



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R7.0 as an Evolution Server, Avaya Aura® Session Manager R7.0 and Avaya Session Border Controller for Enterprise R7.1 to support TalkTalk Business SIP Trunk Service - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between TalkTalk Business Next Generation Voice Services (NGVS) platform 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. TalkTalk 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.

1. Introduction

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between TalkTalk's NGVS SIP Trunk 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 TalkTalk NGVS 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 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 TalkTalk. 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 the G.711A and G.711MU codecs.
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using G.711 pass-through transmission.
- 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 and Avaya Communicator for Windows 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.

2.2. Test Results

Interoperability testing of the test configuration was completed with successful results for TalkTalk's NGVS SIP Trunk service with the following observations:

- G.729 codec is not supported by TalkTalk and therefore was not tested.
- T.38 fax transmission is not supported by TalkTalk and therefore was not tested.
- When performing call re-direction or EC500 calls, TalkTalk require the P-Asserted-Identity Header to be populated with the CLID of the set that is performing the call re-direction/EC500 call or of a known TalkTalk CLID on the PABX. In order for call redirection/EC500 to complete successfully, a SigMa script was created on the Avaya SBCE to copy the CLID from the Diversion Header and populate the P-Asserted-Identity Header with the Diversion Header CLID. The details of the Sigma Script are outlined in **Section 7.2.7**.
- Avaya EC500 features such as on-net and off-net calling were not tested as the From Header CLID containing the EC500 mobility number on inbound calls to TalkTalk NGVS SIP trunk was automatically changed by TalkTalk to a CLID number recognizable to the TalkTalk network.
- All unwanted Avaya proprietary SIP headers and MIME was stripped on outbound calls using the Adaptation Module in Session Manager.
- 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 with the Emergency Services Operator.

2.3. Support

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

For technical support on the TalkTalk Business NGVS SIP Trunk Service, please contact TalkTalk at ngvssupport@talktalkbusiness.co.uk

3. Reference Configuration

The following equipment in **Figure 1** illustrates the test configuration. The test configuration shows an Enterprise site connected to TalkTalk's NGVS SIP Trunk 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 were Avaya One-X® Communicator and Avaya Communicator for Windows soft phones running on a laptop PC.

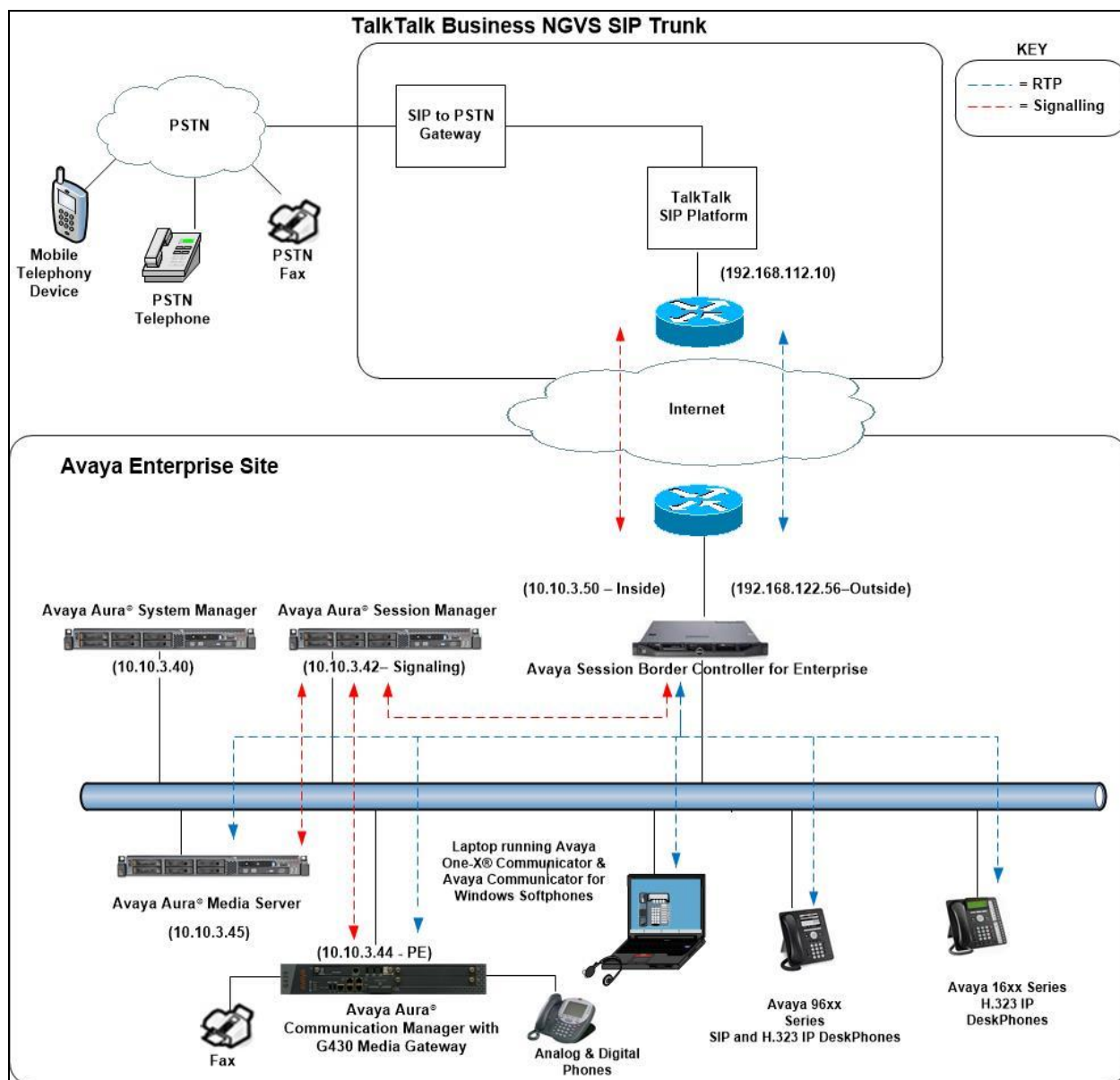


Figure 1: Test Setup TalkTalk Business NGVS SIP Trunk to Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Dell PowerEdge R620 running System Manager on VM Version 8	7.0.1.2 - Build No. - 7.0.0.0.16266 Software Update Revision No: 7.0.1.2.086007 Service Pack 2
Dell PowerEdge R620 running Session Manager on VM Version 8	7.0.1.2.701230
Avaya S8300D Server running Avaya Aura® Communication Manager	R017x.00.0.441.0 (23523)
Avaya G430 Media Gateway	7.0.1.0 (g430_sw_37_41_0)
Avaya Aura® Media Server	7.7.0.375
Avaya Session Border Controller for Enterprise	7.1.0.1-07-12090
Avaya 1600 IP Deskphone (H.323)	1.3.10
Avaya 9670 IP DeskPhone (H.323)	6.6
Avaya 96x0 IP DeskPhone (H.323)	6.6
Avaya 9611 IP DeskPhone (SIP)	7.0
Avaya 9608 IP DeskPhone (SIP)	7.0
Avaya 9621 IP DeskPhone (SIP)	7.0
Avaya 9608 IP DeskPhone (SIP)	7.0
Avaya one-X® Communicator (H.323 & SIP)	6.2.12.04-FP12
Avaya Communicator for Windows	2.1.3.0
Analogue Handset	N/A
Analogue Fax	N/A
TalkTalk	
Asterisk Softswitch	Version 1.8.9.3
Kamailio SIP Server	Version 4.2

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 TalkTalk 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 TalkTalk 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 TalkTalk SIP 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:			41000	0
Maximum Video Capable IP Softphones:			18000	0
Maximum Administered SIP Trunks:			4000	10
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, **SM100** and **10.10.3.42** 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	
SM100	10.10.3.42	
default	0.0.0.0	
procr	10.10.3.44	
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
    H.323 IP ENDPOINTS      AUDIO RESOURCE RESERVATION PARAMETERS
        H.323 Link Bounce Recovery? y      RSVP Enabled? n
        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 TalkTalk 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

TalkTalk SIP trunk supports pass-through for transmission of fax. Navigate to **Page 2** and define fax properties as follows:

- Set the **FAX - Mode** to **pass-through**.

change ip-codec-set 1 Page 2 of 2

IP CODEC SET

Allow Direct-IP Multimedia? n

	Mode	Redundancy	Packet Size (ms)
FAX	pass-through	0	
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 TalkTalk SIP Trunk network. During test, this was configured to use TCP and port 5060 to facilitate tracing and fault analysis. 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 **SM100** 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).
- Set **Direct IP-IP Audio Connections** to **y**.
- Set **Initial IP-IP Direct Media** to **y** to enable Direct Media.
- Set **H.323 Station Outgoing Direct Media** to **y**.
- Leave **DTMF over IP** at default value of **rtp-payload** (Enables **RFC2833** for DTMF transmission from Communication Manager).

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: SM100
Near-end Listen Port: 5060		Far-end Listen Port: 5060
Far-end Domain:		Far-end Network Region: 1
Incoming Dialog Loopbacks: eliminate		Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload		RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3		Direct IP-IP Audio Connections? y
Enable Layer 3 Test? y		IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? y		Initial IP-IP Direct Media? y
		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 TalkTalk to prevent unnecessary SIP messages during call setup. During the compliance testing, **Preferred Minimum Session Refresh Interval (sec)** was set to **600**.

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): 600			
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 **private**.

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

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 TalkTalk.
- 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	
	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? y	
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 private-numbering x** command to configure Communication Manager to send the calling party number in the format required. This calling party number is sent in the SIP From, Contact and PAI headers, and displayed on display-equipped PSTN telephones.

change private-numbering					Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT					
Ext	Ext	Trk	CPN	Total	
Len	Code	Grp(s)	Prefix	CPN	
				Len	
4	6010	1	0203xxxxx95	11	Total Administered: 6
4	6102	1	0330xxxxx17	11	Maximum Entries: 240

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 TalkTalk's SIP trunk. 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		
Auto Route Selection (ARS) - Access Code 1: 9		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 TalkTalk 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 TalkTalk correlate to the internal extensions assigned within Communication Manager. The entries displayed below translate incoming DDI numbers **0330xxxxx17** and **0203xxxxx95** 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/	Number	Number	Del Insert		
Feature	Len	Digits			
public-ntwrk	10	0330xxxxx17	all	6102	
public-ntwrk	10	0203xxxxx95	all	6010	

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	Application	Dial	CC	Phone Number	Trunk	Config	Dual	
Extension		Prefix			Selection	Set	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

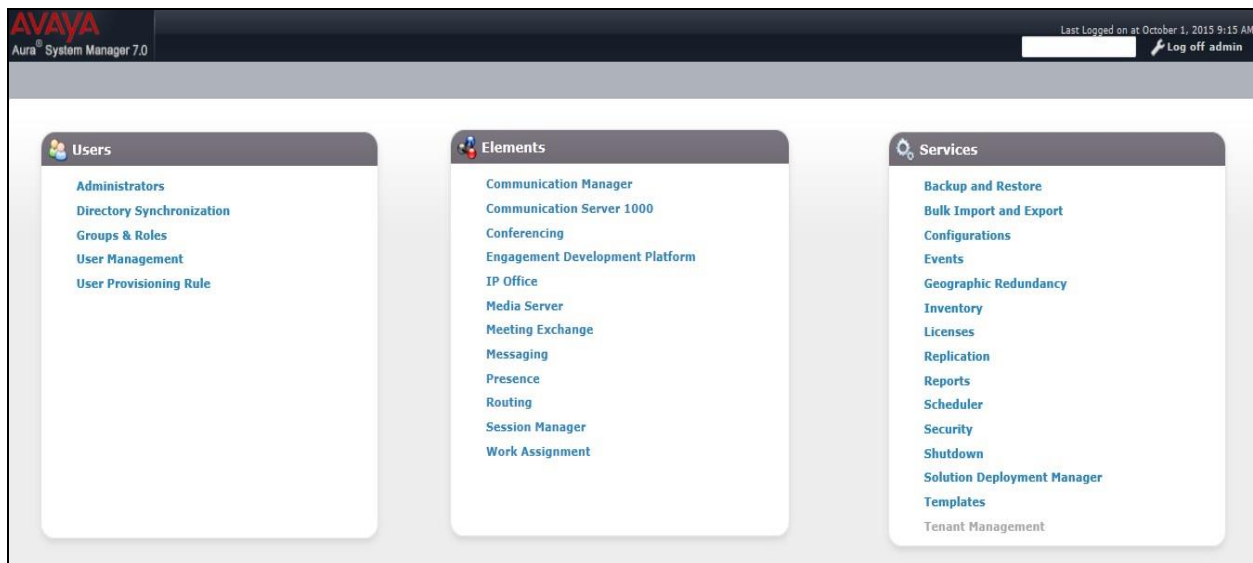
This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

- 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.

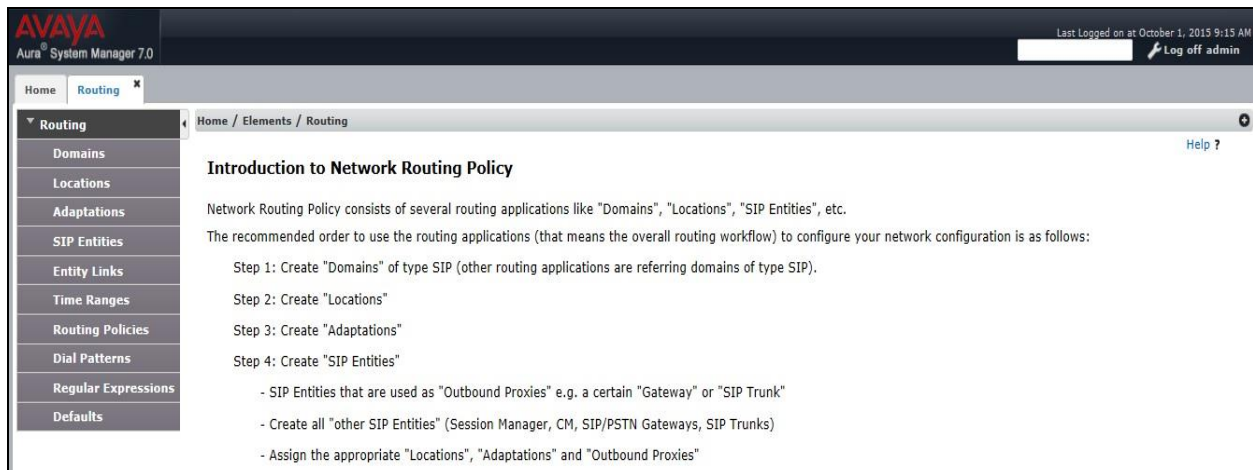
It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session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

6.1. Log in to Avaya Aura® System Manager

Access 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.

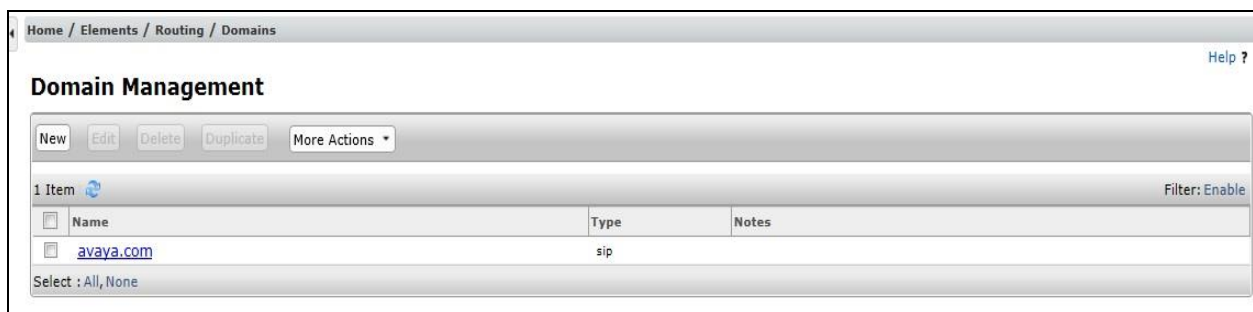


6.2. Administer SIP Domain

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 **SM_7** defined for the compliance testing.

The screenshot shows the 'Location Details' form for a location named 'SM_7'. The form is divided into several sections:

- General:** Contains fields for 'Name' (set to 'SM_7') and 'Notes'.
- Dial Plan Transparency in Survivable Mode:** Includes a checkbox for 'Enabled' (unchecked), and fields for 'Listed Directory Number' and 'Associated CM SIP Entity'.
- Overall Managed Bandwidth:** Includes a dropdown for 'Managed Bandwidth Units' (set to 'Kbit/sec'), and fields for 'Total Bandwidth' and 'Multimedia Bandwidth'.
- Audio Calls Can Take Multimedia Bandwidth:** A checkbox (checked).
- Location Pattern:** A table with 3 items. The table has columns for 'IP Address Pattern' and 'Notes'. The patterns listed are '10.10.3.*', '10.10.5.*', and '10.10.8.*'. Below the table is a 'Select' dropdown set to 'All, None'.

Buttons for 'Commit' and 'Cancel' are present at the top right and bottom right of the form.

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. In order to improve interoperability with third party elements, Session Manager 7.0 incorporates the ability to use Adaptation modules to remove specific SIP headers that are either Avaya proprietary or deemed excessive/unnecessary for non-Avaya elements.

For the compliance test, an Adaptation named “**TalkTalk**” was created to block the following headers from outbound messages, before they were forwarded to the Avaya SBCE: AV-Global-Session-ID, AV-Correlation-ID, Alert-Info, Endpoint-View, P-AV-Message-ID, P-Charging-Vector, and P-Location. These headers contain private information from the enterprise, which should not be propagated outside of the enterprise boundaries. They also add unnecessary size to outbound messages, while they have no significance to the service provider.

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 **Adaptation Details → General**:

- **Adaption Name:** Enter an appropriate name such as **TalkTalk**.
- **Module Name:** Select **DigitConversionAdapter**.
- **Modular Parameter Type:** Select **Name-Value Parameter**.

Click **Add** to add the name and value parameters.

- **Name:** Enter **eRHdrs**. This parameter will remove the specific headers from messages in the egress direction.
- **Value:** Enter “**AV-Global-Session-ID, AV-Correlation-ID, Alert-Info, Endpoint-View, P-AV-Message-ID, P-Charging-Vector, P-Location**”.
- **Name:** Enter **fromto**. Modifies From and To header of a message.
- **Value:** Enter **true**.
- **Name:** Enter **MIME**. Remove MIME message bodies from Session Manager.
- **Value:** Enter **no**.

Home / Elements / Routing / Adaptations

Adaptation Details [Commit] [Cancel] [Help ?]

General

* **Adaption Name:** TalkTalk

* **Module Name:** DigitConversionAdapter

Module Parameter Type: Name-Value Parameter

Name	Value
<input type="checkbox"/> eRHdrs	"Alert-Info, P-Charging-Vector, AV-Global-Session-ID, P-Location, P-AV-Message-id, Endpoint-View"
<input type="checkbox"/> fromto	true
<input type="checkbox"/> MIME	no

Select : All, None

Egress URI Parameters:

Notes:

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 navigation at the top is 'Home / Elements / Routing / SIP Entities'. The page title is 'SIP Entity Details' with 'Commit' and 'Cancel' buttons. The 'General' tab is selected. The form contains the following fields:

- Name:** Session Manager
- FQDN or IP Address:** 10.10.3.42
- Type:** Session Manager (dropdown)
- Notes:** (empty text area)
- Location:** SM_7 (dropdown)
- Outbound Proxy:** (empty dropdown)
- Time Zone:** Europe/Dublin (dropdown)
- Credential name:** (empty text area)
- SIP Link Monitoring:** Use Session Manager Configuration (dropdown)

The 'SIP Link Monitoring' section is also visible at the bottom.

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 'Listen Ports' configuration section. It includes fields for 'TCP Failover port' and 'TLS Failover port'. Below these are 'Add' and 'Remove' buttons. A table lists the configured listen ports:

Listen Ports	Protocol	Default Domain	Notes
5060	TCP	avaya.com	
5060	UDP	avaya.com	
5061	TLS	avaya.com	

At the bottom, there is a 'Select : All, None' option and a 'Filter: Enable' button.

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.

The screenshot shows the 'SIP Entity Details' configuration page. The breadcrumb navigation at the top is 'Home / Elements / Routing / SIP Entities'. The page has a 'Commit' button and a 'Cancel' button in the top right corner. The 'General' tab is selected. The configuration fields are as follows:

- Name:** Communication_Manager
- * FQDN or IP Address:** 10.10.3.44
- Type:** CM (dropdown menu)
- Notes:** (empty text field)
- Adaptation:** (dropdown menu)
- Location:** SM_7 (dropdown menu)
- Time Zone:** Europe/Dublin (dropdown menu)
- * SIP Timer B/F (in seconds):** 4
- Credential name:** (empty text field)
- Securable:** ☐
- Call Detail Recording:** none (dropdown menu)

The 'Loop Detection' tab is also visible, with the following field:

- Loop Detection Mode:** Off (dropdown menu)

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

The screenshot shows the 'SIP Link Monitoring' configuration page. The breadcrumb navigation at the top is 'Home / Elements / Routing / SIP Entities'. The page has a 'Commit' button and a 'Cancel' button in the top right corner. The 'SIP Link Monitoring' tab is selected. The configuration fields are as follows:

- SIP Link Monitoring:** Use Session Manager Configuration (dropdown menu)

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.

The screenshot shows a web-based configuration interface for SIP Entities. The breadcrumb navigation at the top reads 'Home / Elements / Routing / SIP Entities'. The main heading is 'SIP Entity Details', with 'General' selected as the tab. In the top right corner are 'Commit' and 'Cancel' buttons. The 'General' section contains the following fields: 'Name' (Avaya_SBCE), 'FQDN or IP Address' (10.10.3.50), 'Type' (SIP Trunk), 'Notes' (empty), 'Adaptation' (TalkTalk), 'Location' (SM_7), 'Time Zone' (Europe/Dublin), 'SIP Timer B/F (in seconds)' (4), 'Credential name' (empty), 'Securable' (checkbox), and 'Call Detail Recording' (none). Below this is the 'Loop Detection' section with 'Loop Detection Mode' (On) and 'Loop Count Threshold' (5).

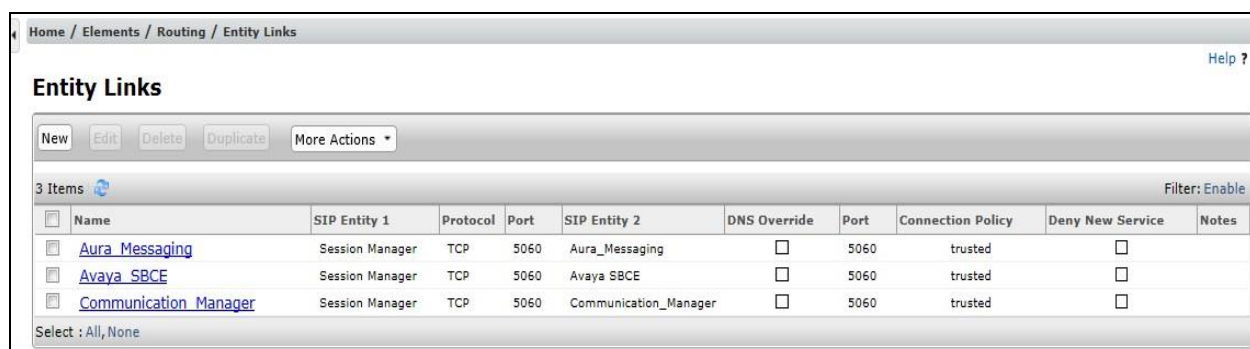
SIP Entity Details		Commit	Cancel
General			
* Name:	Avaya_SBCE		
* FQDN or IP Address:	10.10.3.50		
Type:	SIP Trunk		
Notes:			
Adaptation:	TalkTalk		
Location:	SM_7		
Time Zone:	Europe/Dublin		
* SIP Timer B/F (in seconds):	4		
Credential name:			
Securable:	<input type="checkbox"/>		
Call Detail Recording:	none		
Loop Detection			
Loop Detection Mode:	On		
Loop Count Threshold:	5		

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.



	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy	Deny New Service	Notes
<input type="checkbox"/>	Aura_Messaging	Session Manager	TCP	5060	Aura_Messaging	<input type="checkbox"/>	5060	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	Avaya SBCE	Session Manager	TCP	5060	Avaya SBCE	<input type="checkbox"/>	5060	trusted	<input type="checkbox"/>	
<input type="checkbox"/>	Communication_Manager	Session Manager	TCP	5060	Communication_Manager	<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' web interface. At the top, there is a breadcrumb trail: 'Home / Elements / Routing / Routing Policies'. On the right, there is a 'Help ?' link. Below the breadcrumb, the title 'Routing Policy Details' is displayed, followed by 'Commit' and 'Cancel' buttons. The 'General' section contains the following fields: 'Name' (with a red asterisk) set to 'to_Communication Manager', 'Disabled' (checkbox), 'Retries' (with a red asterisk) set to '0', and 'Notes'. The 'SIP Entity as Destination' section has a 'Select' button above a table. The table has columns: 'Name', 'FQDN or IP Address', 'Type', and 'Notes'. It contains one row: 'Communication_Manager', '10.10.3.44', 'CM'. The 'Time of Day' section has 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. Below these is a table with 1 item. The table has columns: 'Ranking', 'Name', 'Mon', 'Tue', 'Wed', 'Thu', 'Fri', 'Sat', 'Sun', 'Start Time', 'End Time', and 'Notes'. The row shows '0' in Ranking, '24/7' in Name, and checkmarks for all days of the week. The Start Time is '00:00' and the End Time is '23:59'. The Notes column says '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: to_Communication Manager

Disabled: ☐

* Retries: 0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Communication_Manager	10.10.3.44	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 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

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

The screenshot shows the 'Routing Policy Details' configuration page. The breadcrumb trail is 'Home / Elements / Routing / Routing Policies'. The page title is 'Routing Policy Details' with 'Commit' and 'Cancel' buttons. The 'General' tab is active, showing fields for 'Name' (to_Avaya_SBCE), 'Disabled' (checkbox), 'Retries' (0), and 'Notes'. The 'SIP Entity as Destination' section has a 'Select' dropdown and a table with one entry: 'Avaya_SBCE' with FQDN or IP Address '10.10.3.30' and Type 'SIP Trunk'. The 'Time of Day' section has 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. It shows '1 Item' with a table for scheduling. The table has columns for Ranking, Name, days of the week (Mon-Sun), Start Time, End Time, and Notes. The entry shows '24/7' for Name, all days checked, and '00:00' to '23:59' for the time range. A 'Filter: Enable' link is present. At the bottom, it says 'Select : All, None'.

Name	FQDN or IP Address	Type	Notes
Avaya_SBCE	10.10.3.30	SIP Trunk	

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

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: 15

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL- ▼

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"/>	SM_7		to_Avaya_SBCE	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: 0203

* Min: 4

* Max: 16

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL- ▼

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"/>	SM_7		to_Communication Manager	0	<input type="checkbox"/>	Communication_Manager	

Select : All, None

7. Configure Avaya Session Border Controller for Enterprise

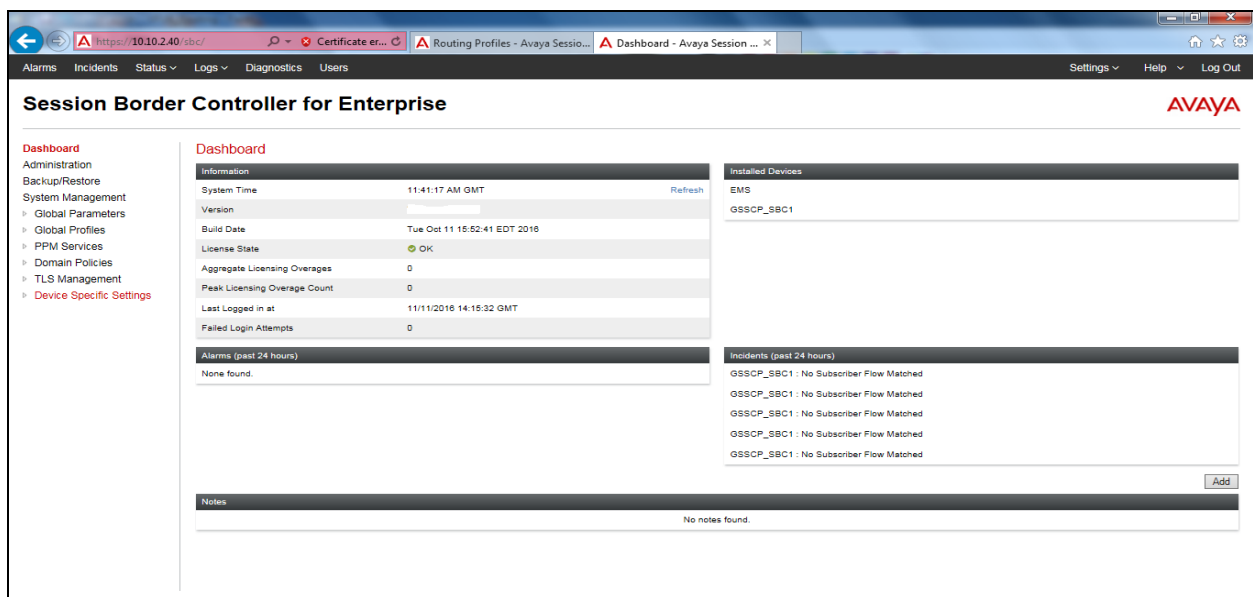
This section describes the configuration of the Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide an interface to the Service Provider's SIP Trunk that is standard where possible and adapted to the Service Provider's SIP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

7.1. Accessing Avaya Session Border Controller for Enterprise

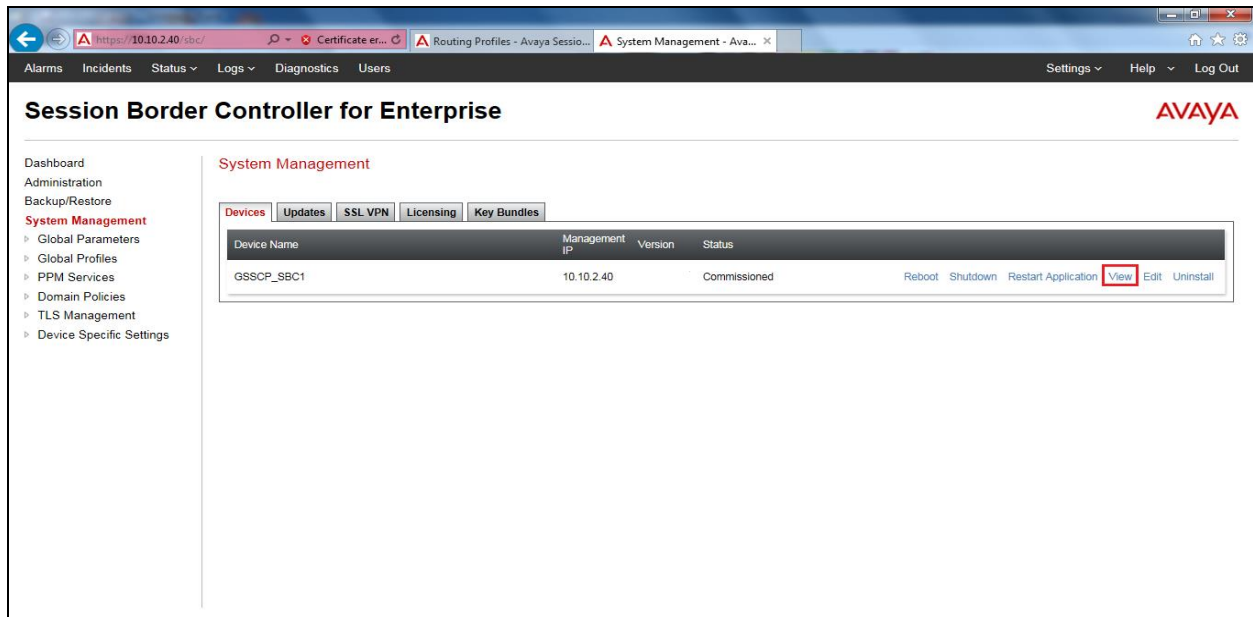
Access the Avaya SBCE using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation and enter the **Username** and **Password**.



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 **GSCCP-SBC1** 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.

System Information: GSCCP_SBC1

General Configuration

Appliance NameGSCCP_SBC1
Box TypeSIP
Deployment ModeProxy

Device Configuration

HAModeNo
Two Bypass ModeNo

License Allocation

Standard Sessions
Requested: 00
Advanced Sessions
Requested: 00
Scopia Video Sessions
Requested: 00
CES Sessions
Requested: 00
Transcoding Sessions
Requested: 00
Encryption☒

Network Configuration

IP	Public IP	Network Prefix or Subnet Mask	Gateway	Interface
10.10.3.50	10.10.3.50	255.255.255.0	10.10.3.1	A1
192.168.122.56	192.168.122.56	255.255.255.0	192.168.122.9	B1

DNS Configuration

Primary DNS8.8.8.8
Secondary DNS10.10.7.100
DNS LocationDMZ

Management IP(s)

IP #1 (IPv4)10.10.2.40

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**.

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

The screenshot shows a configuration window titled "General" with various SIP-related settings. The settings are as follows:

Setting	Value
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 (dropdown menu)
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 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	
Has Remote SBC	<input checked="" type="checkbox"/>
Route Response on Via Port	<input type="checkbox"/>
Relay INVITE Replace for SIPREC	<input type="checkbox"/>
DTMF	
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 – TalkTalk

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**.

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

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

The screenshot shows a configuration window titled "General" with various SIP-related settings. The settings are as follows:

Setting	Value
Hold Support	<input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3284 - 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 (dropdown menu)
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 checked="" type="checkbox"/>
Re-Invite Handling	<input type="checkbox"/>
Prack Handling	<input type="checkbox"/>
Allow 18X SDP	<input type="checkbox"/>
T.38 Support	<input type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3281 <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	
Has Remote SBC	<input checked="" type="checkbox"/>
Route Response on Via Port	<input type="checkbox"/>
Relay INVITE Replace for SIPREC	<input type="checkbox"/>
DTMF	
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, TalkTalk 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** 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 **10.10.3.42** (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.

IP Address / FQDN	Port	Transport
10.10.3.42	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 several configuration options: 'Enable DoS Protection' (checkbox), 'Enable Grooming' (checkbox), 'Interworking Profile' (dropdown menu set to 'Avaya'), 'Signaling Manipulation Script' (dropdown menu set to 'None'), 'Securable' (checkbox), 'Enable FGDN' (checkbox), 'TCP Failover Port' (text input field), and 'TLS Failover Port' (text input field). At the bottom right, there is a 'Finish' button.

7.2.4. Server Configuration – TalkTalk

To define the TalkTalk SBC as a Trunk Server, navigate to **Global Profiles → Server Configuration** and click on **Add** 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 **192.168.112.10** (TalkTalk SBC IP Address).
- For **Port**, enter **5060**.
- For **Transport**, select **UDP**.
- 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. At the top, there is a blue message box that says: 'Server Type can not be changed while this Server Configuration profile is associated to a Server Flow.' Below this, there are fields for 'Server Type' (dropdown menu set to 'Trunk Server'), 'SIP Domain' (text input field), and 'TLS Client Profile' (dropdown menu set to 'None'). There is an 'Add' button to the right of these fields. Below these fields is a table with three columns: 'IP Address / FQDN', 'Port', and 'Transport'. The table contains one row with the values '192.168.112.10', '5060', and 'UDP'. There is a 'Delete' button to the right of the table. At the bottom, there is a 'Finish' button.

On the Advanced tab:

- Select **TalkTalk** for Interworking Profile.
- Select **TalkTalk** for **Signaling Manipulation Script** (Section 7.2.7).
- Click **Finish**.

Server Configuration Profile - Advanced

Enable DoS Protection ☐

Enable Grooming ☐

Interworking Profile TalkTalk

Signaling Manipulation Script TalkTalk

Securable ☐

Enable FGDN ☐

TCP Failover Port

TLS Failover Port

Finish

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 TalkTalk 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

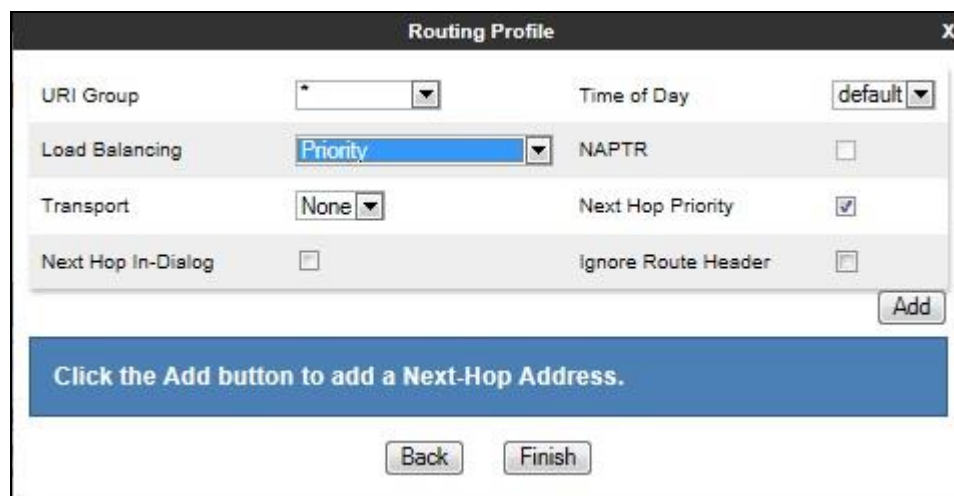
Create a Routing Profile for Session Manager.

- Navigate to **Global Profiles → Routing** and select **Add**.
- Enter a **Profile Name** and click **Next**.



The screenshot shows a window titled "Routing Profile" with a close button (X) in the top right corner. Inside the window, there is a text input field labeled "Profile Name" containing the text "Avaya". Below the input field is a button labeled "Next".

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



The screenshot shows a window titled "Routing Profile" with a close button (X) in the top right corner. The window contains several settings:

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

Below the settings is an "Add" button. At the bottom of the window, 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 10.10.3.42:5060 (TCP)** from drop down menu.
- Click **Finish**.

Priority / Weight	Server Configuration	Next Hop Address	Transport
1	Avaya	10.10.3.42:5060 (TCP)	None

7.2.5.2 Routing – TalkTalk

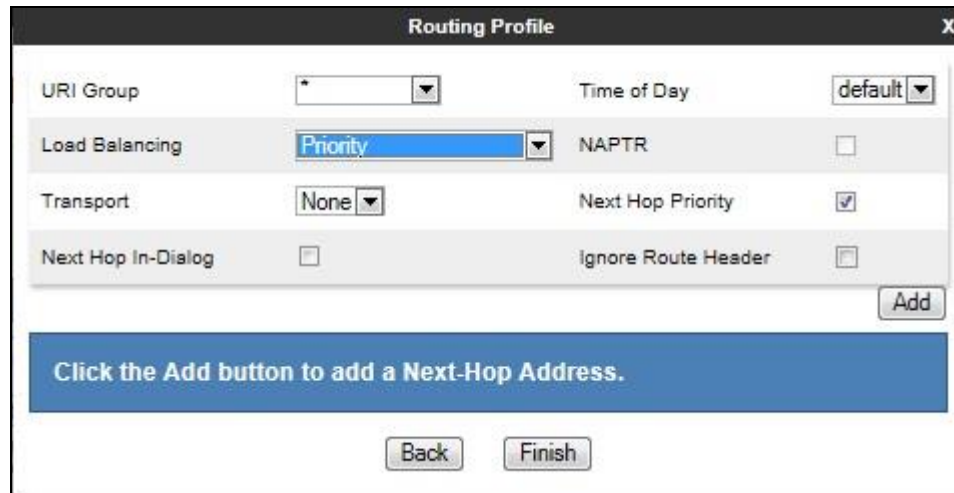
Create a Routing Profile for TalkTalk.

- Navigate to **Global Profiles → Routing** and select **Add Profile**.
- Enter a **Profile Name** and click **Next**.

Profile Name: TalkTalk

Next

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

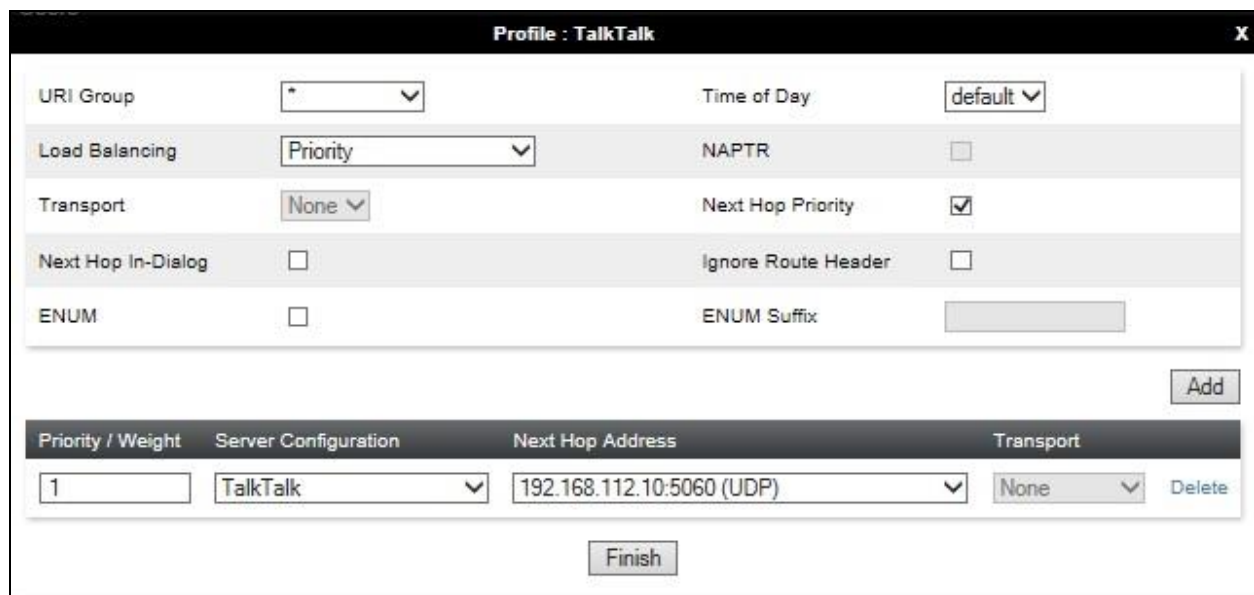


The screenshot shows the 'Routing Profile' window. It contains the following fields and controls:

- 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:** An unchecked checkbox.
- Transport:** A dropdown menu with 'None' as the selected value.
- Next Hop Priority:** A checked checkbox.
- Next Hop In-Dialog:** An unchecked checkbox.
- Ignore Route Header:** An unchecked checkbox.
- Add:** A button located at the bottom right of the configuration area.
- Message:** A blue banner at the bottom states 'Click the Add button to add a Next-Hop Address.'
- Back:** A button at the bottom left.
- Finish:** A button at the bottom right.

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

- **Priority/Weight = 1.**
- **Server Configuration = TalkTalk** (Section 7.2.4) from drop down menu.
- **Next Hop Address = Select 192.168.112.10:5060 (UDP)** from drop down menu.
- Click **Finish**.



The screenshot shows the 'Profile : TalkTalk' window. It contains the following fields and controls:

- 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:** An unchecked checkbox.
- Transport:** A dropdown menu with 'None' as the selected value.
- Next Hop Priority:** A checked checkbox.
- Next Hop In-Dialog:** An unchecked checkbox.
- Ignore Route Header:** An unchecked checkbox.
- ENUM:** An unchecked checkbox.
- ENUM Suffix:** An empty text input field.
- Add:** A button located at the bottom right of the configuration area.
- Table:** A table with the following columns: 'Priority / Weight', 'Server Configuration', 'Next Hop Address', 'Transport', and 'Delete'.

Priority / Weight	Server Configuration	Next Hop Address	Transport	Delete
1	TalkTalk	192.168.112.10:5060 (UDP)	None	Delete
- Finish:** A button at the bottom center.

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).

The screenshot shows the 'Topology Hiding Profiles: Avaya' configuration page. On the left, a sidebar lists 'Topology Hiding Profiles' with options: 'default', 'cisco_th_profile', 'Avaya' (selected), and 'TalkTalk'. The main area has a blue header bar with 'Click here to add a description.' and buttons for 'Add', 'Rename', 'Clone', and 'Delete'. Below this is a 'Topology Hiding' tab containing a table with the following data:

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

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

To define Topology Hiding for TalkTalk, 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 TalkTalk 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: TalkTalk

Add

Topology Hiding Profiles

default

cisco_th_profile

Avaya

TalkTalk

Click here to add a description.

Topology Hiding

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

Edit

Rename

Clone

Delete

7.2.7. Signaling Manipulation

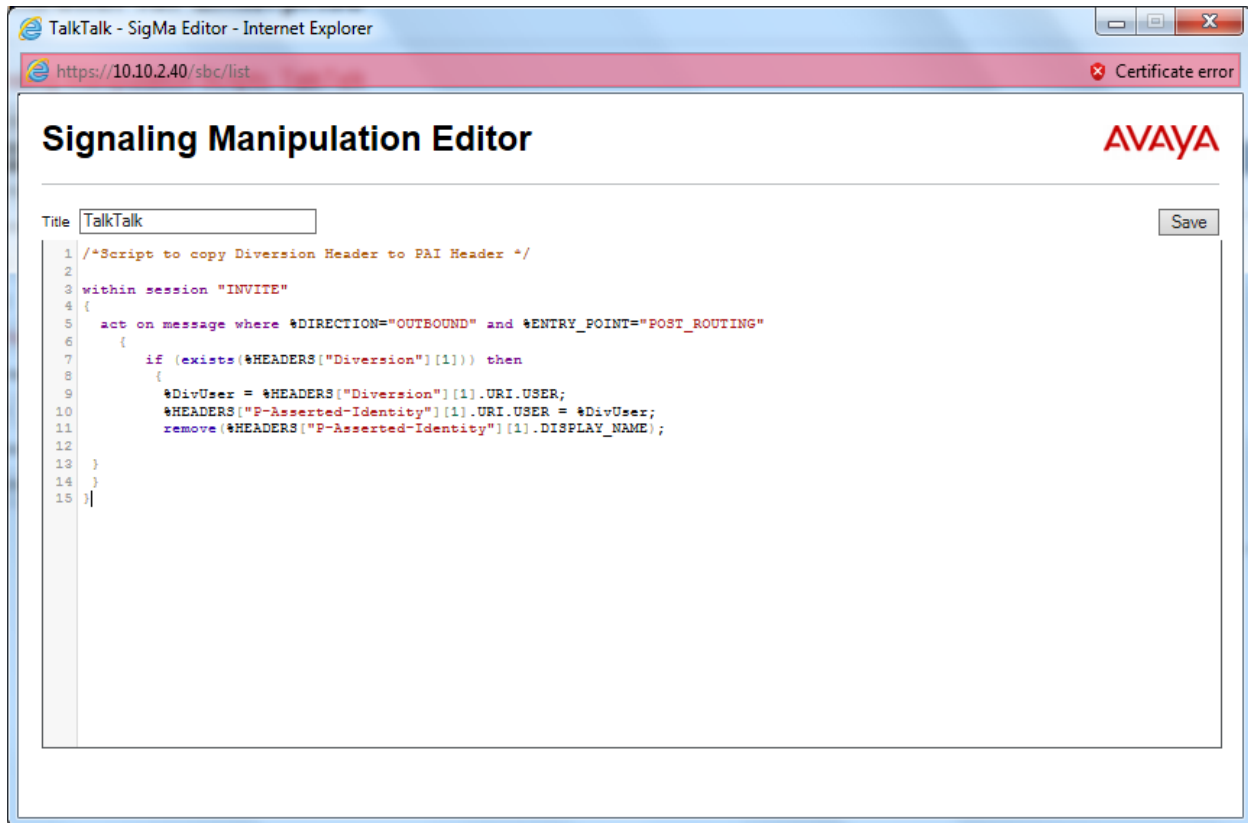
The Signaling Manipulation feature allows the ability to add, change and delete any of the headers in a SIP message. This feature will add the ability to configure such manipulation in a highly flexible manner using a proprietary scripting language called SigMa. The SigMa scripting language is designed to express any of the SIP header manipulation operations to be done by the Avaya SBCE.

When performing call re-direction or EC500 calls, TalkTalk require the P-Asserted-Identity Header to be populated with the CLID of the set that is performing the call re-direction/EC500 call or of a known TalkTalk CLID on the PABX. In order for call redirection/EC500 to complete successfully, a SigMa script was created on the Avaya SBCE to copy the CLID from the Diversion Header and populate the P-Asserted-Identity Header with the Diversion Header CLID.

To define the signalling manipulation to delete the Avaya proprietary parameter from the Contact Header, navigate to **Dashboard → Global Profiles → Signaling Manipulation** and click on **Add** and enter a title. A new blank SigMa Editor window will pop up. The script text is as follows:

```
/*Script to copy Diversion Header to PAI Header */  
  
within session "INVITE"  
{  
  act on message where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"  
  {  
    if (exists(%HEADERS["Diversion"][1])) then  
    {  
      %DivUser = %HEADERS["Diversion"][1].URI.USER;  
      %HEADERS["P-Asserted-Identity"][1].URI.USER = %DivUser;  
      remove(%HEADERS["P-Asserted-Identity"][1].DISPLAY_NAME);  
    }  
  }  
}
```

Once entered and saved, the script appears as shown in the following screenshot:



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.

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: GSSCP_SBC1

Devices | Interfaces | Networks

GSSCP_SBC1

Add

Name	Gateway	Subnet Mask / Prefix Length	Interface	IP Address	Edit	Delete
Internal_A1	10.10.3.1	255.255.255.0	A1	10.10.3.50	Edit	Delete
External_B1	192.168.122.9	255.255.255.0	B1	192.168.122.56	Edit	Delete

Select the **Interface Configuration** tab and click on the **Status** of the physical interface to toggle the state. Change the state to **Enabled** where required.

Network Management: GSSCP-SBC1

Devices | Interfaces | Networks

GSSCP-SBC1

Add VLAN

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

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 address 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 **UDP** port number, **5060** is used for TalkTalk SIP Trunk service.

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

Signaling Interface: GSSCP_SBC1

Devices

GSSCP_SBC1

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	10.10.3.50 Internal_A1 (A1, VLAN 0)	5060	5060	---	None	Edit Delete
Ext_Signalling	192.168.122.56 External_B1 (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** and enter details in the pop-up menu.
- 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** and enter details in the pop-up menu.
- 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: GSSCP_SBC1

Devices

GSSCP_SBC1

Media Interface

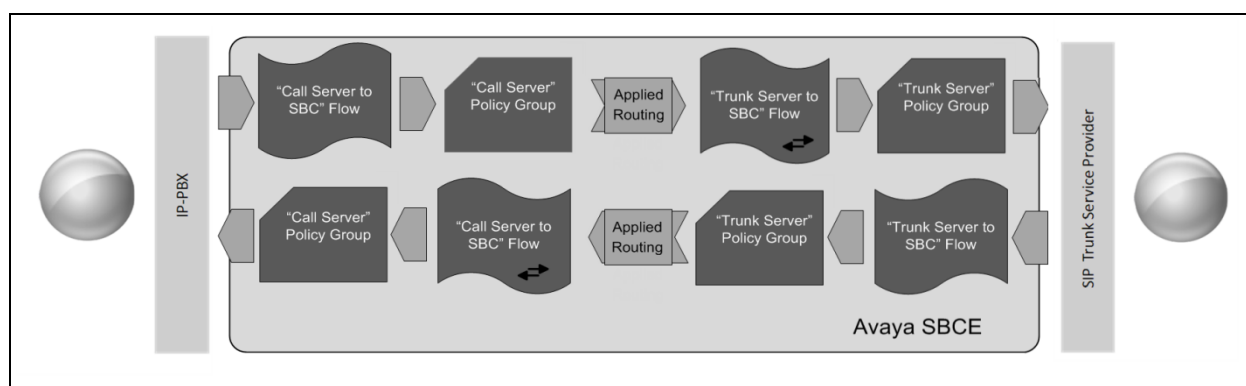
Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from [System Management](#).

Add

Name	Media IP Network	Port Range	
Int_Media	10.10.3.50 Internal_A1 (A1, VLAN 0)	35000 - 40000	Edit Delete
Ext_Media	192.168.122.56 External_B1 (B1, VLAN 0)	35000 - 40000	Edit Delete

7.4.3. Server Flows

Server Flows combine the previously defined profiles into outgoing flows from Session Manager to TalkTalk's SIP Trunk and incoming flows from TalkTalk'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 TalkTalk network and vice versa. The following screen illustrates the flow through the Avaya SBCE to secure a SIP Trunk call.



The following screenshot shows all configured flows.

Subscriber Flows

Server Flows

Add

Hover over a row to see its description.

Server Configuration: Avaya

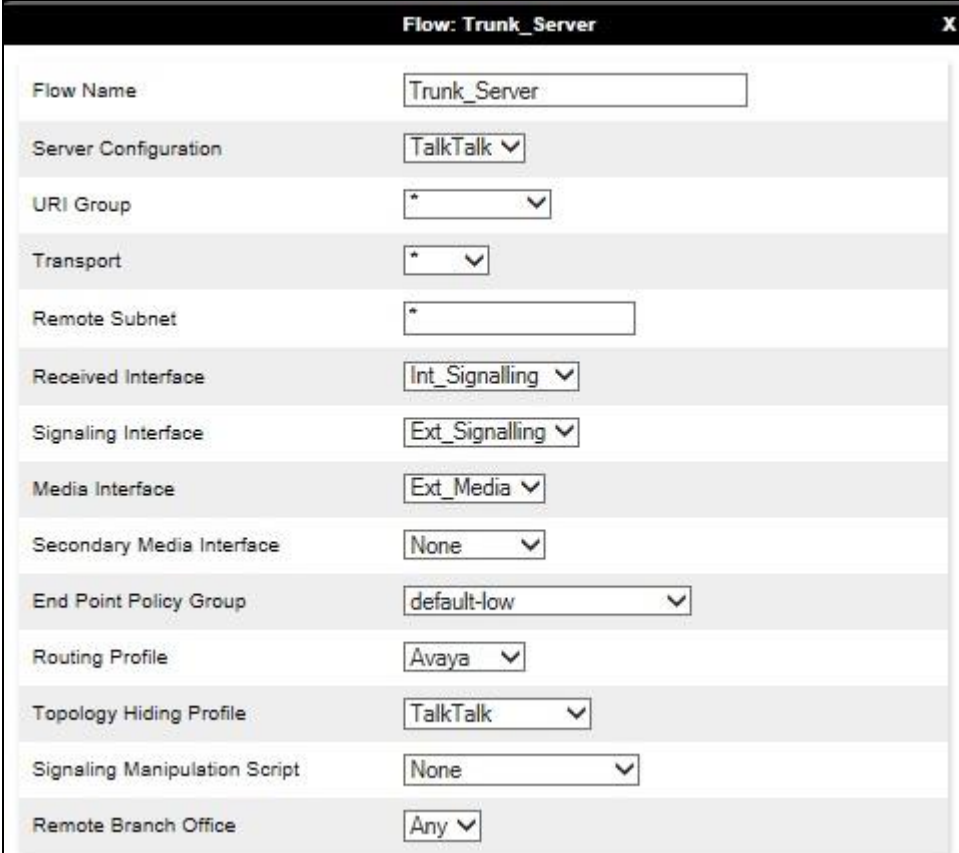
Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Call_Server	*	Ext_Signalling	Int_Signalling	default-low	TalkTalk	View	Clone	Edit	Delete

Server Configuration: TalkTalk

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 TalkTalk SIP Trunk, navigate to **Device Specific Settings** → **End Point Flows**.

- Click on the **Server Flows** tab (shown above).
- Select **Add** and enter details in the pop-up menu.
- In the **Flow Name** field enter a descriptive name for the server flow for TalkTalk SIP Trunk, in the test environment **Trunk_Server** was used.
- In the **Server Configuration** drop-down menu, select the TalkTalk 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 TalkTalk 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 TalkTalk 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 TalkTalk SIP Trunk is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of Session Manager Office defined in **Section 7.2.5**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of TalkTalk SIP Trunk defined in **Section 7.2.6** and click **Finish**.

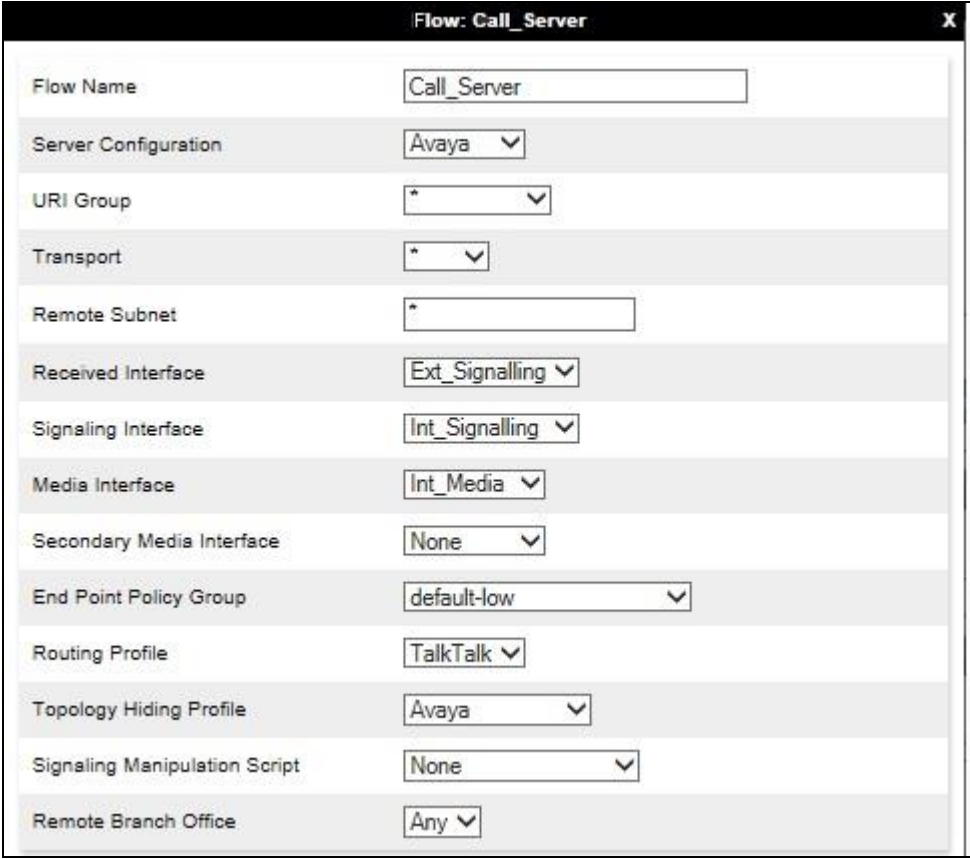


The screenshot displays a configuration window titled "Flow: Trunk_Server". It contains the following fields and values:

Field	Value
Flow Name	Trunk_Server
Server Configuration	TalkTalk
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Int_Signalling
Signaling Interface	Ext_Signalling
Media Interface	Ext_Media
Secondary Media Interface	None
End Point Policy Group	default-low
Routing Profile	Avaya
Topology Hiding Profile	TalkTalk
Signaling Manipulation Script	None
Remote Branch Office	Any

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

- Click on the **Server Flows** tab.
- Select **Add** and enter details in the pop-up menu.
- In the **Flow 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 TalkTalk 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 displays a configuration window titled "Flow: Call_Server". It contains the following fields and values:

Field	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
Secondary Media Interface	None
End Point Policy Group	default-low
Routing Profile	TalkTalk
Topology Hiding Profile	Avaya
Signaling Manipulation Script	None
Remote Branch Office	Any

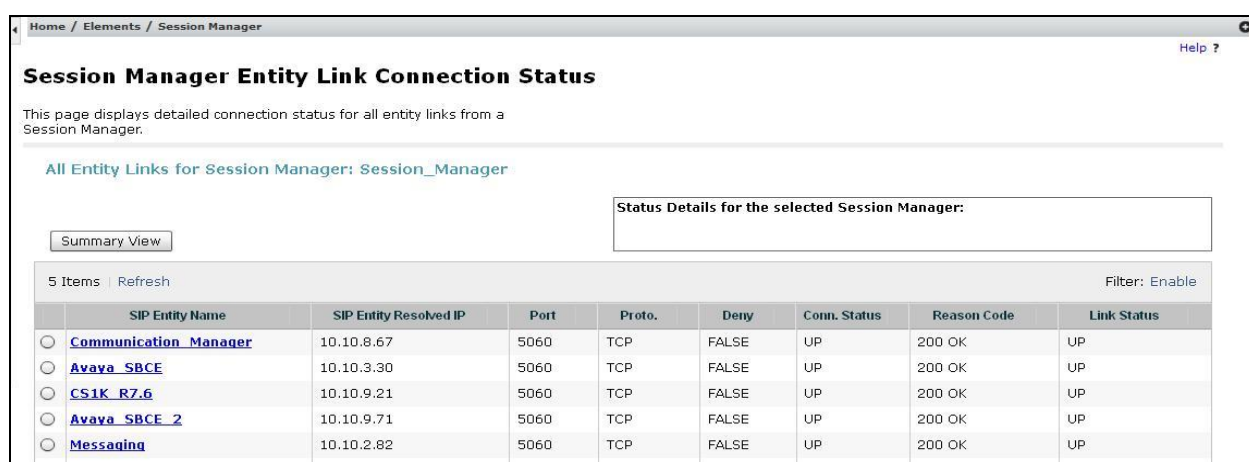
8. TalkTalk Business NGVS SIP Trunk Service Configuration

The configuration of the TalkTalk equipment used to support TalkTalk's NGVS SIP trunk is outside of the scope of these Application Notes and will not be covered. To obtain further information on TalkTalk equipment and system configuration please contact an authorized TalkTalk 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
<input type="radio"/> Communication Manager	10.10.8.67	5060	TCP	FALSE	UP	200 OK	UP
<input type="radio"/> Avaya SBCE	10.10.3.30	5060	TCP	FALSE	UP	200 OK	UP
<input type="radio"/> CS1K R7.6	10.10.9.21	5060	TCP	FALSE	UP	200 OK	UP
<input type="radio"/> Avaya SBCE 2	10.10.9.71	5060	TCP	FALSE	UP	200 OK	UP
<input type="radio"/> 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
```

TRUNK GROUP STATUS			
Member	Port	Service State	Mtce Connected Ports Busy
0001/001	T00001	in-service/idle	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: GSSCP-SBC1

Devices

GSSCP-SBC1

Packet Capture
Captures

Packet Capture Configuration

Status	Ready
Interface	B1 ▾
Local Address IP:Port	All ▾ : <input type="text"/>
Remote Address *, *.Port, IP, IP:Port	* <input type="text"/>
Protocol	All ▾
Maximum Number of Packets to Capture	<input type="text" value="10000"/>
Capture Filename <small>Using the name of an existing capture will overwrite it.</small>	<input type="text" value="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: GSSCP-SBC1

Devices

GSSCP-SBC1

Packet Capture

Captures

Refresh

File Name	File Size (bytes)	Last Modified	
Test_20160413085450.pcap	0	April 13, 2016 8:54:50 AM CDT	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 TalkTalk network.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager R7.0 as an Evolution Server, Avaya Aura® Session Manager R7.0 and Avaya Session Border Controller for Enterprise R7.1 to TalkTalk Business NGVS SIP Trunk solution. TalkTalk's Business NGVS SIP Trunk 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*, Oct 2016
- [2] *Avaya Aura® Communication Manager 7.0 Documentation library*, Oct 2016
- [3] *Avaya Aura® System Manager using VMware® in the Virtualized Environment Deployment Guide Release 7.0* Oct 2016
- [4] *Implementing Avaya Aura® System Manager Release 7.0*, Aug 2016
- [5] *Upgrading Avaya Aura® System Manager to Release 7.0*, Aug 2016
- [6] *Administering Avaya Aura® System Manager Release 7.0*, Aug 2016
- [7] *Avaya Aura® Session Manager using VMware® in the Virtualized Environment Deployment Guide Release 7.0*, 2016
- [8] *Implementing Avaya Aura® Session Manager Release 7.0*, Nov 2016
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