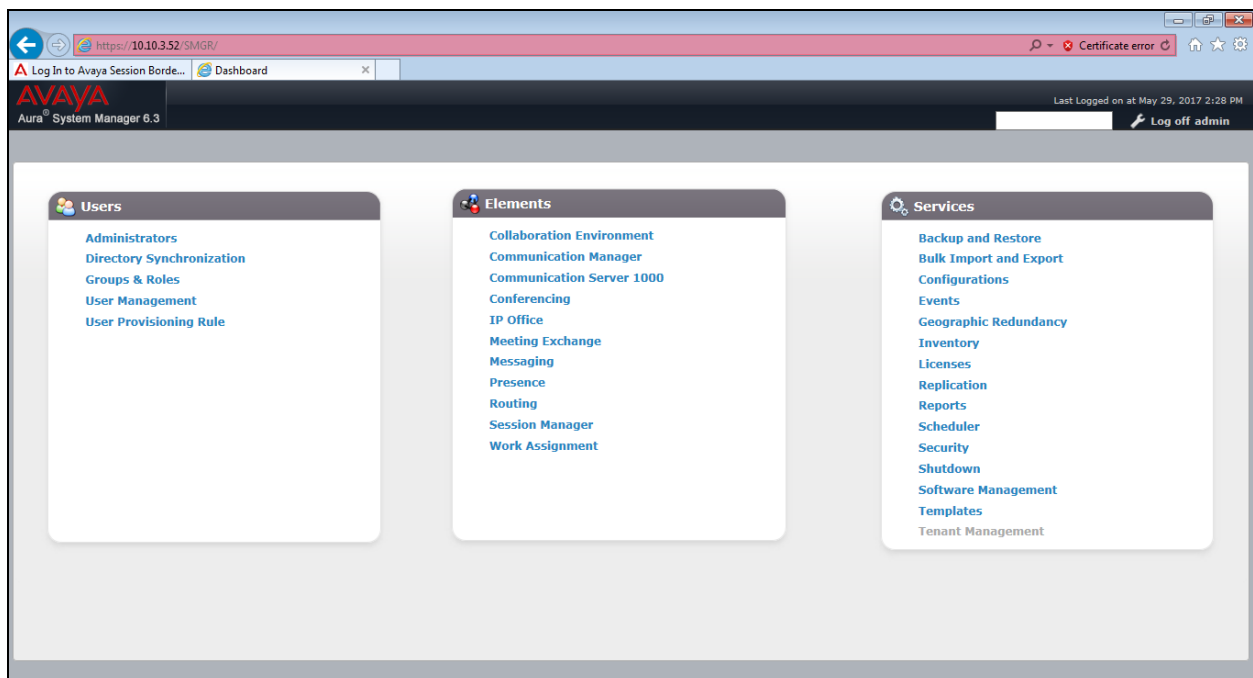
This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

- Log in to Avaya Aura® System Manager.
- Administer SIP Domain.
- Administer SIP Location.
- Administer Adaptations.
- Administer SIP Entities.
- Administer Entity Links.
- Administer Routing Policies.
- Administer Dial Patterns.

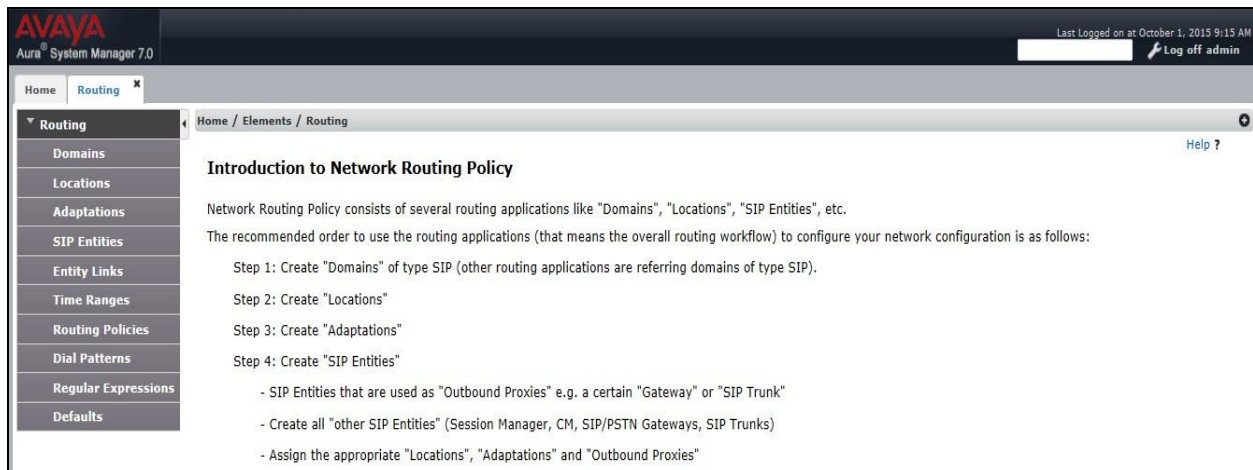
It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session Manager itself. However, each item should be reviewed to verify the configuration.

### 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.



Most of the configuration items are performed in the Routing Element. Click on **Routing** in the Elements column shown above to bring up the **Introduction to Network Routing Policy** screen.



## 6.2. Administer SIP Domain

Create a SIP domain for each domain for which Session Manager will need to be aware in order to route calls. Expand **Elements** → **Routing** and select **Domains** from the left navigation menu, click **New** (not shown). Enter the following values and use default values for remaining fields.

- **Name** Enter a Domain Name. In the sample configuration, **avaya.com** was used.
- **Type** Verify **SIP** is selected.
- **Notes** Add a brief description [Optional].

Click **Commit** to save. The screen below shows the SIP Domain defined for the sample configuration.



## 6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. To add a location, navigate to **Routing → Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description (optional).

The Location Pattern is used to identify call routing based on IP address. Session Manager matches the IP address against the patterns defined in this section. If a call is from a SIP Entity that does not match the IP address pattern then Session Manager uses the location administered for the SIP Entity.

In the **Location Pattern** section, click **Add** and enter the following values.

- **IP Address Pattern** Enter the logical pattern used to identify the location.
- **Notes** Add a brief description [Optional].

Click **Commit** to save. The screenshot below shows the Location **SMGRVL3** defined for the compliance testing.

The screenshot displays the Avaya Session Manager Administration console. The breadcrumb navigation at the top reads 'Home / Elements / Routing / Locations'. The main section is titled 'Location Details' and includes 'Commit' and 'Cancel' buttons. The 'General' tab is active, showing the 'Name' field set to 'SMGRVL3' and an empty 'Notes' field. Below this is the 'Dial Plan Transparency in Survivable Mode' section with an 'Enabled' checkbox (unchecked), a 'Listed Directory Number' field, and an 'Associated CM SIP Entity' dropdown menu. The 'Overall Managed Bandwidth' section shows 'Managed Bandwidth Units' set to 'Kbit/sec', 'Total Bandwidth' and 'Multimedia Bandwidth' fields, and a checked 'Audio Calls Can Take Multimedia Bandwidth' checkbox. The 'Location Pattern' section at the bottom has an 'Add' button and a 'Remove' button. It shows a table with 3 items, each with an 'IP Address Pattern' and a 'Notes' field. The patterns listed are '\*10.10.3.\*', '\*10.10.5.\*', and '\*10.10.8.\*'. A 'Filter: Enable' link is on the right. At the bottom of the 'Location Pattern' section are 'Commit' and 'Cancel' buttons.

IP Address Pattern	Notes
*10.10.3.*	
*10.10.5.*	
*10.10.8.*	

## 6.4. Administer Adaptations

Adaptations can be used to modify the called and calling party numbers to meet the requirements of the service. The called party number present in the SIP INVITE Request URI is modified by the **Digit Conversion** in the Adaptation. The example below was applied to the Avaya SBCE SIP Entity and was used in test to convert numbers being passed between the Avaya SBCE and Session Manager.

To add an adaptation, under the **Routing** tab select **Adaptations** on the left hand menu and then click on the **New** button (not shown). Under **Adaption Details** → **General**:

- In the **Adaptation name** field enter an informative name.
- In the **Module name** field click on the down arrow and then select the **<click to add module>** entry from the drop down list and type **DigitConversionAdapter** in the resulting New Module Name field.
- **Module parameter** **MIME=no** Strips MIME message bodies on egress from Session Manager  
**fromto=true** Modifies from and to headers of a message

Home / Elements / Routing / Adaptations

Adaptation Details

Commit Cancel

General

\* Adaptation Name:

Module Name:

Module Parameter Type:

Add Remove

<input type="checkbox"/>	Name	Value
<input type="checkbox"/>	fromto	True
<input type="checkbox"/>	MIME	no

Select : All, None

Egress URI Parameters:

Notes:

Scroll down the page and under **Digit Conversion for Outgoing Calls from SM**, click the **Add** button and specify the digit manipulation to be performed as follows:

- Enter the leading digits that will be matched in the Matching Pattern field.
- In the **Min** and **Max** fields set the minimum and maximum digits allowed in the digit string to be matched.
- In the **Delete Digits** field enter the number of leading digits to be removed.
- In the **Insert Digits** field specify the digits to be prefixed to the digit string.
- In the **Address to modify** field specify the digits to manipulate by the adaptation. In this configuration the dialed number is the target so **both** have been selected.

Digit Conversion for Outgoing Calls from SM

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
<input type="checkbox"/>	*00	*2	*16		*2	+	both		

Select : All, None

Commit Cancel

This will ensure any outbound numbers will have the 00 digits removed and + digit inserted to convert to E.164 format before being presented to the Swisscom SIP trunk.

## 6.5. Administer SIP Entities

A SIP Entity must be added for each SIP-based telephony system supported by a SIP connection to 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 signalling 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 **SIP Trunk** for the Avaya SBCE 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
- Avaya SBCE SIP Entity

### 6.5.1. 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 signalling interface and **Type** is **Session Manager**. Set the **Location** to that defined in **Section 6.3** and the **Time Zone** to the appropriate time.

The screenshot shows the 'SIP Entity Details' configuration page. The breadcrumb trail is 'Home / Elements / Routing / SIP Entities'. The page has 'Commit' and 'Cancel' buttons in the top right. The 'General' tab is selected. The form contains the following fields:

- Name:** Session Manager
- FQDN or IP Address:** 10.10.3.55
- Type:** Session Manager (dropdown)
- Notes:** (empty text area)
- Location:** SMGRVL3 (dropdown)
- Outbound Proxy:** (empty dropdown)
- Time Zone:** Europe/Dublin (dropdown)
- Credential name:** (empty text area)
- SIP Link Monitoring:** Use Session Manager Configuration (dropdown)

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 the domain added in **Section 6.2** as the default domain.

The screenshot shows the 'Port' configuration section. It includes input fields for 'TCP Failover port' and 'TLS Failover port', and 'Add' and 'Remove' buttons. Below is a table with 3 items, showing a list of configured ports.

	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	avaya.com	
<input type="checkbox"/>	5061	TLS	avaya.com	
<input type="checkbox"/>	5061	UDP	avaya.com	

At the bottom, there is a 'Select : All, None' option.



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

The following screen shows the SIP entity for Communication Manager which is configured as an Evolution Server. The **FQDN or IP Address** field is set to the IP address of the interface on Communication Manager that will be providing SIP signalling and **Type** is **CM**. Set the **Location** to that defined in **Section 6.3** and the **Time Zone** to the appropriate time.

Home / Elements / Routing / SIP Entities

**SIP Entity Details** Commit Cancel

**General**

\* **Name:** Communication Manager

\* **FQDN or IP Address:** 10.10.8.67

**Type:** CM

**Notes:**

**Adaptation:**

**Location:** SMGRVL3

**Time Zone:** Europe/Dublin

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

**Credential name:**

**Call Detail Recording:** none

**Loop Detection**

**Loop Detection Mode:** Off

**SIP Link Monitoring**

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

Other parameters can be set for the SIP Entity as shown in the following screenshot, but for test, these were left at default values.

**Loop Detection**

**Loop Detection Mode:** Off

**SIP Link Monitoring**

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

### 6.5.3. Avaya Session Border Controller for Enterprise SIP Entity

The following screen shows the SIP Entity for the Avaya SBCE. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE private network interface (see **Figure 1**). Set **Type** to **SIP Trunk**. Set **Adaptation** to the adaptation defined in **Section 6.4**. Set the **Location** to that defined in **Section 6.3** and the **Time Zone** to the appropriate time zone.

Home / Elements / Routing / SIP Entities

**SIP Entity Details** Commit Cancel

**General**

\* **Name:** Avaya\_SBCE

\* **FQDN or IP Address:** 10.10.3.45

**Type:** SIP Trunk

**Notes:**

**Adaptation:** Swisscom

**Location:** SMGRVL3

**Time Zone:** Europe/Dublin

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

**Credential name:**

**Call Detail Recording:** egress

**Loop Detection**

**Loop Detection Mode:** Off

**SIP Link Monitoring**

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

In the SIP Link Monitoring section:

- **SIP Link Monitoring:** Select **Link Monitoring Enabled**.
- Set **Proactive Monitoring Interval (in seconds)** to **15**.
- Set **Reactive Monitoring Interval (in seconds)** to **15**.

Click Commit (not shown) to save Avaya SBCE SIP Entity definition.

**SIP Link Monitoring**

**SIP Link Monitoring:** Link Monitoring Enabled ▼

\* Proactive Monitoring Interval (in seconds): 15

\* Reactive Monitoring Interval (in seconds): 15

\* Number of Tries: 1

Supports Call Admission Control: ☐

Shared Bandwidth Manager: ☐

Primary Session Manager Bandwidth Association: ▼

Backup Session Manager Bandwidth Association: ▼

## 6.6. 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 (not shown). 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 **Session Manager**.
- In the **Protocol** field enter the transport protocol to be used to send SIP requests.
- 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 6.5**.
- In the **Port** field enter the port number to which the other system expects to receive SIP requests.
- Select **Trusted** from the drop-down menu to make the other system trusted.

Click **Commit** to save changes. The following screenshot shows the Entity Links used in this configuration.

Home / Elements / Routing / Entity Links

Help ?

Entity Links


New

Edit

Delete

Duplicate

More Actions ▾

3 Items 

Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy	Deny New Service	Notes
<input type="checkbox"/>	<a href="#">Avaya_SBCE</a>	Session Manager	TCP	5060	Avaya_SBCE	<input type="checkbox"/>	5060	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">Session Manager Communication Manager 5060 TCP</a>	Session Manager	TCP	5060	Communication Manager	<input type="checkbox"/>	5060	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">Session Manager Messaging 5060 TCP</a>	Session Manager	TCP	5060	Messaging	<input type="checkbox"/>	5060	trusted	<input type="checkbox"/>	

Select : All, None

## 6.7. 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.
- Under **Time of Day**, click **Add**, and then select the time range.

The following screen shows the routing policy for Communication Manager.

The screenshot shows the 'Routing Policy Details' form. At the top, there is a breadcrumb trail 'Home / Elements / Routing / Routing Policies' and a 'Help ?' link. Below the title 'Routing Policy Details' are 'Commit' and 'Cancel' buttons. The form is divided into three sections: 'General', 'SIP Entity as Destination', and 'Time of Day'. In the 'General' section, the 'Name' field is set to 'to\_Communication Manager', 'Disabled' is unchecked, 'Retries' is set to '0', and the 'Notes' field is empty. The 'SIP Entity as Destination' section has a 'Select' button. Below it is a table with one row: 'Communication Manager' with FQDN or IP Address '10.10.8.67' and Type 'CM'. The 'Time of Day' section has 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. Below these is a table with one item, '24/7', which is selected. The table has columns for Ranking, Name, days of the week (Mon-Sun), Start Time, End Time, and Notes. The '24/7' item has checkboxes for all days of the week, a start time of '00:00', an end time of '23:59', and a note 'Time Range 24/7'. At the bottom, there is a 'Select : All, None' option.

Home / Elements / Routing / Routing Policies [Help ?](#)

Routing Policy Details [Commit](#) [Cancel](#)

**General**

\* Name:

Disabled: ☐

\* Retries:

Notes:

**SIP Entity as Destination**

[Select](#)

Name	FQDN or IP Address	Type	Notes
Communication Manager	10.10.8.67	CM	

**Time of Day**

[Add](#) [Remove](#) [View Gaps/Overlaps](#)

1 Item [Filter: Enable](#)

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input checked="" 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

Select : All, None

The following screen shows the Routing Policy for the Avaya SBCE.

Home / Elements / Routing / Routing Policies Help ?

Routing Policy Details Commit Cancel

**General**

\* Name:

Disabled: ☐

\* Retries:

Notes:

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type	Notes
Avaya_SBCE	10.10.3.45	SIP Trunk	

**Time of Day**

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/> 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

Select : All, None

## 6.8. 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 dialled number or prefix to be matched.
- In the **Min** field enter the minimum length of the dialled number.
- In the **Max** field enter the maximum length of the dialled number.
- In the **SIP Domain** field select **ALL** or alternatively one of those configured in **Section 6.2**.

Under **Originating Locations and Routing Policies**:

- Click **Add**, in the resulting screen (not shown).
- Under **Originating Location**, select the location defined in **Section 6.3** or **ALL**.
- Under **Routing Policies** select one of the routing policies defined in **Section 6.7**.
- Click **Select** button to save.

The following screen shows an example dial pattern configured for the Avaya SBCE.

Home / Elements / Routing / Dial Patterns Help ?

**Dial Pattern Details** Commit Cancel

**General**

\* Pattern: 00

\* Min: 2

\* Max: 16

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: avaya.com

Notes:

**Originating Locations and Routing Policies**

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	SMGRVL3		to_AvayaSBCE	0	<input type="checkbox"/>	Avaya_SBCE	

Select : All, None

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

Home / Elements / Routing / Dial Patterns Help ?

**Dial Pattern Details** Commit Cancel

**General**

\* Pattern: +41

\* Min: 3

\* Max: 16

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: avaya.com

Notes:

**Originating Locations and Routing Policies**

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	SMGRVL3		to_Communication Manager	0	<input type="checkbox"/>	Communication Manager	

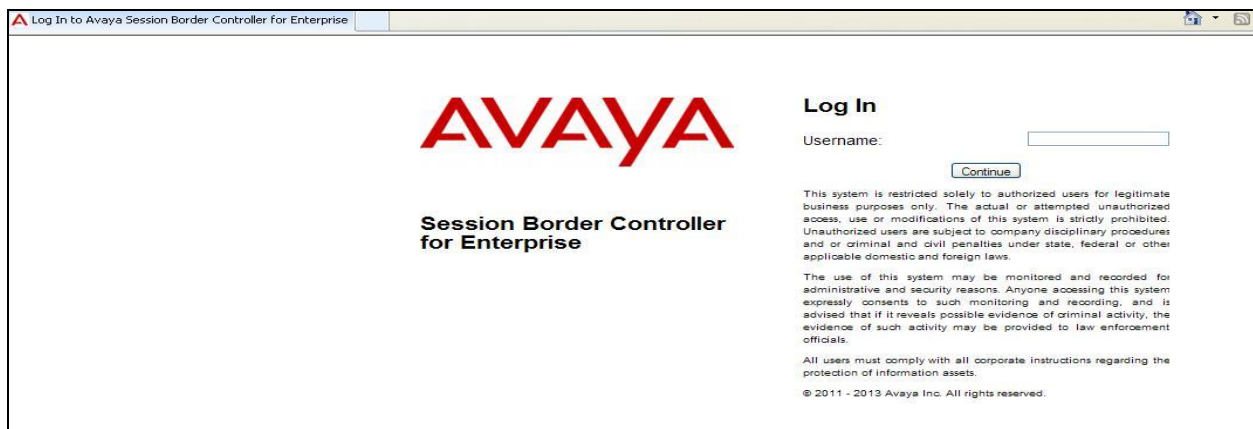
Select : All, None

## 7. Configure Avaya Session Border Controller for Enterprise

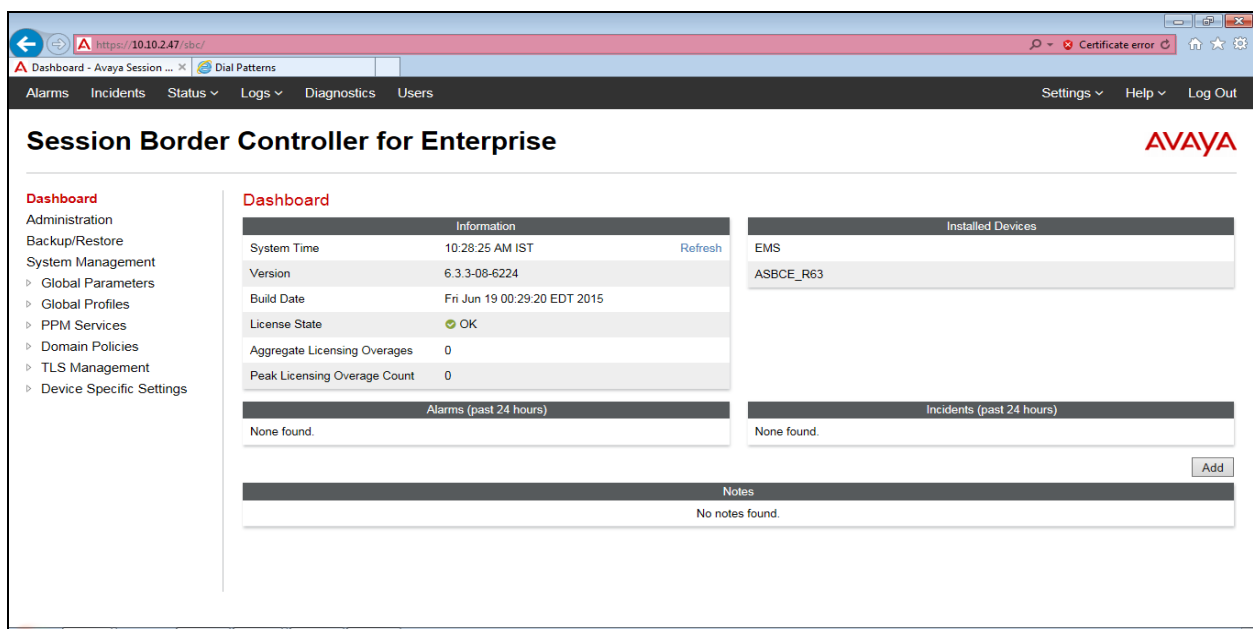
This section describes the configuration of the Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide an interface to the Service Provider's SIP trunk that is standard where possible and adapted to the Service Provider's SIP implementation where necessary.

### 7.1. Accessing Avaya Session Border Controller for Enterprise

Access the Avaya SBCE using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation and enter the **Username** and **Password**.

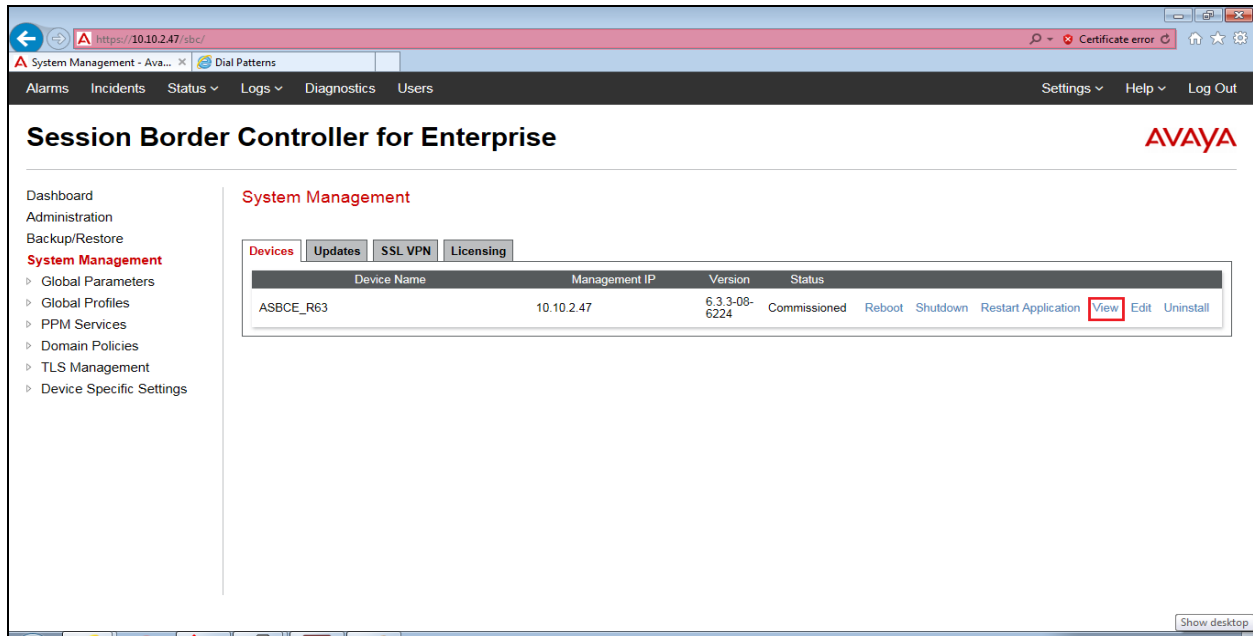


Once logged in, a dashboard is presented with a menu on the left-hand side. The menu is used as a starting point for all configuration of the Avaya SBCE.

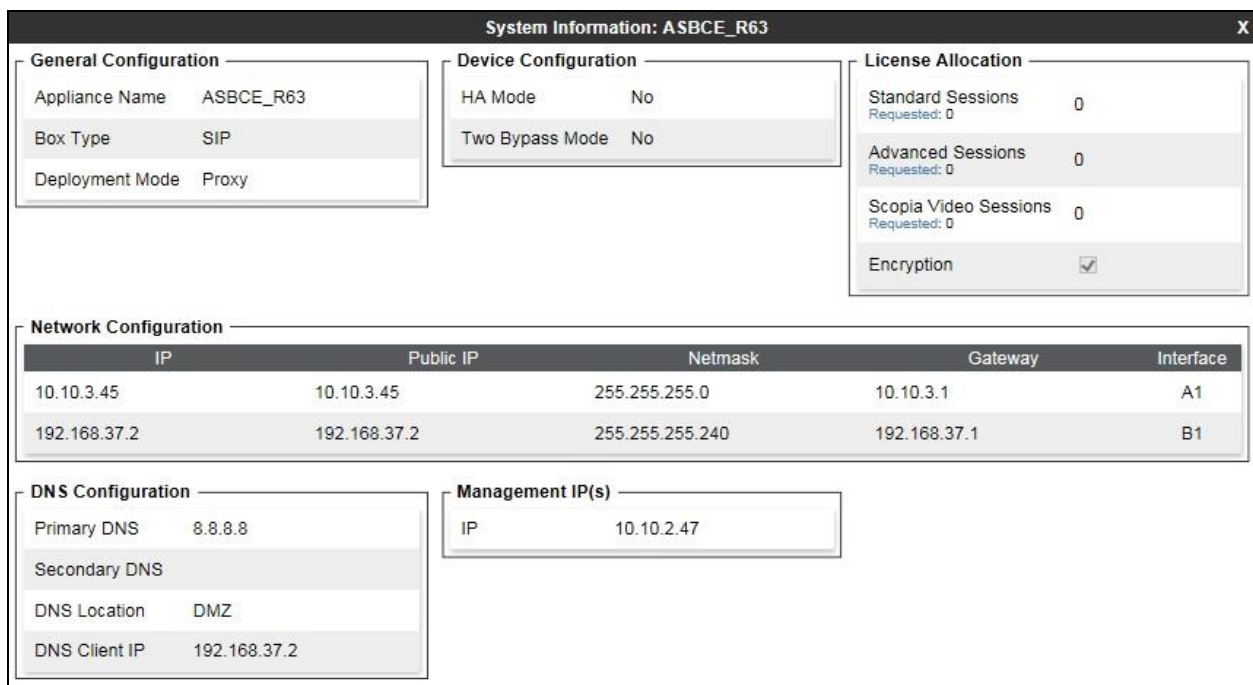




To view system information that was configured during installation, navigate to **System Management**. A list of installed devices is shown in the right pane. In the case of the sample configuration, a single device named **ASBCE\_R63** is shown. To view the configuration of this device, click **View** (the third option from the right).



The **System Information** screen shows the **General Configuration**, **Device Configuration**, **License Allocation**, **Network Configuration**, **DNS Configuration** and **Management IP** information.



## 7.2. Global Profiles

When selected, Global Profiles allows for configuration of parameters across all Avaya SBCE appliances.

### 7.2.1. Server Interworking Avaya

Server Interworking allows the configuration and management of various SIP call server-specific capabilities such as call hold and T.38. From the left-hand menu select **Global Profiles** →

**Server Interworking** and click on **Add** (not shown).

- Enter profile name such as Avaya and click **Next** (Not Shown).
- Check **Hold Support** = **None**.
- Check **T.38 Support**.
- All other options on the **General** Tab can be left at default.

General	
Hold Support	<input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
URI Group	None ▾
Send Hold	<input type="checkbox"/>
Delayed Offer	<input type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
Re-Invite Handling	<input type="checkbox"/>
Prack Handling	<input type="checkbox"/>
Allow 18X SDP	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

Default values can be used for the **Advanced Settings** window. Click **Finish**.

Record Routes	<input checked="" type="radio"/> None <input type="radio"/> Single Side <input checked="" type="radio"/> Both Sides <input type="radio"/> Dialog-Initiate Only (Single Side) <input type="radio"/> Dialog-Initiate Only (Both Sides)
Include End Point IP for Context Lookup	<input checked="" type="checkbox"/>
Extensions	Avaya ▼
Diversion Manipulation	<input type="checkbox"/>
Diversion Condition	None ▼
Diversion Header URI	
Has Remote SBC	<input checked="" type="checkbox"/>
Route Response on Via Port	<input type="checkbox"/>
Relay INVITE Replace for SIPREC	<input type="checkbox"/>
<b>DTMF</b>	
DTMF Support	<input checked="" type="radio"/> None <input type="radio"/> SIP Notify <input type="radio"/> SIP Info <input type="radio"/> Inband
<input type="button" value="Finish"/>	

### 7.2.2. Server Interworking – Swisscom

Server Interworking allows the configuration and management of various SIP call server-specific capabilities such as call hold and T.38. From the left-hand menu select **Global Profiles** → **Server Interworking** and click on **Add** (not shown).

- Enter profile name such as Swisscom and click **Next** (Not Shown).
- Check **Hold Support** = **None**.
- Check **T.38 Support**.
- All other options on the **General** Tab can be left at default.

Click on **Next** on the following screens.

General	
Hold Support	<input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
URI Group	None ▼
Send Hold	<input type="checkbox"/>
Delayed Offer	<input type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
Re-Invite Handling	<input type="checkbox"/>
Prack Handling	<input type="checkbox"/>
Allow 18X SDP	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

Default values can be used for the **Advanced Settings** window. Click **Finish**.

Record Routes	<input type="radio"/> None <input type="radio"/> Single Side <input checked="" type="radio"/> Both Sides <input type="radio"/> Dialog-Initiate Only (Single Side) <input type="radio"/> Dialog-Initiate Only (Both Sides)
Include End Point IP for Context Lookup	<input checked="" type="checkbox"/>
Extensions	Avaya ▼
Diversion Manipulation	<input type="checkbox"/>
Diversion Condition	None ▼
Diversion Header URI	<input type="text"/>
Has Remote SBC	<input checked="" type="checkbox"/>
Route Response on Via Port	<input type="checkbox"/>
Relay INVITE Replace for SIPREC	<input type="checkbox"/>
<b>DTMF</b>	
DTMF Support	<input checked="" type="radio"/> None <input type="radio"/> SIP Notify <input type="radio"/> SIP Info <input type="radio"/> Inband
<input type="button" value="Finish"/>	

### 7.2.3. Server Configuration– Avaya

Servers are defined for each server connected to the Avaya SBCE. In this case, Swisscom is connected as the Trunk Server and Session Manager is connected as the Call Server.

The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow the configuration and management of various SIP call server-specific parameters such as TCP and UDP port assignments, IP Server type, heartbeat signalling parameters and some advanced options.

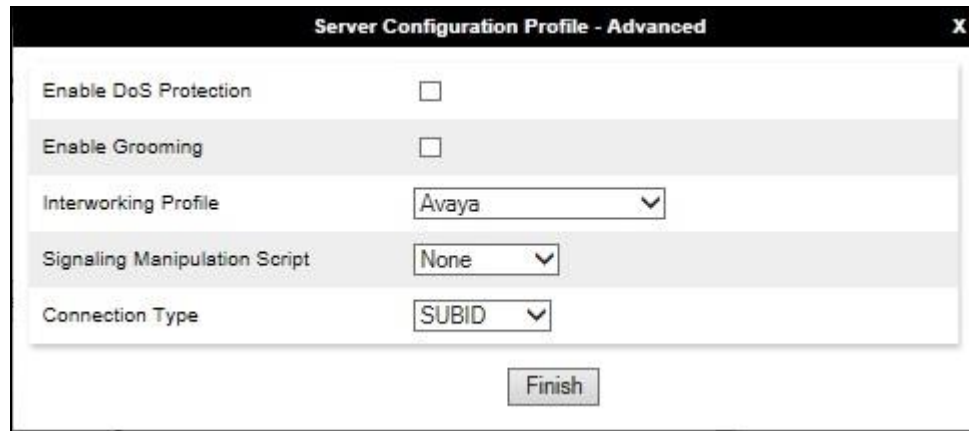
From the left-hand menu select **Global Profiles → Server Configuration** and click on **Add** (not shown) and enter a descriptive name. On the **Add Server Configuration Profile** tab, set the following:

- Select **Server Type** to be **Call Server**.
- Enter **IP Address / FQDN** to **10.10.3.55** (Session Manager IP Address).
- For **Port**, enter **5060**.
- For **Transport**, select **TCP**.
- Click on **Next** (not shown) to use default entries on the **Authentication** and **Heartbeat** tabs.

Server Configuration Profile - General		
IP Address / FQDN	Port	Transport
10.10.3.55	5060	TCP

On the **Advanced** tab:

- Select **Avaya** for **Interworking Profile**.
- Click **Finish**.



The screenshot shows the 'Server Configuration Profile - Advanced' dialog box. It contains the following settings:

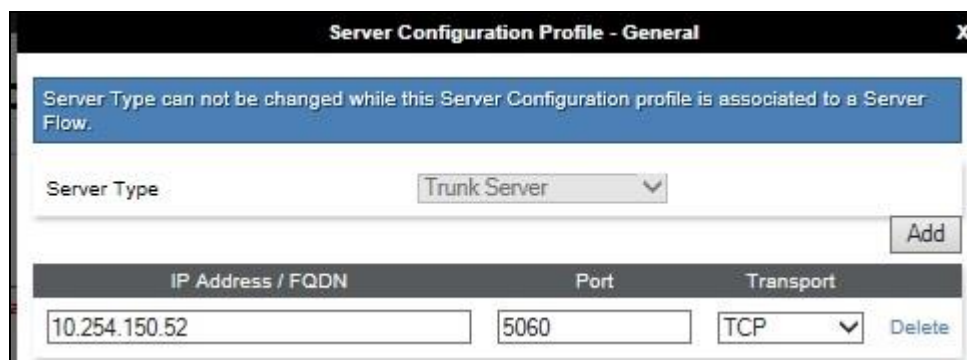
Option	Value
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	Avaya
Signaling Manipulation Script	None
Connection Type	SUBID

At the bottom right, there is a 'Finish' button.

#### 7.2.4. Server Configuration – Swisscom

To define the Swisscom SBC as a Trunk Server, navigate to **Global Profiles → Server Configuration** and click on **Add** (not shown) and enter a descriptive name. On the **Add Server Configuration Profile** tab, set the following:

- Select **Server Type** to be **Trunk Server**.
- Enter **IP Address / FQDN** to **10.254.150.52** (Swisscom SBC IP Address).
- For **Port**, enter **5060**.
- For **Transport**, select **TCP**.
- Click on **Next** (not shown) to use default entries on the **Authentication** and **Heartbeat** tabs.



The screenshot shows the 'Server Configuration Profile - General' dialog box. It contains the following settings:

Option	Value
Server Type	Trunk Server

Below the table, there is a table with the following data:

IP Address / FQDN	Port	Transport	Action
10.254.150.52	5060	TCP	Delete

At the bottom right, there is an 'Add' button.

On the Advanced tab:

- Select **Swisscom** for Interworking Profile.
- Click **Finish**.

Server Configuration Profile - Advanced	
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	Swisscom ▼
Signaling Manipulation Script	None ▼
Connection Type	SUBID ▼
<div>Finish</div>	



## 7.2.5. Routing

Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport settings, name server addresses and resolution methods, next hop routing information, and packet transport types.

Routing information is required for routing to Session Manager on the internal side and Swisscom addresses on the external side. The IP addresses and ports defined here will be used as the destination addresses for signalling. If no port is specified in the **Next Hop IP Address**, default 5060 is used.

### 7.2.5.1 Routing – Avaya

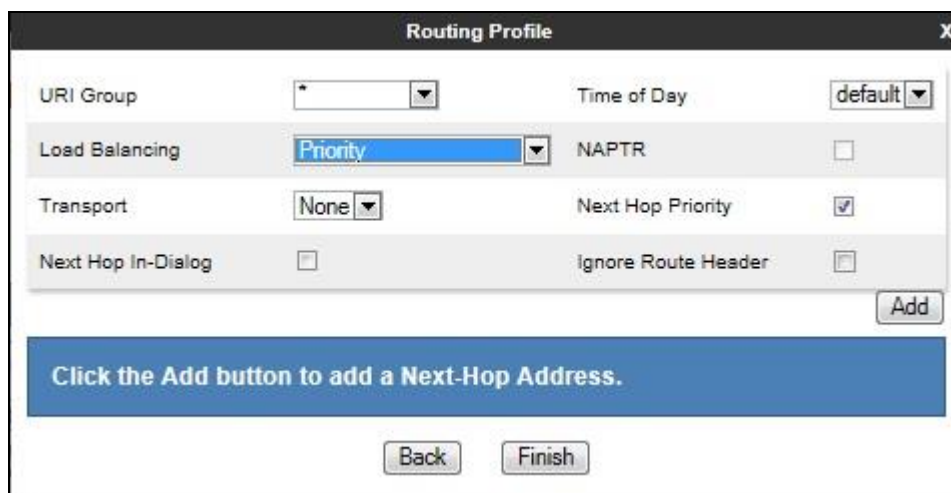
Create a Routing Profile for Session Manager.

- Navigate to **Global Profiles → Routing** and select **Add Profile** (not shown).
- Enter a **Profile Name** and click **Next**.



The image shows a 'Routing Profile' window. It has a title bar with 'Routing Profile' and a close button 'X'. Inside, there is a text input field labeled 'Profile Name' containing the text 'Avaya'. Below the input field is a 'Next' button.

The Routing Profile window will open. Use the default values displayed and click **Add**.



The image shows a 'Routing Profile' window with various settings. The title bar has 'Routing Profile' and a close button 'X'. The settings are as follows:

URI Group	Time of Day
*	default
Load Balancing	NAPTR
Priority	<input type="checkbox"/>
Transport	Next Hop Priority
None	<input checked="" type="checkbox"/>
Next Hop In-Dialog	Ignore Route Header
<input type="checkbox"/>	<input type="checkbox"/>

Below the settings is an 'Add' button. At the bottom, there is a blue banner with the text 'Click the Add button to add a Next-Hop Address.' and two buttons: 'Back' and 'Finish'.

On the **Next Hop Address** window, set the following:

- **Priority/Weight = 1.**
- **Server Configuration = Avaya** (Section 7.2.3) from drop down menu.
- **Next Hop Address = Select 10.10.3.55:5060 TCP** from drop down menu.
- Click **Finish**.

Priority / Weight	Server Configuration	Next Hop Address	Transport
1	Avaya	10.10.3.55:5060 (TCP)	None

### 7.2.5.2 Routing – Swisscom

Create a Routing Profile for Swisscom.

- Navigate to **Global Profiles → Routing** and select **Add Profile** (not shown).
- Enter a **Profile Name** and click **Next**.

Profile Name: Swisscom

Next

The Routing Profile window will open. Use the default values displayed and click **Add**.



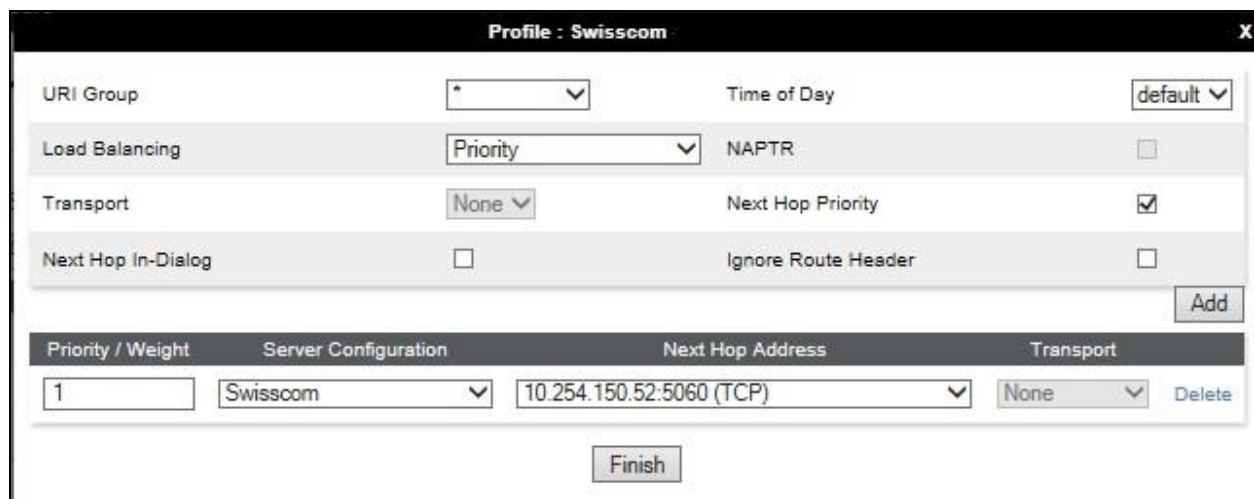
The Routing Profile window is a dialog box with a title bar that says "Routing Profile" and a close button (X). It contains several configuration fields:

- URI Group: A dropdown menu with an asterisk (\*) as the selected value.
- Time of Day: A dropdown menu with "default" as the selected value.
- Load Balancing: A dropdown menu with "Priority" as the selected value.
- NAPTR: A checkbox that is currently unchecked.
- Transport: A dropdown menu with "None" as the selected value.
- Next Hop Priority: A checkbox that is currently checked.
- Next Hop In-Dialog: A checkbox that is currently unchecked.
- Ignore Route Header: A checkbox that is currently unchecked.

At the bottom right, there is an "Add" button. Below the configuration fields, there is a blue banner with the text "Click the Add button to add a Next-Hop Address." At the very bottom, there are "Back" and "Finish" buttons.

On the **Next Hop Address** window, set the following:

- **Priority/Weight = 1.**
- **Server Configuration = Swisscom** (Section 7.2.4) from drop down menu.
- **Next Hop Address = Select 10.254.150.52:5060 TCP** from drop down menu.
- Click **Finish**.



The Profile : Swisscom window is a dialog box with a title bar that says "Profile : Swisscom" and a close button (X). It contains the same configuration fields as the Routing Profile window, but with an additional "Add" button at the bottom right. Below the configuration fields, there is a table with the following data:

Priority / Weight	Server Configuration	Next Hop Address	Transport	
1	Swisscom	10.254.150.52:5060 (TCP)	None	Delete

At the bottom, there is a "Finish" button.

### 7.2.6. Topology Hiding

Topology hiding is used to hide local information such as private IP addresses and local domain names. The local information can be overwritten with a domain name or IP addresses. The default **Replace Action** is **Auto**, this replaces local information with IP addresses, generally the next hop. Topology hiding has the advantage of presenting single Via and Record-Route headers externally where multiple headers may be received from the enterprise. In some cases where Topology Hiding can't be applied, in particular the Contact header, IP addresses are translated to the Avaya SBCE external addresses using NAT.

To define Topology Hiding for Session Manager, navigate to **Global Profiles → Topology Hiding** from menu on the left hand side. Click on **Add** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- Enter a descriptive Profile Name such as **Avaya**.
- If the required Header is not shown, click on **Add Header**.
- Under the **Header** field for **To**, **From** and **Request Line**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For Overwrite value, insert **avaya.com**.
- Click **Finish** (not shown).

Topology Hiding Profiles: Avaya

Add

Topology Hiding Profiles

default

cisco\_th\_profile

**Avaya**

Swisscom

RenameCloneDelete

It is not recommended to edit the defaults. Try cloning or adding a new profile instead.

Topology Hiding

Header	Criteria	Replace Action	Overwrite Value
Referred-By	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
To	IP/Domain	Overwrite	avaya.com
From	IP/Domain	Overwrite	avaya.com
Via	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---
Refer-To	IP/Domain	Auto	---
Request-Line	IP/Domain	Overwrite	avaya.com

Edit

To define Topology Hiding for Swisscom, navigate to **Global Profiles → Topology Hiding** from the menu on the left hand side. Click on **Add** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- In the **Profile Name** field enter a descriptive name for Swisscom and click **Next**.
- If the required Header is not shown, click on **Add Header**.
- Under the **Header** field for **To**, **From** and **Request Line**, select **IP/Domain** under **Criteria** and **Auto** under **Replace Action**.
- Click **Finish** (not shown).

### Topology Hiding Profiles: Swisscom

Add

Topology Hiding Profiles

default

cisco\_th\_profile

Avaya

Swisscom

Rename

Clone

Delete

Click here to add a description.

Topology Hiding

Header	Criteria	Replace Action	Overwrite Value
Referred-By	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
To	IP/Domain	Auto	---
From	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---
Refer-To	IP/Domain	Auto	---
Request-Line	IP/Domain	Auto	---

Edit

### 7.3. Define Network Information

Network information is required on the Avaya SBCE to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one interface assigned.

To define the network information, navigate to **Device Specific Settings → Network Management** from the menu on the left-hand side and click on **Add**. Enter details in the blank box that appears at the end of the list.

- Define the internal IP address with screening mask and assign to interface **A1**.
- Select **Save** to save the information.
- Click on **Add**.
- Define the external IP address with screening mask and assign to interface **B1**.
- Select **Save** to save the information.
- Click on **System Management** in the main menu.
- Select **Restart Application** indicated by an icon in the status bar (not shown).

Network Management: ASBCE\_R63

Devices: ASBCE\_R63

Interfaces: Networks

Name	Gateway	Subnet Mask	Interface	IP Address	
A1_Internal	10.10.3.1	255.255.255.0	A1	10.10.3.45	Edit Delete
B1_External	192.168.37.1	255.255.255.240	B1	192.168.37.2	Edit Delete

Add

Select the **Interface Configuration** Tab and click on the **Status** of the interface to toggle from **Disabled** to **Enabled**.

Network Management: ASBCE\_R63

Devices: ASBCE\_R63

Interfaces: Networks

Interface Name	VLAN Tag	Status
A1		Enabled
A2		Disabled
B1		Enabled
B2		Disabled

Add VLAN

## 7.4. Define Interfaces

When the IP addresses and masks are assigned to the interfaces, these are then configured as signalling and media interfaces.

### 7.4.1. Signalling Interfaces

To define the signalling interfaces on the Avaya SBCE, navigate to **Device Specific Settings** → **Signaling Interface** from the menu on the left hand side. Details of transport protocol and ports for the internal and external SIP signalling are entered here.

To enter details of transport protocol and ports for the SIP signalling on the internal interface:

- Select **Add** and enter details of the internal signalling interface in the pop-up menu (not shown).
- In the **Name** field enter a descriptive name for the interface.
- For **Signaling IP**, select the **internal** signalling interface IP addresses defined in **Section 7.3**.
- Select **TCP** port number, **5060** is used for Session Manager.

To enter details of transport protocol and ports for the SIP signalling on the external interface:

- Select **Add** and enter details of the external signalling interface in the pop-up menu (not shown).
- In the **Name** field enter a descriptive name for the external signalling interface.
- For **Signaling IP**, select the **external** signalling interface IP address defined in **Section 7.3**.
- Select **TCP** port number, **5060** is used for the Swisscom SIP trunk.

The following screen shows the Signalling Interfaces created in the sample configuration for the inside and outside IP interfaces.

Signaling Interface: ASBCE\_R63

Devices

ASBCE\_R63

Signaling Interface

Modifying or deleting an existing signaling interface will require an application restart before taking effect. Application restarts can be issued from [System Management](#).

Add

Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile	
Int_Signalling	10.10.3.45 A1_Internal (A1, VLAN 0)	5060	5060	---	None	Edit Delete
Ext_Signalling	192.168.37.2 B1_External (B1, VLAN 0)	5060	5060	---	None	Edit Delete

## 7.4.2. Media Interfaces

To define the media interfaces on the Avaya SBCE, navigate to **Device Specific Settings** → **Media Interface** from the menu on the left hand side. Details of the RTP and SRTP port ranges for the internal and external media streams are entered here. The IP addresses for media can be the same as those used for signalling.

To enter details of the media IP and RTP port range on the internal interface to be used in the server flow:

- Select **Add Media Interface** and enter details in the pop-up menu (not shown).
- In the **Name** field enter a descriptive name for the internal media interface.
- For **Media IP**, select the **internal** media interface IP address defined in **Section 7.3**.
- Select **RTP port** ranges for the media path with the enterprise end-points.

To enter details of the media IP and RTP port range on the external interface to be used in the server flow.

- Select **Add Media Interface** and enter details in the pop-up menu (not shown).
- In the **Name** field enter a descriptive name for the external media interface.
- For **Media IP**, select the **external** media interface IP address defined in **Section 7.3**.
- Select **RTP port** ranges for the external media path.

The following screen shows the Media Interfaces created in the sample configuration for the inside and outside IP interfaces.

Media Interface: ASBCE\_R63

Devices

ASBCE\_R63

Media Interface

Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from [System Management](#).

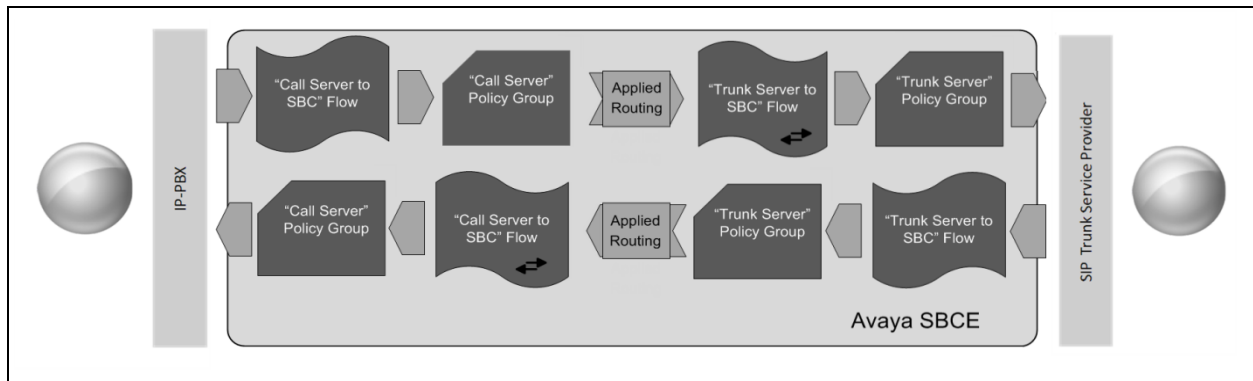
Add

Name	Media IP Network	Port Range	Edit	Delete
Int_Media	10.10.3.45 A1_Internal (A1, VLAN 0)	35000 - 40000	Edit	Delete
Ext_Media	192.168.37.2 B1_External (B1, VLAN 0)	35000 - 40000	Edit	Delete



## 7.5. Server Flows

Server Flows combine the previously defined profiles into outgoing flows from Session Manager to Swisscom's SIP trunk and incoming flows from Swisscom's SIP trunk to Session Manager. This configuration ties all the previously entered information together so that signalling can be routed from Session Manager to the PSTN via the Swisscom network and vice versa. The following screen illustrates the flow through the Avaya SBCE to secure a SIP trunk call.



This configuration ties all the previously entered information together so that calls can be routed from Session Manager to Swisscom SIP trunk and vice versa. The following screenshot shows all configured flows.

Subscriber Flows

Server Flows

Add

Hover over a row to see its description.

Server Configuration: Avaya

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Call_Server	*	Ext_Signalling	Int_Signalling	default-low	Swisscom	View	Clone	Edit	Delete

Server Configuration: Swisscom

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Trunk_Server	*	Int_Signalling	Ext_Signalling	default-low	Avaya	View	Clone	Edit	Delete

To define a Server Flow for the Swisscom SIP trunk, navigate to **Device Specific Settings** → **End Point Flows**.

- Click on the **Server Flows** tab.
- Select **Add Flow** and enter details in the pop-up menu.
- In the **Name** field enter a descriptive name for the server flow for Swisscom SIP trunk, in the test environment **Trunk\_Server** was used.
- In the **Server Configuration** drop-down menu, select the Swisscom server configuration defined in **Section 7.2.4**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 7.4.1**. This is the interface that signalling bound for the Swisscom SIP trunk is received on.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.4.1**. This is the interface that signalling bound for Swisscom SIP trunk is sent on.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 7.4.2**. This is the interface that media bound for Swisscom SIP trunk is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of Session Manager defined in **Section 7.2.5**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the Swisscom SIP trunk defined in **Section 7.2.6** and click **Finish**.

Flow: Trunk_Server	
Flow Name	Trunk_Server
Server Configuration	Swisscom
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Int_Signalling
Signaling Interface	Ext_Signalling
Media Interface	Ext_Media
End Point Policy Group	default-low
Routing Profile	Avaya
Topology Hiding Profile	Swisscom
File Transfer Profile	None
Signaling Manipulation Script	None
Remote Branch Office	Any
<b>Finish</b>	

To define a Server Flow for Session Manager, navigate to **Device Specific Settings → End Point Flows**.

- Click on the **Server Flows** tab.
- Select **Add Flow** and enter details in the pop-up menu.
- In the **Name** field enter a descriptive name for the server flow for Session Manager, in the test environment **Call\_Server** was used.
- In the **Server Configuration** drop-down menu, select the Session Manager server configuration defined in **Section 7.2.3**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 7.4.1**. This is the interface that signalling bound for Session Manager is received on.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.4.1**. This is the interface that signalling bound for Session Manager is sent on.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 7.4.2**. This is the interface that media bound for Session Manager is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of the Swisscom SIP trunk defined in **Section 7.2.5**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of Session Manager defined in **Section 7.2.6** and click **Finish**.

The screenshot shows a configuration window titled "Flow: Call\_Server" with a close button (X) in the top right corner. The window contains a list of configuration fields, each with a label and a corresponding input field or dropdown menu. The fields are as follows:

Field Label	Value
Flow Name	Call_Server
Server Configuration	Avaya
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Ext_Signalling
Signaling Interface	Int_Signalling
Media Interface	Int_Media
End Point Policy Group	default-low
Routing Profile	Swisscom
Topology Hiding Profile	Avaya
File Transfer Profile	None
Signaling Manipulation Script	None
Remote Branch Office	Any

At the bottom of the window, there is a "Finish" button.

## 8. Swisscom Enterprise SIP Service Configuration

The configuration of the Swisscom equipment used to support Swisscom's Enterprise SIP service is outside of the scope of these Application Notes and will not be covered. To obtain further information on Swisscom equipment and system configuration please contact an authorized Swisscom 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 Entities from the list and observe if the **Conn Status** and **Link Status** are showing as **UP**.

SIP Entity Name	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
Communication Manager	10.10.8.67	5060	TCP	FALSE	UP	200 OK	UP
Avaya SBCE	10.10.3.30	5060	TCP	FALSE	UP	200 OK	UP
CS1K R7.6	10.10.9.21	5060	TCP	FALSE	UP	200 OK	UP
Avaya SBCE 2	10.10.9.71	5060	TCP	FALSE	UP	200 OK	UP
Messaging	10.10.2.82	5060	TCP	FALSE	UP	200 OK	UP

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

```
status trunk 1
```

Member	Port	Service State	Mtce Connected Ports Busy
0001/001	T00001	<b>in-service/idle</b>	no
0001/002	T00002	in-service/idle	no
0001/003	T00003	in-service/idle	no
0001/004	T00004	in-service/idle	no
0001/005	T00005	in-service/idle	no
0001/006	T00006	in-service/idle	no
0001/007	T00007	in-service/idle	no
0001/008	T00008	in-service/idle	no
0001/009	T00009	in-service/idle	no
0001/010	T00010	in-service/idle	no

3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
5. Verify that the user on the PSTN can end an active call by hanging up.
6. Verify that an endpoint at the enterprise site can end an active call by hanging up.
7. Should issues arise with the SIP trunk, use the Avaya SBCE trace facility to check that the OPTIONS requests sent from Session Manager via the Avaya SBCE to the network SBCs are receiving a response.

To define the trace, navigate to **Device Specific Settings → Advanced Options → Troubleshooting → Trace** in the main menu on the left hand side and select the **Packet Capture** tab.

- Select the SIP trunk interface from the **Interface** drop down menu.
- Select **All** from the **Local Address** drop down menu.
- Enter the IP address of the network SBC in the **Remote Address** field or enter a \* to capture all traffic.
- Specify the **Maximum Number of Packets to Capture**, 10000 is shown as an example.
- Specify the filename of the resultant pcap file in the **Capture Filename** field.
- Click on **Start Capture**.

Trace: ASBCE\_R63

Devices

ASBCE\_R63

Packet Capture

Captures

Packet Capture Configuration

Status

Ready

Interface

B1

Local Address  
IP[:Port]

All

Remote Address  
\*, \*Port, IP, IP:Port

\*

Protocol

All

Maximum Number of Packets to Capture

10000

Capture Filename  
Using the name of an existing capture will overwrite it.

test.pcap

Start Capture

Clear

To view the trace, select the **Captures** tab and click on the relevant filename in the list of traces.

Trace: ASBCE\_R63

Devices

ASBCE\_R63

Packet Capture

Captures

Refresh

File Name	File Size (bytes)	Last Modified	
test_20170601095310.pcap	4,096	June 1, 2017 3:53:28 PM IST	Delete

The trace is viewed as a standard pcap file in Wireshark. If the SIP trunk is working correctly, a SIP response in the form of a 200 OK will be seen from the Swisscom network.

## 10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager R6.3 as an Evolution Server, Avaya Aura® Session Manager R6.3 and Avaya Session Border Controller for Enterprise to Swisscom Enterprise SIP service. Swisscom Enterprise SIP service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**.

## 11. Additional References

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

- [1] *Avaya Aura® Communication Manager using VMware® in the Virtualized Environment Deployment Guide*, May 2016
- [2] *Avaya Aura® Communication Manager 6.3 Documentation library*, May 2016
- [3] *Avaya Aura® System Manager using VMware® in the Virtualized Environment Deployment Guide Release 6.3* May 2016
- [4] *Implementing Avaya Aura® System Manager Release 6.3*, Apr 2015
- [5] *Upgrading Avaya Aura® System Manager to Release 6.3*, May 2016
- [6] *Administering Avaya Aura® System Manager Release 6.3*, Feb 2017
- [7] *Avaya Aura® Session Manager using VMware® in the Virtualized Environment Deployment Guide Release 6.3*, Nov 2014
- [8] *Implementing Avaya Aura® Session Manager Release 6.3*, Aug 2014
- [9] *Upgrading Avaya Aura® Session Manager Release 6.3*, Aug 2014
- [10] *Administering Avaya Aura® Session Manager Release 6.3*, May 2016
- [11] *Deploying Avaya Session Border Controller for Enterprise Release 6.3*, Aug 2015
- [12] *Upgrading Avaya Session Border Controller for Enterprise Release 6.3*, Oct 2014
- [13] *Administering Avaya Session Border Controller for Enterprise Release 6.3*, Feb 2016
- [14] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>

---

**©2017 Avaya Inc. All Rights Reserved.**

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at [devconnect@avaya.com](mailto:devconnect@avaya.com).